DEPLOYMENT GUIDE

UC Software 5.7.0 | December 2017 | 3725-49078-020A

Polycom® UC Software with Skype for Business

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Contents

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 Polycom devices with Skype for Business.

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.

UC Software Device Compatibility

Polycom UC Software supports the following devices with Skype for Business:

- Polycom® VVX® 201 business media phones
- Polycom® VVX® 300, 301, 310, 311 business media phones
- Polycom® VVX® 400, 401, 410, 411 business media phones
- Polycom® VVX® 500 and 501 business media phones
- Polycom® VVX® 600 and 601 business media phones
- Polycom® SoundStructure® VoIP Interface.

If you are using previous versions of UC Software to register SoundStructure VoIP Interface with Lync Server, see *Polycom SoundStructure VoIP Interface for Use with Microsoft Lync Server* at Polycom SoundStructure on Polycom Support.

Web Info: To register Polycom Trio solution with Skype for Business, see *Polycom Trio with Skype for Business - Deployment Guide* at Polycom Trio on Polycom Support.

Polycom VVX phones and SoundStructure VoIP interface support Skype for Business and Lync Server 2013. Note that Microsoft now supports multiple clients:

- Skype for Business 2016 (v16.x)
- Lync 2013 / Skype for Business 2015 (v15.x)

Microsoft Qualified Phones

As of UC Software 5.3, Polycom offers devices with an Open SIP or a Skype Base Profile. As of UC Software 5.4.0A, Polycom offers devices already configured for use with Skype for Business on-premises deployments or Skype for Business Online. These devices include Microsoft-qualified UC Software with a feature license included and enable you to start up the phone and register with default 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.

Skype for Business Topologies

Polycom support for a Skype for Business topology varies by environment.

Supported Skype for Business Topologies

The following table lists Polycom support for each Skype for Business topology.

Polycom-Supported Skype for Business Topologies

Topology	Active Directory	Skype for Business	Exchange
On-premises			
	On-premises	On-premises	On-premises
Hybrid Voice/Cloud Connector Edition			
	On-premises	Online	Online
Office 365 Multi-tenant (O365MT)			
	Online	Online	Online
Hybrid (Split-Domain)			
	On-premises	On-premises	Online
	On-premises	Online	Online

Unsupported Skype for Business Topologies

The following table lists Skype for Business topologies Polycom does not support.

Unsupported Skype for Business Topologies

Topology	Active Directory	Skype for Business	Exchange
Unsupported Hybrid (Split-Domain)			
	On-premises	Online	On-premises

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.

Skype for Business On-Premises and Online Features

The following table lists Polycom UC Software support for Skype for Business on-premises and Online features.

Polycom with Skype for Business Online Feature Support

Skype for Business Feature	Polycom with Skype for Business On-Premises	Polycom with Skype for Business Online
Resiliency - Branch Office		na
Resiliency - Data Center Outage	✓	na
Call Park	✓	×
PIN Authentication	✓	×
Attendant Console	✓	×
Cross Pool	✓	×
Media Bypass	✓	×
Response Groups	✓	×
Private Line	✓	×
Web Sign In	×	✓
Common Area Phone (CAP)	✓	✓
Host Desking	✓	✓
Enhanced Feature Line Key	✓	✓
Enhanced 911 (E.911)	✓	✓
Web Proxy Auto Discovery	✓	✓
Quality of Service for Audio Calls	✓	✓
Device Lock	✓	✓
Distribution Lists	✓	✓
Quality of Experience (QoE)	✓	✓
User Log Upload	✓	✓
BToE Manual Pairing	✓	✓
Device Update	✓	✓
Inband Provisioning	✓	✓
Call Handling	✓	✓
Call Forward	✓	✓
Call Transfer	✓	✓
Conference Calls	✓	✓
Local Call Logs	✓	✓
Exchange Call Logs	✓	✓
Federated Calls	✓	✓
Simultaneous Ring	✓	✓
Dual Tone Multi Frequency	✓	✓
Emergency 911	✓	✓
Call Admission Control	✓	✓
Monitoring (Device Inventory)	✓	✓

Polycom with Skype for Business Online Feature Support

Skype for Business Feature	Polycom with Skype for Business On-Premises	Polycom with Skype for Business Online
Delegates	✓	✓
Team Call	✓	✓
Message Waiting Indicator	✓	✓
Exchange Integration	✓	✓
Exchange Calendar		
Extended Presence	✓	✓
Visual Voicemail	✓	✓
Boss-Admin	✓	✓

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

Polycom offers several methods to register your Polycom phones with Skype for Business. Regardless of the method you choose, you must complete three major tasks to register your phones correctly:

- Configure the Network
- Set Up Polycom UC Software
- Provisioning the Phones

As of UC Software 5.3.0, Polycom phones ordered with the Skype SKU are shipped with Skype for Business-qualified software that enables you to start up the phone and register with default settings.

Note: 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 three tasks.

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.
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.

Also set up PIN Authentication type if you are using any of the following devices in your deployment: VVX 201, 300/310, 301/311, 400/410, 401/411, 500/501, 600/601 business media phones, and SoundStructure VoIP Interface.

Set Up Polycom UC Software

The latest UC Software for Microsoft deployments is available at Polycom UC Software for Skype for Business Deployments. All UC Software versions are available on the Polycom UC Software Support Center.

If you are setting up your own provisioning server or want to customize feature settings, Polycom provides template configuration files you can use to provision your Polycom phones for use with Skype for Business. You can find the Skype for Business configuration files in your UC Software download.

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.

6. (Optional) To use the BToE feature, download the Polycom BToE connector application and enable BToE. For complete instructions on setting up BToE, see the latest *Polycom VVX Business Media Phones for Skype for Business - User Guide* on Polycom UC Software for Microsoft Deployments.

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 with Polycom UC Software, see the *Polycom UC Software Administrator Guide* on Polycom UC Software for Microsoft Deployments.

Tip: If you are using Polycom UC Software 5.1.1 or later, the Web Configuration Utility is disabled by default and you cannot register phones with the Web Configuration Utility. If you want to use a phone's Web Configuration Utility after the phone is registered with Skype for Business Server, refer to Accessing the Web Configuration Utility.

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 to Skype Using MKC During Startup. Set the Base Profile to Skype using an MKC method during phone startup. This is the fastest manual provisioning method.
- Set the Base Profile to Skype from the Setup Menu During Startup. Set the Base Profile to Skype during startup using the phone boot Setup menu.
- Set the Base Profile Using MKC. Set the Base Profile to Skype using MKC during normal phone functioning.
- 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 to Skype Using MKC During Startup

You can set the Base Profile of a phone to Skype during the phone startup cycle in two ways: by using an MKC method during startup or from the phone boot Setup menu. The MKC during startup is the fastest manual provisioning method.

If your phones are not brand new and directly from the manufacturer, ensure that you reset the phones to factory default settings as shown in Phone Default Settings.

To set the Base Profile to Skype Using MKC during startup:

1 Power on the phone or restart it after you have reset the phone to factory default settings.

2 A few seconds into the device's startup cycle, the phone displays the message 'Starting Application', press Cancel to interrupt and a Cancel soft key. Press the **Cancel** soft key.

3 When the phone displays three soft keys—Start, Setup, and About—press and hold the following key combinations on the phone keypad for about 3 seconds to enter the MKC for the phone model:

   - VVX 300, 310, 400, 410, 500, 600, press **1**, **4**, **9**

4 Press and hold the MKC keys to cause the Base Profile Password menu to display. Enter the password (default 456) to change the Base Profile and press **Ok**.

   The **Base Profile** menu displays.

5 Press the **Edit** soft key, use the keypad keys to set the Base Profile to **Skype**, and press **Ok > Exit.**

6 Highlight **Save & Reboot** and press the **Select** soft key.

   The phone reboots and displays the Sign In screen. Users can now sign in.

Set the Base Profile to Skype from the Setup Menu During Startup

When you boot up the phone, you can set the Base Profile to Skype using the Setup menu available during the phone startup process.

To set the Base Profile to Skype from the phone boot Setup menu:

1 Power on the phone or restart after you have reset the phone to factory default settings.

2 A few seconds into the device power-up cycle, the phone displays the message 'Starting Application, press Cancel to interrupt' and a Cancel soft key. Press the **Cancel** soft key.

3 When the phone displays three soft keys—Start, Setup, and About—press the **Setup** soft key, enter the password (default 456), and press **Ok**.

   The phone displays a diagram of keypad keys you can use to navigate the Setup menu. You will need to use these keys in the next few steps.

4 Press the **Setup** soft key and the Setup menu displays.

5 Using the keypad keys, scroll down, highlight **Base Profile**, and select the **Edit** soft key.

6 Using the keypad keys, set the Base Profile to **Skype**, and press **Ok > Exit.**

7 Highlight **Save & Reboot** and press the **Select** soft key.

8 The phone reboots and displays the Sign In screen. Users can now sign in.

Set the Base Profile Using MKC

This section shows you two ways to set the Base Profile to Skype from the Settings menu when the phone is idle, and how to sign in and register a line.

To set the Base Profile to Skype using MKC:

1 Press the phone's **Home/Menu** key.

2 From the idle screen, press and hold the following key combinations on the phone keypad for about 3 seconds. MKC keys vary by phone.

   ➢ VVX 300, 310, 400, 410, 500, and 600, press **1**, **4**, **9**

3 Press and hold the MKC keys to cause the Base Profile screen to display. Enter the password (default 456) and press **Enter**.

4 In the **Base Profile** menu, select **Skype**.

   The phone automatically restarts and displays the Sign In screen. Users can now sign in using one of the Sign In Methods.

   Note: If the phone does not restart, choose **Settings > Basic > Restart**, or power the phone off and then on.

If your phone supports PIN authentication, you will be prompted for authentication. Otherwise, you will be prompted for Skype for Business sign-in credentials.

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 all Microsoft-compatible UC Software from UC Software for Microsoft Deployments.
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.

Polycom provides the UC Software download in two file formats:

- **Split files**. Enable you to choose UC Software for specific phone models. The split files are smaller in size with faster update times, and they reduce internal network traffic during reboots and updates.
- **Combined file**. A large directory that contain software files for all Polycom phone models.

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.

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.

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.

Configuring Music on Hold

You can enable or disable the music on hold (MoH) feature using configuration files. Music on hold enables music to play when users place a call on hold. If you place multiple calls on hold, only the first call placed on hold hears the music. By default MoH is enabled on the phone when registered with Skype for Business. When MoH is enabled, you can turn on or off the music the phone plays when an active call is placed on hold.

You specify on the provisioning server which file the phone plays when you place an active call on hold. The phone downloads the file you place on the server and stores the file on its internal flash drive.

The default MoH file size is 540 KB and the maximum file size is 600 KB. You can use the parameter res.quotas.tone to increase the maximum MoH file size to 1024 KB. The phone supports the following .wav audio file formats:

- mono G.711 (8 bits/sample, 8-khz sample rate)
- mono L16/16000 (16 bits/sample, 16-kHz sample rate)
- mono L16/48000 (16 bits/sample, 48-kHz sample rate)

Upload a Music File

You can upload a music file to the phone using the phone's Web Configuration Utility.

To upload a music file:

1 Enter the IP address of the phone to a web browser.
2 Log into the Web Configuration Utility as Administrator.
3 Go to **Preferences > Additional Preferences > Music On Hold**.
4 Select **MOH Status Enable** and **Save**.
5 Select **Add** and select a file from your computer or enter a URL.
6 Click **Save**.

Configuring Music on Hold

The following table lists parameters that configure MoH.

Music on Hold Parameters

Parameter Template	Permitted Values
`feature.moh.enabled` features.cfg	Music on hold enables VVX phone users to stream music when they place a caller on hold. 0 (default) - Music on hold is disabled. 1 - Music on hold is enabled and you must specify a music file in `feature.moh.filename`.
`feature.moh.filename` features.cfg	Specify the file the music file you want the phone to play when users place an active call on hold. NULL (default) String, maximum of 256 characters
`feature.moh.payload` features.cfg	Specify the payload for RTP packets when music on hold is playing. For best phone performance, set to 80. In PSTN calls using a media gateway that does not support a payload value of 80, set to 20. 80 (default) 20, 40, 60, 80
`res.quotas.tone`	Set the maximum sample tone file size. 1024 KB 600 - 1024 KB

Music on Hold Error Messages

If a music file fails to play, one of the following messages display on the phone screen.

MoH Error Messages

Failure Scenario	Error Message
Phone fails to download the MoH file because the current file was active	'Download failed' 'Current MoH is Active'
MoH file download failed	'Download Failed'
MoH file size is zero	'Download Failed'
MoH file size exceeds the maximum size of 500KB	'File size exceeded 500KB'
An incorrect .wav file format is specified	'Unsupported .wav file format'
A network failure occurs while the phone downloads the MoH file	'Download failed' 'Network is down'

Supported Audio Codecs

The following table details the supported audio codecs and priorities for Polycom phone models.

Note the following limitations when using the Opus codec:

- VVX 301, 311, 401, 411, 500, 501, 600, and 601 business media phones support a single Opus stream. Users can establish only one call at a time when using the Opus codec on these phones.
- VVX 500 and 600 do not support video when using Opus.
- VVX 500 and 600 do not support local conferences when using Opus.
- Opus is not compatible with G.729 and iLBC. If you set Opus to the highest priority, G.729 and iLBC are not published; if you set G.729 and iLBC to the highest priority, Opus is not published.

On the VVX 500/501 and 600/601, when you enable video, the G.722.1C codec is disabled.

Audio Codec Priority

Phone	Supported Audio Codecs	Priority
VVX 101, 201	G.711µ-law	6
	G.711a-law	7
	G.722	4
	G.729AB	8
	iLBC (13.33kbps, 15.2kbps)	0, 0
VVX 300/301, 310/311, 400/401, 410/411 * Note: VVX 301, 311, 401, 411 support a single Opus stream. VVX 300, 310, 400, 410 do not support Opus.	G.711µ-law	6
	G.711a-law	7
	G.722	4
	G.722.1 (24kbps, 32kbps)	5
	G.729AB	8
	iLBC (13.33kbps, 15.2kbps)	0, 0
	Opus*	0
	Siren 7	0
VVX 500/501, 600/601 • VVX 500 and 600 support a single Opus stream. • VVX 500 and 600 do not support both Opus and video.	G.711 µ -law	6
	G.711a-law	7
	G.722	4
	G.722.1 (24kbps, 32kbps)	5
	G.722.1C (48kbps)	2
	G.729AB	8
	Opus*	0
	iLBC (13.33kbps, 15.2kbps)	0, 0
	Siren 7	0
VVX 501, 601	SILK	0

Audio Codec Priority

Phone	Supported Audio Codecs	Priority
VVX 1500	G.711 µ -law	6
	G.711a-law	7
	G.719 (64kbps)	0
	G.722	4
	G.722.1 (24kbps, 32kbps)	5
	G.722.1C (48kbps)	2
	G.729AB	8
	Siren14 (48kbps)	3
	iLBC (13.33kbps, 15.2kbps)	0, 0
SoundStructure VoIP Interface • SoundStructure VoIP Interface supports a single Opus stream. • SoundStructure VoIP Interface does not support both Opus and video.	G.711 µ -law	6
	G.711a-law	7
	G.722	4
	G.722.1 (24kbps, 32kbps)	5
	G.722.1C (48kbps)	2
	G.729AB	8
	iLBC (13.33kbps, 15.2kbps)	0, 0
	Siren 7	0

Supported Audio Codec Specifications

The following table summarizes the specifications for audio codecs supported on Polycom phones.

Audio Codec Specifications

Algorithm	Reference	Raw Bit Rate	Maximum IP Bit Rate	Sample Rate	Default Payload Size	Effective Audio Bandwidth
G.711 µ -law	RFC 1890	64 Kbps	80 Kbps	8 Ksps	20 ms	3.5 KHz
G.711 a-law	RFC 1890	64 Kbps	80 Kbps	8 Ksps	20 ms	3.5 KHz
G.719	RFC 5404	32 Kbps 48 Kbps 64 Kbps	48 Kbps 64 Kbps 80 Kbps	48 Ksps	20 ms	20 KHz
G.711	RFC 1890	64 Kbps	80 Kbps	16 Ksps	20 ms	7 KHz
G.722[1]	RFC 3551	64 Kbps	80 Kbps	16 Ksps	20 ms	7 KHz

Audio Codec Specifications

Algorithm	Reference	Raw Bit Rate	Maximum IP Bit Rate	Sample Rate	Default Payload Size	Effective Audio Bandwidth
G.722.1	RFC 3047	24 Kbps 32 Kbps	40 Kbps 48 Kbps	16 Ksps	20 ms	7 KHz
G.722.1C	G7221C	224 Kbps 32 Kbps 48 Kbps	40 Kbps 48 Kbps 64 Kbps	32 Ksps	20 ms	14 KHz
G.729AB	RFC 1890	8 Kbps	24 Kbps	8 Ksps	20 ms	3.5 KHz
Opus	RFC 6716	8 - 24 Kbps	24 - 40 Kbps	8 Ksps 16 Ksps	20 ms	3.5 KHz 7 KHz
Lin16	RFC 1890	128 Kbps 256 Kbps 512 Kbps 705.6 Kbps 768 Kbps	132 Kbps 260 Kbps 516 Kbps 709.6 Kbps 772 Kbps	8 Ksps 16 Ksps 32 Ksps 44.1 Ksps 48 Ksps	10 ms	3.5 KHz 7 KHz 14 KHz 20 KHz 22 KHz
Siren 7	SIREN7	16 Kbps 24 Kbps 32 Kbps	32 Kbps 40 Kbps 48 Kbps	16 Ksps	20 ms	7 KHz
Siren14	SIREN14	24 Kbps 32 Kbps 48 Kbps	40 Kbps 48 Kbps 64 Kbps	32 Ksps	20 ms	14 KHz
Siren22	SIREN22	32 Kbps 48 Kbps 64 Kbps	48 Kbps 64 Kbps 80 Kbps	48 Ksps	20 ms	22 KHz
iLBC	RFC 3951	13.33 Kbps 15.2 Kbps	31.2 Kbps 24 Kbps	8 Ksps	30 ms 20 ms	3.5 KHz
SILK	Skype 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 (16ksps), the RTP clock rate advertised for the G.722 payload format is 8,000 Hz because that value was erroneously assigned in RFC 1890 and must remain unchanged for backward compatibility.

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

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.

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.

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

- Change `voice.codecPref.G7221.24kbps` from 0 to 5.
- Change `voice.codecPref.G7221.32kbps` from 5 to 0.

Add `voice.audioProfile.G7221.24kbps.payloadType` and set it to 112.

After the phone is connected, 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.
- VVX business media phones registered with Skype for Business display a one-touch Join button that allows you to join a Skype for Business conference in a federated environment, even if you haven't configured Transport Neutral Encapsulation Format (TNEF).

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.

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.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.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.
`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.
`feature.voicemail.enabled` features.cfg	1 (default) - Enables the voicemail feature on the phone to receive Exchange or SIP voicemail stored on the server. 0 - Disables voicemail feature on the phone to receive Exchange or SIP voicemails stored on the server.
`up.oneTouchDirectory` features.cfg	1 - The Skype for Business Search icon displays on the Home screen. 0 - The Skype for Business Search icon does not display on the Home screen.
`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.

Common Area Phone (CAP)

You can configure your phone with Common Area Phone (CAP) mode allowing users to access the phone with limited features. When you enable this feature, you can set the phone to either CAP or CAP Admin Mode. By default, the phone is set to CAP mode that allow users to access limited features such as ABS Search. However, you can make other features available to users by enabling parameter for the corresponding feature. The following table lists the features that are disabled by default and can be configured for a user accessing CAP phone.

Features Disabled by CAP

Soft Key / Menu	Available for CAP User	Can Administrator Configure for CAP User?
Status/DND	No	Yes
Call Forward	No	Yes

Features Disabled by CAP

Soft Key / Menu	Available for CAP User	Can Administrator Configure for CAP User?
Device Lock	No	Yes
Call Logs	No	Yes
Local Contact Directory	No	Yes
Exchange Calendar	No	Yes
Exchange Contacts	No	Yes
Exchange Voicemail/Messages	No	Yes
Redial	No	Yes

The following settings are only available for Admin in CAP Admin Mode.

- Basic Settings
- Sign In/Sign Out
- Web Configuration Utility
- My Status (under **Features > Presence > My Status**)

Note that the Boss-Admin and Shared Line Appearance feature behaves the same way irrespective whether the phone is CAP enabled or not. Polycom recommends to not configure these features when the phone is CAP enabled to work seamlessly.

If you wish to enable Exchange features for the user, you must make sure to set the values to 1 for all the required parameters of the corresponding feature. The following table lists the required parameters for the corresponding Exchange features to enable and make available for users.

Exchange Features

Exchange Feature	Parameters

Exchange Calendar	`feature.EWSAutodiscover.enabled`
	`feature.exchangeCalendar.enabled`
	`homeScreen.calendar.enable`

Exchange Contacts	`feature.EWSAutodiscover.enabled` `feature.exchangeContacts.enabled`

Exchange Voicemail / Messages	`feature.voicemail.enabled`
	`feature.EWSAutodiscover.enabled`
	`feature.exchangeVoiceMail.enabled`
	`feature.exchangeSipVMPlay.enabled`

Exchange Call Logs	`feature.callList.enabled`
	`feature.exchangeCallLog.enabled`
	`feature.EWSAutodiscover.enabled`

Enable CAP Admin Mode

You can enable the Common Area Phone (CAP) Admin mode from the phone after CAP mode is enabled.

To enable CAP Admin mode from phone's user interface:

1. On the VVX business media phone, navigate to **Settings > Advanced**, and enter the default password.
2. Select **Administration Settings > Common Area Phone Settings > CAP Admin Mode**.
3. Choose **Enable**.

Disable CAP Admin Mode

You can disable the Common Area Phone (CAP) Admin mode from the phone.

To enable CAP Admin mode from phone's user interface:

1. On the VVX business media phone, navigate to **Settings> Advanced**, and enter the default password.
2. Select Administration **Settings > Common Area Phone Settings> CAP Admin Mode**.
3. Choose **Disable**.

Common Area Phone Parameters

The following table lists the parameter to configure the Common Area Phone user profile.

Common Area Phone Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.CAP.enable`	0 (default) - Disables the Common Area Phone mode. 1 - Enables the Common Area Phone mode.	No

Hot Desking

You can configure your phone allowing a Hot Desking (HD) or guest user to sign-in on top of a host user signed in to the phone or a common area phone. You must enable this feature on both the Skype for Business server and on your provisioning server using the feature.HotDesking.enable parameter. When this feature is enabled, a **Guest** soft key displays on the phone. By default, this feature is enabled on the provisioning server. However, the user can choose to enable or disable the feature from the phone.

Note: When the phone is CAP enabled, user do not have the permission to enabled or disable this feature.

If the user disables Hot Desking from the phone, the user setting overrides the Skype for Business server setting and the feature is disabled. When you enable this feature, the guest user can sign-in to the host phone by pressing the **Guest** soft key. After pressing the **Guest** soft key, the guest user can sign in with one of the following methods even if the phone is CAP-enabled or locked:

- User ID
- Pin Authentication
- Via PC
- Online Web Sign In

When the guest user signs in to the phone, the host/CAP user is logged out automatically and the guest user icon displays on the phone. After the guest user has signed in to the phone, the following details of the previously signed-in host/CAP user are not accessible:

- Call Logs
- Voicemail
- Calendar
- Local Contact Directory

The menu options that are not accessible to the guest user are as follows:

- Headset Settings
- Background
- Screen Saver
- Presence
- Location Info
- Diagnostic logs
- Picture frame
- Power Saving
- Reset to Factory
- Clear browser data
- Network Configuration

However, when the guest user signs out of the phone, all the basic settings of the guest user are removed and the phone is set with original settings of the host user. The following scenarios enable the phone to sign out the guest user automatically and sign in back with the previously signed in user:

- Timeout

  This feature supports hot desking timeout, the period of time after which the phone shall sign-in to the host user when being idle in hot desking mode. This timeout is applicable only when the guest user has signed in successfully.

- ➢ When the phone is idle for hot desking timeout configured on the server.

  When the guest user is signed in and does not perform any activity and the timeout interval configured on the server reached the value, the guest user is signed out.

- ➢ User taps the guest soft key and does not sign in using any sign in methods.

  The timeout interval for hot desking is set to 2 minutes by default. However, the host user does not need to wait for 2 minutes. The host user can sign in by pressing the **Host** soft key on the phone screen.

- BToE Mode

  When a guest user is signed in to the phone and the phone is in BToE mode, the following scenarios lead to sign in the host user after logging out the guest user automatically:

  - ➢ Guest user unpairs the BToE pairing from the device.
  - ➢ Guest user unpairs the BToE pairing using BToE client.
  - ➢ Guest user signs out from the paired Skype for Business client.

When the phone is in idle state and any one of the scenario occurs, the phone signs out the guest user.

Hot Desking Parameters

The following table lists the parameters that configure the Hot Desking feature.

Hot Desking Parameters

Template	Parameter	Permitted Value	Change Causes Restart or Reboot
`features.cfg`	`feature.HotDesking.enabled`	1 (default) - Enables the Hot Desking feature. 0 - Disables the Hot Desking feature.	No

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. On VVX phones, an up-to-date call lists history is defined in <user>-calls.xml. This list is retained each time the user logs in to their phone. 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
- **Sign In with Better Together over Ethernet (BToE).** If you use the BToE feature in your deployment, you can sign in to the phone from the PC client when the phone and computer are connected through the BToE application.
- **Web Sign In CAP**. Use this method to securely sign-in from a browser on your computer or mobile device when Common Area Phone mode feature is enabled along with Online Web Sign In and the phone is set to CAP Admin Mode.The phone generates a unique pairing code used to sign in on a secure Office 365 website.
- **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. Polycom UC Software 5.1.1 introduces PIN authentication for SoundStructure VoIP Interface registered with Skype for Business.

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.

Sign In with Better Together over Ethernet (BToE)

You can use this sign-in method with the Better Together over Ethernet (BToE) feature. The BToE feature enables you to place, answer, and hold audio and video calls from your Polycom VVX phone and Skype for Business client on your computer. This sign in method is available after you download the BToE connector application and pair your computer and phone. To download the application and for detailed instructions, see the *Polycom VVX Business Media Phones - User Guide* at Polycom UC Software for Microsoft Deployments.

Web Sign In CAP for Skype for Business Online

When Common Area Phone mode feature is enabled along with Online Web Sign In and the phone is set to CAP Admin mode, you can sign in to the phone registered with Skype for Business Online and securely login to Skype for Business from the phone or from a computer or mobile web browser.

Note that this sign in method is not applicable when the phone is signed in as a guest user.

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.

Polycom VVX business media phones currently support Okta and Ping Federate.

Disabling the Sign-In and Sign-Out Soft Keys

If your VVX business media phones are used as shared devices 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.

Use the following parameters to remove the sign-out soft key, or the sign-in and sign-out keys.

Sign-In and Sign-Out Soft Key Parameters

Parameter Template	Permitted Values
`feature.lync.hideSignInSignOut` features.cfg	0 (default) - The Sign In and Sign Out soft keys display on the Home screen and phone menus. 1 - The Sign In and Sign Out soft keys are removed from the Home screen and phone menus, and users are not able to sign in or out. Administrators can sign in and out with the Web Configuration Utility.
`feature.lync.hideSignOut` features.cfg	0 (default) - The Sign Out soft key displays on the Home screen and phone menus. 1 - The Sign Out soft key is removed from the Home screen and phone menus, and users are not able to sign out. Administrators can sign out of the phone from the Advanced menu or Web Configuration Utility.
`softkey.feature.simplifiedSignIn` lync.cfg	0 (default) - The Sign In and Sign Out soft keys are removed from the Home screen and display in the Features menu. 1 - The Sign In and Sign Out soft keys displays on the Home screen and phone menus.

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 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.

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.

Call Controls

This section shows you how to configure call control features.

Presence Status

You can enable users to monitor the status of other remote users and phones. By adding remote users to a buddy list, users can monitor changes in the status of remote users in real time or they can monitor remote users as speed-dial contacts. Users can also manually specify their status in order to override or mask automatic status updates to others and can receive notifications when the status of a remote line changes.

Polycom phones support a maximum of 64 buddies for Open SIP server platforms and 200 contacts on the Skype for Business server. For information on Skype for Business contacts, refer to the *Polycom UC Software with Skype for Business - Deployment Guide* on Polycom Voice Support.

Configuring Presence Status

Use the parameters in the following table to enable the presence feature and display the **MyStatus** and **Buddies** soft keys on the phone.

Presence Status Parameters

Parameter Template	Permitted Values
`feature.presence.enabled` features.cfg	0 (default) - Disable the presence feature—including buddy managements and user status. 1 - Enable the presence feature with the buddy and status options.
`pres.idleSoftkeys` features.cfg	1 (default) - The **MyStat** and **Buddies** presence idle soft keys display. 0 - The **MyStat** and **Buddies** presence idle soft keys do not display.
`pres.reg` features.cfg	The valid line/registration number that is used for presence. This registration sends a SUBSCIRBE for presence. If the value is not a valid registration, this parameter is ignored. 1 (default) 1 - 34

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.

This feature is available on the following phones:

- VVX 401/411
- VVX 5xx series
- VVX 6xx series
- VVX 1500

For a list of supported USB devices, see *Technical Bulletin 38084: Supported USB Devices for Polycom SoundPoint IP 650 and VVX Phones* at Polycom Engineering Advisories and Technical Notifications.

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.

Call Forwarding with Skype for Business

The Skype for Business server automatically sends call forwarding functionality in-band to the phones. When Call Forwarding is enabled on the Skype for Business server, you can override Microsoft settings using the following Polycom parameters from a provisioning server or from the Web Configuration Utility:

- `feature.forward.enable` Enable or disable the call forwarding from the phone menu.
- `homeScreen.forward.enable` Enable or disable call forwarding icon on the Home screen.
- `softkey.feature.forward` Display or remove the Forward soft key.

In this case:

- If call forwarding is disabled on the Microsoft server then call forward feature is also disabled on the phone and the user cannot override Polycom parameters from a provisioning server or the Web Configuration Utility. To disable call forwarding sent in-band from the Microsoft server, you must disable the settings for call forwarding and simultaneous ring on the Microsoft server.
- To configure `softkey.feature.forward` parameter, you must configure `feature.enhancedFeatureKeys.enabled="1"`.

Enhanced Feature Line Key (EFLK)

This feature enables users with Microsoft-registered VVX 3xx, 4xx, 5xx, and 6xx business media phones to assign contacts to specific line keys on a VVX phone or expansion module. After an administrator enables ELFK using `feature.flexibleLineKey.enable` users can enable and disable the feature from the phone menu.

This feature is disabled by default and the phone displays registrations and contacts in the following order:

- Registration
- Enhanced Feature Key (EFK) as line key
- Shared Line Appearance (SLA) or Boss contacts
- Skype for Business favorites
- Favorites (Local contacts)

After you enable EFLK on the server, the user must sign into the phone and enable Custom Line Keys from the phone menu. The option to customize line keys is not available during active calls. After a user enables custom line keys on the phone, contacts on the phone's local contact directory are not available.

- Assign a Skype for Business contact to a line
- Clear a contact assigned to a line key or clear all customizations
- Delete a line key and the contact assigned to it
- Insert an empty line above or below a line key

Note the following points when using EFLK:

- Changes users make in Customized mode do not affect contacts in Default Mode.
- Deleting a contact from the Skype for Business client does not delete the contact from the phone.
- If a customized contact exists in both Boss Admin and self-contacts, then Boss Admin relation will be given higher precedence.

User customizations are uploaded to the phone and server as a .csv file in the following format:

- <MACaddress>-<sign-in address>.csv

The user .csv customization files cannot be edited manually. To apply a common customization to multiple phones, administrators can rename any user file by replacing the <MACaddress> part of the user file name with <000000000000>-<sign-in address>.csv. You must use centralized provisioning to share custom .csv files.

EFLK Limitations

Note the following limitations when using EFLK:

- The .csv file is always stored in the root directory and you cannot use a sub-directory.
- The phone does not load the .csv file when checking the server for updates using check sync.
- The user cannot configure Speed Dials and Enhanced Feature Key (EFK) as line key.
- The previous FLK feature using `lineKey.reassignment.enabled` does not work with UC Software 5.4.1 or later on phones using a Skype for Business Base Profile. The later EFLK feature requires UC Software 5.4.1 or later.

Configuring EFLK

Use the following parameters to configure the Enhanced Feature Line Key feature for devices registered with Skype for Business.

EFLK Parameters

Parameter Template	Permitted Values
`feature.flexibleLineKey.enable` features.cfg	0 (default) - The EFLK feature is disabled. 1 - The EFLK feature is enabled and Line Key Customization is added to the phone at Settings > Basic > Line Key Customization.

Configuring Boss-Admin

The Boss-Admin feature enables users to assign delegates to share a line so that both can place, answer, hold, transfer calls, and set ringtones on the delegate line. Phones in a Boss-Admin group can receive up to five incoming calls at the same time. Bosses can assign up to 25 delegates to their line, and a delegate can be assigned up to 15 bosses depending on the availability of line keys on the phone. If a VVX Expansion Module is connected to the phone, all VVX phones can support up to 18 bosses.

Boss-Admin is supported with Skype for Business, Lync 2013, and Lync 2010.

You set up Boss-Admin from the Skype for Business client application on a computer.

- For instructions on Boss-Admin functions, see the *Polycom VVX Business Media Phone - User Guide* on Polycom Voice Support.
- If you are using Lync Server 2010, refer to Configure Boss-Admin for Lync Server 2010 for necessary additional steps.
- To enable the Safe Transfer feature and soft key for Boss-Admin, refer to Safe Transfer for Boss-Admin.

The following table includes the maximum number of line keys available that can be used as Boss lines.

Maximum Delegate Line Keys for Assigned Bosses

VVX Phone	Maximum Bosses Assigned
201	1
300/301/310/311	5
400/401/410/411	11
500/501	11
600/601	15

Configure Boss-Admin for Lync Server 2010

If you are using Lync Server 2010, an administrator must complete the following procedure.

To configure Boss-Admin for Lync Server 2010:

1 Add the following SQL write operation command to a row in a static SQL database table:

```
osql -E -S se.fabrikam.com\RTC -Q "use rtc;exec RtcRegisterCategoryDef N'dialogInfo'"
```

You need to substitute the path to the RTC presence back end, shown as <se.fabrikam.com> in this example.

The SQL server operation is sent to the presence back end and must be run in every pool you need to enable.

2 Run the command.

3 Run the following command to verify that the category is registered

```
osql -E -S se.fabrikam.com\RTC -Q "use rtc;select * from CategoryDef"
```

You must substitute the path to the RTC presence back end, shown as <se.fabrikam.com> in this example.

Safe Transfer for Boss-Admin

A safe transfer transfers a call to another party and allows you to continue monitoring the call with the option to resume before the call goes to voicemail. If the call is answered by the other party, you are disconnected from the call.

Configuring Safe Transfer

The following parameters configure safe transfer for the Boss-Admin feature.

Configure Safe Transfer

Parameter Template	Permitted Values
`feature.lyncSafeTransfer.enabled` features.cfg	0 (default) - Disable the safe transfer feature and display of the Safe Transfer soft key. 1 - Enable the safe transfer feature and display of the Safe Transfer soft key.

Busy Options to Manage Incoming Calls

Busy Options enables users to manage incoming calls when a call or conference is already in progress. After you enable and configure the Busy Options on the Skype for Business server, Busy Options settings take effect on all Skype for Business call devices and clients. You can enable one of the following predefined settings on the devices:

- BusyonBusy: Rejects an incoming call and sends a notification to the caller stating that the user is busy on another call.
- VoicemailonBusy: Forwards an incoming call to voicemail, when the user is either busy or does not answer the call.

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.

This feature is not supported on VVX 201 phones. Check with your system administrator to find out if this feature is available on your phone.

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

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.international Dialing.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.international Prefix.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.

Configure the Local Digit Map

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	dialplan.applyToCallListDial	Choose whether the dial plan applies to numbers dialed from the received call list or missed call list, including sub-menus. 1 (default) 0	Yes
site.cfg	dialplan.applyToDirectoryDial	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

Configure the Local Digit Map

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

Configure the Local Digit Map

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

Configure the Local Digit Map

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

Configure the Local Digit Map

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

Configure the Local Digit Map

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.

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
site.cfg	`dialplan.TranslationInAutoComp`	1 (default) - The translated string displays in the auto-complete list. 0 - The translated string does not display in the auto-complete list.	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)

Enhanced 911 (E.911)

This E.911 feature allows you to configure one of three sources the phone obtains location information from:

- LLDP-MED
- DHCP via DHCP option 99
- LIS compliant with RFC 5985

Configuring the source of location information allows the phone to share its location details with 911 operators dispatching emergency services.

Enhanced 911 (E.911) Parameters

Use the following parameters to configure E.911.

E.911 Parameters

Template	Parameters	Permitted Values	Change Causes Restart or Reboot
site.cfg	feature.E911.HELD.server	NULL (default) Set the IP address or hostname of the Location Information Server (LIS) address. For example, host.domain.com or https://xxx.xxx.xxx.xxx.	No
site.cfg	feature.E911.HELD.username	NULL (default) Set the user name used to authenticate to the Location Information Server.	No
site.cfg	feature.E911.HELD.password	NULL (default) Set the password used to authenticate to the Location Information Server.	No
site.cfg	feature.E911.HELD.identity	Set the vendor-specific element to include in a location request message. For example, ‘companyID’. NULL (default) String 255 character max	No
site.cfg	feature.E911.HELD.identityValue	Set the value for the vendor-specific element to include in a location request message. NULL (default) String 255 character max	No
site.cfg	feature.E911.locationRetryTimer	Specify the retry timeout value in seconds for the location request sent to the Location Information Server (LIS). The phone stops retries after receiving location information received the LIS. 60 seconds (default) 60 - 86400 seconds	No
site.cfg	feature.E911.HELD.nai.enable	You can include or omit the Network Access Identifier (NAI) containing the SIP user information used to subscribe to the Location Information Server (LIS). 0 (default) – The NAI is omitted as a device identity in the location request sent to the LIS. 1 - The NAI is included as a device identity in the location request sent to the LIS.	No

E.911 Parameters

Template	Parameters	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`locInfo.source`	Specify the source of phone location information. This parameter is useful for locating a phone in environments that have multiple sources of location information. LLDP (default for Generic Base Profile) – Use the network switch as the source of location information. MS_E911_LIS (default for Lync Base Profile)– Use the Skype for Business Server as the source of location information. CONFIG – You can manually configure the source of location information. Skype only. LIS – Use the location information server as the source of location information. Generic Base Profile only. DHCP – Use DHCP as the source of location information. Generic Base Profile only. If location information is not available from a default or configured source, the fallback priority is as follows: Generic Base Profile: No fallback supported for Generic Base Profile Lync Base Profile: MS_E911_LIS > CONFIG > LLDP	No
`site.cfg`	`feature.E911.enabled`	0 (default) – Disable the E.911 feature. The INVITE sent for emergency calls from the phone does not include the geolocation header, geolocation option in supported header, geolocation-routing header, or the GEOPRIV location object. 1 – Enable the E.911 feature. The INVITE sent for emergency calls from the phone includes the geolocation header defined in RFC 6442 and PIDF presence element as specified in RFC3863 with a GEOPRIV location object specified in RFC4119 for in Open SIP environments. Note: This parameter is mutually exclusive of the GENBAND E.911 feature and if this parameter and `feature.genband.E911.enabled` are enabled, this parameter takes precedence.	No

E.911 Parameters

Template	Parameters	Permitted Values	Change Causes Restart or Reboot
site.cfg	feature.E911.HELD.requestType	Any (default) - Send a request to the Location Information Server (LIS) to return either 'Location by Reference' or 'Location by Value'. Note this is not the 'Any' value referred to in RFC 5985. Civic – Send a request to the LIS to return a location by value in the form of a civic address for the device as defined in RFC 5985. RefID – Send a request to the LIS to return a set of Location URIs for the device as defined in RFC 5985.	No
site.cfg	voIpProt.SIP.header.priority.enable	0 (default) – Do not include a priority header in the E.911 INVITE message. 1 - Include a priority header in the E.911 INVITE message.	No
site.cfg	voIpProt.SIP.header.geolocation-routing.enable	0 (default) – Do not include the geolocation-routing header in the E.911 INVITE message. 1 - Include the geolocation-routing header in the E.911 INVITE message.	No
site.cfg	feature.E911.HELD.secondary.server	Set the IP address or hostname of the secondary Location Information Server (LIS) address. For example, host.domain.com or https://xxx.xxx.xxx.xxx. NULL (default) Dotted-decimal IP address Hostname Fully-qualified domain name (FQDN)	No
site.cfg	feature.E911.HELD.secondary.username	Set a user name to authenticate to the secondary Location information Server (LIS). NULL (default) String	
site.cfg	feature.E911.HELD.secondary.password	Set a password to authenticate to the secondary LIS. NULL (default) String	
site.cfg	feature.E911.usagerule.retransmission	0 (default) - The recipient of this Location Object is not permitted to share the enclosed Location Information, or the object as a whole, with other parties. 1 - Distributing this Location is permitted.	No

System Display

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

Skype for Business User Interface Enhancements

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

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

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.

Web Proxy Auto Discovery (WPAD)

The Web Proxy Auto-Discovery Protocol (WPAD) feature enables Polycom phones to locate the URL of a Proxy Auto- Configuration (PAC) file you configure. Microsoft recommends using Blue Coat proxy with this feature.

You can configure WPAD using configuration parameters on your provisioning server, DHCP Option 252, or DNS-A protocol mechanism to discover the PAC file location. When using a provisioning server or DHCP, the phone looks for the file name you specify. If using DNS-A, the phone looks only for the wpad.dat file.

The priority for PAC file searching is as follows, from first to last:

- Provisioning server
- DHCP Option 252
- DNS-A

Note: If the proxies you configure in the PAC file or configuration file are either invalid or unreachable with a working fallback proxy, the time to register with Skype for Business is delayed and the responsiveness of features that support WPAD degrade.

Polycom phones support Digest and NTLM Authentication mechanisms to authenticate with a proxy server. To allow you to configure proxy-specific credentials common to all users, Basic Authentication is supported only when using the following parameters on a provisioning server:

- `feature.wpad.proxy.username`
- `feature.wpad.proxy.password`

Polycom supports the following list of HTTP/HTTPS services with Skype for Business:

- Registration Services
- Address Book Service (ABS)
- Location Information Sever (LIS)
- Device Update (Note: To ensure reliable software updates, device update is direct in case a proxy is not available.)
- Server Log Upload
- Core File Upload
- Exchange Services Provisioning

View WPAD Diagnostic Information

You can access important WPAD diagnostic information to track HTTP and HTTPS traffic flowing via the proxy you configure for WPAD. You can view diagnostic information on a pre-phone basis by logging into the Web Configuration Utility.

From the WPAD setting, you can:

- View if the WPAD PAC file fetch is successful
- View the configured method used to fetch the PAC file and source URLs
- View the DNS domain if configured

- View PAC file expiry details
- View the Exchange and Upload proxy
- Download the PAC file

To view WPAD diagnostic information:

1. Enter your phone's IP address into a web browser.
2. Select **Admin** as the login type, enter the administrator password (the default is 456), and click **Submit**.
3. Go to **Diagnostics >Skype for Business Status > WPAD**.

WPAD Configuration Parameters

The following parameters configure the WPAD feature.

WPAD Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.wpad.enabled`	1 (default) - Enable WPAD. 0 - Disable WPAD. You can configure values for this parameter from your provisioning server or from the phone.	Yes
`features.cfg`	`feature.wpad.curl`	Enter the Proxy Auto-Configuration (PAC) file location.	Yes
`features.cfg`	`feature.wpad.proxy`	Configure the web proxy server address. If you configure this parameter with a proxy address, the VVX phones do not discover DHCP or DNS-A or fetch the PAC file even if you configure a PAC file location using `feature.wpad.curl`. You can specify multiple proxies using this parameter by separated each with a semicolon the same way you specify them in the PAC file. For example: PROXY 0.10.1.1:8080; PROXY 10.12.2.1:8080	Yes

WPAD Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.wpad.proxy.username	Enter the user name to authenticate with the proxy server.	Yes
features.cfg	feature.wpad.proxy.password	Enter the password to authenticate with the proxy server. The credentials you can use depend on how authentication is enabled on the proxy server. You can use administrator or user credentials. If Skype for Business Active Directory is integrated with the proxy server, you do not need to configure user name or password credentials.	Yes

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

STUN / TURN / ICE Parameters

Template	Parameter Template	Permitted Values	Change Causes Restart or Reboot
firewall-nat.cfg	tcpIpApp.ice.turn.server	Enter the IP address of the TURN server. NULL (default)	No
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.

Polycom Manual BToE PC Pairing

This feature enables users to manually pair their VVX business media phone with their computer using the Polycom Better Together over Ethernet Connector application.

You can pair and unpair the VVX business media phone with the BToE application installed in a Citrix Virtual Desktop Infrastructure. For more information, see *Polycom® Better Together over Ethernet Connector 3.7.0 Release Notes* available on Polycom Support.

When you enable this feature users can select Auto or Manual pairing mode in the Web Configuration Utility or in the Administrator Settings menu on the phone. However, the manual pairing feature no longer requires you to connect the Ethernet cable from your computer to the PC port on your phone. By default, BToE and BToE pairing are enabled for phones registered with Skype for Business. When an administrator disables BToE pairing, users cannot pair their VVX phone with their computer using BToE. When the phone is set to manually pair with your computer connected to a reachable network, the phone generates a pairing code that users must enter into the Polycom BToE Connector application to pair.

To use the Manual Pairing feature, users must update to UC Software version supported to the corresponding BToE Connector application version. The following table lists the supported UC Software version for the corresponding BToE Connector application for Manual Pairing

Supported UC Software Version for Manual Pairing

UC Software Version	BToE Application Version	Manual Pairing	Automatic Pairing
UC Software version 5.7.0	BToE version 3.7.0	Yes	Yes
UC Software version 5.5.1	BToE version 3.4.0	Yes	Yes
UC Software version 5.5.1	BToE version earlier to 3.4.0	No	Yes
UC Software version earlier to 5.5.1	BToE version 3.4.0	No	Yes

BToE Widget

By default, users can access BToE settings from the phone menu at **Settings > Features > BToE**. You can configure a BToE widget to display on the phone's Home screen that allows direct user access to BToE settings. Enabling the BToE widget does not remove access via the phone menu.

BToE Widget Parameter

The following table includes the parameter to configure the BToE Widget.

BToE Widget Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`homeScreen.BToE.enable`	1 (default) - Displays the BToE widget on the phone's home screen. 0 - Does not display the BToE widget on the phone's home screen.	No

Enable or Disable BToE PC Pairing from the Phone

You can enable or disable the BToE PC Pairing feature for Better Together over Ethernet from the phone.

To enable or disable BToE PC Pairing from the phone:

1 On the phone, go to **Settings > Advanced**, and enter the administrator password.
2 Select **Administration Settings > BToE PC Pairing**.
3 Select **Enable** or **Disable**.

Enable or Disable BToE PC Pairing from the Web Configuration Utility

You can enable or disable the BToE PC Pairing feature from the Web Configuration Utility.

To enable or disable BToE PC Pairing in the Web Configuration Utility:

1 Enter the IP address of the phone into a web browser.
2 Log in as an **Admin**.
3 Go to **Settings > BToE PC Pairing**.
4 Check or uncheck **Enable BToE PC Pairing**.

Configuring Better Together over Ethernet (BToE) Firewall Ports

The following table lists ports used by BToE application and the communication direction.

BToE Firewall Ports

Port Number	Type	Description	Direction
24802	UDP	Used for audio streaming	Phone (24802) <=> PC (24802)
6000	TCP	Used for Secure Shell (SSH) client connections to the BToE application (plink.exe)	PC (BToE service) (Dynamic) => PC (plink service) (6000) (Within PC)
Dynamic	TCP	plink.exe uses a dynamic port to connect to VVX business media phones	PC (Dynamic) => Phone (22)
22	TCP	VVX business media phones use this port to connect securely with computer applications	PC (Dynamic) => Phone (22)
2081	UDP	VVX business media phones use this port for discovery packet broadcasts	Phone(2081) => PC (2081)
24801	TCP	VVX business media phones and the BToE computer application communicate with each other using this non-secure port	Phone (plink service) => Phone (BToE service) (24801)

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.

- As of UC Software 5.3.0, you can update software with the user-controlled software update feature explained in Configuring Automatic Software Update.
- If you are updating software from UC Software 4.0.x, refer to Polycom UC Software Update.

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

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:

- VVX 201, 300/310, 301/311, 400/410, 401/411, 500/501, and 600/601 business media phones
- SoundStructure VoIP Interface using Polycom UC Software 5.1.1 or later

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
`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

Experience Cloud Parameters

Parameter Template	Permitted Values
`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.

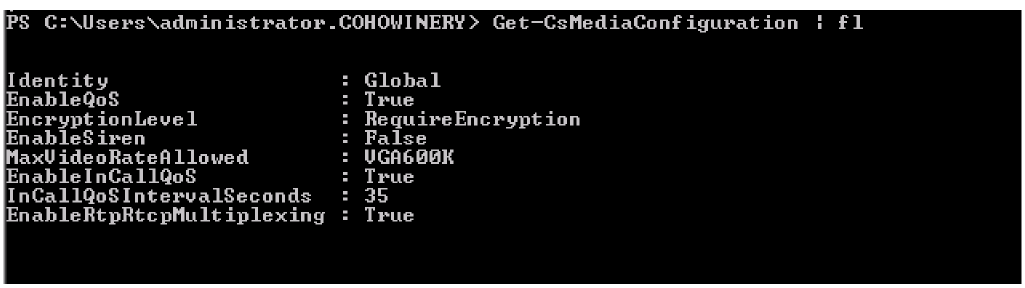
- `EnableQoE`. Set to 'True' to enable QoE on the server and automatically assign the URI to which QoE reports are published. If set to 'False' no QoE reports are published. Note that the URI maps to the in-band element '`qosUri`'. To get the current value of `EnableQoE`, run the command `Get-CsQoEConfiguration` 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
`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
	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

Polycom-Supported Skype for Business QoE Elements

Parent Element	Child Elements/Attributes
	InboundStream
	OutboundStream
Description	Connectivity
	Security
	Transport
	LocalAddr
	RemoteAddr
	v3:ReflexiveLocalIPAddress
	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

Polycom-Supported Skype for Business QoE Elements

Parent Element	Child Elements/Attributes
BurstGapLoss	BurstDensity
	BurstDuration
	GapDensity
	GapDuration
Delay	RoundTrip
	RoundTripMax
Utilization	Packets
Payload	Audio
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

Polycom-Supported Skype for Business QoE Elements

Parent Element	Child Elements/Attributes
	SendSignalLevelCh1
	SendNoiseLevelCh1
QualityEstimates.Audio	RecvListenMOS
	RecvListenMOSMin
	NetworkMOS
NetworkMOS	OverallAvg
	OverallMin

Quality of Service for Audio Calls

When the Quality of Service (QoS) setting is enabled on the Skype for Business server, VVX phones receive the Differentiated Services Code Point (DSCP) value from the server for Quality of Service (QoS) of audio calls placed or received from phones registered to Skype for Business server.

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.

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.
8 Confirm that the phone is running a Skype for Business-enabled Polycom UC Software version.
   - On the VVX 500 Business Media phone, choose **Settings > Status > Platform > Application > Main**. The UC Software version displays beside Version.

Web Info: You can use the Web Configuration Utility to update your Polycom UC Software. For details on how to update the phone software using the Web Configuration Utility, see Feature Profile 67993: Using the Software Upgrade Option in the Web Configuration Utility.

Configuring UC Software Automatic Updates

When you register VVX phones running UC Software 5.x.x, by default the phones poll the Skype for Business Server for software updates and automatically download updated software. This automatic software update feature is available on all devices using UC Software 5.0.0 and later registered with Skype for Business Server. As of UC Software 5.3, when you use automatic software updates, the phone notifies users of the software and prompts users to choose when to update the software. The user options are detailed in the *Polycom VVX Business Media Phones for Skype for Business - User Guide* on Polycom UC Software Support Center.

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

Automatic Software Update Parameters

Parameter Template	Permitted Values
`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.
`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

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**.