DEPLOYMENT GUIDE

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Polycom® UC Software with Skype for Business

For Polycom® Polycom Trio™ 8800 and 8500 Systems and Polycom Trio Visual+ System

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Contents

Conventions Used in Polycom Guides 6
Information Elements 6
Typographic Conventions 7

Getting Started 8
Audience and Purpose of This Guide 8
Microsoft Qualified Phones 8
Polycom Interoperability with Microsoft 8
Prerequisites - On-Premises Deployments 9
Polycom UC Software, Template Files, and Documentation 9
Limitations 10
Get Help 10

Deploying Polycom Phones with Skype for Business 11
Power the Polycom Trio System 11
Configure the Network 11
Set Up Polycom UC Software 14
Provisioning the Phones 15
Configuring In-Band Provisioning Settings 19
Provision and Update with a USB Device 20

Audio Features 23
Polycom NoiseBlock™ 23
Microphone Mute 23
Audio Output Options 24
USB Audio Calls 25
Polycom Trio System Audio Codec Support with Skype for Business 30

Video Features 32
Video and Camera Options 32
Toggling Between Audio-only or Audio-Video Calls 38
Video-based Screen Sharing Support for Polycom Trio Solution 39
Skype for Business Video Layouts on Polycom Trio System 40
Forward Error Correction 40
Video Simulcast 41

Supported Video Codecs 42

Content 46

Content Sharing 46
Create Conference Room Accounts for Skype for Business 49
Screen Mirroring 50

User Accounts and Contacts 58

Smart Login 58
Microsoft Exchange Integration 58
Private Meetings in Microsoft Exchange 66
Skype for Business User Profiles 67
Unified Contact Store 69
Sign In Methods 69
Contact Directories 73
Call Logs 74
Administrator Menu on Polycom Trio Systems 74

Call Controls 76

Skype for Business Local and Centralized Real-Time Audio and Video Calling 76
Using an API to Join a Skype for Business Meeting 76
Join a Meeting with a SIP URI 77
Skype for Business Private Meeting Parameters 78
USB Mode for Skype Room Systems and Surface Hub 80
Local Call Recording 82
Configuring Shared Line Appearance (SLA) for Skype for Business 83
Hybrid Line Registration 83
International Dialing Prefix 86
Centralized Conference Control Protocol (CCCP) 87
Local Digit Map 87
Dial Plans 93

System Display 100

Capture Your Device's Current Screen 100
Time and Date Wizard 101
Polycom Trio System Screen 102
Phone Theme 105
Polycom Trio System Display Name 106
Polycom Trio Solution IP Address 108
Configure the Polycom Trio System Phone Number and Label 109
Status Messages 110

System Name for Wireless Content Connections 111
Time Zone Location Description 113

Network 116
Near Field Communication (NFC)-Assisted Bluetooth 116
Wireless Network Connectivity (Wi-Fi) 117
Extended Link Layer Discovery Protocol (LLDP) 119
STUN / TURN / ICE Parameters 120

Hardware and Accessories 124
Powering the Polycom Trio 8500 and 8800 Systems 124
Power the Polycom Trio System with the Optional Power Injector 125
Pairing the Polycom Trio Visual+ with Polycom Trio 125
Polycom Trio System Power Management 130
Consumer Electronics Controls (CEC) over HDMI 131

Device and Software Support 135
In-Band Provisioning 135
Data Center Resiliency 135
Polycom Experience Cloud 136
Client Media Port Ranges for QoE 137
Microsoft Quality of Experience Monitoring Server Protocol (MS-QoE) 137
User Log Upload 144
Polycom UC Software Update 145
Phone Default Settings 149
Change the Base Profile from the Phone 149
Inbound and Outbound Ports for Polycom Trio System with Skype for Business 150
Real-Time Transport Protocol (RTP) Port Parameters for Skype for Business 152

Conventions Used in Polycom Guides

Polycom guides contain terms, graphical elements, and a few typographic conventions. Familiarizing yourself with these terms, elements, and conventions will help you successfully perform tasks.

Information Elements

Polycom guides may include any of the following icons to alert you to important information.

Information Elements

Name	Icon	Description
Note		The Note icon highlights information of interest or important information needed to be successful in accomplishing a procedure or to understand a concept.
Important		Important highlights information of interest or important information needed to be successful in accomplishing a procedure or to understand a concept.
Caution		The Caution icon highlights information you need to know to avoid a hazard that could potentially impact device performance, application functionality, or successful feature configuration.
Web Info		The Web Info icon highlights supplementary information available online such as documents or downloads on support.polycom.com or other locations.

Typographic Conventions

A few typographic conventions, listed next, are used in Polycom guides to distinguish types of in-text information.

Typographic Conventions

Convention	Description
Bold	Highlights interface items such as menus, menu selections, window and dialog names, soft keys, file names, and directory names when they are involved in a procedure or user action. Also used to highlight text to be entered or typed.
Italics	Used to emphasize text, to show example values or inputs (in this form: <*example*>), and to show titles of reference documents available from the Polycom Support Web site and other reference sites.
Blue Text	Used for cross references to other sections within this document and for hyperlinks to external sites and documents.
`Courier`	Used for code fragments, parameter names and permitted values.

Getting Started

This guide provides general guidance on installing and provisioning with Polycom UC Software and shows you how to deploy the Polycom Trio 8800 and 8500 systems.

The Polycom Trio Visual+ accessory is supported with the Polycom Trio 8500 and 8800 systems. This guide shows you how to set up a Polycom Trio system with the Polycom Trio Visual+ system to provide video and content-sharing capabilities to your calls.

The Polycom Trio 8800 and 8500 systems are also known as Polycom RealPresence Trio 8800 and 8500 systems.

Audience and Purpose of This Guide

This guide provides information for mid-level administrators with experience in networking who understand the basics of open SIP networks, VoIP endpoint environments, and Microsoft servers and environments.

Microsoft Qualified Phones

Polycom offers Polycom Trio systems already configured for use with Skype for Business on-premise deployments or Skype for Business Online. Polycom Trio systems include Microsoft-qualified UC Software with a Polycom feature license that enable you to start up and register with default Microsoft settings.

Feature Licenses

Polycom devices purchased and shipped with a Skype or Lync Base Profile include a Polycom feature license to register with Skype for Business, Lync Server, and Office 365. If you do not purchase devices with a configured Skype or Lync Base Profile, you can use Polycom phones in a Skype for Business, Lync Server, or Office 365 environment for trial purposes, without purchasing a license, for a maximum of 30 days.

For information about purchasing a Polycom feature license, talk to your Polycom reseller or Polycom sales representative.

Polycom Interoperability with Microsoft

Polycom Trio systems support integration with:

- Skype for Business on-premises.
- Skype for Business Online including Microsoft Skype for Business Cloud PBX in Microsoft Office 365.
- Polycom Trio Visual+ system interoperability with Skype for Business on-premises and Online (O365) is Polycom supported but not Microsoft qualified.
- The migration of users from Skype for Business on-premises to Office 365
- Microsoft Exchange on-premises 2013 and 2010
- Exchange Online services
- Lync Server 2013 and Lync Server 2010 (audio only) for on-premise deployments.

Web Info: For full lists of Polycom-supported Microsoft environments, features, client versions, and products tested, see the *Polycom Trio Solution - Release Notes* on Polycom Trio.

Prerequisites - On-Premises Deployments

Before you set up Polycom devices for an on-premises Skype for Business deployment, ensure that you complete the following tasks:

- Set the server log levels to capture only low-level events.
- Disable automatic device update by setting:
  - `Set-CsIPPhonePolicy -EnableDeviceUpdate $False`

    For more information see Set-CsIPPhonePolicy on Microsoft TechNet.
  - `device.prov.lyncDeviceUpdateEnabled.set=0`
  - `device.prov.lyncDeviceUpdateEnabled=0`

Polycom UC Software, Template Files, and Documentation

Polycom offers UC Software for Skype for Business in two file formats:

- Combined or Split **sip.ld**.
- Polycom offers UC Software in CAB file format. This Microsoft Windows archive file format, recommended by Microsoft for customer premises equipment (CPE), safely compresses data and embeds digital certificates.

This Deployment Guide refers to two configuration file templates:

- `SkypeTrioSharedExample.cfg`. This template file for the Polycom Trio 8800 system provides you with all required parameters and default settings to register with Skype for Business Server.
- `000000000000.cfg`. Use this master configuration file to load `SkypeTrioSharedExample.cfg` and all other configuration files containing settings you want to load to devices from a central provisioning server. For details on using template and master configuration files, see the *Polycom Trio Solution - Administrator Guide*.

Web Info: All template files, Polycom UC Software, and supporting documentation is available on Polycom Trio Support.

Limitations

The Polycom Trio 8800 and 8500 systems support audio-only conference calls in a Microsoft Lync Server 2010 environment.

Get Help

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

The Polycom Community

The Polycom Community gives you access to the latest developer and support information. Participate in discussion forums to share ideas and solve problems with your colleagues. To register with the Polycom Community, simply create a Polycom online account. When logged in, you can access Polycom support personnel and participate in developer and support forums to find the latest information on hardware, software, and partner solutions topics.

Deploying Polycom Phones with Skype for Business

Complete four major tasks to register your phones:

- Power the Polycom Trio System
- Configure the Network
- Set Up Polycom UC Software
- Provisioning the Phones

Note: Polycom phones ordered with the Skype or Lync Base Profile are shipped with Skype for Business-qualified software that enables you to start up the phone and register with Skype for Business Server with default settings. If you are using Polycom phones shipped with Skype for Business-qualified UC Software and want to keep default settings with no change, you need only complete the task Set Up the Network. If you want to customize default settings, complete all four tasks.

Power the Polycom Trio System

For instructions on powering the Polycom Trio 8800 and 8500 systems, and the Polycom Trio Visual+ system, refer to Hardware and Accessories.

Configure the Network

Configure the following network settings to register Polycom devices with Skype for Business.

To configure the network:

1 Set up or verify Domain Name System (DNS) service (SRV) records to allow the devices to discover Skype for Business server automatically. For information on creating and verifying DNS SRV records, see Required DNS Records for Automatic Client Sign-In on Microsoft TechNet.

2 (Optional) If you are setting Microsoft Call Admission Control (CAC) refer to Microsoft Plan for call admission control in Skype for Business Server 2015 for required bandwidth guidelines.

3 Obtain a root certificate authority (CA) security certificate using one of the following methods:

Certificate Method	Description
Lightweight Directory Access Protocol (LDAP) Domain Name System (DNS)	Polycom devices running UC Software 5.3.0 or later that you are registering with Skype for Business automatically fetch the root certificate using a LDAP DNS query. Phones you register with Skype for Business are enabled with this feature by default and no additional configuration is required.
Dynamic Host Configuration Protocol (DHCP) Option 43	When provisioning phones from within an enterprise, you can use DHCP Option 43 to download a private CA root security certificate used by Skype for Business. The security certificate is required to support secure HTTPS and TLS connections. In conjunction with DHCP Option 43, ensure that your devices can access Skype for Business Server Certificate Provisioning Web service over HTTP (TCP 80) and HTTPS (TCP 443). Note: If you configure DHCP Option 43 in on-premises Skype for Business deployments, the phone displays only the PIN Authentication menu to users. For more details and troubleshooting information on DHCP Option 43, see Set Up DHCP for Devices on Microsoft TechNet. For a list of DHCP options and sub-options supported by Polycom Trio solution, refer to Supported DHCP Options and Sub-Options on the Polycom Trio System.
DHCP Option 66	Use this method if you are using a provisioning server or set DHCP options using one of the following methods: • DHCP Option 160. If you are using Polycom devices with a Skype or Lync Base Profile, use Option 161 with the address (URL or IP address) of the provisioning server. You can set the provisioning server address or URL on the device menu. • DHCP Option 161. If you are using Polycom devices with an Open SIP Base Profile, use Option 160 with the address (URL or IP address) of the provisioning server. You can set the provisioning server address or URL on the device menu or set the Base Profile using the Web Configuration Utility.

4 Set up each user with a Skype for Business account and credentials.

Supported DHCP Options and Sub-Options on the Polycom Trio System

The following table lists the individual options and sub-options for DHCP Option 43 supported on the Polycom Trio system.

DHCP Options and Sub-Options

Option	*Result*
Option 1- Subnet mask	The phone parses the value from Option 43
Option 2 - Time offset	The phone parses the value.
Option 3 - Router	The phone parses the value.

DHCP Options and Sub-Options

Option 4 - Time server	The phone parses the value.
Option 6 - Domain Name Server	The phone parses the value.
Option 7 - Domain Log server	The phone parses the value.
Option 15 - Domain Name	The phone parses the value.
Option 42 - Network Time Protocol server	The phone parses the value.
Option 66 - TFTP Server Name	The phone parses the value.
Sub-options configured in Option 43	
Options 1, 2, 3, 4, 5, 6, 7, 15, 42, and 66	The phone parses the value.

Skype for Business DHCP Options on Polycom Trio Solution

The following table lists the Skype for Business options and sub-options for DHCP Option 43 supported on Polycom Trio systems. For more detailed information on sub-options and URL format, see Setting Up DHCP for Devices on Microsoft TechNet.

DHCP Options and Sub-Options

Option	*Result*
Option 1- Subnet mask	The phone parses the value from Option 43
Option 2 - Time offset	The phone parses the value.
Option 3 - Router	The phone parses the value.
Option 4 - Time server	The phone parses the value.
Option 6 - Domain Name Server	The phone parses the value.
Option 7 - Domain Log server	The phone parses the value.
Option 15 - Domain Name	The phone parses the value.
Option 42 - Network Time Protocol server	The phone parses the value.
Option 66 - TFTP Server Name	The phone parses the value.
Sub-options configured in Option 43	
Sub-option 1 - UC Identifier	
Sub-option 2 - URL Scheme	
Sub-option 3 - Web Server FQDN	

DHCP Options and Sub-Options
Sub-option 4 - Port
Sub-option 5 - Relative Path for Certificate Provisioning Web Service.
Example URL: `https://lyncsvrWebPoolFQDN:443/CertProv/CertProvisioningService.svc`

Set Up Polycom UC Software

The latest UC Software for Polycom Trio is available on Polycom Trio Support.

Caution: Do not provision phones with UC Software from both a Microsoft server and your own provisioning server. This places the phones in a reboot cycle.

To set up Polycom UC Software:

1. Set up a provisioning server on your computer and create a root directory to hold all of the required UC Software, configuration files, and subdirectories. Name the directory to identify it as containing the Polycom UC Software release.

   To set up your own provisioning server, you need an XML editor, such as XML Notepad, installed on your computer. Your provisioning, or boot server must support one of the FTP, FTPS, TFTP, HTTP, or HTTPS protocols, FTP being the most common. FileZilla Server is a free FTP solution.

2. Decide if you are provisioning your phones from Skype for Business Server or using your own provisioning server.

   Deploying UC Software in CAB file format provisions the phones and enables default feature functionality, including the automatic software update feature. However, if you want to change or customize default functionality of the phone features, you need to set up and edit Polycom UC Software configuration files on your own provisioning server and send the custom settings to the phones.

   - To use Skype for Business Server to push software to the phones, complete the steps in the section Deploy UC Software from Skype for Business Server.

3. Download, save, and extract UC Software to the root directory you created.

   - If you are deploying UC Software from Skype for Business Server, download the CAB file version of Polycom UC Software.
   - If you are deploying phones from your own provisioning server, download the split or combined version of Polycom UC Software in XML format.

4. After the UC Software directory is extracted, open the folder in your root directory.

5. Configure a Call Park Orbit Policy. You must configure a call park orbit policy to enable the call park feature. See Configuring Call Park on the Microsoft web site.

Provisioning the Phones

Polycom provides manual per-phone provisioning methods and centralized provisioning methods. The method labeled device.set is an advanced method for users familiar with Polycom configuration files and uses centralized provisioning to set the Base Profile for multiple phones.

The Base Profile is a provisioning option available on Skype for Business-enabled Polycom devices that simplifies the process of registering your devices with Skype for Business. The Base Profile displays in the phone's menu system and varies by phone model. The Base Profile automates registration with a default set of configuration parameters and settings; you cannot modify or customize the Base Profile or feature settings. Because you can provision only a single phone at a time from the local phone menu, Polycom recommends using centralized provisioning for deployments of greater than 20 devices requiring only default Skype for Business settings.

For complete information on provisioning Polycom Trio with UC Software, see the Polycom Trio Systems - Administrator Guide on Polycom Trio Support.

Manual Provisioning Methods

Polycom provides five per-phone manual methods you can use to register Polycom devices with Skype for Business. All manual provisioning methods set the Base Profile of a phone to Skype. The Base Profile is a feature on each Polycom phone that, when set to Skype, automatically provisions the phone with the default parameters required to work with Skype for Business.

You can set the Base Profile of a phone to Skype in the following ways:

- Set the Base Profile from the Settings Menu. Set the Base Profile to Skype from the phone's Settings menu during normal phone functioning.
- Set the Base Profile Using the Web Configuration Utility. Use the Polycom Web Configuration Utility to set the Base Profile from a web browser. This is particularly useful when working remotely.

Note: When you use configuration files to provision the phones with Skype for Business, the phone Base Profile stays set to Generic. You do not need to set the Base Profile feature on the phones to Skype for Business when provisioning with configuration files.

Manually Reboot the Phone

When you change the Base Profile using any of these methods, the phone reboots. If the phone does not reboot, you can manually reboot by powering off/on the phone or manually rebooting the phone from the Settings menu.

To manually reboot the phone:

1. Go to **Settings > Advanced**.
2. Enter the password (default 456).
3. Press **Enter**.
4. Choose **Reboot Phone**.

   When the phone completes the reboot cycle, the Sign In screen displays.

Set the Base Profile from the Settings Menu

You can set the Base Profile to Skype from the phone Settings menu.

To set the Base Profile to Skype from the Settings Menu:

1. Go to **Settings > Advanced > Administration Settings > Network Configuration**, and set Base Profile to **Skype**.
2. Select **Back > Save Configuration**. The phone automatically restarts and displays the Sign In screen. Users can now sign in.

Set the Base Profile Using the Web Configuration Utility

If your phone is not shipped with the Base Profile set to Skype for Business, you can use the Web Configuration Utility to manually set a phone's Base Profile to Skype. As part of a UC Software security update, phone access to the Web Configuration Utility is disabled by default when the phone registers with Skype for Business Server. To enable access, refer to Access to the Web Configuration Utility. Note you cannot configure sign-in credentials using the Polycom Web Configuration Utility.

To set the Base Profile to Skype using the Web Configuration Utility:

1. Provide power to your phones and allow the phones to complete the power-up process.
2. Obtain the IP address of each phone in your deployment by going to **Settings > Status > Platform > Phone**. The IP address displays in the IP: field.
3. Enter the phone's IP address in the address bar of a web browser.
   The Web Configuration Utility login screen displays.
4. Choose **Admin** to log in as an administrator, and then enter the administrator password (default 456) and click **Submit**.
5. In the Home page, navigate to the **Simple Setup** menu.
6. From the Base Profile drop-down, choose **Skype**, and click **Save** at the bottom of the page.
7. In the confirmation dialog, choose **Yes**. The phone automatically restarts.
   Users can now sign in.

Centralized Provisioning

Polycom strongly recommends using a central provisioning server when provisioning multiple phones to:

- Configure multiple devices automatically
- Facilitate automated software updates
- Receive automatic log files
- Add, remove, or manage features and settings to multiple phones simultaneously
- Create phone groups and modify features and settings for each phone group

Caution: Using an existing server to deploy your provisioning server can affect performance of your Skype for Business deployment. Misconfiguration or nonstandard deployment of the Microsoft Internet Information Services (IIS) web server may affect your ability to obtain accurate Microsoft support.

Centralized Provisioning Methods

Use one of the following methods to centrally deploy multiple devices:

- Set Up Polycom with Skype for Business Online and Microsoft® Exchange Online. Use Skype for Business Online or Microsoft Exchange Online to set up phones and configure features.
- Deploy UC Software from Skype for Business Server. Download UC Software in CAB file format and place the software on Skype for Business Server. Default feature settings are applied to all your phones.
- Deploy UC Software from a Provisioning Server. This method requires you to set up your own provisioning server. Setting up your own provisioning server enables you to customize feature settings using the template configuration files included in the UC Software download. With this method, users can sign in with their credentials from the phone's interface.
- Set the Base Profile with device.* Parameters. Polycom recommends using device.* parameters to configure multiple devices and only if you are familiar with Polycom centralized provisioning and configuration files.

Set Up Polycom with Skype for Business Online and Microsoft® Exchange Online

Skype for Business Online and Microsoft Exchange Online provide applications and services including email and social networking, Exchange Server, SharePoint, Yammer, MS Office web applications, and Microsoft Office software. Polycom offers Skype for Business Online and Exchange Online for:

- Polycom Trio 8800 and 8500 systems
- VVX 201, 300/310, 301/311, 400/410, 401/411, 500/501, and 600/601 business media phones

If you need to configure media ports for Skype for Business Online deployments, see Skype for Business Online for specific port numbers.

When using Skype for Business Online and Microsoft Exchange Online, note the following:

- You must use TLS-DSK to authenticate Polycom phones
- Polycom phones support use of ZTP staging for software upgrades

You can configure and manage VVX business media phones from the Office 365 online interface without the need for a separate provisioning server. After you set up phones, the first time users log in to a phone, users are prompted by a menu to set the time zone.

To set up Exchange online:

1. Install and open the Skype for Business Online, Windows Powershell Module.
2. Type the command `Import-Module SkypeOnlineConnector`.
3. Connect to the Skype for Business tenancy using the command

```
$session=New-CsOnlineSession -Credential $cred
```

4 When the Powershell credential request dialog displays, enter your Skype for Business user name and password.

5 Import the session with the command

```
Import-PSSession $session -Verbose -AllowClobber
```

6 Set policies with the command `CsIPPhonePolicies`.

Deploy UC Software from Skype for Business Server

If you downloaded UC Software files in CAB format, complete the following procedure to deploy UC Software from Skype for Business Server.

To deploy UC Software from Skype for Business Server:

1 Download and save UC Software in CAB file format to your computer.

You can obtain UC Software for Polycom Trio on Polycom Trio Support.

2 Go to Skype for Business Server and copy the CAB file to a C: drive directory.

3 Use the Skype for Business Server Management Shell to go to a particular directory.

4 In the Skype for Business Server Management Shell, run the following import command:

```
Import-CsDeviceUpdate -Identity service:1-WebServices-1 -FileName UCUpdates.cab
```

5 In the Skype for Business Control Panel, go to **Clients > Device Update** to view UC Software versions available on Skype for Business Server.

6 Go to **Clients > Action > Approve** to approve the UC Software.

Deploy UC Software from a Provisioning Server

Setting up your own provisioning server enables you to customize feature settings using the template configuration files included in the UC Software download. All configuration files are saved in compressed ZIP file format and you must unzip (extract) the files before use.

To deploy UC Software from a provisioning server:

1 Locate the following Skype for Business configuration files templates on Polycom Trio Support:

- `SkypeTrioSharedExample.cfg`. This template file contains all default settings you need to register with Skype for Business Server.
- `000000000000.cfg`. This is the master configuration file. In the CONFIG_FILES field, enter the names of all the configuration files containing settings you want to apply to the phones.

2 Place these configuration files in your root provisioning directory, create a copy of each file, and rename them keeping the suffix `.cfg`.

Using edited copies of the template files ensures that you have unedited template files containing the default values.

3 If you are manually installing a root CA security certificate, go to step 4. If not, go to step 5.

4 Open your renamed file `SkypeTrioSharedExample.cfg`. If you are manually configuring a root CA certificate, configure the following two parameters:

- Enter the root CA certificate, in Base64 format, in `sec.TLS.customCaCert.1`.

- Set the application profile in `sec.TLS.profileSelection.SIP`.

5 Open the master configuration file `000000000000.cfg`. In the CONFIG_FILES field, enter the name of your Skype for Business configuration file and save.

  Ensure that multiple configuration file names are comma separated.

  Configuration files you enter in the CONFIG_FILES field are read left to right. If you have configured the same setting in two configuration files, the setting listed first (left) is applied. Ensure that you do not have the same parameter in more than one configuration file.

  If you do not want to use the Microsoft Autodiscover service, use the following parameters to disable the feature and manually set the Skype for Business server address and SIP signaling port using:

  - Disable Autodiscover: `reg.1.serverAutoDiscovery=0`
  - Server: `reg.1.server.1.address=<server_address>`
  - Port: `reg.1.server.1.port=<port_number>`

6 Power on your phones. Your phones display the Skype for Business Sign In screen.

Set the Base Profile with device.* Parameters

This section shows you how to provision multiple devices using parameters in the `device.cfg` template configuration file included in your UC Software download. Polycom recommends using device.* parameters to configure multiple devices and only if you are familiar with centralized provisioning and configuration files.

To set the Base Profile using device.* parameters:

1 Locate the `device.cfg` template configuration file and place the `device.cfg` file on your provisioning server.

2 Locate and change the values of the following parameters:

  - `device.baseProfile=<Base Profile value>`
  - `device.set=1`
  - `device.baseProfile.set=1`

3 Rename and save the file.

4 Power on the phones.

5 Once boot-up is complete, remove `device.set` from the template configuration file and save the file again after removing `device.set`.

Configuring In-Band Provisioning Settings

Skype for Business in-band provisioning device settings take precedence over the same settings configured on-premises. To avoid configuration conflicts, ensure that the following parameters are applied to phones from one source or the other. If you are provisioning in-band, remove these parameters from your on-premises configuration. If you are provisioning on-premises, it is best practice to disable (block) these in-band provisioning device settings.

Use the parameter `lync.provisionDeviceParams.enabled=0` to disable the following in-band provisioning device settings sent from the Skype for Business Server:

- `EnableDeviceUpdate`

- `IPPhoneAdminPasswd`
- `LocalProvisioningServerAddress`
- `LocalProvisioningServerUser`
- `LocalProvisioningServerPassword`
- `LocalProvisioningServerType`

In-band Provisioning Device Settings

Parameter	Permitted Values
`lync.provisionDeviceParams.enabled`	1 (default) - Enable (accept) in-band provisioning device settings sent from Skype for Business. 0 - Disable (block) in-band provisioning device settings sent from Skype for Business.

Provision and Update with a USB Device

You can store configuration files and settings on a USB flash memory device and provision or update Polycom Trio 8800, 8500, or Polycom Trio Visual+ during normal functioning or in recovery mode. Recovery mode enables you to recover the Polycom Trio systems or Polycom Trio Visual+ to a normal provisioning state when other methods are not working or not available.

Note: The Polycom Trio system automatically provisions and updates a connected and paired Polycom Trio Visual+ system. You can, however, provision and update the Polycom Trio Visual+ separately, for example, if you need to support IEEE 802.1x or provision on networks without DHCP.

Polycom Trio 8800 and 8500 support only File Allocation Table (FAT) file systems and Polycom recommends using FAT32.

If other USB devices are attached to Polycom Trio system, you must remove them and ensure that Polycom Trio system correctly recognizes the USB device you want to install from.

If you use a USB device to provision while centralized provisioning server is in use, the USB configuration files override server settings. When you remove the USB device, the device returns to settings you configured on the server. Note, however, that the original server settings are subject to direct.set changes initiated by the USB device. The direct.set changes can alter parameters on the provisioning server and change basic provisioning settings.

When you attach a USB device, you are prompted for the administrator password (default 456). The Polycom Trio system downloads and installs the configuration files and you can remove the USB when complete.

Provision or Update Software Manually with a USB Device

You can manually provision the Polycom Trio system, one at a time, with a USB during normal phone functioning.

To provision or update software manually with a USB device:

1. Format a USB flash drive as FAT32. Polycom recommends that you use a USB 2.0 flash drive.

   If you are using a drive that is already formatted, ensure that previous files are deleted from the flash drive.
2. Download the UC Software from Polycom Trio on Polycom Support.
3. Copy the configuration files you want to use to the root of the USB device. The minimum required configuration files are as follows:
   - Master configuration file: `000000000000.cfg`
   - Polycom Trio 8800: `3111-65290-001.sip.ld`
   - Polycom Trio 8500: `3111-66700-001.sip.ld`

   If you are using the Polycom Trio Visual+, do one of the following:
   - Rename the Polycom Trio 8500 or 8800 `sip.ld` file to use the Polycom Trio Visual+ part number: `3111-66420-001.sip.ld.`
   - Rename the Polycom Trio 8500 or 8800 `3111-65290-001.sip.ld` file to “`sip.ld`” (delete “`3111-65290-001.`”) to use the same file for the Polycom Trio system and Polycom Trio Visual+.
4. Insert the USB to the Polycom Trio 8500, 8800, or Polycom Trio Visual+, follow the prompt for the Administrator password, and power cycle the device. Allow time for the devices to reboot.

Place the Polycom Trio 8800 or 8500 System into Recovery Mode

You can place the Polycom Trio 8800 or 8500 system into recovery mode when you want to provision with a USB and the provisioning process is not working during normal phone functioning.

To place Polycom Trio into recovery mode:

1. Ensure that the phone is powered off.
2. Plug in a USB device.
3. Power up the phone.
4. When the Polycom logo displays, press and hold with four fingers the four corners of the LCD screen until the LEDs blink. (Blinking rotates between orange/red/green/off).
5. Remove fingers from the LCD screen. Recovery process is complete when the device reboots.

Place Polycom Trio Visual+ into Recovery Mode

You can place the Polycom Trio Visual+ into recovery mode when you want to provision with a USB and the provisioning process is not working during normal phone functioning.

To place the Polycom Trio Visual+ into recovery mode:

1. Ensure that the phone is powered off.
2. Plug in a USB device.

3. Power up the phone.
4. When the LED initially turns from on to off, press and hold the pairing button until the pairing LED turns orange and release the button. The pairing LED blinks. (Blinking rotates between orange/red/green/off).

   Recovery process is complete when the device reboots.

Audio Features

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

Polycom NoiseBlock™

Polycom NoiseBlock technology automatically mutes the microphone during video calls when a user stops speaking, silencing noises, such as paper shuffling, food wrappers, and keyboard typing that interrupt conversations. When a user speaks, the microphone is automatically unmuted.

The Polycom NoiseBlock feature is now enabled by default on Polycom Trio systems. Polycom NoiseBlock is disabled and not available when the Polycom Trio system Base Profile is set to `SkypeUSB`.

Configuring Polycom NoiseBlock

The following parameters configure the Polycom NoiseBlock feature.

Polycom NoiseBlock Parameters

Parameter Template	Permitted Values
`voice.ns.hf.blocker` new.cfg	1 (default) - Enable the NoiseBlock feature. 0 - Disable the NoiseBlock feature.

Microphone Mute

You can configure the Polycom 8800 and 8500 systems to play an audible tone when the mute status of the device is changed either from any of the mute buttons of the system (device and any connected devices) or far-end (remote mute) system. This allows to know if the microphone of the system are in mute or un-mute state. In addition, you can set a periodic reminder which plays a tone periodically when the phone is in mute state. The time interval can be set using configuration parameter and the value must not be less than five seconds.

Microphone Mute Parameters

The following parameters configure microphone mute status alert tones.

Mute Status Alert Tone Parameters

Template	Parameter	Permitted Description	Change Causes Restart or Reboot
`features.cfg`	`se.touchFeedback.enabled`	0 - Does not play an alert tone when the mute status is changed on the Polycom 8800 or 8500 system. 1 - An alert tone is played when the mute status is changed either from the Polycom 8800, 8500, or far-end system.	No
`features.cfg`	`call.mute.reminder.period`	The time interval in seconds to play an alert tone periodically when the Polycom 8800 or 8500 system is in the mute state. 5 (default) 5 - 3600	No

Audio Output Options

Polycom Trio 8500 and 8800 offer audio output and routing options.

By default, audio plays out on the Polycom Trio 8800 and 8500 system speaker. When you add video capability by connecting and pairing the Polycom Trio 8500 or 8800 with a Polycom Trio Visual+ system, you can choose to play out audio on connected external speaker and/or the TV/monitor speakers. You can choose audio output options using the parameter `up.audio.networkedDevicePlayout`.

You can configure the following audio routing options:

- Polycom Trio 8800 or 8500 speaker only
- Polycom Trio™ Expansion Microphones

  The expansion microphones include a 2.1 m | 7 ft cable that you can attach directly to the Polycom Trio 8500 or 8800 to broaden its audio range to a total of 70 ft.
- Polycom Trio Visual+ using HDMI or a connected 3.5mm analog output
- Any combination of outputs available with the Polycom Trio system and Polycom Trio Visual+

Use the parameters in the following table to choose where audio is routed to for audio and video calls.

Configure Audio Output

Use the parameters in the following table to route audio.

Audio Output Parameters

Parameter Template	Permitted Values
`up.audio.networkedDevicePlayout` new.cfg	PhoneOnly (default) - Audio plays out on the Polycom Trio 8800 speakers. TvOnly - Audio plays out on the TV/monitor speakers connected by HDMI to a paired Polycom Trio Visual+ and, if connected, external speakers connected to the 3.5mm port of a paired Polycom Trio Visual+. Auto - Audio-only calls play out on the Polycom Trio system speakers. Video-call audio plays out on the TV/monitor speakers connected by HDMI to a paired Polycom Trio Visual+ and, if connected, external speakers connected to the 3.5mm port of a paired Polycom Trio Visual+.

USB Audio Calls

You can use Polycom Trio 8800 or 8500 as an audio device for your tablet or laptop by connecting your device to the Polycom Trio system with the USB cable supplied in the box.

When you connect a Microsoft® Windows® computer to the Polycom Trio system using a USB cable, you can control the volume of audio and video calls from the computer or Polycom Trio system, and the volume on both devices is synchronized.

You can use the Polycom Trio 8800 or 8500 system as an audio speakerphone when connected by USB to Mac computers running one of the following software versions:

- macOS 10.9.x (Mavericks)
- macOS 10.10.x (Yosemite)
- macOS 10.11.x (El Capitan)

Configuring USB Calls

You can configure settings for USB calls.

USB Call Parameters

Parameter **Template**	**Permitted Values**
`device.baseProfile` device.cfg	Generic - Disables the Skype for Business graphic interface. Lync - Use this Base Profile for Skype for Business deployments. SkypeUSB - Use this Base Profile when you want to connect Polycom Trio to a Microsoft Room System or a Microsoft Surface Hub.
`voice.usb.holdResume.enable` feature.cfg	0 (default) - The Hold and Resume buttons do not display during USB calls. 1 - The Hold and Resume buttons display during USB calls. This parameter applies only when Polycom Trio Base Profile is set to ‘SkypeUSB’.

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		0 to 27		No
	voice.codecPref.G711_A		7	
	voice.codecPref.G711_Mu		6	
	voice.codecPref.G719.32kbps		0	
	voice.codecPref.G719.48kbps		0	
	voice.codecPref.G719.64kbps		0	
	voice.codecPref.G722		4	
	voice.codecPref.G7221.24kbps		0	
	voice.codecPref.G7221.32kbps		0	
	voice.codecPref.G7221_C.24kbps		5	
	voice.codecPref.G7221_C.32kbps		0	
	voice.codecPref.G7221_C.48kbps		2	
	voice.codecPref.G729_AB		8	
	voice.codecPref.iLBC.13_33kbps		0	
	voice.codecPref.iLBC.15_2kbps		0	
	voice.codecPref.Lin16.8ksps		0	
	voice.codecPref.Lin16.16ksps		0	
	voice.codecPref.Lin16.32ksps		0	
	voice.codecPref.Lin16.44_1ksps		0	
	voice.codecPref.Lin16.48ksps		0	
	voice.codecPref.Siren7.16kbps		0	
	voice.codecPref.Siren7.24kbps		0	
	voice.codecPref.Siren7.32kbps		0	
	voice.codecPref.Siren14.24kbps		0	
	voice.codecPref.Siren14.32kbps		0	
	voice.codecPref.Siren14.48kbps		3	
	voice.codecPref.Siren22.32kbps		0	
	voice.codecPref.Siren22.48kbps		0	
	voice.codecPref.Siren22.64kbps		1	
	voice.codecPref.SILK.8ksps		0	
	voice.codecPref.SILK.12ksps		0	
	voice.codecPref.SILK.16ksps		0	
	voice.codecPref.SILK.24ksps		0	

SILK Audio Codec Parameters

The VVX 501 and 601 business media phones support the following SILK audio codec parameters.

SILK Audio Codec Parameters

Template	Parameters	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 (kbps/s) for the supported SILK sample rate. 20 kbps (default) 6 - 20 kbps	No
`site.cfg`	voice.audioProfile.SILK.12ksps.encMaxAvgBitrateKbps	Set the maximum average encoder output bitrate in kilobits per second (kbps/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 (kbps/s) for the supported SILK sample rate. 30 kbps (default) 8 - 30 kbps	No

SILK Audio Codec Parameters

Template	Parameters	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	voice.audioProfile.SILK.24ksps.encMaxAvgBitrateKbps	Set the maximum average encoder output bitrate in kilobits per second (kbps/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
`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

SILK Audio Codec Parameters

Template	Parameters	Permitted Values	Change Causes Restart or Reboot
`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

Polycom Trio System Audio Codec Support with Skype for Business

The following tables detail Polycom Trio system audio codec support and priority when registered with Skype for Business.

Phone	***Supported Audio Codec***	***Priority***
Polycom Trio systems	SILK	0
	Siren 7 (16 kbps)	0
	G.722.1 (24 kbps)	0
	G.722	4
	G.711 µ-law	6
	G.711a-law	7

The following table summarizes the audio codecs supported on Polycom Trio systems with Skype for Business.

Algorithm	***Reference***	***Raw Bit Rate***	***Maximum IP Bit Rate***	***Sample Rate***	***Default Payload Size***	***Effective Audio Bandwidth***
G.722.1	RFC 3047	24 Kbps	40 Kbps	16 Ksps	20 ms	7 KHz
Siren 7	SIREN7	16 Kbps	32 Kbps	16 Ksps	20 ms	7 KHz
G.722[1]	RFC 3551	64 Kbps	80 Kbps	16 Ksps	20 ms	7 KHz

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.711a-law	RFC 1890	64 Kbps	80 Kbps	8 Ksps	20 ms	3.5 KHz
SILK	SILK	6 - 20 Kbps 7 - 25 Kbps 8 - 30 Kbps 12 - 40 Kbps	36 Kbps 41 Kbps 46 Kbps 56 Kbps	8 Ksps 12 Ksps 16 Ksps 24 Ksps	20 ms	3.5 KHz 5.2 KHz 7 KHz 11 KHz

[1] Per RFC 3551. Even though the actual sampling rate for G.722 audio is 16,000 Hz (16 Ksps), 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.

Video Features

After you set up Polycom phones on your network with the default configuration, users can place and answer video calls, if supported. This section provides information on making custom configurations to optimize video calling for Polycom phones.

Video and Camera Options

By default, at the start of a video call, the Polycom Trio 8500 or 8800 paired with a Polycom Trio Visual+ with the Logitech C930e camera transmits an RTP encapsulated video stream from the local camera. Users can stop and start video by pressing the 'Stop My Video' and 'Start My Video' buttons. When users stop video during a video call, video is reset and displays again at the start of the next video call.

You can use the parameters in the following sections to configure video transmission, the video and local camera view, and video camera options.

Configuring Video Transmission

Use the parameters in the following table to configure video transmission.

Video Transmission Parameters

Parameter Template	Permitted Values
`video.quality` video.cfg	The optimal quality for video that is sent in a call or a conference. Motion — for outgoing video that has motion or movement. Sharpness — for outgoing video that has little or no movement. NULL (default) — 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.
`video.autoFullScreen` video.cfg	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
`video.callRate` video.cfg	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

Video Transmission Parameters

Parameter Template	Permitted Values
`video.forceRtcpVideoCodecControl` video.cfg	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.
`video.maxCallRate` video.cfg	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

Configuring the Video and Camera View

Use the parameters in the following table to set the video and local camera view settings.

Video and Camera View Parameters

Parameter Template	Permitted Values
`call.singleKeyPressCameraControls` site.cfg	1 (default) - Tapping Camera in the call view directly shows the Camera Controls menu for EagleEye MSR camera. 0 - Tapping Camera in the call view shows a menu with camera preferences, presets, and camera controls menu items for EagleEye MSR camera.
`homeScreen.camera.enable` features.cfg	Applies to the EagleEye MSR camera. 0 (default) - A Camera menu item is shown on the main menu. 1 - A Camera menu item displays on the Home Screen allowing users to pan, tilt or zoom.
`up.arrow.repeatDelay` features.cfg	Choose the milliseconds (ms) an arrow button must be held before the arrow starts repeating in the Camera Controls menu for EagleEye MSR camera. 500 ms (default) 100 – 5000 ms

Video and Camera View Parameters

Parameter Template	Permitted Values
`up.arrow.repeatRate` features.cfg	Choose the milliseconds (ms) between repeated simulated presses while an arrow button is being held down. This applies to the arrows in the Camera Controls menu for EagleEye MSR camera. 80 ms (default) 50 - 2000 ms
`video.camera.controlStyle` features.cfg	Controls whether EagleEye MSR camera pan and tilt is controlled by directional arrow buttons or separate pan/tilt sliders. Simple (default)
`video.camera.invertPanControl` video.cfg	Use to correct the color of video captured by the EagleEye MSR camera. 0 (default) 1000
`video.camera.preset.x.label` features.cfg	Enter a label for the EagleEye MSR camera preset. String 0 – 12 characters
`video.camera.preset.x.pan` features.cfg	Set the pan for the EagleEye MSR camera presets, where x equals the preset. 0 (default) 0 - 1000
`video.camera.preset.x.tilt` features.cfg	Set the tilt for the EagleEye MSR camera presets, where x equals the preset. 0 (default) 0 - 1000
`video.camera.preset.x.zoom` features.cfg	Set the zoom for the EagleEye MSR camera presets, where x equals the preset. 0 (default) 0 - 1000
`video.localCameraView.callState` video.cfg	Applies to the EagleEye MSR camera and Logitech C930e webcam. This parameter applies only when `video.localCameraView.userControl` is set to PerSession or Hidden. 1 (default) - The local camera view displays on the Polycom Trio Visual+ monitor. 0 - The local camera view does not display on the Polycom Trio Visual+ monitor.

Video and Camera View Parameters

Parameter Template	Permitted Values
`video.localCameraView.fullscreen.enabled` video.cfg	Applies to the EagleEye MSR camera and Logitech C930e webcam. Determines whether the local camera view is shown in the full screen layout. 1 (default) — The local camera view is shown. 0 — The local camera view is not shown.
`video.localCameraView.fullscreen.mode` video.cfg	Applies to the EagleEye MSR camera and Logitech C930e webcam. Determines how the local camera view is shown. Side-by-side (default) — The local camera view displays side-by-side with the far end window. PIP — The local camera view displays as a picture-in-picture with the far end window
`video.localCameraView.idleState` video.cfg	Applies to the EagleEye MSR camera and Logitech C930e webcam. This parameter applies only when `video.localCameraView.userControl` is set to PerSession or Hidden. 1 (default) - The local camera view displays on the Polycom Trio Visual+ monitor. 0 - The local camera view does not display on the Polycom Trio Visual+ monitor.
`video.localCameraView.userControl` video.cfg	Applies to the EagleEye MSR camera and Logitech C930e webcam. Persistent (default) - The local camera view user setting is available in the phone menu and overrides the default you specify with `video.localCameraView.fullScreen.enabled`. PerSession: The local camera view user setting is available in the phone menu and overrides the default you specify with video.localCameraView.callState on a per-call basis. Changes the user makes from the phone menu revert to the default specified by `video.localCameraView.idleState` after the phone returns to the idle state. Hidden: The user control in the phone menu to show or hide the self view is not available.
`video.screenModeFS` video.cfg	Specify the view of the video window in full screen viewing mode. normal (default)

Video and Camera View Parameters

Parameter Template	Permitted Values
`video.screenMode` video.cfg	Specify the view of the video window in normal viewing mode. normal (default) full crop

Video Camera Parameters

Use the parameters in the following table to configure the video camera options.

Video Camera Parameters

Parameter Template	Permitted Values
`video.camera.autoWhiteBalance` video.cfg	For the EagleEye MSR camera and Logitech C930e webcam. 1 (default) – Auto white balance is enabled and the value of the video.camera.whiteBalance parameter is not used. 0 – Auto white balance is disabled, so the value of the video.camera.whiteBalance is used for white balance.
`video.camera.backlightCompensation` video.cfg	0 (default) – Disable EagleEye MSR camera backlight compensation. 1 - Enable EagleEye MSR camera backlight compensation.
`video.camera.brightness` video.cfg	Sets the brightness level of video captured by the EagleEye MSR camera and Logitech C930e webcam. The value range is from 0 (Dimmest) to 1000 (Brightest). NULL (default) - Take the default value from the attached camera device. 0 - 1000
`video.camera.contrast` video.cfg	Sets the contrast level of video captured by the EagleEye MSR camera and Logitech C930e webcam. The value range is from 0 (no contrast increase) to 3 (most contrast increase), and 4 (noise reduction contrast). NULL (default) - Take the default value from the attached camera device. 0 - 1000
`video.camera.flickerAvoidanced` video.cfg	Sets the flicker avoidance for EagleEye MSR camera and Logitech C930e webcam. 0 (default) — flicker avoidance is automatic. 1 — 50hz AC power frequency flicker avoidance (Europe/Asia). 2 — 60hz AC power frequency flicker avoidance (North America).

Video Camera Parameters

Parameter Template	Permitted Values
`video.camera.frameRate` video.cfg	Sets the target frame rate (frames per second). 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.
`video.camera.gamma` video.cfg	Set the factor to use for gamma correction applied to each frame of video captured by the EagleEye MSR camera. You can use this setting to correct for video that appears too dark or too light. 0 (default) 1000
`video.camera.hue` video.cfg	Use to correct the color of video captured by the EagleEye MSR camera. 0 (default) 1000
`video.camera.menuLocation` features.cfg	Specify if camera settings display under the Advanced menu for administrators or the Basic menu for users. Camera settings displayed in the menu apply to the EagleEye MSR camera and Logitech C930e webcam. Basic (default) Advanced
`video.camera.saturation` video.cfg	Sets the saturation level of video captured by the EagleEye MSR camera and Logitech C930e webcam. NULL (default) - Take the default value from the attached camera device. 0 - 1000
`video.camera.sharpnes` video.cfg	Sets the sharpness level of video captured by the EagleEye MSR camera and Logitech C930e webcam. NULL (default) - Take the default value from the attached camera device. 0 - 1000
`video.camera.whiteBalance` video.cfg	Use to correct the white balance tint of video captured by the EagleEye MSR camera and Logitech C930e webcam. NULL (default) - Take the default value from the attached camera device. 0 - 1000

Toggling Between Audio-only or Audio-Video Calls

When this feature is enabled on the Polycom Trio 8500 or 8800 system using Polycom Trio Visual+ video capabilities, you can toggle calls between audio-only or audio-video.

This feature applies only to outbound calls from your phone; incoming video calls to your phone are answered using video even when you set the feature to use audio-only.

When the phone is registered with Skype for Business, you can:

- Escalate a point-to-point Skype for Business content session to a Skype for Business conference call by adding a participant.
- Use `video.callMode.default` to set the initial call to audio-video or audio only. By default, calls begin as audio-video. If you set this parameter to audio, users can press a button on the Polycom Trio to add 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.

Configuring Audio-only or Audio-Video Calls

The following parameters configure whether the phone starts a call with audio and video.

Audio/Video Toggle Parameters

Parameter Template	Permitted Values
`up.homeScreen.audioCall.enabled` features.cfg	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.
`video.autoStartVideoTx` video.cfg	1 (default) - Automatically begin video to the far side when you start a call. 0 - Video to the far side does not begin.
`video.callMode.default` video.cfg	**Polycom Trio** Allow the user to begin calls as audio-only or with video. video (default) - Set the initial call to audio and video. audio - Set the initial call to audio only and video may be added during a call. On Polycom Trio, you can combine this parameter with `video.autoStartVideoTx`.

Video-based Screen Sharing Support for Polycom Trio Solution

Polycom Trio 8500 and 8800 allows you to use Video-based Screen Sharing (VbSS) with Skype for Business clients that enables both application and desktop sharing. In previous releases, Trio systems supported only Remote Desktop Protocol (RDP) for receiving content. Only systems registered to Skype for Business support VbSS content sharing.

Note: The Polycom Trio solution can only receive Skype for Business VbSS-based content. You cannot transmit VbSS-based content from the Polycom Trio solution.

The advantages of VbSS content sharing over RDP are as follows:

- The video experience is faster, with an improvement in frames-per-second.
- Works better in low bandwidth conditions, even when receiving high motion content, such as 3-D graphics.

However, if any participant in a Skype for Business conference does not support VbSS, the Skype for Business server content switches from VbSS to Remote Desktop Protocol (RDP) content.

Video-based Screen Sharing Parameters

The following table lists parameter that configure Video-based Screen Sharing with the Polycom Trio solution.

Video-based Screen Sharing Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`content.vbss.enable`	1 (default) - Polycom 8500 and 8800 systems use Video-based Screen Sharing (VbSS) to receive Skype for Business content. 0 - Polycom 8500 and 8800 systems use Remote Desktop Protocol (RDP) to receive Skype for Business content.	No

Skype for Business Video Layouts on Polycom Trio System

When using the Polycom Trio 8500 or 8800 system with Polycom Trio Visual+ connected to a monitor, you can set how participants and content display during video calls. The Polycom Trio supports the Gallery View layout for video and content during video calls in standard H.264 video meetings or point-to-point calls.

Configuring Skype for Business Video Layouts

The following parameters configure video layouts on the Polycom Trio system registered with Skype for Business.

Video Layout Parameters

Parameter Template	Permitted Values
`video.conf.displayLayout.gallery.allowContent` new.cfg	1 (default) - Enable Gallery View layout for video and content. Content is scaled to fit into the 720p window of a gallery window. 0 - Disable Galley View layout. Content displays in a full screen window.
`video.conf.displayLayout.PIP.peopleMode` new.cfg	Choose what the PIP screen displays. selfView (default) - Display your own video. recentTalker - Display video from the current or most recent talker.
`video.conf.galleryView.overlayTimeout` new.cfg	Set the timer for the participant name overlay on the Visual+ monitor when using the Gallery View. 0 (default) - The overlay does not time out. 0 - 60000 ms

Forward Error Correction

The Polycom Trio system supports Forward Error Correction (FEC) DV0 and DV1 with Skype for Business Server 2015, Skype for Business 2015 client, and Lync 2013 environments for H.264 SVC. The scheme introduces recovery packets on the transmitter which recover lost video packets on the receiver.

FEC performance and quality improvements with this release may vary depending on network conditions.

Use the parameter `video.codecPref.XUlpFecUC` to set the forward error correction codec priority.

Forward Error Correction Parameters

Parameter	Permitted Values
`video.codecPref.XUlpFecUC` video.cfg	Set the forward error correction (FEC) codec priority. 5 (default) 0 - 7

Video Simulcast

Polycom Trio systems registered Skype for Business can simultaneously send a low resolution video stream and a second higher-resolution video stream to conference participants in a Skype for Business AVMCU meeting. Simulcast is enabled by default.

Skype for Business AVMCU-based video meetings are driven by endpoint requests to receive video called a video source request (VSR). The VSR specifies the resolution (among other constraints) and the participant(s) whose video the endpoint would like to display. The requested resolution in Skype for Business client video calls is based on the size of the video window and new VSRs are sent when the size of the window changes.

Configure Video Resolution from the Phone

You can configure video resolution from the phone.

To configure video resolution from the phone:

» On the Polycom Trio system, go to **Settings > Basic > Video > Video Call Settings > Centralized Conferencing Profile**.

Configure Video Resolution using Centralized Provisioning

You can configure video resolution using centralized provisioning parameters.

Video Resolution Parameters

Parameter Template	Permitted Values
`video.callRate` video.cfg	The default call rate (in kbps) to use when initially negotiating bandwidth for a video call. 2048 (default) 128 - 6144
`video.camera.flickerAvoidance` video.cfg	0 (default) Camera flicker avoidance is automatic. 1 - Optimize camera flicker avoidance for 50Hz AC power frequency environments (Europe/Asia). 2 - Optimize camera flicker avoidance for 60Hz AC power frequency environments (North America).
`video.conf.profile` video.cfg	Set the video resolution requested for the large video window in all layouts. 540p (default) 1080p 720p 360p 240p 180p

Supported Video Codecs

Use the optional Polycom Trio Visual+ and Logitech C930E USB webcam to add video to your Polycom Trio calls. Polycom supports the following video standards and codecs:

- H.264 advanced video coding (AVC) Baseline Profile and High Profile
- Lync 2013 and Skype for Business H.264 scalable video coding (SVC) (X-H264UC)

The following table lists video codec parameters and specifies the video codec preferences for the Polycom Trio 8800 system. To disable codecs, set the value to 0. A value of 1 indicates the codec is the most preferred and has highest priority.

Video Codec Parameters

`Parameter` Template	`Permitted Values`
`video.codecPref.H264` video.cfg	Set the H.264 video codec preference priority. 6 (default) 0 - 8
`video.codecPref.H264HP` video.cfg	Set the H.264 High Profile video codec preference priority. 2 (default) - Indicates the codec is the most preferred and has highest priority. 0 - 8

Video Codec Parameters

Parameter Template	Permitted Values
`video.codecPref.H264HP.packetizationMode0` video.cfg	5 (default) 0 - 8
`video.codecPref.Xdata` video.cfg	7 (default) 0 - 5 Set the Remote Desktop Protocol (RDP) codec preference priority. 1 indicates the codec is the most preferred and has highest priority. The value 4 is for Skype Base Profile.
`video.codecPref.XH264UC` video.cfg	Set the Microsoft H.264 UC video codec preference priority. 1 (default) 0 - 8
`video.codecPref.XUlpFecUC` video.cfg	Set the forward error correction (FEC) codec priority. 8 (default) 0 - 7
`video.conf.addVideoWhenAvailable` new.cfg	0 (default) 1 - When Polycom Trio system is added to a conference by another participant via digit dialing, the Trio system adds video if video is available on the conference.
`video.conf.profile` video.cfg	Configure the video media profile request sent to the video conference server. 3 (default) - Identifies the 540p profile. 2 - 7
`video.conf.simulcast.enabled` video.cfg	1 (default) - Allow sending two different video media resolutions on a Skype for Business video conference call.
`video.profile.H264.payloadType.packetizationMode` video.cfg	99 (default) 0 - 127 Set the H.264 payload type with packetization mode set to 0.
`video.profile.H264.payloadType.packetizationMode1` video.cfg	109 (default) 0 - 127 Set the H.264 payload type with packetization mode set to 1.
`video.profile.H264HP.jitterBufferMax` video.cfg	2000 (default) 533 - 2500
`video.profile.H264HP.jitterBufferMin` video.cfg	150 (default) 33 - 1000
`video.profile.H264HP.jitterBufferShrink` video.cfg	70 (default) 33 - 1000

Video Codec Parameters

Parameter Template	Permitted Values
`video.profile.H264HP.payloadType` video.cfg	Specify the RTP payload format type for H264/90000 MIME type (High Profile). 100 (default) 0 - 127
`video.profile.H264HP.payloadType.packetizationMode1` video.cfg	100 (default) 0 - 127 Set the H.264 high profile payload type with packetization mode set to 1.
`video.profile.H264HP.profileLevel` video.cfg	4.1 (default) String (1 - 5 characters)
`video.profile.H264M.payloadType.packetizationMode0` video.cfg	113 (default) 0 - 127 Set the H.264 high profile payload type with packetization mode set to 0.
`video.profile.Xdata.payloadType` video.cfg	Specify the RTP payload format type for x-data/90000 MIME type. This parameter is for Remote Desktop Protocol (RDP) content sharing. 127 (default) 0 - 127
`video.profile.XH264UC.jitterBufferMax` video.cfg	The largest jitter buffer depth to support. Jitter above this size always causes lost packets. This parameter should be set to the smallest possible value that supports the expected network jitter. 2000 (default) 533 - 2500
`video.profile.XH264UC.jitterBufferMin` video.cfg	The smallest jitter buffer depth that must be achieved before play out begins for the first time. After this depth has been achieved initially, the depth may fall below this point and play out still continues. This parameter should be set to the smallest possible value which is at least two packet payloads, and larger than the expected short term average jitter. 150 (default) 33 - 1000
`video.profile.XH264UC.jitterBufferShrink` video.cfg	70 (default) 33 - 1000
`video.profile.XH264UC.jitterBufferShrink` video.cfg	The minimum duration (in milliseconds) of Real-time Transport Protocol (RTP) packet Rx with no packet loss that will trigger jitter buffer size shrinks. Use smaller values to minimize the delay on known good networks. 70 (default) 33 - 1000

Video Codec Parameters

Parameter Template	Permitted Values
`video.profile.XH264UC.mstMode` video.cfg	Specify the multi-session transmission packetization mode. The value of NI-TC identifies non-interleaved combined timestamp and CS-DON mode. This value should not be modified for operation with other Skype for Business devices. NI-TC (default) string
`video.profile.XH264UC.payloadType` video.cfg	Specify the RTP payload format type for X-H264UC/90000 MIME type. 122 (default) 0 - 127
`video.profile.XH264UC.payloadType` video.cfg	RTP payload format type for H.264 MIME type. 122 (default) 0 - 127
`video.quality` video.cfg	The optimal quality for video that you send in a call or a conference. Note: If motion is not selected, moderate to heavy motion can cause some frames to be dropped. NULL (default) - Use sharpness or Null if your outgoing video has little or no movement. motion - Use motion if your outgoing video has motion or movement. sharpness - Use sharpness or Null if your outgoing video has little or no movement.
`video.rtcpbandwidthdetect.enable` video.cfg	0 (default) 1 - Polycom Trio uses an estimated bandwidth value from the RTCP message to control Tx/Rx video bps.

Content

Polycom offers several content sharing options.

Content Sharing

You can show content from a computer during in-person meetings, video conference calls, and point-to-point video calls on the Polycom Trio Visual+ system monitor when paired with a Polycom Trio 8500 or 8800 system. To share content:

- The Polycom Visual+ system must be paired with the Polycom Trio system
- The computer and Polycom Trio solution must be able to communicate on the same IP network

You can use the following Polycom applications to share content:

- Polycom® People+Content® (PPCIP)
- Polycom® Polycom® Desktop for Windows® or Mac®
- Polycom® Polycom® Mobile application

You can download People+Content IP and Polycom Desktop from Polycom Support and Polycom Mobile from your mobile application store.

For information about using PPCIP on the Polycom Trio solution registered with Skype for Business, see the *Polycom Trio - User Guide* at Polycom Trio on Polycom Support.

Important: The default port used by Group Paging when enabled conflicts with the UDP port 5001 used by Polycom® People+Content™ on the Polycom Trio system. Since the port used by People+Content is fixed and cannot be configured, configure one of the following workarounds:

- Configure a different port for Group Paging using parameter `ptt.port`.
- Disable People+Content IP using parameter `content.ppcipServer.enabled="0"`.

Configuring Content Sharing

Use the parameters in the following table to configure content sharing options.

To enable device pairing with the Polycom Trio solution, use the `smartpairing*` parameters. Note that People+Content IP does not support ultrasonic SmartPairing.

Content Sharing Parameters

Parameter Template	Permitted Values
`content.autoAccept.rdp` new.cfg	1 (default) - Content shown by far-end users is automatically accepted and displayed on the Polycom Trio solution. 0 - Near-end users are prompted to accept meeting content sent to Polycom Trio solution from a far-end user.
`content.bfcp.enabled` new.cfg	1 (default) - Enable content sharing by offering or accepting the Binary Floor Control Protocol (BFCP) in Session Description Protocol (SDP) negotiation during SIP calls. Does not apply to Skype for Business calls. 0 - Disable content sharing using BFCP.
`content.bfcp.port` sip-interop.cfg	15000 (default) - 0 - 65535 -
`content.bfcp.transport` features.cfg	UDP (default) - TCP -
`content.ppcipServer.enabled` new.cfg	1 (default) - Enable Polycom People+Content IP. 0 - Disable Polycom People+Content IP.
`content.ppcipServer.meetingPassword` site.cfg	NULL (default) - String (0 - 256 characters) -
`smartPairing.mode` features.cfg	Enables users with People+Content IP or Polycom Desktop on a computer or Polycom Mobile on a tablet to pair with the Polycom Trio conference phone using SmartPairing. Disabled (default) - Users cannot use SmartPairing to pair with the conference phone. Manual - Users must enter the IP address of the conference phone to pair with it.
`smartPairing.volume` features.cfg	The relative volume to use for the SmartPairing ultrasonic beacon. 6 (default) 0 - 10

Polycom People+Content IP over USB

You can use Polycom® People+Content® IP (PPCIP) to share video or data from a Windows® or Mac® computer connected by USB to the Polycom Trio system when in or out of a call. When you install PPCIP version 1.4.2 and run it unopened in the background, the PPCIP application pops up immediately when you connect the computer to Polycom Trio solution via USB.

Keep the following points in mind:

- Showing content with People+Content IP over USB provides content to a maximum of 1080p resolution on a connected Windows or Mac computer.
- Audio content is not shared.

- Content sent from People+Content is sent over USB, and no network connection is needed. This is useful for environments where guest IP access is not allowed. You can show content with People+Content IP on a computer connected by USB to Polycom Trio to a maximum of 1080p resolution on a Windows computer. You must use UC Software 5.4.3AA or later to share your desktop at up to 1080p resolution using a Mac computer connected by USB to the Polycom Trio solution.

Important: The default port used by Group Paging when enabled `ptt.pageMode.enable="1"` conflicts with the UDP port 5001 used by Polycom® People+Content™ on the Polycom Trio system. Since the port used by People+Content is fixed and cannot be configured, configure one of two workarounds:

- Configure a different port for Group Paging using parameter `ptt.port` or
- Disable People+Content IP using parameter `content.ppcipServer.enabled="0"`.

Configuring Polycom People+Content IP over USB

This following table lists parameters that configure the People+Content over USB feature.

Polycom People+Content Content Sharing Parameters

Parameter Template	Permitted Values
`feature.usb.device.content`	1 (default) - Enable content sharing using the People+Content IP application on a computer connected by USB to Polycom Trio solution. 0 - Disable content sharing using the People+Content IP application on a computer connected by USB to Polycom Trio solution.

Polycom People+Content IP

You can share content from a computer over IP using Polycom® Polycom® Desktop Software, Polycom® People+Content IP (PPCIP), and Polycom® Polycom® Mobile Software. Sharing content with Polycom People+Content IP from a computer connected over IP supports 1080p resolution with about five frames per second on the Polycom Visual+ monitor. The computer and Polycom Trio solution must be able to communicate on the same IP network and you must pair your Polycom software application with the Polycom Trio system.

When Polycom Trio is registered with Skype for Business, you can use these applications to share content only to a local monitor. You cannot share content from a Polycom Trio system over a Skype for Business call. For instructions, see the *Polycom Trio - User Guide* at Polycom Trio on Polycom Support.

Configuring Polycom People+Content IP for Polycom Trio Solution

The following table lists parameters that configure content sharing with the Polycom Trio solution.

Polycom People+Content IP Parameters

Parameter Template	Permitted Values
`content.autoAccept.rdp` new.cfg	1 (default) - Content shown by far-end users is automatically accepted and displayed on the Polycom Trio solution. 0 - Near-end users are prompted to accept meeting content sent to Polycom Trio solution from a far-end user.
`content.bfcp.port`	15000 (default) 0 - 65535
`content.bfcp.transport`	UDP (default TCP
`content.ppcipServer.enabled`	1 (default) - Enable Polycom People+Content IP content server. 0 - Disable Polycom People+Content IP content server.
`content.ppcipServer.enabled.Trio8500`	1 (default) - Enable Polycom People+Content IP content server for Polycom Trio. 0 - Disable Polycom People+Content IP content server for Polycom Trio.
`content.ppcipServer.enabled.Trio8800`	1 (default) - Enable Polycom People+Content IP content server. 0 - Disable Polycom People+Content IP content server.
`content.ppcipServer.meetingPassword`	NULL (default) String (0 - 256 characters)

Create Conference Room Accounts for Skype for Business

The Polycom Trio system enables you to use Remote Desktop Protocol (RDP) with Skype for Business clients, which enables both application and desktop sharing. To maximize the benefits of RDP content sharing, Polycom recommends creating a Skype for Business Room System or CsMeetingRoom account to allow sharing from in-room clients. When you create a conference room account, the Skype for Business Room System prompts content presenters to mute the microphone and speaker to avoid audio feedback.

Note: The Polycom Trio solution can only receive Skype for Business RDP-based content. You cannot transmit RDP-based content from the Polycom Trio solution.

Create a Skype for Business Room System Account

To create a Skype for Business Room System account, complete the following procedure and update your account name and server details on your Exchange Server Management Shell.

To create a Skype for Business Room System account:

1 Within your Exchange Management Shell, set the following:

➢ `New-Mailbox -Name 'Trio Room01' -Alias 'Trio.Room01' -UserPrincipalName 'Trio.Room01@domain.com' -SamAccountName 'Trio.Room01' -FirstName 'Trio' -Initials '' -LastName 'Room01' -Room`

➢ `Set-CalendarProcessing -Identity Trio.Room01 -AutomateProcessing AutoAccept -AddOrganizerToSubject $false -RemovePrivateProperty $false -DeleteSubject $false`

➢ `Set-Mailbox -Identity Trio.Room01@domain.com -MailTip "This room is equipped with a Polycom Trio 8800, please make it a Skype for Business Meeting to take advantage of the enhanced meeting experience."`

➢ `Set-ADAccountPassword -Identity Trio.Room01`

➢ `Enable-ADAccount -Identity Trio.Room01`

2 Within your Skype for Business Management Shell, set:

➢ `Enable-CsMeetingRoom -SipAddress "sip:Trio.Room01@domain" -domaincontroller dc.domain.local -RegistrarPool pool01.domain.local -Identity Trio.Room01`

Screen Mirroring

The Polycom Trio 8800 system provides screen mirroring locally from Apple® AirPlay®-certified devices and the Wireless Display feature for Miracast®-certified Android™ and Windows® devices.

Screen Mirroring with AirPlay Certified Devices

This section provides information you need to set up and configure the Polycom Trio 8800 system to work with AirPlay certified devices.

The following information applies to using AirPlay certified devices with the Polycom Trio 8800 system:

- You can display local content only from your AirPlay certified device to the Polycom Trio 8800 system monitor.
- Sharing content from direct streaming sources, such as YouTube™ or web links, is not supported.
- If you share content during a point-to-point or conference call, the content is not sent to far-end participants.
- Audio-only content is not supported. If you only want to share audio, consider using Bluetooth or USB connectivity.
- Apple Lossless Audio Codec (ALAC) is not supported.

The Polycom Trio 8800 system supports the following AirPlay certified devices:

- Apple® iPhone®
- iPad®
- iPad Pro™
- MacBook Pro®

The Polycom Trio 8800 system supports a maximum resolution and frame rate of 720p@60fps or 1080p@30fps. If configured for 1080p resolution, an iPad often sends 60fps video, which can result in latency in mirroring, visual artifacts, or both.

When the Polycom Trio 8800 system receives content from a Skype for Business client using the Remote Desktop Protocol (RDP) at the same time as content from an AirPlay certified device, the AirPlay content takes precedence and displays. When you end AirPlay content, available Skype for Business content displays.

Requirements

You must meet the following requirements to use the screen mirroring feature on an AirPlay certified device with the Polycom Trio 8800 system:

- Polycom Trio 8800 collaboration kit running UC Software version 5.4.4AA or later
- The Polycom Trio 8800 system and Apple devices are on the same subnet.

  The devices can be on different subnets if the devices are routable and multicast DNS (Bonjour) is bridged between the subnets for discovery. The devices can also be on different subnets if AirPlay Discovery over Bluetooth is enabled, the subnets are routable to each other, and the device is within Bluetooth range.
- The screen mirroring feature uses the following ports:
  - Discovery: UDP port 5353
  - Sessions: TCP ports 7000, 7100, 8009, and 47000; UDP port 1900

Polycom Trio 8800 for AirPlay Parameters

Use the following parameters to configure the Polycom Trio 8800 system for AirPlay certified devices.

Polycom Trio 8800 for AirPlay Parameters

Template	Parameter	Permitted Values	Restart Causes Restart or Reboot
features.cfg	content.airplayServer.authType	none (default) - No security code for AirPlay certified devices is required. passcode - Use a security code to authenticate AirPlay certified devices.	No
features.cfg	content.airplayServer.discovery.bluetooth.enabled	Set to allow Airplay discovery over Bluetooth. 1 (default) - Enables Polycom Trio 8800 to be discoverable to AirPlay-certified devices over Bluetooth. Turns Bluetooth radio on when feature.bluetooth.enabled=1. 0 - Polycom Trio 8800 is not discoverable to AirPlay-certified devices over Bluetooth. Note: To use this feature, enable the parameter feature.bluetooth.enabled.	No
features.cfg	content.airplayServer.enabled	0 (default) - Disable the content sink for AirPlay certified devices. 1 - Enable the content sink for AirPlay certified devices.	No
features.cfg	content.airplayServer.maxResolution	Set the content resolution. 720p (default) 1080p 1024x1024 960x960 480x480	No
features.cfg	content.airplayServer.name	Specify a system name for the local content sink for AirPlay certified devices. If left blank the previously configured or default system name is used. NULL (default)	No

Polycom Trio 8800 for AirPlay Parameters

Template	Parameter	Permitted Values	Restart Causes Restart or Reboot
`features.cfg`	`content.local.authChangeInterval`	Set the interval in minutes between changes to the local content authentication credentials. 1440 (default) 0 - 65535 0 - Do not change	No
`features.cfg`	`content.local.authChangeMode`	Specify when the security code for content sharing with AirPlay certified devices changes. session (default) - Code changes at the end of each content sharing session. relativeTime - Code changes at an interval specified by the `content.local.authChangeInterval` parameter.	No

Troubleshooting

This section provides solutions to common issues you may have using the Polycom Trio 8800 system with AirPlay certified devices.

The Polycom Trio 8800 system does not advertise on my device

The Polycom Trio may not be broadcasting for discovery, or the broadcasts are being blocked.

- Ensure your Apple device is on the same subnet as the Polycom Trio 8800 system and that Polycom Trio has screen mirroring enabled.

Configuring Debugging Logs

If you experience further issues using AirPlay certified devices with the Polycom Trio 8800 system, you can enable the following logging parameters on your Polycom Trio to get extended debugging data.

Screen Mirroring Debugging Parameters

Log Component	Permitted Values
airp	Session management and communication specifically for AirPlay certified devices.
airpl	Protocol library for AirPlay certified devices
airps	Android service AirPlay certified devices
lc	Local Content (including for AirPlay certified devices and PPCIP) session management

Screen Mirroring with Miracast-Certified Devices

The Wireless Display feature lets you display content locally from your Miracast-certified Android or Windows device to the Polycom Trio 8800 system monitor. Windows or Android devices can discover and connect directly with the Polycom Trio 8800 system and do not have to be on the same network.

The Polycom Trio 8800 system supports content sharing from the following Android and Windows devices:

- Miracast-certified devices running Windows 10
- Samsung Galaxy smartphones and tablets running Android version 4.4 or earlier

Polycom cannot guarantee connectivity with all Miracast-certified devices, but connectivity has been validated to work well with Samsung smartphones and tablets using Android version 4.4 or later and the Microsoft Surface® 3 Pro and Surface® 4 Pro running Windows 10.

To send content from your device, you must first connect your device wirelessly to the Polycom Trio 8800 system.

The Polycom Trio 8800 system can display content to a maximum resolution and frame rate of 720p@60fps or 1080p@30fps. If the Polycom Trio 8800 system is configured to auto-negotiate the frame rate of transmitted content, some tablets might send 1080p@60fps video, which can result in latency in mirroring, visual artifacts, or both.

Requirements

You must meet the following requirements to use the Wireless Display feature on a Miracast-certified device with the Polycom Trio 8800 system:

- Polycom Trio 8800 collaboration kit running UC Software version 5.4.4AA or later

If you do not allow auto-negotiation, some devices might fail to pick the best possible video stream parameters.

Configuring the Polycom Trio 8800 for Miracast-Certified Devices

Use the following parameters to configure Wireless Display on the Polycom Trio 8800 system.

Wireless Display Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	content.wirelessDisplay.sink.authorizationType	Auto (Default) - Content is automatically accepted and displays on the Polycom Trio 8800 system. Button - Users must confirm content acceptance on a popup message.	No

Wireless Display Parameters

features.cfg	content.wirelessDisplay.sink.bitrate	Set the content maximum bitrate in Mbps 30 (default) 0 - 60 0 allows auto-negotiation.	No
features.cfg	content.wirelessDisplay.sink.enabled	0 (default) - Disable Wireless Display. 1 - Enable Wireless Display.	No
features.cfg	content.wirelessDisplay.sink.fps	Set the content frame rate in frames per second. 30 (default) 0 - 60 0 allows auto-negotiation	No
features.cfg	content.wirelessDisplay.sink.height	Set the maximum content height in pixels. 1080 (default) 0 - 1200 0 allows auto-negotiation	No
features.cfg	content.wirelessDisplay.sink.name	NULL - default Specify a system name for the local content sink for Android or Windows devices. If left blank the previously configured or default system name is used.	No
features.cfg	content.wirelessDisplay.sink.width	Set the maximum content width in pixels. 1920 (Default and Maximum) 0 allows auto-negotiation	No

Troubleshooting

This section provides solutions to common issues you may have using Wireless Display on the Polycom Trio 8800 system.

My Polycom Trio 8800 system does not advertise on my smartphone or tablet

If the Polycom Trio 8800 system does not advertise on your smartphone or tablet device, check the following:

- Ensure Wi-Fi is enabled on your device and the band is set to 2.4GHz or Auto. The Auto setting allows the connecting device better access to a free wireless channel.
- Ensure the correct country of operation is set and that both bands are selected on the Polycom Trio 8800 system by configuring the following:

```
device.wifi.country.set="1"
device.wifi.country="CA"
```

```
device.wifi.radio.band2_4GHz.enable.set="1"
device.wifi.radio.band2_4GHz.enable="1"
device.wifi.radio.band5GHz.enable.set="1"
device.wifi.radio.band5GHz.enable="1"
device.wifi.enabled.set="1"
device.wifi.enabled="0"
device.net.enabled.set="1"
device.net.enabled="1"
```

The WLAN operating mode on the Polycom Trio 8800 system is mutually exclusive of the Wireless Display feature. You can enable Wireless Display only if wired Ethernet is used for calling and conferencing. Ensure that wired Ethernet is used for calling and conferencing by configuring the following:

Video Quality is Poor

Incorrect image resolution can cause content delays and video artifacts. Note that the Polycom Trio 8800 system does not accept 1080@60fps video resolution.

- To resolve video quality issues, configure the following for the Polycom Trio 8800 system:

```
content.wirelessDisplay.sink.width="0"
content.wirelessDisplay.sink.height="0"
content.wirelessDisplay.sink.fps="0"
```

- In addition, you can set a limit on the live stream parameters by setting:

```
content.wirelessDisplay.sink.fps="30"
```

Configuring Debugging Logs

If you experience further issues using Wireless Display on the Polycom Trio 8800 system, you can enable the following logging parameters on your Polycom Trio 8800 system to get extended debugging data.

Wireless Display Debugging Parameters

Parameters	Permitted Values
wdisp	Wireless Display session management and communication with the Wireless Display source
apps	Wireless Display support for Android
lc	Local Content (including Wireless Display and PPCIP) session management

Access Diagnostic Information

If you experience issues using Wireless Display on the Polycom Trio 8800 system, you can access diagnostic information from the Polycom Trio 8800 menu.

Procedure:

1 On the phone menu, go to one of the following settings:
- **Settings > Status > Diagnostics > Local Content Media Statistics**.
- **Settings > Status > Diagnostics > Graphs > Local Video Content Statistics**.
- **Settings > Status > Diagnostics > Graphs > Networked Devices Graphs**.
- **Settings > Status > Diagnostics > Networked Devices > Statistics**.

User Accounts and Contacts

After you set up Polycom phones on your network with the default configuration, you can configure user accounts and user contact list features.

Smart Login

Smart Login, available with the Polycom Trio 8800 and 8500 system, determines if a network environment is capable of PIN Authentication. If the STS-URI is not configured via DHCP Option43 or manually through configuration files, then PIN Authentication will not be enabled for the phone or in the Web Configuration Utility for a Skype for Business sign in.

Microsoft Exchange Integration

Exchange Integration is available for Skype for Business, Office 365, and Lync Server 2010 and 2013 deployments. This feature enables set up of 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.

When you register a Polycom Trio 8800 or 8500 system with Skype for Business, a Calendar icon displays on the phone Home screen that enables users to access features. Users can view and join Outlook calendar events directly from Polycom Trio system. This displays the day and meeting view for scheduled events; the month view is not currently available. Note you cannot schedule calendar events or view email from the phone.

When you pair Polycom Trio 8500 or 8800 with Polycom Trio Visual+, the system automatically displays the Calendar and up to five meetings scheduled within the next 24-48 hours on the Home screen of connected monitor. You can configure whether or not users receive reminder notifications on the display monitor and whether or not an alert sound accompanies reminder notifications.

After the phone is connected, you can:

- Verify which Exchange Server services are not working on each phone by going to **Status > Diagnostics > Warnings** on the phone.
- View the status of each service in the Web Configuration Utility.

Enabling Microsoft Exchange Integration

You can enable Exchange integration using one of the following methods:

- Exchange Server autodiscover.
- Centralized provisioning.

- On a per-phone basis with the Web Configuration Utility.
- When using a UC Software release prior to 5.3.0, you can enable the exchange calendar using centralized provisioning or with the Web Configuration Utility. To enable the Web Configuration Utility, refer to Accessing the 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.

Enable Microsoft Exchange Calendar Using Centralized Provisioning

You have the option to enable Skype for Business Exchange calendar using the following parameters on your central provisioning server.

If you are using Polycom Trio Solution, parameters are included in *Example Configuration File for Polycom Trio 8800 Collaboration Kit with Skype for Business* on Polycom Trio > Documentation > Setup Documents.

To enable the exchange calendar from a provisioning server:

» Add the following parameter 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 have the option to use the Web Configuration Utility to manually enable Skye for Business Exchange Calendar. This is useful for troubleshooting if autodiscovery is not working or misconfigured. This method applies only to a single phone at a time.

To enable the exchange calendar manually:

1. Ensure that you enable Accessing the Web Configuration Utility.
2. Log in to the Web Configuration Utility as Admin (default password 456).
3. Go to **Settings > Applications > Exchange Applications**, and expand **Exchange Applications**, as shown next.
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.

Setting Up Calendar Features

- Visual voicemail. On the server, 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.
- Call log synchronization. On the server, enable the option to save calls logs to each user's conversation history in Outlook.
- ABS adaptive search. On the server, enable the ABS service. 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.

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.

Calendar Meeting Details

You can use `exchange.meeting.show*` parameters to show or hide the following meeting details from the calendar display on the Polycom Trio 8500 and 8800 system screen and on the monitor connected to the Polycom Trio Visual+:

- Subject.
- Location.
- Invitee(s).
- Agenda/Notes. When you hide Agenda/Notes, a message indicates the meeting is private.
- Meeting organizer. The organizer does not display for meetings displayed on the monitor.
- Show More Actions. If multiple numbers are available to dial into a meeting, More Actions displays in Meeting Details to allow users to choose the dial-in number.

Meeting Reminder Messages

A meeting reminder displays on the Polycom Trio system screen at five minutes and one minute before the start of a meeting. The five-minute reminder disappears after 30 seconds if not dismissed. If the one-minute reminder has not been dismissed, the reminder message displays on the Polycom Trio system Home Screen during the duration of the meeting. The one minute reminder disappears when the meeting ends or when the next meeting reminder pops up, whichever comes first.

When multiple meetings are booked at the same time or overlap, a message displays available meetings. Users can tap the message to display the calendar day view and choose which meeting to join.

You can also show or hide all day events, configure the maximum number of future meetings, or configure a user requirement to enter the Skype for Business conference ID when a meeting organize marks a meeting as ‘private’. When meeting organizers mark a meeting invitation as Private in Outlook, the Polycom Trio system displays the meeting invite on the Trio phone calendar and TV screens with ‘Private Meeting’ in the subject line and a lock icon. The conference ID is included in the Outlook invitation.

Configuring Microsoft Exchange Integration

The following table lists parameters that configure Microsoft Exchange integration.

Exchange Integration Parameters

Parameter Template	Permitted Values
`exchange.meeting.alert.followOfficeHours` applications.cfg	1 - Audible alerts occur during business hours. 0 - Audible alerts occur at all times.
`exchange.meeting.alert.tonePattern` applications.cfg	positiveConfirm (default) - Set the tone pattern of the reminder alerts using any tone specified by `se.pat.*`. See section Customize Audio Sound Effects in the UC Software Administrator Guide.
`exchange.meeting.alert.toneVolume` applications.cfg	10 (default) - Set the volume level of reminder alert tones. 0 - 17
`exchange.meeting.allowScrollingToPast` applications.cfg	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.
`exchange.meeting.hideAllDayNotification` applications.cfg	0 (default) - All day meeting notifications display on the Calendar screen. 1 - All day meeting notifications are hidden from the Calendar screen.
`exchange.meeting.parseOption1` applications.cfg	Indicates the field in the meeting invite from which the VMR or meeting number should be fetched. Location (default) All LocationAndSubject Description
`exchange.meeting.parseWhen` applications.cfg	NonSkypeMeeting (default) - Disable number-searching on the Calendar to look for additional numbers to dial in Skype Meeting calendar entries. Always - Enables number-searching on the Calendar to look for additional numbers to dial even for Skype Meetings.

Exchange Integration Parameters

Parameter Template	Permitted Values
`exchange.meeting.phonePattern` applications.cfg	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).
`exchange.meeting.reminderEnabled` applications.cfg	1 (default) - Meeting reminders are enabled. 0 - Meeting reminders are disabled.
`exchange.meeting.reminderInterval` applications.cfg	300 seconds (default) 60 - 900 seconds Set the interval at which phones display reminder messages.
`exchange.meeting.reminderSound.enabled` applications.cfg	1 - 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.
`exchange.meeting.reminderType` applications.cfg	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.
`exchange.meeting.showAttendees` applications.cfg	1 (default) - Show the names of the meeting invitees. 0 - Hide the names of the meeting invitees.
`exchange.meeting.showDescription` applications.cfg	1 (default) - Show Agenda/Notes in Meeting Details that displays after you tap a scheduled meeting on the Polycom Trio system calendar. 0 - Hide the meeting Agenda/Notes.
`exchange.meeting.showLocation` applications.cfg	1 (default) - Show the meeting location. 0 - Hide the meeting location.
`exchange.meeting.showMoreActions` applications.cfg	1 (default) - Show More Actions in Meeting Details to allow users to choose a dial-in number. 0 - Hide More Actions in Meeting Details.
`exchange.meeting.showOnlyCurrentOrNext` applications.cfg	0 (default) - Disabled the limitation to display only the current or next meeting on the Calendar. 1 - Enables the limitation to display only the current or next meeting on the Calendar.

Exchange Integration Parameters

Parameter Template	Permitted Values
`exchange.meeting.showOrganizer` applications.cfg	1 (default) - Show the meeting organizer in the meeting invite. 0 - Hide the meeting organizer in the meeting invite.
`exchange.meeting.showSubject` applications.cfg	1 (default) - Show the meeting Subject. 0 - Hide the meeting Subject.
`exchange.meeting.showTomorrow` applications.cfg	1 (default) - Show meetings scheduled for tomorrow as well as meetings scheduled for today. 0 - Do not show meetings scheduled for tomorrow.
`exchange.menu.location` applications.cfg	Features (default) - Displays the Calendar in the global menu under Settings > Features. Administrator - Displays the Calendar in the Admin menu at Settings > Advanced > Administration Settings.
`exchange.reconnectOnError` applications.cfg	1 (default) - The phone attempts to reconnect to the Exchange server after an error. 0 - The phone does not attempt to reconnect to the Exchange server after an error.
`exchange.server.url` applications.cfg	NULL (default) string The Microsoft Exchange server address.
`exchange.showSeparateAuth`[1] applications.cfg	0 (default) - Allow users to sign in with their Skype for Business account and access an associated calendar. 1 - Provides users an option to sign in with credentials in addition to their Skype for Business account to access a different calendar. These additional fields (Exchange Email, Exchange Domain, Exchange User, and Exchange Password) are located in the Sign In > Advanced menu.
`feature.EWSAutodiscover.enabled` applications.cfg	If you configure `exchange.server.url` and set this parameter to 1, preference is given to the value of `exchange.server.url`. 1 (default) - 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.

Exchange Integration Parameters

Parameter Template	Permitted Values
`feature.exchangeCalendar.enabled` features.cfg	For the Polycom Trio 8800 solution, VVX 300/301, 310/311, 400/401, 410/411, 500/501, 600/601 and 1500 phones, and the CX5500 Unified Conference Station. 1 (default) - 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. 0 (default) - The calendaring feature is disabled.
`feature.exchangeCallLog.enabled` features.cfg	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.
`feature.exchangeContacts.enabled` features.cfg	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. 0 - The Exchange call log feature is disabled and the user call logs history cannot be retrieved from the Exchange server. You must also enable the parameter `feature.exchangeCallLog.enabled` to use the Exchange call log feature.
`feature.exchangeVoiceMail.enabled` features.cfg	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.
`feature.exchangeVoiceMail.skipPin.enabled` features.cfg	1 (default) - Enable PIN Auth for Exchange Voicemail. 0 - Disable PIN Auth for Exchange Voicemail.

Exchange Integration Parameters

Parameter Template	Permitted Values
`feature.lync.abs.enabled` features.cfg	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.
`feature.lync.abs.maxResult` features.cfg	20 (default) 5 - 50 The value for this parameter defines the maximum number of contacts to display in a Skype for Business address book service contact search.
`features.contacts.readonly` features.cfg	0 (default) - Skype for Business Contacts are editable. 1 - Skype for Business are read-only.
`up.oneTouchVoiceMail`[1] features.cfg	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.

[1] Change causes phone to restart or reboot.

Private Meetings in Microsoft Exchange

When a Skype for Business meeting is set to Private, you can choose which meeting information to show or hide.

Configuring Private Meetings

The following parameters configure Skype for Business private meetings.

Private Meeting Parameters

Parameter Template	Permitted Values
`exchange.meeting.private.showAttendees` applications.cfg	0 (default) – Meetings marked as private in Outlook do not show the list of meeting attendees and invitees on the Polycom Trio calendar. 1 – Meetings marked as private in Outlook show the list of meeting attendees and invitees on the Polycom Trio calendar.
`exchange.meeting.private.showDescription` applications.cfg	0 (default) – Meetings marked as private in Outlook do not display a meeting description on the Polycom Trio calendar. 1 - Meetings marked as private in Outlook display a meeting description on Polycom Trio calendar.
`exchange.meeting.private.showLocation` applications.cfg	0 (default) – Meetings marked as private in Outlook do not display the meeting location on the Polycom Trio calendar. 1 - Meetings marked as private in Outlook display the meeting location on the Polycom Trio calendar.
`exchange.meeting.private.showSubject` applications.cfg	0 (default) – Meetings marked as private in Outlook do not display a subject line on Polycom Trio calendar. 1 – Meetings marked as private in Outlook display a subject line on Polycom Trio calendar.
`exchange.meeting.private.showMoreActions` applications.cfg	1 (default) – Meetings marked as private in Outlook display the ‘More Actions’ button, when applicable. 0 – Meetings marked as private in Outlook do not display the ‘More Actions’ button.
`exchange.meeting.private.showOrganizer` applications.cfg	1 (default) – Meetings marked as private in Outlook display the name of the meeting organizer on the Polycom Trio calendar. 0 – Meetings marked as private in Outlook display the name of the meeting organizer on the Polycom Trio calendar.

Parameter Template	Permitted Values
exchange.meeting.private.enabled applications.cfg	1 (default) – The Polycom Trio considers the private meeting flag for meetings marked as private in Outlook. 0 – Treat meetings marked as private in Outlook the same as other meetings.
exchange.meeting.private.promptForPIN applications.cfg	0 (default) - Disable the Skype for Business Conference ID prompt that allows users to join meetings marked as 'private'. 1 - Enable the Skype for Business Conference ID prompt that allows users to join meetings marked as 'private'.

Skype for Business User Profiles

You can enable users to access their personal settings from any phone in the organization registered to Skype for Business. For example, users can access their contact directory and speed dials, as well as other phone settings, even if they temporarily change work areas. This feature is particularly useful for remote and mobile workers who do not use a dedicated work space and conduct business in multiple locations. The user profile feature is also useful if an office has a common conference phone from which multiple users need to access their personal settings.

You must decide whether to require users to always log in to a phone or not. If you do not require users to log in, users have the option to use the phone as is-without access to their personal settings-or they can log in to display their personal settings. You can also specify if, after the device restarts or reboots, a user is automatically logged out.

You can choose to define default credentials. If you specify a default user ID and password, the phone automatically logs itself in each time an actual user logs out or the device restarts or reboots. When the device logs itself in using the default login credentials, a default profile displays, and users retain the option to log in and view their personal settings.

You can configure the phones so that anyone can call authorized and emergency numbers when not logged in to a phone using the parameter dialplan.routing.emergency.outboundIdentity.

Polycom recommends that you create a single default user password for all users. 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.

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.

To set up the user profile feature, you must:

- Create a phone configuration file or update an existing file to enable the feature's settings, and configure attributes for the feature.
- Create a user configuration file in the format <user>.cfg to specify each user's password and registration and other user-specific settings that you want to define.

Create a User Profile Configuration File

Create a configuration file if you want to add or edit user login or feature settings for multiple phones.

To create a user profile configuration file:

1. Create a configuration file for the phone and place it on the provisioning server.

   You can create your own or base this file on the sample configuration template in the UC Software, for example, `site.cfg`. To find the file, navigate to `<provisioning server location>/Config/site.cfg`.
2. In `site.cfg`, open the `<prov.login/>` attribute, and then add and set values for the user login parameters.
3. Copy these attributes for each user and enter user-specific values.

Create a User Configuration File

Create a user-specific configuration file that stores user names, passwords, and registrations.

To create a user configuration file:

1. On the provisioning server, create a user configuration file for each user to log in to the phone. The name of the file is the user's ID 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`.
2. In each `<user>.cfg` file, you can add and set values for the user's login password (optional).
3. Add and set values for any user-specific parameters, such as:
   - Registration details (for example, the number of lines the profile displays and line labels).
   - Feature settings (for example, browser settings).

Note: 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. For information on which parameters cause the phone to restart or reboot, see the reference section Configuration Parameters.

After a user logs in, with their user ID and password (The default password is 123.), users can:

- Log in to a phone to access their personal phone settings.
- Log out of a phone after they finish using it.
- Place a call to an authorized number from a phone that is in the logged out state.
- Change their user password.

If a user changes any settings while logged in, the settings save and display the next time the user logs in. When a user logs out, the user's personal phone settings no longer display.

Stored User Settings

If a user updates their password or other user-specific settings using the Main Menu 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. Configuration parameter precedence (from first to last) for a phone that has the user profile feature enabled is:

- `<user>-phone.cfg`
- Web Configuration Utility
- Configuration files listed in the master configuration file (including <user>.cfg)
- Default values

Unified Contact Store

Administrators can unify users' contacts with Microsoft Exchange Server to enable users to access and manage contacts from any application or device synchronized with the Exchange Server including Polycom Trio 8800 and 8500 systems, VVX business media phones, Skype for Business client, Outlook, or Outlook Web Application from a mobile device. For example, if a user deletes a contact from a phone, the contact is also deleted on the Skype for Business client. Note users can manage (move, copy) contacts across Groups only on the Skype for Business client and Group contacts on the phone stay unified.

When an administrator enables Unified Contact Store, users can:

- Add a contact
- Delete a contact
- Add and delete a Distribution List (DL) group
- Manage contacts or groups

To set up this feature, administrators must use a PowerShell command using the instructions on the Microsoft TechNet web site Planning and deploying unified contact store in Lync Server 2013.

Sign In Methods

You can configure users to sign in or out of the phone using one of the following methods:

- **User ID.** Use this to sign in with user credentials on the Sign In screen. You cannot configure login credentials using the Polycom Web Configuration Utility.
- **PIN Authentication**. Use this to sign in on the phone or from the Web Configuration Utility. As of UC Software 5.1.1, this sign in method is available on the SoundStructure VoIP Interface. This option is available in on-premises Skype for Business deployments when you configure DCHP Option 43, and is not available for online deployments.
- **Web Sign In for Skype for Business Online.** For online deployments only, this method enables secure sign-in from a browser on your computer or mobile device. The phone generates a unique pairing code used to sign in on a secure Office 365 website. For more information about Web Sign In, refer to Web Sign In for Skype for Business Online
- **Single Sign-On Solutions (SSO)**. Allows you to use the same login credentials across multiple cloud-based applications such as Microsoft Exchange and Skype for Business.

Note that the maximum length of the user name or sign in address (Name + Domain) is limited to 45 characters.

Note: You cannot configure login credentials using the Polycom Web Configuration Utility.

Configuring a Skype for Business Sign In Method and Credentials

The following parameters configure the type of sign in on the phones and user credentials.

Skype for Business Sign In Method Parameters

Parameter Template	Permitted Values
`reg.1.auth.loginCredentialType` reg-advanced.cfg	Configure a login type and user credentials. You cannot log in to the phone with Microsoft credentials if the parameter reg.1.auth.loginCredentialType is set to the default value. LoginCredentialNone (default) usernameAndPassword - Set credentials to sign-in address, user name, domain, and password in the required format. extensionAndPIN - Set credentials to extension and PIN.
`reg.1.auth.useLoginCredentials` reg-advanced.cfg	You can use this method in the configuration file to automatically sign in users after the phone powers up. 1 (default) - SSI Login credentials, BToE Sign in, and Web Sign types are available for authentication with the server. 0 - SSI Login credentials, BToE Sign in, and Web Sign types are not available for authentication with the server.
`reg.1.auth.usePinCredentials` reg-advanced.cfg	You can use this method in the configuration file to automatically sign in users after the phone powers up. To use this sign-in method, you must enable DHCP Option 43 or dhcp.option43.override.stsUri. 1 (default) - PIN authentication sign in method is available for authentication on the server. 0 (default) - PIN authentication sign in method is not available for authentication on the server.

Example Sign In Configurations

You can set PIN Authentication or SSI login credentials in the configuration file to log in users automatically after the phone powers up.

The following example sets PIN Auth user credentials in the configuration file:

- `reg.1.auth.usePinCredentials="1"`
- `reg.1.auth.loginCredentialType="extensionAndPIN"`
- `device.set="1"`
- `device.logincred.extension.set="1"`

- `device.logincred.extension="xxxx"`
- `device.logincred.pin.set="1"`
- `device.logincred.pin="xxxx"`

The following example sets SSI login credentials in the configuration file:

- `reg.1.auth.loginCredentialType="usernameAndPassword"`
- `reg.1.address="xxxx@domain.com"`
- `device.set="1"`
- `device.logincred.user.set="1"`
- `device.logincred.user="xxxx"`
- `device.logincred.password.set="1"`
- `device.logincred.password="xxxxx"`
- `device.logincred.domain.set="1"`
- `device.logincred.domain="domain"`

PIN Authentication

You can sign in to Skype for Business using PIN authentication.

To use PIN authentication, you must enable the Web Configuration Utility, which is disabled by default. Refer to the section Accessing the Web Configuration Utility. After you enable the Web Configuration Utility, you can enable or disable PIN authentication using `reg.1.auth.usePinCredentials`.

If you configure DHCP Option 43 in on-premises Skype for Business deployments, the phone displays only the PIN Authentication menu to users. The PIN Auth menu does not display and is not available for Skype for Business Online.

PIN Authentication Parameters

Parameters listed in the following table configure PIN Authentication sign-in method.

PIN Authentication Parameters

Parameter Template	Permitted Values
`device.logincred.extension` device.cfg, features.cfg	NULL (default) - Phones will not trigger registration. 0 to 32 - Enter a user phone extension number or string to a maximum of 32 characters. The phone reads this extension when you configure PIN-Auth as the phone registration method.
`device.logincred.pin` device.cfg, features.cfg	NULL (default) - If the default value is set, phones will not trigger registration. 0 to 32 -Enter a user phone PIN to a maximum of 32 characters. The phone reads this PIN when you configure PIN-Auth as the phone registration method.

Web Sign In for Skype for Business Online

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

Note that this sign in method generates a pairing code that expires within a few minutes after the Skype for Business server sends the code to the phone. Users must sign in before the pairing code expires.

Configuring Web Sign In for Skype for Business Online

The following table lists parameters that configure Web Sign In for Skype for Business Online deployments.

Skype for Business Online Web Sign-In Parameters

Template	Parameter	Permitted Values
features.cfg	`feature.webSignIn.enabled`	1 (default) – In Skype for Business Base Profile, the web sign in option is displayed on the phone for the user. 0 – In Skype for Business Base Profile, the web sign in option is not displayed on the phone for the user.
reg-advancd.cfg	`reg.1.auth.loginCredentialType`	Specify the credential type the user must provide to log in. You cannot log in to the phone with Microsoft credentials if `reg.1.auth.loginCredentialType` is set to the default value. `LoginCredentialNone` (default) `onlineDeviceAuth` - Enables users to sign in to the phone using Web Sign In. `usernameAndPassword` - Provide description of this value.

Single Sign-On (SSO) Solutions

The Third-party Single Sign-On (SSO) is an authentication method that allows users to use the same login credentials to log in to multiple cloud-based applications, such as Microsoft Exchange and Skype for Business, at the same time. SSO enables users to switch between different cloud-based applications during a single session, without being prompted to enter login credentials every time.

If your Polycom Trio system is used as a shared device in your organization, you can remove the sign-out soft key to prevent users from signing others out. Or, you can remove both the sign-in and sign-out soft keys.

Contact Directories

When the Polycom Trio solution Base Profile is set to Skype, you can access contacts using the Skype for Business contact list and set the contact list to be editable or read-only.

The local contact directory is disabled by default when the Polycom Trio system Base Profile is set to Skype and you can enable it using the parameter `feature.directory.enabled`. You can protect the local contact directory with user credentials and set contacts to be read-only or editable. If the Polycom Trio system Base Profile is set to 'SkypeUSB', no local contact directory is available and you can access the Skype for Business contact list.

The maximum number of contacts you can configure is 3000 and the maximum file size of the local Contact Directory is 4MB. To reduce use of phone memory, use the parameter `dir.local.contacts.maxNum` to set a lower maximum number of contacts.

Configuring Contacts

The following parameters configure the local contact directory on the Polycom Trio solution.

Local Contact Directory Parameters

Parameter	Permitted Values
`dir.local.contacts.maxNum` features.cfg	2,000 (default) - Number of contacts that can be stored by default in the local Contact Directory. 3,000 - Maximum number of contacts that can be stored in the local Contact Directory.
`dir.local.passwordProtected` features.cfg	0 (default) - Disable password protection of the local Contact Directory. 1 - Enables password protection of the local Contact Directory.
`dir.local.readOnly` features.cfg	0 (default) - Disable read only protection of the local Contact Directory. 1 - Enable read-only protection of the local Contact Directory.
`feature.contacts.readonly` features.cfg	0 (default) - Skype for Business Contacts are editable. 1 -Skype for Business are read-only.
`feature.corporateDirectory.enabled` features.cfg	0 (default) - Disable the corporate directory. 1 - Enable the corporate directory.
`feature.directory.enabled` features.cfg	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.
`feature.pauseAndWaitDigitEntryControl.enabled` features.cfg	1 (default) - Enable processing of control characters in the contact phone number field. When enabled, ',' 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.

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 server. 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).

Configuring Call Logs

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

Call Log Parameters

`Parameter` Template	Permitted Values

`feature.exchangeCallLog.enabled`
features.cfg
	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.

Administrator Menu on Polycom Trio Systems

For the Polycom Trio 8800 and 8500 systems, you can add a new ‘Advanced' menu containing a subset of administrator settings. The added 'Advanced' menu item does not require a password but one can be assigned to it.

After enabling this feature, the added 'Advanced' menu provides access to all administrator features except:

- Line Configuration
- Call Server Configuration
- TLS Security
- Test Automation

Administrator Menu Parameters

The following table lists the parameters to enable the new Administrator menu.

Admin Menu Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg, site.cfg	device.auth.localAdvancedPassword.set	Set a password for the Advanced menu. 0 (default) – You cannot set a password for the added Advanced menu. 1 – You can set a password for the added Administrator menu.	No
device.cfg, site.cfg	device.auth.localAdvancedPassword	Enter a password for the added Administrator menu. Null (default) String (0 to 64 characters)	No
feaures.cfg	feature.advancedUser.enabled	0 (default) - The normal password-protected Advanced menu displays. 1 - Causes the 'Advanced' menu item to be renamed 'Admin' and adds a menu item 'Advanced' that contains a subset of administrator features. The new 'Advanced' menu does not require a password but you have the option to assign one to it.	No

Call Controls

This section shows you how to configure call control features.

Skype for Business Local and Centralized Real-Time Audio and Video Calling

You can set up ad hoc 'Meet Now' or scheduled, local, or centralized calls on Polycom Trio 8800 and 8500 registered with Skype for Business.

To enable video calls, you must pair a Polycom Trio 8500 or 8800 system to Polycom Trio Visual+ and connect Polycom Trio Visual+ to a display screen and a Logitech Webcam C930e USB camera. When the devices are successfully paired, the display screen shows a small self-view window.

After you pair a Polycom Trio system with Polycom Trio Visual+, you can set up local or centralized calls with audio and video or audio-only, and share content using the Skype for Business client. You can start each new call with audio and video or audio-only. If you set to audio-only, users can add video during a call if the far-end is video capable.

Related topics:

- For instructions on pairing, refer to Pairing the Polycom Trio Visual+ with Polycom Trio.
- For instructions on configuring video layouts, refer to Skype for Business Video Layouts on Polycom Trio System.
- For details on supported video codecs, refer to Supported Video Codecs.

Using an API to Join a Skype for Business Meeting

This feature allows third-party application developers to join the Polycom Trio 8800 and 8500 system to a scheduled Skype for Business meeting using an API. The third-party application requires independent access to the Skype for Business meeting information.

The following example API illustrates how you can use a third-party application on your computer or mobile device to join the Polycom Trio system to a Skype for Business meeting.

REST API Command: WebCallControl.Dial

Structure

```
{

"data":api/v1/callctrl/dial
```

```
{
 "data":
 {
    "Dest": "<SIP URI of meeting organizer>;gruu;opaque=app:conf:focus:id:<SfB Meeting
Focus ID>",
     "Line": "1",
     "Type":"SIP"
 }
```

Join a Meeting with a SIP URI

When you set up a meeting in the Calendar, the Polycom Trio 8800 system displays a meeting reminder pop up. If a dial-in number is available for the meeting, the reminder pop-up presents a Join button that joins you to the meeting. If a meeting lists multiple dial-in numbers or URIs for the meeting, by default the Join button automatically dials the first number.

You now have the option of configuring the Polycom Trio 8800 system to offer users a list of available numbers when they tap the Join button instead of dialing the first number.

Polycom Trio system provide multiple dial-in options including a SIP URI by selecting a dial-in information from a list of numbers below:

- SIP URI
- Tel URI
- PSTN number
- IP dial

Parameters to Join a Meeting with a SIP URI

The following table lists the parameters to configure the dial-in information.

SIP URI Dial-in Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`applications.cfg`	`exchange.meeting.join.promptWithList`	Specifies the behavior of the Join button on meeting reminder pop-ups. 0 (default) - Tapping Join on a meeting reminder should show a list of numbers to dial rather than immediately dialing the first one. 1 - A meeting reminder does not show a list of numbers to dial.	No
`applications.cfg`	`exchange.meeting.parseWhen`	Specifies when to scan the meeting's subject, location, and description fields for dialable numbers. NonSkypeMeeting (default) Always Never	Yes
`applications.cfg`	`exchange.meeting.parseOption`	Specifies where to search for a dialable number. All (default)	Yes
`applications.cfg`	`exchange.meeting.parseEmailsAsSipUris`	List instances of text like `user@domain` or `user@ipaddress` in the meeting description or subject under the More Actions pane as dialable SIP URIs. 0 (default) - it does not list the text as a dialable SIP URI 1 - it treats `user@domain` or user@ipaddress as a dialable SIP URI.	Yes
`applications.cfg`	`exchange.meeting.parseAllowedSipUriDomains`	List of comma-separated domains that will be permitted to be interpreted as SIP URIs Null (default) String (maximum of 255 characters)	Yes

Skype for Business Private Meeting Parameters

Use the following parameters to configure Skype for Business private meetings.

Private Meeting Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
	exchange.meeting.private.showAttendees	0 (default) – Meetings marked as private in Outlook do not show the list of meeting attendees and invitees on the Polycom Trio calendar. 1 – Meetings marked as private in Outlook show the list of meeting attendees and invitees on the Polycom Trio calendar.	
	exchange.meeting.private.showDescription	0 (default) – Meetings marked as private in Outlook do not display a meeting description on the Polycom Trio calendar. 1 - Meetings marked as private in Outlook display a meeting description on Polycom Trio calendar.	
	exchange.meeting.private.showLocation	0 (default) – Meetings marked as private in Outlook do not display the meeting location on the Polycom Trio calendar. 1 - Meetings marked as private in Outlook display the meeting location on the Polycom Trio calendar.	
	exchange.meeting.private.showSubject	0 (default) – Meetings marked as private in Outlook do not display a subject line on Polycom Trio calendar. 1 – Meetings marked as private in Outlook display a subject line on Polycom Trio calendar.	
	exchange.meeting.private.showMoreActions	1 (default) – Meetings marked as private in Outlook display the ‘More Actions’ button, when applicable. 0 – Meetings marked as private in Outlook do not display the ‘More Actions’ button.	

Private Meeting Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
	`exchange.meeting.private.showOrganizer`	1 (default) – Meetings marked as private in Outlook display the name of the meeting organizer on the Polycom Trio calendar. 0 – Meetings marked as private in Outlook display the name of the meeting organizer on the Polycom Trio calendar.	
	`exchange.meeting.private.enabled`	1 (default) – The Polycom Trio considers the private meeting flag for meetings marked as private in Outlook. 0 – Treat meetings marked as private in Outlook the same as other meetings.	
	`exchange.meeting.private.promptForPIN` `applications.cfg`	0 (default) - Disable the Skype for Business Conference ID prompt that allows users to join meetings marked as 'private'. 1 - Enable the Skype for Business Conference ID prompt that allows users to join meetings marked as 'private'.	

USB Mode for Skype Room Systems and Surface Hub

Calls and call controls display on the Polycom Trio 8800 and 8500 system and Skype Room System or Microsoft Surface Hub when you connect Polycom Trio system via USB cable and set the Polycom Trio system Base Profile to 'SkypeUSB'.

Note

- You must use UC Software 5.4.4AD or later to successfully connect with a Skype Room System or Surface Hub.
- You must update the Microsoft Surface Hub to the latest release version to successfully connect to the Polycom Trio system.

The Polycom Acoustic Echo Cancellation (AEC) feature is not available when the Polycom Trio system is connected to Microsoft Surface Hub.

You can use the Polycom Trio system as a speaker and microphone with a Skype Room System and Surface Hub to:

- Answer calls

- End calls
- Mute/unmute audio (microphones)
- Adjust loudspeaker volume

When you set the Base Profile to SkypeUSB, the following functions are not available:

- Dial a number
- Contacts
- Meetings
- Recent Calls
- Meet Now
- Automatic Gain Control (AGC)
- Bluetooth

Connecting Polycom Trio System to a Skype Room System or Microsoft Surface Hub

Complete the following steps to connect Polycom Trio 8800 and 8500 system with a Skype Room System or Surface Hub.

To connect Polycom Trio with a Skype Room System or Surface Hub:

1. Update the Microsoft Surface Hub to the latest release version to successfully connect to the Polycom Trio solution.
2. Power the Polycom Trio system. Refer to Powering the Polycom Trio 8500 and 8800 Systems.
3. Use or update to UC Software 5.4.4AD or later to successfully connect with a Skype Room System or Surface Hub. Refer to Polycom UC Software Update.
4. If you did not purchase a Polycom Trio system Skype Room System edition, configure the Polycom Trio system Base Profile to ‘SkypeUSB’. Otherwise, omit this step. Refer to Configuring Polycom Trio System with a Skype Room System or Surface Hub.
5. Connect the Polycom Trio system to a Skype Room System or Surface Hub via USB cable.

   When you connect the USB cable while the Skype Room System or Surface Hub is in a call, both systems are placed on hold until you re-join the call from either system.
6. On the Skype Room System or Surface Hub you are connecting, configure Polycom Trio as the audio device - microphone and speaker.
7. (Optional) When the Base Profile of a phone is set to ‘Skype’ or ‘SkypeUSB’, access to the Web Configuration Utility is disabled by default. To enable access to a phone's Web Configuration Utility, refer to Accessing the Web Configuration Utility.

Configuring Polycom Trio System with a Skype Room System or Surface Hub

The following parameters configure the Polycom Trio 8800 and 8500 system with Skype Room Systems and Surface Hub.

USB Mode Parameters

Parameter Template	Permitted Values
`device.baseProfile` device.cfg	Generic (default) - Disables the Skype for Business graphic interface. Skype - Use this Base Profile for Skype for Business deployments. SkypeUSB - Use this Base Profile when you want to connect Polycom Trio to a Skype Room System or Surface Hub.
`voice.usb.holdResume.enable` features.cfg	0 (default) - The Hold and Resume buttons do not display during USB calls. 1 - The Hold and Resume buttons display during USB calls. This parameter applies only when the Polycom Trio system Base Profile is set to ‘SkypeUSB’.

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.

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.

Configuring Local Call Recording

Use the parameters in the following table to configure local call recording.

Local Call Recording Parameters

Parameter Template	Permitted Values
`feature.callRecording.enabled` features.cfg	0 (default) - Enables audio call recording. 1 - Disables audio call recording.

Configuring Shared Line Appearance (SLA) for Skype for Business

Shared Line Appearance (SLA) feature enables user to share a single line with other contacts as a member of a group. Each shared line can receive only one incoming call at a time, and users cannot make outgoing calls from the shared line, including 911 emergency calls.

An incoming call to the shared line is received by all phones sharing the line. Any SLA group member can place, answer, hold, or resume calls on the line, and all group members can view the status of a call on the shared line on their phones.

The following features are not supported on SLA lines:

- BToE
- Conference class
- Call Park

Administrators must install the Shared line Application on the Microsoft Front End server and configure SLA groups in Windows PowerShell.

Administrators can configure a ring tone type, and users can set a ring type from the phone's Basic Settings menu.

SLA for Skype for Business Parameters

Parameter Template	Permitted Values
`up.SLA.ringType`	Set the ring type for the share line so that users can distinguish between incoming calls to a private, primary line and the group SLA line. Note that users can set this ring type from the phone, which overrides the value you set here. 1 (default) 0 - 25

Hybrid Line Registration

The Polycom Trio 8500 and 8800 system supports hybrid (Skype for Business / Open SIP) registration. You can simultaneously register one line with Skype for Business or Open SIP and a second line with another Open SIP server. Similarly, you can choose to register all lines with Open SIP sever. You can also choose the number of lines you want to use by setting the value in `reg.limit` parameter.

If you plan to configure and register Skype for Business on one line, make sure to always use Line 1 for Skype for Business. You cannot simultaneously register two Skype for Business lines.

In addition, you can configure the line switching feature based on dial plan when the phone is on-hook. The line switching feature enables the dialed number to switch to the corresponding line. For example, when you place a call from the phone and the number corresponds to an Open SIP line, the line switching feature enables the dialed number to switch to the corresponding line.

Moreover, for dial plan based line switching, when all the lines are registered to Open SIP, the value defined in the global parameter for a dial plan takes the priority. For example, `dialplan.impossibleMatchHandling` and `dialplan.conflictMatchHandling`. Similarly, if the line is registered to Skype for Business, the value defined in the per-registration dial plan parameter takes priority over general dial plan parameter. For example, `dialplan.1.conflictMatchHandling` and `dialplan.1.impossibleMatchHandling`.

When more than one digit maps are getting matched to the dialed number - a conflict match - and the `dialplan.conflictMatchHandling` parameter is disabled, the first matching digit map starting from left to right takes priority. However, if `dialplan.conflictMatchHandling` parameter is enabled, the matching digit map having the lowest timeout value takes priority.

However, line switching is configurable based on dial plan when the phone is off-hook. By default, line switching for on-hook and off-hook dialing is disabled.

Note that the Presence feature is available only on the Skype for Business line and will display the Device status. The following table list the Presence status for specific environment.

Presence Status Indicators for Hybrid Line Registration

Use Cases	Presence State on SfB Line	Presence String	Presence State on Open SIP Line
Non-Skype line in a call	Busy	In a call	Not Supported
Skype line in a call	Busy	In a call	Not Supported
Content shared over PPCIP	Busy	In a call	Not Supported
Non-Skype line in conference	Busy	In a conference	Not Supported
Skype line in conference	Busy	In a conference	Not Supported
DND on Skype line	DND	Do Not Disturb	Not Supported
DND on Open SIP line	Available	Available	Not Supported

Hybrid Line Registration Limitations

The Hybrid Registration feature include the following limitations:

- Merging of local conference is only supported with open SIP registrations and not supported with Skype for Business (bridging a Skype for Business with an open SIP line is not supported).
- Local merging of two point-to-point calls made using two different lines between two Polycom Trio systems is not supported.
- Only call transfers between different SIP registrations with the same SIP call servers is supported. Call transfer between SIP registrations on different SIP call servers is not supported.
- Transport Layer Security (TLS) encryption of Real-time Transport Protocol (RTP) media for secure communication in hybrid Open SIP registrations is not supported.

Hybrid Line Registration Parameters

The following tables lists the parameters to configure dial plan and line switching for Hybrid Registration.

Dial Plan and Digit Map Parameters for Hybrid Registrations

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`dialplan.digitmap.lineSwitching.enable`	0 (default) - Disable the line switching in dial plan to switch the call to the dial plan matched line. 1 - Enable the line switching in dial plan to switch the call to the dial plan matched line. This is not applicable for off-hook dialing.	No
`reg.cfg`	`reg.limit`	Specify the maximum number of lines to use for registration. 1 (default) 12 maximum (1-3 supported)	No
`sip-interop.cfg`	`reg.1.mergeServerDigitMapLocally`	1 (default) - Allows the dialplans from `dialplan.1.digitmap` to append on top of the dialplans received from the server. 0 - Does not allow the dial plans from `dialplan.1.digitmap` to append on top of the dialplans received from the server.	No

Configure Hybrid Line Registration using the Web Configuration Utility

You can configure the phone to support the Hybrid (Skype for Business/ Open SIP) Registration from phone's Web Configuration Utility page after enabling the feature using configuration parameter.

Make sure the to set the Base profile as Skype for Business on the Polycom Trio 8800 system.

To configure a hybrid line registration using Web Configuration Utility:

1. Sign in to the Polycom Trio 8800 system's Web configuration Utility page using Admin account.

   If configuring Skype for Business on Line 1, sign in to the Web Configuration Utility as Skype for Business user.
2. On the Web Configuration Utility page, navigate to Settings > Line.

   The number of lines enabled to configure is displayed.
3. Configure the Skype for Business registration on Line 1.
4. Configure the Open SIP registration on Line 2.

   You can configure other lines with Open SIP registration.

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.

Configuring International Dialing Prefixes

The following parameters configure the international dialing prefixes.

International Dialing Prefix Parameters

Parameter Template	Permitted Values
`call.internationalDialing.enabled` site.cfg	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.
`call.internationalPrefix.key` site.cfg	The phone supports international call prefix (+) with both '0' and '*'. 0 (default) - Set the international prefix with *. 1 - Set the international prefix with 0.

Centralized Conference Control Protocol (CCCP)

CCCP is enabled by default when the phone Base Profile is set to Skype. CCCP enables users to initiate conference calls with Skype for Business contacts from their phone, manage conference participants, enable announcements, and lock a conference. Users can manage a maximum of 24 Skype for Business conference calls at a time on their phone. However, users can have only one active conference call in progress on their phone.

Centralized Conference Control Protocol (CCCP) Parameters

The following parameters configure CCCP.

CCCP Parameters

Parameter Template	Permitted Values
`feature.cccp.enabled`	1 (enabled) - Enable use of CCCP. 0 - Disable use of CCCP.

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.

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.

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	Choose whether the dial plan is applied to numbers dialed from the directory or speed dial, including auto-call contact numbers. 0 (default) 1	Yes
site.cfg	dialplan.applyToForward	Choose whether the dial plan applies to forwarded calls. 0 1	Yes
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)	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
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.	No
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. (Default)[2-9]11|0T|+011xxx.T|0[2-9]xxxxxxxxx|+1[2-9]xxxxxxxx|[2-9]xxxxxxxxx|[2-9]xxxT 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	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.	No
`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=1: 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 1.	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

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
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
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. 0 (default for Generic Profile) 0-99 seconds You can apply dialplan.userDial.timeOut only when its value is lower than up.IdleTimeOut.	No

Dial Plans

This section on dial plans includes information on dial plan normalization, multiple emergency number dial plans, parameters you can configure on your provisioning server, and examples of supported and unsupported dial plans.

Dial Plan Normalization

Dial Plan Normalization enables you to configure dial plans on the Skype for Business server or on your provisioning server.

For more information on regular expressions used on Skype for Business server, see .NET Framework Regular Expressions on Microsoft Developer Network.

Multiple Emergency Number Dial Plan

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

Polycom phones receive emergency numbers through in-band provisioning and can conflict with the emergency dial string and mask. When a phone receives both multiple emergency numbers and emergency dial string and mask, the client and phone use multiple emergency numbers.

For instructions on creating a multiple emergency number dial plan, see Configure Multiple Emergency Numbers in Skype for Business 2015 on Microsoft TechNet.

SIP URI Dialing

When making a URI call, Polycom Trio 8500 and 8800 systems allow dial plan matching for SIP URI calls to append strings to the dialed number. SIP URI dial plan can also be used with auto line switching in Hybrid registration scenarios to automatically select the line based on dial plan.

The following examples illustrate the semantics of the syntax:

- `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 enable dial plan-based line switching, and the third line's dial plan is `sip\:xxxx\@registrar\.polycomcsn\.com` then calls will be initiated from the third line if you dial `1234@registrar.polycomcsn.com` because it matches the third line's dial plan.

Dial Plan, Dial Plan Normalization, and Digit Map Parameters

Polycom does not support all regular expression dial plans. The following tables list available parameters and supported and unsupported dial plans with Skype for Business Server. The tables are followed by examples of supported and unsupported dial plans.

Dial Plan, Dial Plan Normalization, and Digit Map Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
	`dialplan.1.digitmap`	x.T	No
	`dialplan.1.digitmap.timeOut`	Specify a timeout in seconds for each segment of digit map. After you press a key, the phone waits the number of seconds you specify to match the digits to a dial plan and dial the call. 4 seconds (default) string of positive integers separated by | for example 3 | 3 | 3 | 3 | 3 | 3 Note: If there are more digit maps than timeout values, the default value is used. If there are more timeout values than digit maps, the extra timeout values are ignored.	No

Dial Plan, Dial Plan Normalization, and Digit Map Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
	`dialplan.1.lyncDigitmap.timeOut`[1]	This parameter applies to lines registered with Skype for Business or Lync Server. Specify a timeout in seconds for each segment of a digit map. After you press a key, the phone waits the number of seconds you specify to match the digits to a dial plan and dial the call. 4 seconds (default) 0 to 99 seconds Note: If there are more digit maps than timeout values, the default value is used. If there are more timeout values than digit maps, the extra timeout values are ignored. Note also that if you configure a value outside of the permitted range, the default value is used. [1] Changes to the value of this parameter cause the phone to restart.	No
	`dialplan.userDial.timeOut`	Specify the time in seconds that the phone waits before dialing a number you enter while the phone is on hook. This parameter applies only when its value is lower than `up.IdleTimeOut`. 4 seconds (default) 0 to 99 seconds	No
sip-interop.cfg	`reg.1.applyServerDigitMapLocally`	1 (default) - Enable dial plan normalization. 0 - Disable dial plan normalization.	No

Dial Plan, Dial Plan Normalization, and Digit Map Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
	reg.1.applyServerDigitMapLocally	0 (default) - Dial plan rules are processed by Lync Server. 1 - Dial plan normalization rules are downloaded from the Lync Server and processed on the phone.	No
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

Supported Dial Plans

Polycom phones support Skype for Business External Access Prefix functionality.

Examples of supported dial plans include the following:

- Support for multiple combination of braces (): ^91(727|813)([2-9]\d{6})$@+9$1$2@0
- Support for 'ext': ^64(\d{2})$@+86411845933$1;ext=64$1@0

Supported Dial Plans

Number	Element	Meaning	Example	Description of Example
1	^	Match at beginning of string	^123	Match the digits 123 at the beginning of the string
2	()	Captures the matched subexpression	(456)	Capture what is between the parentheses into a numbered variable, starting at 1 which can be accessed as $n, for example, $1
3		Specifies zero or more matches	\d(*)	
4	+	Specifies one or more matches	\d(+)	

Supported Dial Plans

Number	Element	Meaning	Example	Description of Example
5	?	Specifies zero or one matches	\d(+)	
6	{n}	Specifies exactly n matches	\d {4}	Match 4 digits
7	Vertical Bar (Pipe)	Matches any one of the terms separated by the (vertical bar) character when all characters are surrounded by brackets or square brackets	(1|2|3) or [1|2|3]	Match either 1, 2, or 3.
8	\d	Matches any decimal digit	^\d	Match any decimal digit (at the beginning of a string)
9	$	The match must occur at the end of the string	^(123)$	Match exactly digits 123 (and not 1234)

Unsupported Dial Plans

Examples of dial plans not supported include the following:

- Braces within the braces with pipes: ^56(12(3|4))((4|5)6)@+1$2$1@0
- Non-sequential $ values in translation patters: ^1(45)(89)@+123$2$1@0

Unsupported Dial Plans

Number	Element	Meaning	Example	Description of Example
1	{,m}	Specifies at most m matches	\d {,6}	Match at most 6 digits
2	{n,}	Specifies at least n matches	\d {3,}	Match at least 3 digits (with no limit to number of digits matched)
3	{n,m}	Specifies at least n, but no more than m, matches	\d {3,6}	Match at least 3 digits but no more than 6 digits
4	$	The match must end at '$'	^(123$\|125$)	Match either the string 123 or the string 125

System Display

This section provides information on setting up features involving the phone’s user interface.

Capture Your Device's Current Screen

You can capture your phone or expansion module's current screen. Note that the Polycom Trio solution does 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.

To capture a device’s current screen:

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

User the following parameters to get a screen capture of the current screen on your phone or expansion module.

Device's Current Screen Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot\
`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

Time and Date Wizard

Users signing into Skype for Business on the Polycom Trio 8800 or 8500 system for the first time are prompted to set the time zone, time format, and date format before they start using the system. This feature is enabled by default.

Use the following parameters to enable or disable the Time and Date Wizard.

Time and Date Wizard Parameters

Parameter	Permitted Values
`device.set`	0 (default) - Do not use any `device.xxx` fields to set any parameters. 1 - Use the `device.xxx` fields that have `device.xxx.set=1`. Set this to 1 only for the initial installation and set back to 0 after the initial installation.
`device.lync.timeZone.set`	0 (default) - Do not use the `device.xxx` value. 1 (default) - Use the `device.xxx` value. For example, if `device.lync.timeZone.set=1`, then use the value set for `device.lync.timeZone`.
`device.lync.timeZone`	1 (default) - Skype for Business Time Zone Control is enabled. 0 - Skype for Business Time Zone Control is disabled.

Polycom Trio System Screen

This section provides an overview of icons and feature buttons you can display or hide on the Polycom Trio 8500 and 8800 system user interface. When using the Polycom Trio system with the Polycom Trio Visual+, you can also configure system information to display on the monitor connected to the Polycom Trio Visual+ system.

For more information about each option, refer to the section for that feature or search for a parameter.

Configuring the Polycom Trio User Interface

The following table lists parameters you can use to hide or display icons and features.

Phone Menu	Configuration Parameter	Permitted Values
Bluetooth	`feature.bluetooth.enabled` features.cfg	1 (default) - Bluetooth connection is enabled and the Bluetooth menu displays. 0 - Bluetooth connection is disabled.
Call Lists	`feature.callList.enabled` features.cfg	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.
Missed Calls	`feature.callListMissed.enabled` features.cfg	1 (default) - Missed calls show in the Missed Calls call list. 0 - Missed calls do not show in the Missed Calls list and you cannot clear existing entries.
Placed Calls	`feature.callListPlaced.enabled` features.cfg	1 (default) - Placed calls show in the Placed Calls call list. 0 - Placed calls do not show in the Placed Calls list and you cannot clear existing entries.
Received Calls	`feature.callListReceived.enabled` features.cfg	1 (default) - Received calls show in the Received Calls call list. 0 - Received calls do not show in the Received Calls list and you cannot clear existing entries.

Phone Menu	Configuration Parameter	Permitted Values
Contacts	`feature.contacts.enabled` features.cfg	1 (default) - Enable display of the Contacts icon displays on the Home screen, the global menu, and in the dialer. 0 - Disable display of the Contacts icon displays on the Home screen, the global menu, and in the dialer. Requires UCS 5.4.2 RevAA or higher.
DND icon	`homeScreen.doNotDisturb.enable` features.cfg	0 (default) - Disable display of the DND icon on the Home screen. 1 - Enable display of the DND icon on the Home screen.
Redial icon	`homeScreen.redial.enable` features.cfg	0 (default) - Disable display of the Redial icon on the Home screen. 1 - Enable display of the Redial icon on the Home screen.
Content	`homeScreen.present.enable` features.cfg	Control whether the Present icon displays on the Home screen when Content Sharing is enabled and the system is paired with Polycom Trio Visual+. 1 (default) 0
IP Address	`up.hideSystemIpAddress` features.cfg	Specify where the IP address of the Polycom Trio system and Polycom Trio Visual+ are hidden from view. You can access the IP address from the phone Advanced menu if you set this parameter to 'Menu' or 'Everywhere'. • Nowhere (default) - The IP addresses display on all user interfaces. • TV - IP addresses are hidden from the TV monitor. • HomeScreen - IP addresses are hidden from the TV monitor and phone menu. • Menus - IP addresses are hidden from the TV monitor, phone Home screen, and menu. • Everywhere - IP addresses are hidden from the TV monitor, phone Home screen, and menu.
Global Address Book	`feature.corporateDirectory.alt.enabled` features.cfg	0 (disable) - The global address book service is disabled. 1 - The global address book service is disabled.

Phone Menu	Configuration Parameter	Permitted Values
Corporate Directory	`feature.corporateDirectory.enabled` features.cfg	0 (default) - The corporate directory feature is disabled and the icon is hidden. 1 (default) - The corporate directory is enabled and the icon shows.
Calendar	`feature.exchangeCalendar.enabled` features.cfg	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.

Outlook Contacts	`feature.exchangeContacts.enabled`
	features.cfg
	`feature.lync.abs.enabled`
		The Outlook Search feature allows you to search and view Outlook Contacts and displays in the Contacts menu when the parameters are set as follows:
		`feature.exchangeContacts.enabled="1"`
		`feature.lync.abs.enabled="0"`

Voicemail menu	`feature.exchangeVoiceMail.menuLocation` features.cfg	Default (default) - Show the Voicemail menu in the global menu only when unread voicemails are available. After the voicemail is accessed, the Voicemail option no longer displays in the global menu and is accessible in the phone menu. Everywhere - Always show the Voicemail menu in the global menu and phone menu. MenusOnly - Show the Voicemail menu only in the phone Features menu.
Calendar	`homeScreen.calendar.enable` features.cfg	1 (default) - The Calendar icon on the Home screen displays. 0 - The Calendar icon does not display on the Home screen and is accessible from the dial pad.
Diagnostics	`homeScreen.diagnostics.enable` features.cfg	0 (default) - A Diagnostics icon does not show on the Home screen. 1 - A Diagnostics icon shows on the Home screen to provide quick access to the Diagnostics menu.

Phone Menu	Configuration Parameter	Permitted Values
Contacts	`homeScreen.directories.enable` features.cfg	1 (default) - Enable display of the Directories menu icon on the phone Home screen. 0 - Enable display of the Directories menu icon on the phone Home screen.
Settings	`homeScreen.settings.enable` features.cfg	1 (default) - The Settings menu icon displays on the Home screen and global menu. 0 - The Settings menu icon does not display on the Home screen and global menu. You require UC Software 5.4.2 RevAA or higher to hide the Settings icon from the global menu
Content-sharing graphic	`mr.bg.showWelcomeInstructions` features.cfg	All (default) - Display both the content-sharing graphic and welcome message on the Polycom Trio Visual+ monitor. TextOnly - Hide the content-sharing graphic. None - Hide both the content-sharing graphic and welcome message.
Basic Settings	`up.basicSettingsPasswordEnabled` features.cfg	0 (default) - No password is required to access the Basic settings menu. 1 - A password is required to access the Basic settings menu.
Date and Time	`up.localClockEnabled` features.cfg	1 (default) - The date and time display. 0 - The date and time do not display.
Voice Mail	`up.oneTouchVoiceMail` features.cfg	0 (default) - The phone displays a summary page with message counts. Users can press Connect to dial the voicemail server. 1 - The phone dials voicemail services directly, if available on the call server, and does not display the voicemail summary page.

Phone Theme

You can set the Polycom Trio system theme, the labels and colors that display on the system screen.

When the Polycom Trio system Base Profile is set to Skype, the Skype for Business theme displays by default.

Phone theme Parameters

Parameter Template	Permitted Values
`up.uiTheme` features.cfg	Default (default) - The phone displays the default Polycom theme. SkypeForBusiness - The phone displays the Skype for Business theme.

Polycom Trio System Display Name

The system name displays in the Global menu of the Polycom Trio 8500 and 8800 systems and on the monitor connected to the Polycom Trio Visual+ accessory paired with a Polycom Trio system. The system name also displays on any devices connected with the system wirelessly, such as Bluetooth, AirPlay-enabled devices.

By default, the system name displays as Polycom Trio <model number> (xxxxxx) where (xxxxxx) is the last six digits of the phone's MAC address. For example, Polycom Trio 8800 (01161C).

You can configure the name that displays on the system, the connected monitor, and any devices wirelessly connected to the system. The name you configure for the system, using any of the following parameters, displays in the subsequent priority order:

- `system.name`
- `displayname` - Set this parameter on the Skype for business server
- `reg.1.displayname`
- `reg.1.label`
- `reg.1.address`
- Default system name

If you set the system name using the `system.name` parameter, the value you set displays for the system unless you configure a name to display for a specific feature.

The system name you set using any of the following feature parameters takes precedence over the name set in `system.name`:

- AirPlay: `content.airplayServer.name`
- Bluetooth: `bluetooth.devName`
- Wireless Display: `content.wirelessDisplay.name`

System Display Name Parameters

Set the system name using one or more of parameters in the following table.

System Display Name Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
	`content.airplayServer.name`	Specify a system name for the local content sink for AirPlay certified devices. If left blank, the previously configured or default system name is used. NULL (default) UTF-8 encoded string	No
	`content.wirelessDisplay.sink.name`	Specify a system name for the local content sink for Android or Windows devices. If left blank the previously configured or default system name is used NULL (default) UTF-8 encoded string	No
`sip-interop.cfg`	`bluetooth.devName`	Enter the name of the system that broadcasts over Bluetooth to other devices. NULL (default) UTF-8 encoded string	
	`reg.1.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	
`reg-advanced.cfg`	`reg.1.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	N

System Display Name Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
	`reg.1.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) UTF-8 encoded string	No
	`system.name`	The system name that displays at the top left corner of the monitor, and at the top of the Global menu of the Polycom Trio system. String	No

Polycom Trio Solution IP Address

You can hide or choose where to show the IP addresses of the Polycom Trio systems and Polycom Trio Visual+ accessory.

Parameter Template	Permitted Values
`up.hideSystemIpAddress` features.cfg	Specify where the IP address of the Polycom Trio system and Polycom Trio Visual+ are hidden from view. You can access the IP address from the phone Advanced menu if you set this parameter to 'Menu' or 'Everywhere'. • Nowhere (default) - The IP addresses display on all user interfaces. • TV - IP addresses are hidden from the TV monitor. • HomeScreen - IP addresses are hidden from the TV monitor and phone menu. • Menus - IP addresses are hidden from the TV monitor, phone Home screen, and menu. • Everywhere - IP addresses are hidden from the TV monitor, phone Home screen, and menu.

Configure the Polycom Trio System Phone Number and Label

You can configure the Polycom Trio system phone number or label to display on the Home screen from the phone or using centralized provisioning parameters.

Configure the Phone Number or Label from the Phone

You can configure display of the phone number or label on the Home screen from the phone.

To configure display of the phone number of label on the Home screen from the phone:

» On the phone, go to **Settings > Advanced > Administration Settings > Home Screen Label.**

Configure the Phone Number or Label Using Centralized Provisioning Parameters

You can configure display of the phone number or label using centralized provisioning parameters.

Phone Number and Label Parameters

Parameter Template	Permitted Values
`homeScreen.customLabel` features.cfg	Specify the label to display on the phone's Home screen when `homeScreen.labelType`=Custom. The label can be 0 to 255 characters. Null (default)
`homeScreen.labelLocation` features.cfg	Specify where the label displays on the screen. StatusBar (default) - The phone displays the custom label in the status bar at the top of the screen. BelowDate - The phone displays the custom label on the Home screen only, just below the time and date.
`homeScreen.labelType` features.cfg	Specify the type of label to display on the phone's Home screen. PhoneNumber (default) • When the phone is set to use Skype Base Profile, the phone number is derived from the Skype for Business server. • When the phone is set to use the Generic Base Profile, the phone uses the number you specify in reg.1.address. Custom - Custom alphanumeric string between 0 and 255 characters. None - Don't display a label.
`reg.1.useteluriAsLineLabel`	1 - If `reg.x.label="Null"` the tel URI/phone number/address displays as the label of the line key. 0 - If `reg.x.label="Null"` 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.
`up.formatPhoneNumbers` features.cfg	1 (default) - Enables automatic number formatting. 0 - Disables automatic number formatting and numbers display separated by "-".

Status Messages

You can choose to display a maximum of five multi-line messages in the Polycom Trio Visual+ Status Bar. Each message can contain a maximum of 64 characters. If the length of the message exceeds the size of the status bar, the message wraps into multiple lines.

When you configure multiple messages, you can adjust the number of seconds each message displays.

Status Message Parameters

Parameter Template	Permitted Values
`up.status.message.flash.rate` features.cfg	Specify the number of seconds to display a message before moving to the next message. 2 seconds (default) 1 - 8 seconds

`up.status.message.1`
`up.status.message.2`
`up.status.message.3`
`up.status.message.4`
`up.status.message.5`
	<message line one>
	<message line two>
	<message line three>
	<message line four>
	<message line five>

System Name for Wireless Content Connections

When the Polycom Trio 8800 system is registered with Skype for Business, you can determine if the Skype for Business display name shows in the list of available devices that supported Apple, Windows, or Android devices can connect with to share content using AirPlay or Miracast.

The Airplay or Miracast devices search for a system name, which you can specify in one or more of the following parameters. The device displays the name you configure for the system in the following priority in order:

- `content.airplayServer.name` for AirPlay or `content.wirelessDisplay.name` for Miracast
- `system.name`
- `displayname` (on the Skype for Business server)
- `reg.1.displayname`
- `reg.1.label`
- `reg.1.address`
- Default system name based on the model and MAC address

System Name for Wireless Content Connections Parameters

Set the system name using one or more of the following parameters.

System Display Name for Wireless Content Connections

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
	`content.airplayServer.name`	Specify a system name for the local content sink for AirPlay certified devices. If left blank, the system name specified by the system.name parameter is used. NULL (default)	No
	`content.wirelessDisplay.sink.name`	Specify a system name for the local content sink for Miracast-certified devices. If left blank, the system name specified by the system.name parameter is used. NULL (default)	No
	`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

System Display Name for Wireless Content Connections

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
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
	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.	No

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.

Use the values in the following table to set the time zone location description. The default value is NULL.

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 32 (GMT -3:00) Salvador 33 (GMT -3:00) Santiago 34 (GMT -2:00) Mid-Atlantic 35 (GMT -1:00) Azores 36 (GMT -1:00) Cape Verde Islands 37 (GMT 0:00) Western Europe Time 38 (GMT 0:00) London,Lisbon 39 (GMT 0:00) Casablanca 40 (GMT 0:00) Dublin	91 (GMT +5:30) Sri Jayawardenepura 92 (GMT +5:45) Kathmandu 93 (GMT +6:00) Astana,Dhaka 94 (GMT +6:00) Almaty 95 (GMT +6:00) Novosibirsk (RTZ 5) 96 (GMT +6:30) Yangon (Rangoon) 97 (GMT +7:00) Bangkok,Hanoi 98 (GMT +7:00) Jakarta 99 (GMT +7:00) Krasnoyarsk (RTZ 6) 100 (GMT +8:00) Beijing,Chongqing
41 (GMT 0:00) Edinburgh 42 (GMT 0:00) Monrovia 43 (GMT 0:00) Reykjavik 44 (GMT +1:00) Belgrade 45 (GMT +1:00) Bratislava 46 (GMT +1:00) Budapest 47 (GMT +1:00) Ljubljana 48 (GMT +1:00) Prague 49 (GMT +1:00) Sarajevo,Skopje 50 (GMT +1:00) Warsaw,Zagreb	101 (GMT +8:00) Hong Kong,Urumqi 102 (GMT +8:00) Kuala Lumpur 103 (GMT +8:00) Singapore 104 (GMT +8:00) Taipei,Perth 105 (GMT +8:00) Irkutsk (RTZ 7) 106 (GMT +8:00) Ulaanbaatar 107 (GMT +9:00) Tokyo,Seoul,Osaka 108 (GMT +9:00) Sapporo,Yakutsk (RTZ 8) 109 (GMT +9:30) Adelaide,Darwin 110 (GMT +10:00) Canberra
51 (GMT +1:00) Brussels 52 (GMT +1:00) Copenhagen 53 (GMT +1:00) Madrid,Paris 54 (GMT +1:00) Amsterdam,Berlin 55 (GMT +1:00) Bern,Rome 56 (GMT +1:00) Stockholm,Vienna 57 (GMT +1:00) West Central Africa 58 (GMT +1:00) Windhoek 59 (GMT +2:00) Bucharest,Cairo 60 (GMT +2:00) Amman,Beirut	111 (GMT +10:00) Magadan (RTZ 9) 112 (GMT +10:00) Melbourne 113 (GMT +10:00) Sydney,Brisbane 114 (GMT +10:00) Hobart 115 (GMT +10:00) Vladivostok 116 (GMT +10:00) Guam,Port Moresby 117 (GMT +11:00) Solomon Islands 118 (GMT +11:00) New Caledonia 119 (GMT +11:00) Chokurdakh (RTZ 10) 120 (GMT +12:00) Fiji Islands
	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

Network

Polycom UC Software enables you to make custom network configurations.

Near Field Communication (NFC)-Assisted Bluetooth

The Polycom Trio 8800 and 8500 systems support Bluetooth.

Only the Polycom Trio 8800 supports near-field communication (NFC)-assisted Bluetooth pairing. This feature is disabled by default.

When you enable Bluetooth, users can connect a Bluetooth-capable device, such as a mobile phone, tablet, or laptop to the Polycom Trio system. You can make calls from the connected device and play audio from calls, video, or music from the Polycom Trio system speaker. Note you can connect one device at a time to the Polycom Trio system via Bluetooth. You cannot connect via Bluetooth during an active call. The Polycom Trio 8800 conference phone can remember up to 10 previously paired devices.

When NFC is enabled on the Polycom Trio 8800 system and you connect a personal device to the Polycom Trio 8800, the NFC logo displays on the device screen. When your device is connected over Bluetooth during an audio or a video call, you can use the Polycom Trio system microphones for audio instead of the microphone(s) of your connected device.

Enable NFC-Assisted Bluetooth from the Phone Menu

You can enable or disable NFC Mode from the Polycom Trio 8800 system.

To enable NFC Mode:

1. Go to **Settings > Advanced > Administrator Settings > NFC Mode**.
2. Press the NFC sensor to the left of the Polycom Trio 8800 screen.

   The phone prompts you to confirm pairing.

Configuring NFC-Assisted Bluetooth

Use the parameters in the following table to configure Bluetooth and NFC Mode on the Polycom Trio 8800 system.

Bluetooth and NFC Parameters

Parameter Template	Permitted Values
`bluetooth.devName` new.cfg, sip-interop.cfg	NULL (default) UTF-8 string Enter the name of the device that broadcasts over Bluetooth to other devices.
`bluetooth.discoverableTimeout` new.cfg, features.cfg	0 (default) - Other devices can always discover this device over Bluetooth. 0 - 3600 seconds Set the time in seconds after which other devices can discover this device over Bluetooth.
`bluetooth.radioOn` features.cfg	0 (default) - The Bluetooth radio (transmitter/receiver) is off. 1 - The Bluetooth radio is on. The Bluetooth radio must be turned on before other devices can connect to this device over Bluetooth.
`feature.bluetooth.enabled` features.cfg	For high security environments. 1 (default)- The Bluetooth feature is enabled. 0 - The Bluetooth feature is disabled.
`feature.nfc.enabled` features.cfg	0 - The NFC pairing feature is disabled. 1 - The NFC pairing is enabled and users can pair NFC-capable devices to the Polycom Trio 8800 solution.

Wireless Network Connectivity (Wi-Fi)

The Polycom Trio 8800 system supports various wireless modes, security options, radio controls, and Quality of Service monitoring. To ensure the best performance in your location, set a proper country code with the parameter `device.wifi.country` before enabling Wi-Fi.

Enabling Wi-Fi automatically disables the Ethernet port. You cannot use Wi-Fi and Ethernet simultaneously to connect Polycom Trio 8800 system to your network. When you connect to your network over Wi-Fi, only audio-only calls are available. Note that Polycom Trio 8800 system does not support Wi-Fi captive portals or Wireless Display (WiDi).

Note: When you provision the Polycom Trio 8800 system via Wi-Fi connection to the network, the Polycom Trio system looks for files on the provisioning server using the LAN MAC address and not the Wi-Fi MAC address.

The Polycom Trio 8800 system supports the following wireless modes:

- 2.4 GHz / 5 GHz operation
- IEEE 802.11a radio transmission standard
- IEEE 802.11b radio transmission standard
- IEEE 802.11g radio transmission standard
- IEEE 802.11n radio transmission standard

Note: You cannot use Polycom Trio Visual+ for video calls when you connect Polycom Trio 8800 to your network using Wi-Fi. The Polycom Trio 8800 and Polycom Trio Visual+ do not pair when the Polycom Trio 8800 is connected to your network using Wi-Fi.

Enable Wi-Fi on the Polycom Trio 8800 System

You can wirelessly connect the Polycom Trio 8800 to your network using Wi-Fi, which is disabled by default. When you enable Wi-Fi, the system reboots.

To enable Wi-Fi from the Polycom Trio 8800:

1 Go to **Settings > Advanced > Administration Settings > Network Configuration > Network Interfaces > Wi-Fi Menu**, and turn Wi-Fi **On**.

   The phone restarts.

2 When the phone completes restart, go to **Settings > Advanced > Administration Settings > Network Configuration > Network Interfaces > Wi-Fi Menu** to view available networks.

3 Select a network you want to connect to and press **Connect**.

Configuring Wi-Fi

The parameters you configure depend on the security mode of your organization and whether or not you enable DHCP. Polycom Trio 8800 system is shipped with a security-restrictive worldwide safe Wi-Fi country code setting.

The Polycom Trio system supports the following Wi-Fi security modes:

- WEP
- WPA PSK
- WPA2 PSK
- WPA2 Enterprise

Configure Wi-Fi Network Parameters

Parameter Function	**template** > parameter
Enable the Wi-Fi radio.	device.wifi.enabled
Enter the two-letter code for the country in which you enable the Wi-Fi radio.	device.wifi.country
Enable DHCP for Wi-Fi.	device.wifi.dhcpEnabled
	device.wifi.dhcpBootServer
The IP address of the wireless device if not using DHCP.	device.wifi.ipAddress
The network mask address of the wireless device if not using DHCP.	device.wifi.subnetMask

Configure Wi-Fi Network Parameters (continued)

Parameter Function	**template** > parameter
The IP gateway address of the wireless device if not using DHCP.	device.wifi.ipGateway
The SSID of the wireless network.	device.wifi.ssid
Specify the wireless security mode.	device.wifi.securityMode
The length of the hexadecimal WEP key.	device.wifi.wep.key
The hexadecimal key or ASCII passphrase.	device.wifi.psk.key
The EAP to use for 802.1X authentication.	device.wifi.wpa2Ent.method
The WPA2-Enterprise user name.	device.wifi.wpa2Ent.user
The WPA2-Enterprise password.	device.wifi.wpa2Ent.password
	device.wifi.radio.enable2ghz
	device.wifi.radio.enable5ghz

Extended Link Layer Discovery Protocol (LLDP)

The Link Layer Discovery Protocol (LLDP) is used by network devices to advertise their identity, capabilities, and neighbors on an IEEE 802 local area network, principally wired Ethernet. LLDP is enabled by default.

Media Endpoint Discover (MED) capabilities include:

- Network policy discover
- Endpoint location identification discovery
- Extender power discovery required for endpoint

Configuring LLDP Fast Start Count

Fast start count enables a device to initially advertise itself over the network at a fast rate for a limited time when an LLDP-MED endpoint has been newly detected or connected to the network.

LLDP Parameters

Parameter Template	Permitted Values
`device.net.lldpFastStartCount` device.cfg, site.cfg	Configure the fast-start LLDP packets that the phone sends when booting up or when the network comes up. 5 (default) 3 - 10 If fast-start packet count is configured > 10 the, the value resets to 10. If the fast-start packet count is < 3, the value resets to 3. If you configure an invalid value-for example, a negative value, string, or character-the value resets to default 5.

STUN / TURN / ICE Parameters

This section lists parameters that configure the following Microsoft network features:

- Session Traversal Utilities for NAT (STUN)
- Traversal Using Relays Around NAT (TURN)
- Interactive Connectivity Establishment (ICE)

STUN / TURN / ICE Parameters

Template	Parameter Template	Permitted Values	Change Causes Restart or Reboot
`firewall-nat.cfg`	`tcpIpApp.ice.password`	Enter the password to authenticate to the TURN server. NULL (default)	No
`firewall-nat.cfg`	`tcpIpApp.ice.mode`	MSOCS (default) Disabled Standard	No
`firewall-nat.cfg`	`tcpIpApp.ice.stun.server`	Enter the IP address of the STUN server. NULL (default)	No
`firewall-nat.cfg`	`tcpIpApp.ice.stun.udpPort`	The UDP port number of the STUN server. 3478 (default) 1 - 65535	No
`firewall-nat.cfg`	`tcpIpApp.ice.tcp.enabled`	1 (default) - Enable TCP. 0 - Disable TCP.	No
`firewall-nat.cfg`	`tcpIpApp.ice.turn.server`	Enter the IP address of the TURN server. NULL (default)	No

STUN / TURN / ICE Parameters

Template	Parameter Template	Permitted Values	Change Causes Restart or Reboot
`firewall-nat.cfg`	`tcpIpApp.ice.turn.tcpPort`	443 (default) 1 - 65535	No
`firewall-nat.cfg`	`tcpIpApp.ice.turn.udpPort`	The UDP port number of the TURN server. 443 (default) 65535	No
`firewall-nat.cfg`	`tcpIpApp.ice.username`	Enter the user name to authenticate to the TURN server. NULL (default)	No
`firewall-nat.cfg`	`tcpIpApp.ice.policy`	The default policy is set as per the phone model. Default (default) DefaultVVX201 DefaultVVX300 DefaultVVX301 DefaultVVX310 DefaultVVX311 DefaultVVX400 DefaultVVX401 DefaultVVX410 DefaultVVX411 DefaultVVX500 DefaultVVX501 DefaultVVX600 Legacy – Support the legacy behavior of ICE stack. Custom – Tune the following ICE parameters according to network conditions: • `tcpIpApp.ice.NetworkMode` • `tcpIpApp.ice.MaxCandidateGatheringInParallel` • `tcpIpApp.ice.MaxConnectivityChecksInParallel` • `tcpIpApp.ice.ConnCheckInetvalPairs` • `tcpIpApp.ice.ConnCheckInetvalRetries` • `tcpIpApp.ice.ReflexiveChecksRequired` • `tcpIpApp.ice.MaxRetries`	No

STUN / TURN / ICE Parameters

Template	Parameter Template	Permitted Values	Change Causes Restart or Reboot
firewall-nat.cfg	tcpIpApp.ice.NetworkMode	TCPUDP (default) – Gathers all the possible UDP and TCP ice candidates. TCPOnly – Gathers all the TCP candidates along with UDP host candidates. UDPOnly - Gathers all the UDP candidates.	No
firewall-nat.cfg	tcpIpApp.ice.MaxCandidateGatheringInParallel	The number of ICE candidates gathering threads run in parallel in the maximum time span of 2 seconds for simultaneous incoming calls only. 2 (default) 2 – 24 The default value for `tcpIpApp.ice.MaxCandidateGatheringInParallel` parameter is set to 3 when using VVX 201 business media phone. For all other VVX platforms, the default value is set to 5.	No
firewall-nat.cfg	tcpIpApp.ice.MaxConnectivityChecksInParallel	The number of ICE connectivity checks threads run in parallel in the maximum time span of 30 seconds (connectivity checks will be complete in 1 sec after answering call in general) for simultaneous incoming calls only. 2 (default) 1 – 24 The default value for `tcpIpApp.ice.MaxConnectivityChecksInParallel` parameter is set as follows when using the corresponding VVX platforms: • For VVX 201, 300, 310, 400 phones, value is set to 1. • For VVX 410 phone, value is set to 2. • For VVX 600 phone, value is set to 3. • For VVX 301, 311, 401, 411, 500 phones, value is set to 5. • For VVX 501 and 601 phones, value is set to 7.	No

STUN / TURN / ICE Parameters

Template	Parameter Template	Permitted Values	Change Causes Restart or Reboot
firewall-nat.cfg	tcpIpApp.ice.ConnCheckInetvalPairs	Time interval to serialize first attempt of connectivity check of identified ice candidate pairs per call. 25 (default) 25 – 100 ms	No
firewall-nat.cfg	tcpIpApp.ice.ConnCheckInetvalRetries	Time interval to serialize the retry attempts of connectivity check for identified pairs per call. 50 (default) 25 – 100 ms The default value for tcpIpApp.ice.ConnCheckInetvalRetries parameter is set to 100 when using any VVX platform.	No
firewall-nat.cfg	tcpIpApp.ice.ReflexiveChecksRequired	To determine whether reflexive candidates to be collected as part of ice candidates collection. 1 (default) - TCP and UDP reflexive candidates will be collected in candidate gathering process. 0 - TCP and UDP reflexive candidates will not be collected in candidate gathering process.	No
firewall-nat.cfg	tcpIpApp.ice.MaxRetries	The maximum number of retry attempts performed on each ICE connectivity check pair identified in case of a request timeout or upon failure. 2 (default) 2 – 24 The default value for tcpIpApp.ice.MaxRetries parameter is set to 5 when using any VVX platform.	No

Hardware and Accessories

This section provides information on configuring phone hardware.

Powering the Polycom Trio 8500 and 8800 Systems

Powering requirements and options vary between the Polycom Trio 8500 and 8800 systems. Read the powering requirements and options carefully to understand powering for your Polycom Trio system.

Powering the Polycom Trio 8800

You can power the Polycom Trio 8800 system with Power over Ethernet (PoE) or PoE+ (IEEE 802.3at Type 2). When the Polycom Trio 8800 system is booting up, an on-screen message indicates the available power supply type. Note that PoE+ provides Polycom Trio systems with full functionality.

The following features are not available on Polycom Trio 8800 system when using PoE:

- The Polycom Trio 8800 system LAN OUT port out does not provide PoE+ power and cannot be used to power the Polycom Trio Visual+.
- No USB charging is provided to devices (mobile phones, tablets) connected to the Polycom Trio 8800 system USB port.
- Maximum peak power to the loudspeaker is limited.

Powering the Polycom Trio 8500

You can power the Polycom Trio 8500 system with Power over Ethernet (PoE). When the Polycom Trio 8500 system is booting up, an on-screen message indicates the available power supply type.

The Polycom Trio 8500 does not support:

- PoE+
- Power Sourcing Equipment (PSE)
- LAN Out / PC Port
- USB

The following features are not available on Polycom Trio 8500 system using PoE:

- No USB charging is provided to devices (mobile phones, tablets) connected to the Polycom Trio 8500 system USB port.

- Maximum peak power to the loudspeaker is limited.

Power the Polycom Trio System with the Optional Power Injector

If your building is not equipped with PoE**+** you can use the optional power injector to enable full functionality on your Polycom Trio 8800 and 8500 system.

If you are using the power injector to power the Polycom Trio 8800 and your setup includes the Polycom Trio Visual+, connect the Polycom Trio 8800 in the following sequence before powering up the Polycom Trio 8800 and Visual+.

1 Plug the AC power cord of the power injector into the wall and use a network cable to connect the power injector to the Polycom Trio 8800.

2 Connect the power injector to the network with a CAT-5E or CAT-6 Ethernet cable.

The power adapter LED is green when the Polycom Trio 8800 is correctly powered. If the LED is yellow, the power injector is bypassed and the Polycom Trio system is drawing PoE power from the outlet.

Note: If the Polycom Trio Visual+ loses power after a Polycom Trio 8800 reboot, unplug both devices and repeat steps 1 and 2.

3 If the power injector LED is yellow, turn off the PoE network port or connect the Polycom Trio solution in the following sequence:

a Power up Polycom Trio 8800 and Visual+ using the power injector but do not plug the devices into the network wall port.

b Wait for the Polycom Trio 8800 and Visual+ systems to boot up

c Plug the devices into the network wall port.

d Ensure the LED indicator on the power injector is green.

Pairing the Polycom Trio Visual+ with Polycom Trio

Pair the Polycom Trio Visual+ with Polycom Trio 8500 or 8800 system to enable video calls and content sharing. You can pair only one Polycom Trio Visual+ to a Polycom Trio 8500 or 8800 system. Polycom recommends you plug both devices into a local gigabit switch.

You can pair the Polycom Trio Visual+ to the Polycom Trio system using configuration files or from the Polycom Trio menu system. To pair, the Polycom Trio system and Polycom Trio Visual+ must be connected to the same subnet and you must unblock the following network components:

- Multicast address 224.0.0.200
- Port 2000

Note: You cannot use Polycom Trio Visual+ for video calls when you connect the Polycom Trio 8800 system to your network using Wi-Fi. The Polycom Trio 8800 system and Polycom Trio Visual+ pair only when the Polycom Trio system is connected to your network over Ethernet.

Pair the Polycom Trio System Manually

You can manually pair the Polycom Visual+ to the Polycom Trio system from the Polycom Trio system menu.

To pair Polycom Trio Visual+ with Polycom Trio system manually:

1 Set up Polycom Trio Visual+. For instructions, refer to the Polycom Trio system Setup Sheet.

The Welcome screen displays on your monitor and indicates steps to pair with the Polycom Trio system.

2 Tap the **Pair** button on Polycom Trio Visual+ to broadcast discovery to the Polycom Trio.

3 On the Polycom Trio system, go to **Settings > Advanced > Networked Devices**, and ensure that **Notification of New Devices** is **On**.

4 Choose one of the following:

- If you have not paired the device before, tap **Pair with New Device**, tap the device you want to pair from the Discovered Devices list, and in the Details screen, tap **Pair**. (All currently paired devices display under Paired Devices.)
- If the device has been paired before, select the device from the **Available Devices** list and tap **Pair**.

5 When you see the message prompting you to complete pairing, do one of the following:

- Tap **Complete**.
- Tap the **Pair** button on the Polycom Visual+

If pairing was successful, a success message displays on the monitor along with a self-view window, the LED light on the Polycom Trio Visual+ device is continuously green, and a paired icon displays on the phone.

If pairing was not successful, a message displays on the monitor that the devices could not pair.

After successful pairing, if devices become disconnected for 60 seconds, a message displays that the devices have temporarily lost connection.

Polycom Trio System Pairing Parameters

To pair using configuration files, enter the MAC address of your Polycom Visual+ as the value for the parameter `mr.pair.uid.1`. The MAC address can be in either of the following formats:

- `00e0d::B09128D`
- `00E0DB09128D`

Use the following parameters to configure this feature and additional feature options.

Pairing Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	mr.audio.srtp.require	If set to 1, SRTP is used to encrypt and authenticate modular room audio signals sent between Polycom Trio 8500 or 8800 and Polycom Trio Visual+. 1 (default) 0	No
features.cfg	mr.bg.selection	Sets the background image for the paired Polycom Trio Visual+ display. HallstatterSeeLake (default) Auto - Automatically cycles through background images 2, 3, 4. The background image changes each time a video call ends. BlueGradient BavarianAlps ForgetMeNotPond Custom - Use a custom background specified by mr.bg.url.	No
features.cfg	mr.bg.showPlcmLogo	1 (default) - The Polycom logo shows on the TV attached to the paired Polycom Trio Visual+. 0 - Hides the Polycom logo on the Polycom Trio Visual+.	No
features.cfg	mr.bg.url	Specifies the HTTP URL location of a background image to use on the TV attached to the paired Polycom Trio Visual+. The system supports PNG and JPEG images up to 2.9 MB. This background image will be used only if mr.bg.selection= "5" Null (default) String (maximum 256 characters)	No

Pairing Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	mr.pair.tls.enabled	1 (default) - Use TLS to communicate between Polycom Trio 8500 or 8800 and Polycom Trio Visual+ systems. 0 - Does not use TLS to communicate between Polycom Trio 8500 or 8800 and Polycom Trio Visual+ systems.	No
features.cfg	mr.pair.uid.1	Enter the MAC address of the Polycom Trio Visual+ you want to pair with. Null (default) String (maximum of 64 characters)	No
features.cfg	mr.PairButton.notification	0 (default) - The Polycom Trio 8500 or 8800 system is notified when you press the Pair button on the Polycom Trio Visual+. 1 - The Polycom Trio 8500 or 8800 system is not notified when you press the Pair button on the Polycom Trio Visual+.	No
features.cfg	mr.video.camera.focus.auto	0 (default) - Disable the camera's automatic focus. 1 - Enable the camera's automatic focus.	No
features.cfg	mr.video.camera.focus.range	Specify the distance to the camera's optimally-focused target. 0 (default) 0 - 255	No
features.cfg	mr.video.iFrame.minPeriod	Choose the minimum time in seconds between transmitted video i-Frames or transmitted i-Frame requests. 2 (default) 1 - 60	No

Pairing Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`smartPairing.mode`	Enables users with Polycom Desktop on a laptop or Polycom Mobile on a tablet to pair with the Polycom Trio system using SmartPairing. disabled (default) manual	No
`features.cfg`	`smartPairing.volume`	The relative volume to use for the SmartPairing ultrasonic beacon. 6 (default) 0 - 10	No

Identify Paired Devices

If you are using multiple Polycom Trio systems and are not sure which Polycom Trio Visual+ it is paired with which, you can identify which devices are paired on the Polycom Trio system screen.

To identify paired devices:

1 On the Polycom Trio system, go to **Settings > Advanced > Networked Devices**, and ensure that **Notification of New Devices** is **On**.
2 Select a device that displays under Paired Devices or Available Devices.
3 Tap **Identify** to flash the LED of the device you selected.

Place the Polycom Trio Visual+ in Pairing Diagnostic Mode

If you are using multiple Polycom Trio systems and are not sure which Polycom Trio Visual+ it is paired with which, you can place the Polycom Trio Visual+ devices in pairing diagnostic mode to distinguish between accessories.

To enter pairing diagnostic mode:

1 Power up the Polycom Trio Visual+ device.
2 Wait for the initial LED on state to turn off.
3 Press and hold the pairing button until the LED turns orange.
4 Release the pairing button.
5 The LED blinks.
6 Wait for the device to reboot.
7 The paired Pod LED is steady green.

Polycom Trio System Power Management

Power available to the Polycom Trio 8500 and 8800 systems is limited and you must choose how to power the system and which features to enable or disable. You need to consider these powering options even if you are powering from a POE+ source.

USB Port Power Management

Device charging with the USB port on the Polycom Trio system is disabled by default and when disabled the USB host port provides 100mA of power for peripheral devices. USB charging is disabled when powering the Polycom Trio Visual+ from a LAN Out port.

To enable USB charging, you must power your Polycom Trio system with an IEEE 802.3at Power over Ethernet **Plus** (PoE**+**) compliant power source. When USB charging is enabled, you can power and charge USB 2.0 compliant devices having a power draw of up to 1.500mA/7.5W.

Using Power over Ethernet (POE) Class 0

Powering the Polycom Trio system from a Power over Ethernet (POE) Class 0 source provides full core functionality and results in the following limitations:

- The LAN Out port does not provide PoE power but otherwise is fully functional.

Using Power Sourcing Equipment Power (PoE PSE Power)

You can use Power Sourcing Equipment Power (PoE PSE Power) to power a Polycom Trio Visual+ system from the LAN OUT port of the Polycom Trio 8800 system.

To use PoE PSE Power, you must power the Polycom Trio 8800 system with an IEEE 802.3at Power over Ethernet **Plus** (PoE**+**) compliant power source.

Note: You cannot enable USB Charging of the USB host port and PSE PoE Power of LAN OUT port at the same time. If both are enabled, the Polycom Trio system uses PSE PoE Power and ignores the USB charging setting.

Polycom Trio System Power Management

Use the parameters listed to manage the Polycom Trio system's power usage.

Parameter template	Permitted Values
`poe.pse.class` new.cfg	Specify the LAN OUT PoE class. 0 (default) 0 - 3
`poe.pse.enabled` new.cfg	1 (default) - The Polycom Trio 8800 LAN OUT interface provides PoE power to a connected device. 0 - PoE power is not provided by the LAN OUT port.
`usb.charging.enabled` new.cfg	0 (default) - You cannot charge USB-connected devices from the USB charging port. 1 - Enable fast charging of devices connected by USB port up to 7.5W power / 1.5A current.

Consumer Electronics Controls (CEC) over HDMI

Consumer Electronics Control (CEC) enables monitor standby on Polycom Trio 8500 and 8800 systems when using the Polycom Trio Visual+ system to connect to CEC-capable monitors with HDMI. Check the feature settings and sub-settings on your monitor to verify that your monitor supports CEC.

When you enable CEC, any connected CEC-capable monitors switch to standby mode to save power when the Polycom Trio system enters standby mode. When the system awakes, the monitors are powered up before displaying Polycom Trio system video. Use of system standby requires CEC-capable monitors. Note that not all HDMI monitors support CEC.

During startup, the Polycom Trio Visual+ system might display messages about potential issues when a monitor is connected to the HDMI port, for example:

- HDMI connectivity issues. The Polycom Visual+ message may indicate monitor capability issues. Check the HDMI connection and replace the HDMI cable if necessary.
- Low resolution monitor. The Polycom Visual+ message indicates when a low resolution monitor is connected. Use a full HD monitor if possible.

Note: CEC features can vary by the brand of monitor. Specifically, some monitors have sub-feature settings under the main CEC setting that control whether or not the monitor responds to CEC commands. Ensure that you enable all CEC features and sub-features on all monitors connected to the Polycom Trio systems.

Configure Consumer Electronics Control (CEC) using the Web Configuration Utility

You can enable or disable CEC on Polycom Trio systems using the Web Configuration Utility.

To enable CEC using the Web Configuration Utility:

1. Enter the IP address of the Polycom Trio 8800 system you are using to a web browser.
2. Log into the Web Configuration Utility as an administrator.
3. Go to **Settings > Networked Devices > Power Saving Settings**.
4. Beside **Consumer Electronic Control**, select **Enable** or **Disable**.

Consumer Electronics Controls (CEC) over HDMI Parameters

The following parameters configure CEC when using HDMI.

CEC Parameters

Parameter Template	Permitted Values
`powerSaving.cecEnable` new.cfg	0 (default) - The Polycom Trio Visual+ display behavior is controlled only by the value set for `powerSaving.tvStandbyMode`. 1 - When the Polycom Trio system enters power-saving mode, the Polycom Trio Visual+ display switches to standby mode and powers up when the Polycom Trio system exits power-saving mode.

Power-Saving

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

You can configure power-saving options for the Polycom Trio 8500 and 8800 system including:

- Turn on power-saving during non-working days and hours
- Configure power-saving around working days and hours
- Configure an idle inactivity time after which the phone enters power-saving mode

When the phone is in power-saving mode, an LED light flashes at intervals to indicate power is on.

By default the Polycom Trio 8500 and 8800 systems enter power-saving mode after a period of idle time to conserve energy. However, Polycom Trio systems do not enter power-saving mode while idle in the Bluetooth menu. To ensure the system enters power-saving mode, you must exit the Bluetooth menu using the **Home** or **Back** key on the Bluetooth menu.

Power-Saving Parameters

Parameter Template	Permitted Values
`powerSaving.enable` site.cfg	1 (default) - Enable the LCD power-saving feature. 0 - Disable The LCD power-saving feature.
`powerSaving.idleTimeout.offHours` site.cfg	The number of idle minutes during off hours after which the phone enters power saving. 1 (default) 1 - 10
`powerSaving.idleTimeout.officeHours` site.cfg	The number of idle minutes during office hours after which the phone enters power saving. 30 (default) 1 - 600
`powerSaving.idleTimeout.userInputExtension` site.cfg	The number of minutes after the phone is last used that the phone enters power saving. 10 (default) 1 - 20

`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

`powerSaving.officeHours.startHour.x` site.cfg	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).
`powerSaving.tvStandbyMode` site.cfg	black (default) - The Polycom Trio Visual+ displays a black screen after entering power-saving mode. noSignal - Power-saving mode turns off the HDMI signal going to the Polycom Trio Visual+ monitor.

Device and Software Support

This section provides information on updating and maintaining your devices and the UC Software.

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.

The Updater, UC Software executable, and configuration files can all be updated using centralized provisioning.

In-Band Provisioning

You can configure Polycom Trio 8500 and 8800 system to accept or block in-band provisioning settings sent from the Skype for Business server.

In-Band Provisioning Parameters

Parameters Template	Permitted Values
`lync.provisionDeviceParams.enabled`	1 (default) - Enable (accept) in-band provisioning device settings sent from Skype for Business. 0 - Disable (block) in-band provisioning device settings sent from Skype for Business. When set to 0, the following in-band provisioning device settings are blocked: • EnableDeviceUpdate • IPPhoneAdminPasswd • LocalProvisioningServerAddress • LocalProvisioningServerUser • LocalProvisioningServerPassword

Data Center Resiliency

Data Center Resiliency ensures that minimum basic call functions remain available in the event of a server shutdown or Wide area network (WAN) outage. This features is available with the following:

- Polycom Trio 8500 and 8800 system

Phones you register with Skype for Business on-premises are enabled with this feature by default and no additional configuration is required.

In the event of an unplanned server shutdown or outage, phone behavior changes to the following:

- The phone displays a scrolling banner message 'Limited functionality due to outage'.
- Your presence status displays as 'Unknown'.
- The presence status of your contacts displays as 'Unknown'.
- You cannot change your presence status.
- You cannot add or delete Skype for Business contacts.
- Phones in the locked state display a message on the Sign In menu 'Limited functionality due to outage'.
- You can access current Call Forwarding settings in read-only mode.

Polycom Experience Cloud

The Polycom Experience Cloud (PEC) service is an experimental feature that allows your Polycom Trio solution to share basic diagnostic and phone usage data including start and stop events, call quality information, packet statistics, call duration, and call logs with Polycom.

Experience Cloud Parameters

Parameter Template	Permitted Values
`log.level.change.apps`	Initial logging level for the Applications log module. 4 (default) 0 - 6
`log.level.change.bfcp`	Initial logging level for the BFCP content log module. 4 (default) 0 - 6
`log.level.change.pec`	Initial logging level for the Polycom Experience Cloud (PEC) log 4 (default) 0 - 6
`log.level.change.mr`	Initial logging level for the Networked Devices log module. 4 (default) 0 - 6
`log.level.change.mraud`	Initial logging level for the Networked Devices Audio log module. 4 (default) 0 - 6
`log.level.change.mrcam`	Initial logging level for the Networked Devices Camera log module. 4 (default) 0 - 6
`log.level.change.mrcon`	Initial logging level for the Networked Devices Connection log module. 4 (default) 0 - 6

Experience Cloud Parameters

Parameter Template	Permitted Values
`log.level.change.mrdis`	Initial logging level for the Networked Devices Display log module. 4 (default) 0 - 6
`log.level.change.mrmgr`	Initial logging level for the Networked Devices Manager log module. 4 (default) 0 - 6
`log.level.change.ppcip`	Initial logging level for the Polycom People+Content IP log module. 4 (default) 0 - 6
`log.level.change.prox`	Initial logging level for the Proximity log module. 4 (default) 0 - 6
`log.level.change.ptp`	Initial logging level for the Precision Time Protocol log module. 4 (default) 0 - 6
`log.level.change.usba`	Set the logging detail level for the USB audio log. 4 (default) 0 - 6
`log.level.change.usbh`	Set the logging detail level for the USB HID log. 4 (default) 0 - 6

Client Media Port Ranges for QoE

To help deploy QoE, you can enable client media ports and configure unique port ranges on the Skype for Business Server.

To configure client media port ranges:

» Enable client media ports as shown in Configuring Port Ranges for your Microsoft Lync Clients in Lync Server 2013. Note that VVX business media phones use only the Audio port and range.

Microsoft Quality of Experience Monitoring Server Protocol (MS-QoE)

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

MS-QoE is compatible with Skype for Business and Lync Server 2010 and 2013.

All parameters for enabling or disabling QoE are included in the in-band provisioning parameters sent from the Skype for Business server. Note that Polycom supports only those elements listed in section Polycom-Supported Skype for Business QoE Elements.

- For a list of all parameters that report QoE data, see Microsoft [MS-QoE] PDF at [MS-QoE]: Quality of Experience Monitoring Server Protocol.

Setting QoE Parameters on the Skype for Business Server

Set the following QoE parameters on the Skype for Business Server.

- `EnableQoE`. Set to 'True' to enable QoE on the server and automatically assign the URI to which QoE reports are published. If set to 'False' no QoE reports are published. Note that the URI maps to the in-band element '`qosUri`'. To get the current value of `EnableQoE`, run the command `Get-CsQoEConfiguration` in Skype for Business Server Powershell.
- `EnableInCallQoS`. Set to 'True' to enable in-call QoE on the server. If set to 'False', only end-call QoE reports are sent. `EnableInCallQoS` maps to the in-band element '`enableInCallQoS`'.
- `InCallQoSIntervalSeconds`. Set the time interval in seconds to publish in-call QoE reports only if there is a transition in call quality. If no change in call quality is detected, no report is sent at the interval time you set. `InCallQoSIntervalSeconds` maps to the in-band element '`inCallQoSIntervalSeconds`'.

  When you enable in-call QoE, you do not need to wait until the end of the call to view call quality data. In-call QoE is off by default and you can enable it on Windows PowerShell using the following command:

  `Set-CsMediaConfiguration -Identity Global -EnableInCallQoS:$TRUE -InCallQoSIntervalSeconds x` (where x is a digit from 1 to 65535).
- `voice.qualityMonitoring.rtcpxr.enable`. Set to 1 (default) to allow the phone to collect RTCP XR metrics.

The following figure illustrates the QoE parameter values you need to set.

QoE Parameters on Server Media Configuration

```
PS C:\Users\administrator.COHOWINERY> Get-CsMediaConfiguration | fl

Identity                    : Global
EnableQoS                   : True
EncryptionLevel             : RequireEncryption
EnableSiren                 : False
MaxVideoRateAllowed         : VGA600K
EnableInCallQoS             : True
InCallQoSIntervalSeconds    : 35
EnableRtpRtcpMultiplexing   : True
```

Query QoE Status from the Web Configuration Utility

Users and administrators can query the in-band QoE status, interval, and URI from the Web Configuration Utility.

To query the in-band QoE status:

1 Enter the IP address of the phone into a web browser and log in as Administrator or User.

2 Go to **Diagnostics > Skype for Business Status > Quality of Experience**.

QoE Parameters

Use the following Polycom parameters to configure MS-QoE from a provisioning server.

QoE Parameters

Parameter Template	Permitted Values
`voice.qoe.event.lossrate.threshold.bad` features.cfg	Defines the threshold for the network loss rate. Total packets lost for an interval/total packets expected for the interval *256 as stated in RFC 2611, section 4.7.1. 38 (default) - Approximately a 15% packet loss. 0 to 100
`voice.qoe.event.lossrate.threshold.poor` features.cfg	Defines the threshold for the network loss rate. Total packets lost for an interval/total packets expected for the interval *256 as stated in RFC 2611, section 4.7.1. 25 ms (default) - Approximately a 10% packet loss. 0 to 100

QoE Parameters

Parameter Template	Permitted Values
`voice.qoe.event.networkmos.threshold.bad` features.cfg	Defines the threshold for Network MOS as follows: The average of MOS-LQO wideband, as specified by [ITUP.800.1] section 2.1.2, based on the audio codec used and the observed packet loss and inter-arrival packet jitter. 19 (default) - Indicates a MOS score of 1.9. 10 - 50 - Indicates a MOS score between 1 - 5. networkMOS > 2.9 signifies good quality networkMOS > 2.9 < 1.9 signifies poor quality networkMOS < 1.9 signifies bad quality
`voice.qoe.event.networkmos.threshold.poor` features.cfg	Defines the threshold for Network MOS as follows: The average of MOS-LQO wideband, as specified by [ITUP.800.1] section 2.1.2, based on the audio codec used and the observed packet loss and inter-arrival packet jitter. 29 (default) - Indicates a MOS score of 2.9. 10 - 50 - Indicates a MOS score between 1 - 5. networkMOS > 2.9 signifies good quality networkMOS > 2.9 < 1.9 signifies poor quality networkMOS < 1.9 signifies bad quality

Polycom-Supported Skype for Business QoE Elements

This section lists the Microsoft Quality of Experience (QoE) elements supported by Polycom phones.

For a list of all parameters that report QoE data, see Microsoft [MS-QoE] PDF at [MS-QoE]: Quality of Experience Monitoring Server Protocol.

Polycom-Supported Skype for Business QoE Elements

Parent Element	Child Elements/Attributes
VQReportEvent	VQSessionReport
VQSessionReport	SessionId
	Endpoint
	DialogInfo
	MediaLine
Endpoint	Name
	v2:OS
	v2:VirtualizationFlag

Polycom-Supported Skype for Business QoE Elements

Parent Element	Child Elements/Attributes
	CorrelationID
	FromURI
	ToURI
	Caller
	LocalContactURI
	RemoteContactURI
	LocalUserAgent
	RemoteUserAgent
	LocalPAI
	RemotePAI
	ConfURI
	v2:CallPriority
	v2:MediationServerBypassFlag
	v2:TrunkingPeer
	v2:RegisteredInside
	CallID
	FromTag
	ToTag
	Start
	End
MediaLine	Description
	InboundStream
	OutboundStream
Description	Connectivity
	Security
	Transport
	LocalAddr
	RemoteAddr
	v3:ReflexiveLocalIPAddress

Polycom-Supported Skype for Business QoE Elements

Parent Element	Child Elements/Attributes
	v3:MidCallReport
LocalAddr, RemoteAddr, RelayAddr	IPAddr
	Port
	SubnetMask
	v2:MACAddr
Connectivity	Ice
	IceWarningFlags (Five flags supported)
	RelayAddress
InboundStream OutboundStream	Network
	Payload
	QualityEstimates
Network	Jitter
	PacketLoss
	BurstGapLoss
	Delay
	Utilization
Jitter	InterArrival
	InterArrivalMax
Packetloss	LossRate
	LossRateMax
BurstGapLoss	BurstDensity
	BurstDuration
	GapDensity
	GapDuration
Delay	RoundTrip
	RoundTripMax
Utilization	Packets
Payload	Audio

Polycom-Supported Skype for Business QoE Elements

Parent Element	Child Elements/Attributes
Payload.Audio	PayloadType
	PayloadDescription
	SampleRate
	v4:JitterBufferSizeAvg
	v4:JitterBufferSizeMax
	v4:JitterBufferSizeMin
	v4:NetworkJitterAvg
	v4:NetworkJitterMax
	v4:NetworkJitterMin
Signal	SignalLevel
	NoiseLevel
	InitialSignalLevelRMS
	RecvSignalLevelCh1
	RecvNoiseLevelCh1
	RenderSignalLevel
	RenderNoiseLevel
	RenderLoopbackSignalLevel
	VsEntryCauses
	EchoEventCauses
	EchoPercentMicIn
	EchoPercentSend
	SendSignalLevelCh1
	SendNoiseLevelCh1
QualityEstimates.Audio	RecvListenMOS
	RecvListenMOSMin
	NetworkMOS
NetworkMOS	OverallAvg
	OverallMin

User Log Upload

To help troubleshoot user issues, administrators can enable or disable for users the ability to upload diagnostic logs from the phone or Web Configuration Utility and set log levels from the phone. This feature is available on the Polycom Trio 8800 and 8500 systems, and all VVX business media phones registered with Skype for Business Server on-premises or online and with Microsoft Lync 2013 or 2010 Server.

Logs are uploaded to the Skype for Business Server at the following location which you can specify in the Skype for Business topology builder or at initial installation:

`<LYNC_SERVER_LOG_PATH>\1-WebServices-1\DeviceUpdateLogs\Client\CELog`

User instructions on uploading log files from the phone or Web Configuration Utility are detailed in the latest user guide for your phone model on Polycom Voice Support.

Configure User Log Upload

The following table lists parameters that configure user log uploading.

Configure User Log Uploading

Parameter Template	Permitted Values
`feature.logUpload.enabled` features.cfg	1 (default) - Enable log uploads. 0 - Disable log uploads.

Send Diagnostic Logs from the Phone

To help troubleshoot issues, you can send diagnostic logs from the phone.

To send diagnostic logs from the phone:

» Go to **Settings > Basic > Diagnostic Logs > Upload Logs**. Files are uploaded as plain text.

If the log upload is successful, the phone displays a message that the upload was successful.

If the log upload fails, the phone displays a message that the log upload failed.

Send Diagnostic Logs from the Web Configuration Utility

To help troubleshoot issues, you can send diagnostic logs from the Web Configuration Utility. This option is available when logged in as Administrator or User.

To send diagnostic logs from the Web Configuration Utility:

1 Enter the IP address of the phone into a web browser and log in as Administrator or User.

2 Go to **Diagnostics > Upload Logs**. Files are uploaded as plain text.

3 View upload URLs at **Skype for Business Status > Skype for Business Parameters**:
   - Update Server Internal URL for on-premises deployments
   - Update Server External URL online deployments.

   If the log upload is successful, the phone displays a message that the upload was successful.

   If the log upload fails, the phone displays a message that the log upload failed.

Setting Log Levels

You can set log levels from the phone or Web Configuration Utility. By default, the phone sends log levels set on the server.

Set Log Levels from the Phone

You can set log levels from the phone.

To set log levels from the phone:

» On the phone, go to **Home > Settings > Basic > Diagnostic Logs > Server Log Level**.

Set Log Levels from the Web Configuration Utility

You can set log levels from the Web Configuration Utility.

To set log levels from the Web Configuration Utility:

1 Enter the IP address of the phone into a web browser and log in as Administrator or User.

2 Go to **Settings > Logging**.

3 In **Server Log Level**, select a log level.

Polycom UC Software Update

You can update the phone's UC Software manually on a per-phone basis. Or, you can use the automatic software update feature to update your phone's software. All UC Software releases compatible with Microsoft are available at Polycom UC Software for Microsoft Deployments.

You can update the Polycom Trio 8800 and 8500 system with a USB flash drive.

Update Software with a USB Drive

You can use an USB flash drive to update the software on the Polycom Trio system or to provision and configure the system.

When you configure the system using a USB drive, the configuration on the USB overrides all previous configurations. However, when the USB drive is removed, the system returns to the previous configuration.

To update or provision the Polycom Trio system using a USB flash drive:

1 Format a USB flash drive as FAT32. Polycom recommends that you use a USB 2.0 flash drive.

If you are using a drive that is already formatted, ensure that previous files are deleted from the flash drive.

2 Download the software package from Polycom Voice Support.

3 Place the `3111-65290-001.sip.ld` file in the root directory of the flash drive. If provisioning the system, place the `000000000000.cfg` or `<MAC>.cfg` file and any configuration files in the root directory as well.

4 Connect the USB flash drive to the USB port on the system.

5 Enter the administrator password.

The system detects the flash drive and starts the update within 30 seconds. The mute keys' indicator lights begin to flash, indicating that the update has started.

The system reboots several times during the update. The update is complete when the indicator lights stop flashing and the Home screen displays.

Update UC Software Manually

This update procedure applies to phones running UC Software 4.1.x or UC Software 5.x.x.

To update UC Software manually:

1 Download and unzip UC Software to a directory on your provisioning server.

2 On the phone, go to **Settings > Advanced**, enter the password (default 456)

3 Go to **Network Configuration > Provisioning Server > DHCP Menu > Boot Server**.

4 In the Boot Server menu, choose **Static** if you are testing or provisioning a few phones, or choose **Option 66** if you are provisioning in a large environment and want phones to use a boot server defined in DHCP. If you choose Option 66, skip step 5 and go to step 6.

5 Go back to **Provisioning Server** and do the following:

➢ Choose a server type in the **Server Type** field.

➢ Enter the server address, for example, `http://server.domain.com/41X or ftp://ftp.domain.com/41X`.

➢ Enter your server user name and server password, if required.

6 Press **Back** until you are prompted to save your settings.

7 Choose **Save Configuration** to save your settings. The phone reboots.

Configuring UC Software Automatic Updates

By default, when a software update is available, an Information pop-up displays on your phone. The Information pop-up provides three options:

- Press **Reboot** to restart the phone and automatically update the phone's software.
- Press **Cancel** to cancel the automatic software update. When you press Cancel, a **DevUpdt** soft key displays on the phone's home screen. Press **Dev Updt** at any time to update your phone's software.

- Press **Details** to view information about current and available software.

When the phone is inactive for a long period of time, the phone automatically reboots and updates the phone's software.

If you want to change the default behavior of the software update any of these parameters, you must configure the parameters in the following table. Note these parameters are not included in the sample configuration files Polycom provides in the Microsoft directory of the UC Software download.

Configuring Automatic Software Update

The following table lists parameters that configure automatic software updates and polling of the provisioning server.

Automatic Software Update Parameters

Parameter Template	Permitted Values
`device.prov.lyncDeviceUpdateEnabled`	0 (default) - The automatic device update is disabled and the phone does not receive software updates from the server. Changing the value of this parameter reboots the phone. 1 (default) - The automatic device update is enabled on the phone and the phone receives software updates from the server.
`device.prov.lyncDeviceUpdateEnabled.set`	0 (default) - Disable automatic device update for all devices. 1 - Enable automatic device update for all devices and use `device.prov.lyncDeviceUpdateEnabled`.
`lync.deviceUpdate.popUpSK.enabled`	0 (disable) - Disable the Information popup that indicates when an automatic software update is available. 1 - Enable the Information popup that indicates when an automatic software update is available.
`lync.deviceUpdate.serverPollInterval`	7200 seconds (default) - The time interval in seconds that the phone sends a software update request to the Skype for Business Server. min=1800 seconds max=28800 seconds
`lync.deviceUpdate.userInactivityTimeout`	900 seconds [15 minutes] (default) - Sets the user inactivity timeout period after which the phone's software is automatically updated. Min=300 seconds Max=1800 seconds
`prov.polling.enabled`	You can choose to automatically poll the provisioning server for software updates. 1 (default) - the phone automatically polls the server for software updates. 0 - Disable automatic polling.

Automatic Software Update Parameters

Parameter Template	Permitted Values
`prov.polling.mode`	Choose the polling mode. 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`. Note that 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 (meaning values such as 86401 are over 2 days) and only between the start and end times. The day within that period is determined by the phone MAC addresses and does not change with a reboot. The time within the start and end is calculated again with every reboot.
`prov.polling.period`	The polling period in seconds. 86400 (default) integer > 3600 The polling period is rounded up to the nearest number of days in absolute and random mode you set in `prov.polling.mode`. In relative mode, the polling period starts once the phone boots. If random mode is set to a time greater than 86400 (one day) polling occurs on a random day based on the phone MAC address.
`prov.polling.time`	Specify the polling start time in absolute or random polling mode you choose with `prov.polling.mode`. 03:00 (default) hh:mm
`prov.polling.timeRandomEnd`	The polling stop time when the polling mode is set to random. NULL (default) hh:mm

Software Update using Windows Update Server

You can use Windows Update Server to update software on the Polycom Trio 8800 and 8500 systems connected as a USB device to a Microsoft Surface Pro or to a computer running Windows 10 when you set the Base Profile to **Skype USB Optimized**.

The Windows Update is enabled by default on the Polycom Trio 8500 and 8800 systems.

Software Update using Windows Update Server Parameters

The following table lists the parameter to configure the Software update using Windows Update feature.

Software Update using Windows Update Server Parameters

Parameter and Template	Permitted Values
`feature.usb.device.msrSoftwareUpdate` features.cfg	1 (default) - Enable the software update using Windows Update feature. 0 - Disable the software update using Windows Update feature.

Manually Update UC Software using Windows Update

You can manually update the software for Polycom Trio 8800 and 8500 systems connected to a Windows 10 computer or Microsoft Surface Pro through Windows Upgrade server.

To manually run the software upgrade for Polycom Trio 8800 system:

1. Connect the Polycom Trio system to a computer running Windows 10 operating system.
2. Right click **Polycom Trio 8800 USB-Interface** under **Universal Serial Bus devices** in the Device Manager.
3. Select **Update Driver Software**.
4. In the dialog, choose **Search automatically for updated driver software**.

   If a software update is available the Polycom Trio system updates automatically.

Phone Default Settings

If the device has already been in use, you can reset settings applied to the phone, or to factory default settings. Before resetting a device, verify that you do not need to keep parameters such as a provisioning server address or credentials.

Polycom devices store settings in up to three locations on a provisioning server that correspond to ways you can apply settings:

- In configuration files stored on the provisioning server
- In a per-device file uploaded to the provisioning server when settings are made using the Web Configuration Utility
- Locally on the phone's memory system

Note: Ensure that you restore default settings from all three configuration sources. Settings that you do not reset to factory defaults may override any new settings you apply.

Restore default settings from each source. You can perform all resets directly from the phone.

Change the Base Profile from the Phone

You can change the phone’s Base Profile from the phone.

To change the phone's Base Profile:

» On the phone, go to **Settings > Advanced**, enter the password (default 456), and go to **Administration Settings > Network Configuration > Base Profile**, and choose **Generic** or **Skype**.

Inbound and Outbound Ports for Polycom Trio System with Skype for Business

This section provides port usage information when configuring network equipment to support the Polycom Trio 8800 and 8500 system with Skype for Business.

For details, see Ports and protocols for internal servers on Microsoft TechNet.

Inbound Ports for Polycom Trio System with Skype for Business

The following table lists the inbound IP ports currently used by the Polycom UC Software running on the Polycom Trio 8800 and 8500 system with Skype for Business.

Inbound IP Port Connections to Polycom Trio Systems with Skype for Business

Inbound Port	*Type*	*Protocol*	*Function*	*Default*	*Configurable Port Number*
22	static	TCP	SSH Administration	Off	No
80	static	TCP	HTTP Pull Web interface, HTTP Push	Off	Yes
443	static	TCP	HTTP Pull Web interface, HTTP Push	On	Yes
1023	static	TCP	Telnet Diagnostics	Off	No
1024 - 65535	Dynamic	TCP/UDP	RTP media packets	On	Yes
1024 - 65535	Dynamic	TCP/UDP	RTCP media packets statistics	On	Yes
5001	static	TCP	People+Content IP	On	No
5060	static	TCP/UDP	SIP signaling	On	No
5061	static	TLS	SIP over TLS signaling	On	No
8001	static	TCP	HTTPS for modular room provisioning	On	Yes mr.deviceMgmt.port

Outbound Ports for Polycom Trio 8800 System with Skype for Business

The following table lists the outbound IP ports currently used by the Polycom UC Software running on the Polycom Trio 8800 system with Skype for Business.

Outbound IP Port Connections to Polycom Trio Systems with Skype for Business

Outbound Port	*Type*	*Protocol*	*Function*	*Default*	*Configurable Port Number*
21	static	TCP	FTP Provisioning, Logs	On	No
22	static	TCP	SSH	On	No
53	static	UDP	DNS	On	No
67	static	UDP	DHCP Server	On	No
68	static	UDP	DHCP Client		No
69	static	UDP	TFTP Provisioning, Logs		No
80	static	TCP	HTTP Provisioning, Logs, Web Interface		No
123	static	UDP	NTP time server		No
389	static	TCP/UDP	LDAP directory query		No
443	static	TCP	HTTPS Provisioning, Logs, Web Interface		No
514	static	UDP	SYSLOG		No
636	static	TCP/UDP	LDAP directory query		No
1024 - 65535	Dynamic	TCP/UDP	RTP media packets	On	Yes
1024 - 65535	Dynamic	TCP/UDP	RTCP media packets statistics	On	Yes
5060		TCP/UDP	SIP signaling	On	
5061		TCP	SIP over TLS signaling	On	
5222	static	TCP	Polycom Resource Manager: XMPP	Off	No
8001	static	TCP	HTTPS for modular room provisioning	On	Yes mr.deviceMgmt.port

Real-Time Transport Protocol (RTP) Port Parameters for Skype for Business

Use the parameters in the following table to configure RTP packets and ports for Polycom Trio 8800 and 8500 systems registered with Skype for Business.

The Polycom Trio 8500 supports audio only and does not support video or content; you cannot configure video and content port ranges for the Polycom Trio 8500 system.

Real-Time Transport Protocol (RTP) Port Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	tcpIpApp.port.rtp.lync.audioPortRangeStart	Determines the start port for the audio port range. 5350 (default) 1024 - 65436	No
site.cfg	tcpIpApp.port.rtp.lync.videoPortRangeStart	Determines the start port for the video port range. 5390 (default) 1024 65486	No
site.cfg	tcpIpApp.port.rtp.lync.contentPortRangeStart	Determines the start port for the content port range. 5430 (default) Min - 1024 Max - 65486	No
site.cfg	tcpIpApp.port.rtp.lync.audioPortRangeEnd	Determines the end port for the audio port range. 5389 (default) Min - 1024 Max - 65485	No
site.cfg	tcpIpApp.port.rtp.lync.videoPortRangeEnd	Determines the end port for the video port range. 5429 (default) Min - 1024 Max - 65535	No
site.cfg	tcpIpApp.port.rtp.lync.contentPortRangeEnd	Determines the end port for the content port range. 5469 (default) Min - 1024 Max - 65535	No

Polycom Trio and Visual+ System Media Transport Parameters

These parameters provide additional RTP port ranges to transmit media between the Polycom Trio system and Polycom Trio Visual+ systems. Use these parameters only when the default port ranges of these parameters overlaps with the audio, video, and content port ranges.

Polycom Trio 8800 and Visual+ Systems Media Transport Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	mr.displayStreamPortStart	Determines the start port range for video streams that display on the Polycom Visual+. 4000 (default) 1024 - 65436	Yes
site.cfg	mr.displayStreamPortEnd	Determines the end port range for video streams that display on the Polycom Visual+. 4019 (default) 1024 - 65436	Yes
site.cfg	mr.cameraStreamPortStart	Determines the start port range for video streams received from the Polycom Visual+ camera. 4020 (default) 1024 - 65436	Yes
site.cfg	mr.cameraStreamPortEnd	Determines the end port range for video streams received from the Polycom Visual+ camera. 4219 (default) 1024 - 65436	Yes
site.cfg	mr.audioStreamPortStart	Determines the start port range for audio streams sent to the Polycom Visual+. 6000 (default) 1024 - 65436	Yes
site.cfg	mr.audioStreamPortEnd	Determines the end port range for audio streams sent to the Polycom Visual+. 6199 (default) 1024 - 65436	Yes