Polycom Documentation

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6001 America Center Drive
San Jose, CA 95002
USA

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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.

Support for Lync 2010 is limited to testing of basic call scenarios. Microsoft support of Lync and Skype for Business is documented on Microsoft's website. Microsoft does not currently support IP phones on Lync 2010. For information, see IP Phones on Microsoft Support.

Audience, Purpose, and Required Skills

This guide is written for a technical audience.

You must be familiar with the following concepts before beginning:

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

Related Topics

Getting Started

UC Software Device Compatibility

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

- Polycom® VVX® 201 business media phones
- Polycom® VVX® 250 business IP phones (on-premise only)
- Polycom® VVX® 300, 301, 310, 311 business media phones
- Polycom® VVX® 350 business IP phones (on-premise only)
- Polycom® VVX® 400, 401, 410, 411 business media phones
- Polycom® VVX® 450 business IP phones (on-premise only)
- 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.

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)

Related Topics

Getting Started

Microsoft Qualified Phones

Polycom offers devices with an Open SIP or a Skype Base Profile. Polycom also 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.

Related Topics

Getting Started

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.

Related Topics

Microsoft Qualified Phones

Skype for Business Topologies

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

Related Topics

Getting Started

Supported Skype for Business Topologies

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

Note that VVX business IP phones support on-premises topologies only.

Table 1. 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

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

Related Topics

Skype for Business Topologies

Unsupported Skype for Business Topologies

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

Table 1. Unsupported Skype for Business Topologies

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

Related Topics

Skype for Business Topologies

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`

Related Topics

Getting Started

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.

Related Topics

Getting Started

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.

Table 1. 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	□	x
PIN Authentication	□	x
Attendant Console	□	x
Cross Pool	□	x
Media Bypass	□	x
Response Groups	□	x
Private Line	□	x
Web Sign In	x	□
Common Area Phone (CAP)	□	□
Host Desking	□	□
Enhanced Feature Line Key (EFLK)	□	□

Skype for Business Feature	Polycom with Skype for Business On-Premises	Polycom with Skype for Business Online
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	□	□

Skype for Business Feature	Polycom with Skype for Business On-Premises	Polycom with Skype for Business Online
Local Call Logs	□	□
Exchange Call Logs	□	□
Federated Calls	□	□
Simultaneous Ring	□	□
Dual Tone Multi Frequency	□	□
Emergency 911	□	□
Call Admission Control	□	□
Monitoring (Device Inventory)	□	□
Delegates	□	□
Team Call	□	□
Message Waiting Indicator	□	□
Exchange Integration	□	□
Exchange Calendar		
Extended Presence	□	□

Skype for Business Feature	Polycom with Skype for Business On-Premises	Polycom with Skype for Business Online
Visual Voicemail	□	□
Boss-Admin	□	□

Related Topics

Getting Started

Get Help

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

Related Topics

Getting Started

Polycom and Partner Resources

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

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

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

Related Topics

Get Help

Documentation Feedback

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

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

Related Topics

Get Help

Deploying Polycom Phones with Skype for Business

Polycom offers several methods to register your Polycom phones with Skype for Business.

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 configure the network. If you want to customize default settings, complete the following tasks:

- 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.

Procedure

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	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

Certificate Method	Description
(LDAP) Domain Name System (DNS)	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 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, VVX 250, 350, and 450 business IP phones, and SoundStructure VoIP Interface.

Related Topics

Deploying Polycom Phones with Skype for Business

Supported DHCP Sub-Options

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

Table 1. DHCP Option 43 Sub-Options

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

Related Topics

Configure the Network

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.

Note	To avoid placing the phone in a continuous reboot cycle, do not provision phones with UC Software from both a Microsoft server and your own provisioning server.

Procedure

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.
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.

Related Topics

Deploying Polycom Phones with Skype for Business

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.

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, you must enable access to the Web Configuration Utility.

For complete information on provisioning with Polycom UC Software, see the *Polycom UC Software Administrator Guide* on Polycom UC Software for Microsoft Deployments.

Related Topics

Deploying Polycom Phones with Skype for Business

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.

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 to **Skype** when provisioning with configuration files.

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

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

Related Topics

Provisioning the Phones

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.

Procedure

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.

Related Topics

Manual Provisioning Methods

Set the Base Profile of VVX Phones 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 before setting the Base Profile manually.

Procedure

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 **1, 4, 9** on the phone keypad for about 3 seconds to enter the MKC.

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.

Related Topics

Manual Provisioning Methods

Set the Base Profile of VVX Phones 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.

Procedure

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.

Related Topics

Manual Provisioning Methods

Set the Base Profile of VVX Phones 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.

Procedure

1. Press the phone's **Home/Menu** key.
2. From the idle screen, press and hold the following key combinations **1, 4, 9** on the phone keypad for about 3 seconds.
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.

   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.

Related Topics

Manual Provisioning Methods

Sign In Methods

Set the Base Profile from the Settings Menu

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

Procedure

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.

Related Topics

Manual Provisioning Methods

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. You cannot configure sign-in credentials using the Polycom Web Configuration Utility.

Procedure

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.

Related Topics

Manual Provisioning Methods

Accessing the Web Configuration Utility

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

Note	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.

Related Topics

Provisioning the Phones

Centralized Provisioning Methods

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

- Use Skype for Business Online or Microsoft Exchange Online to set up phones and configure features.
- 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.
- 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.
- Polycom recommends using device.* parameters to configure multiple devices and only if you are familiar with Polycom centralized provisioning and configuration files.

Related Topics

Centralized Provisioning

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:

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

Note	VVX 250, 350, and 450 business IP phones do not support Skype for Business Online. VVX business IP phones support Skype for Business on-premises only.

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.

Procedure

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` .

Related Topics

Centralized Provisioning

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.

Procedure

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.

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

2. Go to Skype for Business Server and copy the CAB file to a C: drive directory.
3. Use the Skype for Business Server Management Shell to go to a particular directory.
4. In the Skype for Business Server Management Shell, run the following import command:

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

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

Related Topics

Centralized Provisioning

Deploying 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.

Related Topics

Centralized Provisioning

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.

Procedure

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`.

Related Topics

Centralized Provisioning

Configuring In-Band Provisioning Settings

You must provision phones using settings from either the in-band settings or your provisioning server and not both.

Where settings conflict, Skype for Business in-band provisioning device settings take precedence over the same settings configured on your provisioning server. If you are using your own provisioning server, avoid phone update loops by configuring `lync.provisionDeviceParams.enabled=0` to disable the following in-band provisioning device settings sent from the Skype for Business Server or Skype for Business Online:

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

Table 1. In-band Provisioning Device Settings

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
na	`lync.provisionDeviceParams.enabled`	1 (default) - Enable (accept) in-band provisioning device	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		settings sent from Skype for Business. 0 - Disable (block) in-band provisioning device settings sent from Skype for Business.	

Related Topics

Deploying Polycom Phones with Skype for Business

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.
- **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.
- **CAP Web Sign In**. Use this method to securely sign in from a browser on your computer or mobile device when CAP Admin Mode is enabled on the phone. The phone generates a unique pairing code you use to sign in to the CAP Provisioning Portal 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.

When an Admin changes the Active Directory password, the phone de-registers from the Skype for Business server with-in registration expiry value.

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.

While signing in to the phone, the phone displays sign-in progress messages such as **Discovering Skype for Business Server** or **Authentication in progress**. VVX 201 business media phones do not display these messages due to screen size limitations.

Set the Base Profile of VVX Phones Using MKC

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.

Table 1. Skype for Business Sign In Method Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg-advanced.cfg`	`reg.1.auth.loginCredentialType`	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.	No
`reg-advanced.cfg`	`reg.1.auth.useLoginCredentials`	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.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg-advanced.cfg`	`reg.1.auth.usePinCredentials`	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.	No

Related Topics

Sign In Methods

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"`

Related Topics

Configuring a Skype for Business Sign In Method and Credentials

PIN Authentication

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

You can enable PIN authentication for VVX business media phones, VVX business IP phones, and SoundStructure VoIP Interface registered with Skype for Business.

To use PIN authentication, you must enable the Web Configuration Utility, which is disabled by default. 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.

Related Topics

Sign In Methods

PIN Authentication Parameters

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

Table 1. PIN Authentication Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`device.cfg, features.cfg`	`device.logincred.extension`	NULL (default) - The 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.	No
`device.cfg, features.cfg`	`device.logincred.pin`	NULL (default) - If the default value is set, the 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.	No

Related Topics

PIN Authentication

Web Sign In for Skype for Business Online

Web Sign in is enabled by default on phones registered with Skype for Business Online and is available only for Skype for Business Online deployments.

Web Sign In enables users to securely log in to Skype for Business from the phone, a computer, or mobile web browser. The default maximum number of phones users can sign in to concurrently is eight. If a user is signed in to multiple phones and signs out from one phone, that user remains signed in to the other phones.

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.

If you are using Multi-Factor Authentication (MFA), you must use Web Sign In as the user sign in method with Polycom phones. Note that if you are using MFA and you enable the remember Multi-Factor Authentication option on Office 365, the phones automatically sign out after the number of days specified in Office 365. For more information on configuring Office 365, see Microsoft's Configure Azure Multi-Factor Authentication Settings.

Note	This feature is not supported on VVX 250, 350, and 450 business IP phones. VVX business IP phones support on-premises deployments only.

Related Topics

Sign In Methods

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.

Table 1. Skype for Business Online Web Sign-In Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`hidden.cfg`	`device.prov.AutoProvEnabled`	0 - URL registration is disabled. 1 (default) - URL registration is enabled.	
`features.cfg`	`feature.webSignIn.enabled`	1 (default) - In Skype for Business Base Profile, the web sign in option is	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		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.	No

Related Topics

Web Sign In for Skype for Business Online

Sign In with Better Together over Ethernet (BToE) on VVX Phones

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*.

Related Topics

Sign In Methods

Web Sign In CAP for VVX Phones with 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.

This sign in method is not applicable when the phone is signed in as a guest user. This feature is not available on VVX business IP phones.

Related Topics

Sign In Methods

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.

Related Topics

Sign In Methods

Disabling the Sign-In and Sign-Out Soft Keys on VVX Phones

If your 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.

Table 1. Sign-In and Sign-Out Soft Key Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.lync.hideSignInSignOut	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.	No
features.cfg	feature.lync.hideSignOut	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.	No
features.cfg	feature.lyncbtoe.autosignin.signoff.enabled	0 (default) - When the connection between VVX business media phone and BToE application is terminated, the credentials cached on the phone remains as is and the phone continues to stay signed in.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		1 - When the connection between VVX business media phone and BToE application is terminated, the credentials cached on the phone are removed and the phone triggers auto sign-off. Note: The auto sign-off triggers only when the phone was previously signed in using via PC sign-in method.	
`lync.cfg`	`softkey.feature.simplifiedSignIn`	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.	No

Related Topics

Sign In Methods

Microsoft Exchange Integration

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

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

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

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

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

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

Skype for Business

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

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

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

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

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

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

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

Related Topics

Microsoft Exchange Integration

Integrating with Microsoft Exchange

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

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

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

Related Topics

Microsoft Exchange Integration

Provision the Microsoft Exchange Calendar

You can provision your phones with the Microsoft Exchange calendar.

Procedure

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

Related Topics

Integrating with Microsoft Exchange

Enable Microsoft Exchange Calendar Using the Web Configuration Utility

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

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

Procedure

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

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

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

Related Topics

Integrating with Microsoft Exchange

Verify the Microsoft Exchange Integration

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

Procedure

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

Related Topics

Integrating with Microsoft Exchange

Configuring the Microsoft Exchange Server

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

Note	Web Info: For help with Lync Server 2010, refer to Microsoft Configure Exchange Services for the Autodiscover Service. For help with Lync Server 2013, refer to Microsoft Configuring Unified Messaging on Microsoft Exchange Server to work with Lync Server 2013.

Related Topics

Microsoft Exchange Integration

Visual Voicemail

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

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

Related Topics

Configuring the Microsoft Exchange Server

Synchronizing Call Logs

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

Related Topics

Configuring the Microsoft Exchange Server

Call Log Synchronization Parameters

Use the following parameters to configure call logs.

Table 1. Call Log Parameters

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

Related Topics

Synchronizing Call Logs

Directory Search

You can enable the ABS service on the Exchange server.

There are three possible configurations.

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

Related Topics

Configuring the Microsoft Exchange Server

Microsoft Exchange Parameters

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

Table 1. Microsoft Exchange Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`applications.cfg`	`exchange.meeting.alert. followOfficeHours`	1 (default) - Audible alerts occur during business hours. 0 - Audible alerts occur at all times.	No
`applications.cfg`	`exchange.meeting.alert. tonePattern`	positiveConfirm (default) - Set the tone pattern of the reminder alerts using any tone specified by `se.pat.*`.	No
`applications.cfg`	`exchange.meeting.alert. toneVolume`	10 (default) - Set the volume level of reminder alert tones. 0 - 17	No
`applications.cfg`	`exchange.meeting.allowScrollingToPast`	0 (default) - Do not allow scrolling up in the Day calendar view to see recently past meetings.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		1 - Allow scrolling up in the Day calendar view to see recently past meetings.	
applications.cfg	exchange.meeting.parseOption	Indicates the field in the meeting invite from which the VMR or meeting number should be fetched. Location (default) All LocationAndSubject Description	No
applications.cfg	exchange.meeting.phonePattern	NULL (default) string The pattern used to identify phone numbers in meeting descriptions, where "x" denotes any digit and "|" separates alternative patterns (for example, xxx-xxx-xxxx|604.xxx.xxxx).	No
applications.cfg	exchange.meeting.reminderEnabled	1 (default) - Meeting reminders are enabled. 0 - Meeting reminders are disabled.	No
applications.cfg	exchange.meeting.reminderInterval	300 seconds (default) 60 - 900 seconds Set the interval at which phones display reminder messages.	No

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

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

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.exchangeCalendar.enabled`	Available for: • Polycom Trio 8800 and 8500 systems • VVX 300/301, 310/311, 400/401, 410/411, 500/501, 600/601 and 1500 business media phones • VVX 250, 350, and 450 business IP phones • CX5500 Unified Conference Station 1 (default) - Lync Base Profile 0 (default) - Generic Base Profile 0 - The calendaring feature is disabled. 1 - The calendaring feature is enabled. You must enable this parameter if you also enable `feature.exchangeCallLog.enabled` . If you disable `feature.exchangeCalendar.enabled` , also disable `feature.exchangeCallLog.enabled` to ensure call log functionality.	No
`features.cfg`	`feature.exchangeContacts.enabled`	1 (default) - Lync Base Profile 0 (default) - Generic Base Profile 1 - The Exchange call log feature is enabled and users can retrieve the call log histories for missed, received, and outgoing calls.	No

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

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		You must also enable `feature.exchangeCalendar.enabled` to use the Exchange contact feature.	
`features.cfg`	`feature.exchangeVoiceMail. enabled`	1 (default) - The Exchange voicemail feature is enabled and users can retrieve voicemails stored on the Exchange server from the phone. 0 - The Exchange voicemail feature is disabled and users cannot retrieve voicemails from Exchange Server on the phone. You must also enable `feature.exchangeCalendar.enabled` to use the Exchange contact feature.	No
`features.cfg`	`feature.exchangeVoiceMail.skipPin.enabled`	0 (default) - Enable PIN authentication for Exchange Voicemail. Users are required to enter their PIN before accessing Exchange Voicemail. 1 - Disable PIN authentication for Exchange Voicemail. Users are not required to enter their PIN before accessing Exchange Voicemail.	No
`features.cfg`	`feature.lync.abs.enabled`	1 (default) - Lync Base Profile 0 (default) - Generic Base Profile 1 - Enable comprehensive contact search in the Skype for	No

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

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		displaying the voicemail summary.	

Related Topics

Configuring the Microsoft Exchange Server

Audio Features

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

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

This section describes the audio sound quality features and options you can configure for your Polycom phones. Use these features and options to optimize the conditions of your organization's phone network system.

Polycom NoiseBlock

Polycom NoiseBlock technology automatically mutes the microphone during audio-only and audio/ video calls when a user stops speaking.

This feature silences noises that interrupt conversations such as paper shuffling, food wrappers, and keyboard typing. When a user speaks, the microphone is automatically unmuted.

Related Topics

Audio Features

Polycom NoiseBlock Parameters

The following table includes the parameter you can use to configure the Polycom NoiseBlock feature.

Table 1. Polycom NoiseBlock Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
	`voice.ns.hf.blocker`	1 (default) - Enable Polycom NoiseBlock. 0 - Disable Polycom NoiseBlock.	No

Related Topics

Polycom NoiseBlock

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 250, 350, and 450 business IP phones support a single Opus stream. Users can establish only one call at a time when using the Opus codec on these phones.
- VVX 500 and 600 do not support video when using Opus.
- VVX 500 and 600 do not support local conferences when using Opus.
- Opus is not compatible with G.729 and iLBC. If you set Opus to the highest priority, G.729 and iLBC are not published; if you set G.729 and iLBC to the highest priority, Opus is not published.

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

Table 1. Audio Codec Priority

Phone	Supported Audio Codecs	Priority
VVX 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 VVX 250, 350, 450 * Note: VVX 301, 311, 401, 411 support a single Opus stream. VVX 300, 310, 400, 410 do not support Opus.	G.711µ-law	6
	G.711a-law	7
	G.722	4
	G.722.1 (24kbps, 32kbps)	5
	G.729AB	8
	iLBC (13.33kbps, 15.2kbps)	0, 0
	Opus*	0
	Siren 7	0
VVX 500/501, 600/601	G.711 µ -law	6
	G.711a-law	7

Phone	Supported Audio Codecs	Priority
◦ VVX 500 and 600 support a single Opus stream. ◦ VVX 500 and 600 do not support both Opus and video.	G.722	4
	G.722.1 (24kbps, 32kbps)	5
	G.722.1C (48kbps)	2
	G.729AB	8
	Opus*	0
	iLBC (13.33kbps, 15.2kbps)	0, 0
	Siren 7	0
VVX 1500	G.711 µ -law	6
	G.711a-law	7
	G.719 (64kbps)	0
	G.722	4
	G.722.1 (24kbps, 32kbps)	5
	G.722.1C (48kbps)	2
	G.729AB	8
	Siren14 (48kbps)	3
	iLBC (13.33kbps, 15.2kbps)	0, 0
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

Related Topics

Audio Features

Supported Audio Codec Specifications

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

Table 1. Audio Codec Specifications

Algorithm	Reference	Raw Bit Rate	Maximum IP Bit Rate	Sample Rate	Default Payload Size	Effective Audio Bandwidth
G.711 µ - law	RFC 1890	64 Kbps	80 Kbps	8 Ksps	20 ms	3.5 KHz
G.711 a-law	RFC 1890	64 Kbps	80 Kbps	8 Ksps	20 ms	3.5 KHz
G.719	RFC 5404	32 Kbps48 Kbps64 Kbps	48 Kbps64 Kbps80 Kbps	48 Ksps	20 ms	20 KHz
G.711	RFC 1890	64 Kbps	80 Kbps	16 Ksps	20 ms	7 KHz
G.722[1]	RFC 3551	64 Kbps	80 Kbps	16 Ksps	20 ms	7 KHz
G.722.1	RFC 3047	24 Kbps32 Kbps	40 Kbps48 Kbps	16 Ksps	20 ms	7 KHz
G.722.1C	G7221C	224 Kbps32 Kbps48 Kbps	40 Kbps48 Kbps64 Kbps	32 Ksps	20 ms	14 KHz
G.729AB	RFC 1890	8 Kbps	24 Kbps	8 Ksps	20 ms	3.5 KHz
Opus	RFC 6716	8 - 24 Kbps	24 - 40 Kbps	8 Ksps 16 Ksps	20 ms	3.5 KHz 7 KHz
Lin16	RFC 1890	128 Kbps256 Kbps512 Kbps705.6	132 Kbps260 Kbps516 Kbps709.6	8 Ksps16 Ksps32 Ksps44.1 Ksps48 Ksps	10 ms	3.5 KHz7 KHz14 KHz20 KHz22 KHz

Algorithm	Reference	Raw Bit Rate	Maximum IP Bit Rate	Sample Rate	Default Payload Size	Effective Audio Bandwidth
		Kbps768 Kbps	Kbps772 Kbps			
Siren 7	SIREN7	16 Kbps24 Kbps32 Kbps	32 Kbps40 Kbps48 Kbps	16 Ksps	20 ms	7 KHz
Siren14	SIREN14	24 Kbps32 Kbps48 Kbps	40 Kbps48 Kbps64 Kbps	32 Ksps	20 ms	14 KHz
Siren22	SIREN22	32 Kbps48 Kbps64 Kbps	48 Kbps64 Kbps80 Kbps	48 Ksps	20 ms	22 KHz
iLBC	RFC 3951	13.33 Kbps15.2 Kbps	31.2 Kbps24 Kbps	8 Ksps	30 ms20 ms	3.5 KHz
SILK	SILK	Skype SILK	6 - 20 Kbps 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	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.

Note The network bandwidth necessary to send the encoded voice is typically 5-10% higher than the encoded bit rate due to packetization overhead. For example, a G.722.1C call at 48kbps for both the receive and transmit signals consumes about 100kbps of network bandwidth (two-way audio).

Related Topics

Supported Audio Codecs

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.

Table 1. Audio Codec Parameters

Template	Parameter	Permitted Value	Default	Change Causes Restart or Reboot
`site.cfg`	`voice.codecPref.G711_A`	0 to 27	7	No
`site.cfg`	`voice.codecPref.G711_Mu`	0 to 27	6	No
`site.cfg`	`voice.codecPref.G719.32kbps`	0 to 27	0	No
`site.cfg`	`voice.codecPref.G719.48kbps`	0 to 27	0	No
`site.cfg`	`voice.codecPref.G719.64kbps`	0 to 27	0	No
`site.cfg`	`voice.codecPref.G722`	0 to 27	4	No
`site.cfg`	`voice.codecPref.G7221.24kbps`	0 to 27	0	No

Template	Parameter	Permitted Value	Default	Change Causes Restart or Reboot
site.cfg	voice.codecPref.G7221.32kbps	0 to 27	0	No
site.cfg	voice.codecPref.G7221_C.24kbps	0 to 27	5	No
site.cfg	voice.codecPref.G7221_C.32kbps	0 to 27	0	No
site.cfg	voice.codecPref.G7221_C.48kbps	0 to 27	2	No
site.cfg	voice.codecPref.G729_AB	0 to 27	8	No
site.cfg	voice.codecPref.iLBC.13_33kbps	0 to 27	0	No
site.cfg	voice.codecPref.iLBC.15_2kbps	0 to 27	0	No
site.cfg	voice.codecPref.Lin16.8ksps	0 to 27	0	No
site.cfg	voice.codecPref.Lin16.16ksps	0 to 27	0	No
site.cfg	voice.codecPref.Lin16.32ksps	0 to 27	0	No
site.cfg	voice.codecPref.Lin16.44_1ksps	0 to 27	0	No

Template	Parameter	Permitted Value	Default	Change Causes Restart or Reboot
site.cfg	voice.codecPref.Lin16.48ksps	0 to 27	0	No
site.cfg	voice.codecPref.Siren7.16kbps	0 to 27	0	No
site.cfg	voice.codecPref.Siren7.24kbps	0 to 27	0	No
site.cfg	voice.codecPref.Siren7.32kbps	0 to 27	0	No
site.cfg	voice.codecPref.Siren14.24kbps	0 to 27	0	No
site.cfg	voice.codecPref.Siren14.32kbps	0 to 27	0	No
site.cfg	voice.codecPref.Siren14.48kbps	0 to 27	3	No
site.cfg	voice.codecPref.Siren22.32kbps	0 to 27	0	No
site.cfg	voice.codecPref.Siren22.48kbps	0 to 27	0	No
site.cfg	voice.codecPref.Siren22.64kbps	0 to 27	1	No
site.cfg	voice.codecPref.SILK.8ksps	0 to 27	0	No

Template	Parameter	Permitted Value	Default	Change Causes Restart or Reboot
`site.cfg`	`voice.codecPref.SILK.12ksps`	0 to 27	0	No
`site.cfg`	`voice.codecPref.SILK.16ksps`	0 to 27	0	No
`site.cfg`	`voice.codecPref.SILK.24ksps`	0 to 27	0	No

Related Topics

Supported Audio Codecs

SILK Audio Codec

Polycom VVX 501 and 601 business media phones support the SILK audio codec.

Related Topics

Supported Audio Codecs

SILK Audio Codec Parameters

Use the following parameters to configure the SILK audio codec.

Table 1. SILK Audio Codec Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`voice.codecPref.SILK.8ksps`	Set the SILK audio codec preference for the supported codec sample rates.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		0 (default)	
site.cfg	voice.codecPref.SILK.12ksps	Set the SILK audio codec preference for the supported codec sample rates.	No
site.cfg	voice.codecPref.SILK.16ksps	Set the SILK audio codec preference for the supported codec sample rates. 0 (default)	No
site.cfg	voice.codecPref.SILK.24ksps	Set the SILK audio codec preference for the supported codec sample rates. 0 (default)	No
site.cfg	voice.audioProfile.SILK.8ksps.encMaxAvgBitrateKbps	Set the maximum average encoder output bitrate in kilobits per second (kpbs/s) for the supported SILK sample rate. 20 kbps (default) 6 – 20 kbps	No
site.cfg	voice.audioProfile.SILK.12ksps.encMaxAvgBitrateKbps	Set the maximum average encoder output bitrate in kilobits per second (kpbs/s) for the supported SILK sample rate. 25 kbps (default)	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		7 – 25 kbps	
site.cfg	voice.audioProfile.SILK.16ksps.encMaxAvgBitrateKbps	Set the maximum average encoder output bitrate in kilobits per second (kpbs/s) for the supported SILK sample rate. 30 kbps (default) 8 – 30 kbps	No
site.cfg	voice.audioProfile.SILK.24ksps.encMaxAvgBitrateKbps	Set the maximum average encoder output bitrate in kilobits per second (kpbs/s) for the supported SILK sample rate. 40 kbps (default) 12 – 40 kbps	No
site.cfg	voice.audioProfile.SILK.encComplexity	Specify the SILK encoder complexity. The higher the number the more complex the encoding allowed. 2 (default) 0-2	No
site.cfg	voice.audioProfile.SILK.encDTXEnable	0 (default) – Disable Enable Discontinuous transmission (DTX). 1 – Enable DTX in the SILK encoder. Note that DTX reduces the	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		encoder bitrate to 0bps during silence.	
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

Template	Parameter	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

Related Topics

SILK Audio Codec

Music on Hold

Music on Hold (MoH) enables you to play music when you place a call on hold.

You can specify on the provisioning server which music file the phone plays or upload a file using the phone's Web Configuration Utility. When MoH is enabled, you can turn the music on or off while the call is on hold. If you place multiple calls on hold, only the first call placed on hold hears the music.

The default MoH file size is 540 KB and the maximum file size is 600 KB. You can increase the max file size to 1014KB using the parameter `res.quotas.tone`. 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)

Related Topics

Audio Features

Upload a Music File

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

Procedure

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**.

Related Topics

Music on Hold

Configuring Music on Hold

The following table lists parameters that configure Music on Hold.

Table 1. Music on Hold Parameters

Template	Parameter	Permitted Values	
`features.cfg`	`feature.moh.enabled`	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	No

Template	Parameter	Permitted Values	
		feature.moh.filename.	
features.cfg	feature.moh.filename	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	No
features.cfg	feature.moh.payload	Specify the payload for RTP packets when music on hold is playing. For best phoneperformance, 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	No
features.cfg	res.quotas.tone	Set the maximum sample tone file size. 1024 KB 600 - 1024 KB	No

Related Topics

Music on Hold

Music on Hold Error Messages

If a music file fails to play, the phone displays one of the following messages to indicate the problem.

Table 1. MoH Error Messages

Message	Cause
Download failed	Phone failed to download the MoH file because the current file was in use. MoH file size is 0 A network failure occurred during download.
File size exceeded the maximum	File size exceeded the maximum. You can configure the maximum file size using `res.quotas.tone`.
Unsupported file format	The file you are uploading is not a supported file format.
Network is down	A network failure occurred during download.

Related Topics

Music on Hold

Phone Display and Appearances

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

Skype for Business User Interface on VVX Phones

The user interface for VVX 400, 500, and 600 series business media phones and VVX 250, 350, and 450 business IP phones matches 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.

Related Topics

Phone Display and Appearances

Reverse Name Lookup

You can configure the phone to display incoming caller names, outgoing recipient names, and the source the phone obtains names from.

The phone displays all Skype for Business participant names for the following functions:

- CCCP conference calls
- Local and remote participants for Boss-Admin calls
- Response group calls
- Team calls
- Voicemails
- Placed, Received, and Missed call lists

If the phone cannot match the number of the incoming or outgoing name to a name in your organization, the phone displays the name given in the SIP signaling.

If a user saves a contact in the phone's local contact directory, the call lists display that name regardless of the priority you configure.

All VVX phones support this Skype for Business feature except the following:

- VVX 101 business media phones
- VVX 150 business IP phones

Related Topics

Phone Display and Appearances

Reverse Name Lookup Parameters

The following parameters configure Reverse Name Lookup for Skype for Business.

Table 1. Reverse Name Lookup Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`up.rnl.priority`	disabled (default) - Feature is disabled and the phone uses the contact name given in the SIP signaling. This parameter overrides `up.useDirectoryNames` in the Skype Base Profile. Enter a comma-separated string, no spaces, for components you want to enable with Reverse Name Lookup. If you misconfigure the string, the parameter value falls back to the default priority order. The string is not case sensitive and can include any of the following values, listed here in the default priority order the phone looks for a matching name: • Outlook • SIP	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		• ABS • Local For example, if you configure “ABS,SIP,Outlook,Local”, the phone tries to match the incoming number with contact names in the order of components you list. If you do not configure the value "SIP" as one of the values, and the phone does not obtain the contact name using any one of the others values you configure, the phone uses the name given in the SIP signaling.	

Related Topics

Reverse Name Lookup

Time Zone Location Description

The following two parameters configure a time zone location description for their associated GMT offset:

- `device.sntp.gmtOffsetcityID` If you are not provisioning phones manually from the phone menu or Web Configuration Utility and you are setting the `device.sntp.gmtOffset` parameter, then you must configure `device.sntp.gmtOffsetcityID` to ensure that the correct time zone location description displays on the phone menu and Web Configuration Utility. The time zone location description is set automatically if you set the `device.sntp.gmtOffset` parameter manually using the phone menu or Web Configuration Utility.
- `tcpIpApp.sntp.gmtOffsetcityID` If you are not provisioning phones manually from the Web Configuration Utility and you are setting the `tcpIpApp.sntp.gmtOffset` parameter, then you must configure `tcpIpApp.sntp.gmtOffsetcityID` to ensure that the correct time zone location description displays on the Web Configuration Utility. The time zone location

description is set automatically if you set the `tcpIpApp.sntp.gmtOffset` parameter manually using the Web Configuration Utility.

Related Topics

Phone Display and Appearances

Time Zone Location Parameters

The following parameters configure time zone location.

Time Zone Location Parameters

Permitted Values		**Permitted Values**	
0	(GMT -12:00) Eniwetok,Kwajalein	61	(GMT +2:00) Helsinki,Kyiv
1	(GMT -11:00) Midway Island	62	(GMT +2:00) Riga,Sofia
2	(GMT -10:00) Hawaii	63	(GMT +2:00) Tallinn,Vilnius
3	(GMT -9:00) Alaska	64	(GMT +2:00) Athens,Istanbul
4	(GMT -8:00) Pacific Time (US & Canada)	65	(GMT +2:00) Damascus
5	(GMT -8:00) Baja California	66	(GMT +2:00) E.Europe
6	(GMT -7:00) Mountain Time (US & Canada)	67	(GMT +2:00) Harare,Pretoria
7	(GMT -7:00) Chihuahua,La Paz	68	(GMT +2:00) Jerusalem
8	(GMT -7:00) Mazatlan	69	(GMT +2:00) Kaliningrad (RTZ 1)
9	(GMT -7:00) Arizona	70	(GMT +2:00) Tripoli
10	(GMT -6:00) Central Time (US & Canada)		
11	(GMT -6:00) Mexico City	71	(GMT +3:00) Moscow
12	(GMT -6:00) Saskatchewan	72	(GMT +3:00) St.Petersburg
	(GMT -6:00) Guadalajara	73	(GMT +3:00) Volgograd (RTZ 2)

Permitted Values		Permitted Values	
13			
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
	(GMT -3:00) Cayenne,Fortaleza	90	(GMT +5:30) Kolkata,New Delhi

Permitted Values		Permitted Values	
30			
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)
	(GMT +1:00) Budapest		(GMT +8:00) Ulaanbaatar
	(GMT +1:00) Ljubljana		(GMT +9:00) Tokyo,Seoul,Osaka
	(GMT +1:00) Prague		(GMT +9:00) Sapporo,Yakutsk (RTZ 8)

Permitted Values		Permitted Values	
46 47 48 49 50	(GMT +1:00) Sarajevo,Skopje (GMT +1:00) Warsaw,Zagreb	106 107 108 109 110	(GMT +9:30) Adelaide,Darwin (GMT +10:00) Canberra
51 52 53 54 55 56 57 58 59 60	(GMT +1:00) Brussels (GMT +1:00) Copenhagen (GMT +1:00) Madrid,Paris (GMT +1:00) Amsterdam,Berlin (GMT +1:00) Bern,Rome (GMT +1:00) Stockholm,Vienna (GMT +1:00) West Central Africa (GMT +1:00) Windhoek (GMT +2:00) Bucharest,Cairo (GMT +2:00) Amman,Beirut	111 112 113 114 115 116 117 118 119 120	(GMT +10:00) Magadan (RTZ 9) (GMT +10:00) Melbourne (GMT +10:00) Sydney,Brisbane (GMT +10:00) Hobart (GMT +10:00) Vladivostok (GMT +10:00) Guam,Port Moresby (GMT +11:00) Solomon Islands (GMT +11:00) New Caledonia (GMT +11:00) Chokurdakh (RTZ 10) (GMT +12:00) Fiji Islands
		121	(GMT +12:00) Auckland,Anadyr (GMT +12:00) Petropavlovsk-Kamchatsky (RTZ 11)

Permitted Values		Permitted Values	
		122 123 124 125 126	(GMT +12:00) Wellington (GMT +12:00) Marshall Islands (GMT +13:00) Nuku'alofa (GMT +13:00) Samoa

Related Topics

Time Zone Location Description

Capture Your Device's Current Screen

You can capture your phone or expansion module's current screen.

VVX business IP phones and the Polycom Trio solution do not support expansion modules.

Before you can take a screen capture, you must provide power and connect the expansion module to a phone, and enable the phone's web server using the parameter httpd.enabled.

Procedure

1. In the sip-interop.cfg template, locate the parameter up.screenCapture.enabled .

   You can add the sip-interop.cfg template to the CONFIG-FILES field of the master configuration file, or copy the parameter to an existing configuration file.
2. Set the value to 1 and save the configuration file.
3. On the device, go to **Settings** > **Basic** > **Preferences** > **Screen Capture**.

   Note you must repeat step 3 each time the device restarts or reboots.
4. Locate and record the phone's IP address at **Status** > **Platform** > **Phone** > **IP Address**.
5. Set the phone to the screen you want to capture.
6. In a web browser address field, enter https://<phoneIPaddress>/captureScreen where <phoneIPaddress> is the IP address you obtained in step 5.

   The web browser displays an image showing the phone's current screen. You can save the image as a BMP or JPEG file.

Related Topics

Phone Display and Appearances

Capture Your Device's Current Screen Parameters

Use the following parameters to get a screen capture of the current screen on your device.

Table 1. Device's Current Screen Parameters

Template	Parameter	Permitted Values	Change Causes Reboot or Restart
`sip-interop.cfg`	`up.screenCapture.enabled`	0 (Default) - The Screen Capture menu is hidden on the phone. 1 - The Screen Capture menu displays on the phone. When the phone reboots, screen captures are disabled from the Screen Capture menu on the phone.	Yes
`sip-interop.cfg`	`up.screenCapture.value`	0 (Default) - The Screen Capture feature is disabled. 1 - The Screen Capture feature is enabled.	No

Related Topics

Capture Your Device's Current Screen

Port Usage

This section lists ports used by Polycom phones and ports you can configure.

Configuring Better Together over Ethernet (BToE) Firewall Ports for VVX Phones

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

Table 1. 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 phones	PC (Dynamic) => Phone (22)
22	TCP	VVX phones use this port to connect securely with computer applications	PC (Dynamic) => Phone (22)
2081	UDP	VVX phones use this port for discovery packet broadcasts	Phone(2081) => PC (2081)
24801	TCP	VVX phones and the BToE computer application communicate with each other using this non-secure port	Phone (plink service) => Phone (BToE service) (24801)

Related Topics

Port Usage

Configuring Security Options

Polycom UC Software enables you to optimize security settings.

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

Accessing the Web Configuration Utility

When the Base Profile of a phone is set to **Skype** , access to the Web Configuration Utility is disabled by default.

Administrators must enable access to a phone's Web Configuration Utility from the phone menu system or using configuration parameters.

If a phone Base Profile is set to **Skype** , or you use the centralized provisioning method to enter user credentials to the configuration files, the phone displays a screen prompting an administrator to change the default Admin password (456). Polycom strongly recommends that administrators change the default password. This password is not the Skype for Business Sign In password. The password you enter here is the same password administrators use to access the advanced settings on the phone menu and to log in to a phone's Web Configuration Utility as an administrator.

On the SoundStructure VoIP Interface, you must enable the Web Configuration Utility using configuration files on a provisioning server before you set the Base Profile to Skype. If you do not enable the Web Configuration Utility before setting the Base Profile to Skype, the Web Configuration Utility will not be available and you will need to reset the SoundStructure VoIP Interface to factory default settings.

Related Topics

Configuring Security Options

Set the Base Profile Using the Web Configuration Utility

Enable Access to the Web Configuration Utility From the Phone Menu

When the phone's Base Profile is set to Skype, you can enable access to a phone's Web Configuration Utility form the phone's menu system.

Procedure

1. On the phone's menu system, go to **Settings > Advanced**, enter the password (default 456), and go to **Administration Settings > Web Server Configuration**.

   Web Server and Web Config Mode display.
2. Set **Web Server** to **Enabled**.
3. Set **Web Config Mode** to **HTTP Only**, **HTTPS Only**, or **HTTP/HTTPS**.

Related Topics

Accessing the Web Configuration Utility

Configuring the Web Configuration Utility

The security update for Skype for Business includes a device parameter and a corresponding `device.set` parameter

Polycom recommends using `device.*` parameters only if you are familiar with the centralized provisioning method and with Polycom UC Software.

Use the following parameters to enable and configure the Web Configuration Utility.

Table 1. Web Configuration Utility Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`device.cfg, site.cfg`	`device.sec.coreDumpEncryption.enabled`	0 (default) 1	No
`device.cfg, site.cfg`	`device.sec.coreDumpEncryption.enabled.set`	0 (default) 1	No
`lyncSharedExample.cfg, lyncSharedLCExample.cfg`	`httpd.cfg.enabled`	Base Profile = Generic 1 (default) - The Web Configuration Utility is enabled.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		0 - The Web Configuration Utility is disabled. Base Profile = Skype, SkypeUSB 0 (default) - The Web Configuration Utility is disabled. 1 - The Web Configuration Utility is enabled.	
`lyncSharedExample.cfg`, `lyncSharedLCExample.cfg`	`httpd.cfg.secureTunnelRequired`	1 (default) - Access to the Web Configuration Utility is allowed only over a secure tunnel (HTTPS) and non-secure (HTTP) is not allowed. 0 - Access to the Web Configuration Utility is allowed over both a secure tunnel (HTTPS) and non-secure (HTTP).	No
`lyncSharedExample.cfg`, `lyncSharedLCExample.cfg`	`httpd.enabled`	Base Profile = Generic 1 (default) - The web server is enabled. 0 - The web server is disabled. Base Profile = Skype, SkypeUSB 0 (default) - The web server is disabled. 1 - The web server is enabled.	No

Related Topics

Accessing the Web Configuration Utility

Securing Audio Using Master Key Identifier (MKI)

For secure audio communications, Polycom phones offer support for the crypto header with and without MKI in the offer SDP.

The master key identifier (MKI) is an optional parameter to include the crypto header in the SDP that uniquely identifies the SRTP stream within an SRTP session. The far end can choose to include a crypt with or without MKI.

Table 1. MKI Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	sec.srtp.mki.enabled	1 (default) - The Polycom phone offers two cryptos in the SDP offer: one without an MKI, and one with a four-byte MKI parameter in the SDP message of the SIP INVITE / 200 OK. 0 - Polycom phone offers only one non-MKI crypto in the SDL offer.	No

Related Topics

Configuring Security Options

Administrator and User Passwords

You can change the default administrator and user passwords.

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

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

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

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

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

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

Related Topics

Configuring Security Options

Change the Default Administrator Password on the Phone

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

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

Procedure

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

Related Topics

Administrator and User Passwords

Change the Default Passwords in the Web Configuration Utility

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

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

Procedure

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

Related Topics

Administrator and User Passwords

Administrator and User Password Parameters

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

Table 1. Local Administrator and User Password Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`sec.pwd.length.admin`	The minimum character length for administrator passwords changed using the phone. Use 0 to allow null passwords. 1 (default) 0 -32	Yes
`site.cfg`	`sec.pwd.length.user`	The minimum character length for user passwords changed using the phone. Use 0 to allow null passwords. 2 (default) 0 -32	Yes

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

Related Topics

Administrator and User Passwords

Device Lock for Skype for Business

You can configure phones to be protected with a lock code that enables users to access personal settings from different phones.

You can configure Device Lock on the Skype for Business server or using Polycom parameters on a centralized provisioning server. If you enable Device Lock using both methods, centralized provisioning parameters take precedence. You cannot enable or disable Device Lock using the Web Configuration Utility.

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

Administrators can configure phone behavior after six unsuccessful user unlock attempts. If users forget their lock code, they can reset it from the phone when signed in to their Skype for Business account. If users sign in to their Skype for Business account using the Web Sign-In method, they cannot reset their lock code from the phone.

Users must sign into the phone before using Device Lock. If a phone restarts or reboots after a user sets the lock code, the phone is locked after the restart or reboot. Users can lock the phone from the phone screen or Skype for Business client when the phone and computer are connected using BToE. If Device Lock is used in conjunction with BToE, the phone and computer always remain synchronized if either the phone or computer restarts or reboots. If the BToE connection is broken between phone and computer, the phone is locked.

You can also:

- Define authorized outbound emergency numbers from a locked device
- Set up a minimum lock code length on the Skype for Business server

Related Topics

Configuring Security Options

Profile Photo on Device Lock Screen

When a user is signed in to their Skype for Business account on the Polycom Trio systems, VVX 400, 500, and 600 series business media phones, and VVX 250, 350, and 450 business IP phones, the user's Microsoft Exchange or public website profile photo displays on the Lock screen.

This feature is enabled by default when the Microsoft Exchange Service is enabled. Profile photos set using Active Directory are not supported and do not display on the phone.

Related Topics

Device Lock for Skype for Business

Adding Authorized Emergency Contacts on a Locked Device

You can configure emergency contact numbers that users can call on a locked device in one of two ways:

- Create a policy for emergency numbers on the Skype for Business Server. Note that this method must be supported by a voice routing trunk configuration.
- Create an authorized list for a line by configuring the value of the parameter `phoneLock.authorized.x.value` to a Tel URI or SIP URI, for example, `phoneLock.authorized.1.value="cwi57@cohovineyard.com"`.

When the Base Profile of the phone is set to Skype for Business, you can configure the phone to set the order of display for the authorized emergency numbers when the device is locked.

Related Topics

Device Lock for Skype for Business

Device Lock for Skype for Business Parameters

The following parameters configure the Skype for Business Device Lock feature.

Table 1. Device Lock Parameters

Parameter		Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.deviceLock.enable`	Enables or disables the Device Lock feature on the phone. 0 (default) - Device Lock is disabled. 1 - Device Lock is enabled.	No
`features.cfg`	`phoneLock.authorized.x.value`	Specify a registered line for 'x' and an authorized call list when the device is locked using a Tel URI or SIP URI, for example, `phoneLock.authorized.1.value="cwi57@cohovineyard.com"` .	No
`features.cfg`	`up.btoeDeviceLock.timeOut`	Configure a time delay after which the phone locks when the user	No

Parameter		Permitted Values	Change Causes Restart or Reboot
		locks the computer paired with the phone. 10 seconds (default) 0 - 40 seconds	
`features.cfg`	`up.configureDeviceLockAuthList`	EmergencyNumberAtTop (default) - The E911 emergency number will be displayed followed by authorized numbers when the phone is locked. EmergencyNumberAtBottom - The authorized numbers will be displayed followed by the E911 number when the phone is locked. EmergencyNumberDisabled - Only the authorized numbers will be displayed when the phone is locked.	No
`features.cfg`	`up.deviceLock.createLockTimeout`	Specify the timeout in minutes after which the Create Lock Code screen disappears and the user is signed out. 0 (default) - No timeout for the Create Lock Code prompt. 0 - 3 minutes - If the user does not provide input to the Create Lock Code within the time you specify, the Create Lock Code screen disappears and the user is signed out of the phone.	No

Parameter		Permitted Values	Change Causes Restart or Reboot
features.cfg	up.deviceLock.signOutOnIncorrectAttempts	Specify whether to sign out the user from the phone after six unsuccessful attempts to unlock the phone. 0 (default) - After six unsuccessful unlock attempts, the phone displays a message indicating a countdown of 60 seconds after which the user can attempt to unlock the phone. 1 - After six unsuccessful unlock attempts, the user is signed out of the phone, must sign in again, and is prompted to create a new lock code.	No

Related Topics

Device Lock for Skype for Business

Certificates

If you need to set up a remote worker, you must manually enter a certificate to the phone.

You also have the option to create your own XML configuration file and upload it to a phone using the Web Configuration Utility after you Accessing the Web Configuration Utility.

If you need to set up a remote worker, you must manually enter a certificate to the phone. You can add the certificate using parameters included in the `SkypeTrioSharedExample.cfg` file. You also have the option to create your own XML configuration file and upload it to a phone using the Web Configuration Utility.

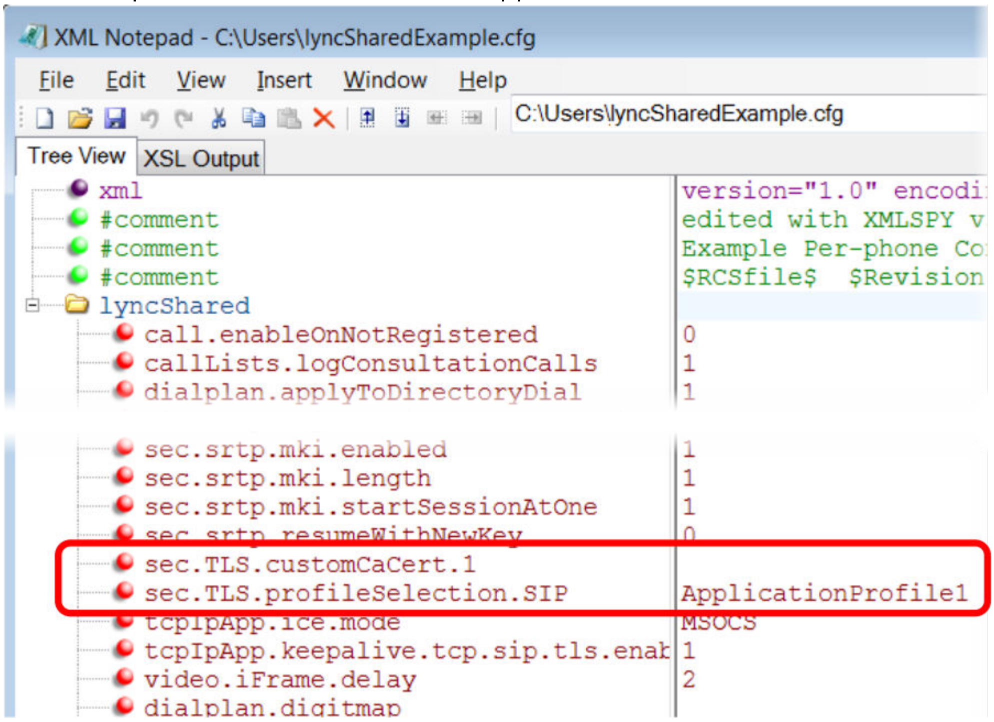
You can manually install certificates on a per-phone basis only. You must use Base64 format.

Install a Certificate on VVX Phones Using Configuration Files

You can manually install a certificate using configuration parameters in the template files available with UC Software.

1 Enter the following two parameters to a configuration file in your Skype for Business directory.
2 Enter the certificate and application profile as values for the two parameters:

- `sec.TLS.customCaCert.1=<enter the certificate>`
- `sec.TLS.profileSelection.SIP=<ApplicationProfile1>`

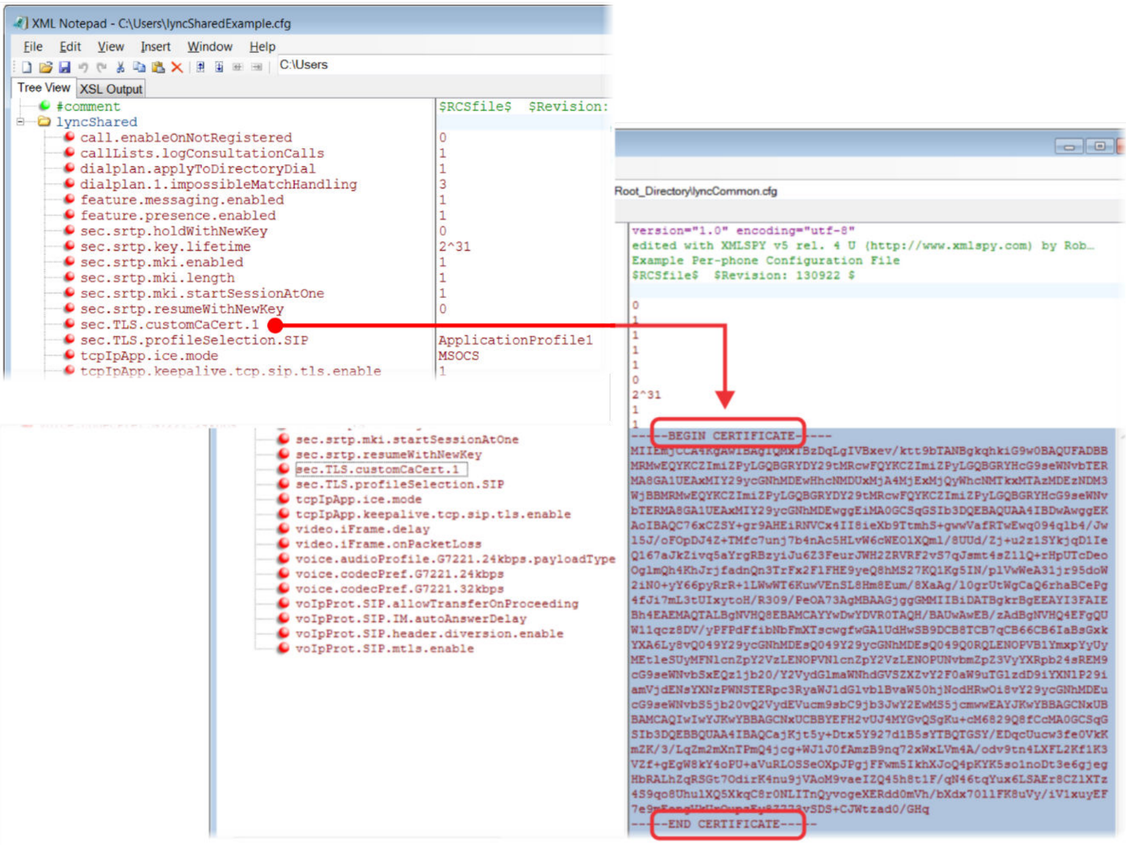


You can also enter the certificate by doing one of the following:

- Add the two parameters in an XML file you create with an XML editor.
- Add the two parameters to an existing configuration file you are using.

Procedure

- Enter the root CA certificate, in Base64 format, in `sec.TLS.customCaCert.1` and set the application profile in `sec.TLS.profileSelection.SIP` .



You have successfully installed a security certificate.

Related Topics

[Certificates]

Manually Install a Certificate with the Web Configuration Utility

You can use the Web Configuration Utility to install a certificate manually after you access the Web Configuration.

Procedure

1 Log into the Web Configuration Utility as an Administrator.
2 Go to **Utilities > Import & Export Configuration**.
3 Under **Import Configuration**, click **Choose File**.
4 In the dialog, choose the XML configuration file you created and click **Import**.

The XML configuration file is successfully loaded to the phone.

5 To verify that the file is loaded, go to **Menu > Settings > Status > Platform > Configuration**.

Related Topics

Certificates

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 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 the Polycom Documentation Library.

VVX business media phone supports Core File Upload when device crashes. The logs are uploaded to Skype for Business server in `tar.gz` format. The Skype for Business server must support `tar.gz` format to decrypt the log file uploaded to server.

Configure User Log Upload

The following table lists parameters that configure user log uploading.

Table 1. Configure User Log Uploading

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.logUpload.enabled`	1 (default) - Enable log uploads. 0 - Disable log uploads.	No

Related Topics

User Log Upload

Send Diagnostic Logs from the Phone

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

Procedure

- 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.

Related Topics

User Log Upload

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.

Procedure

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.

Related Topics

User Log Upload

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.

Related Topics

User Log Upload

Set Log Levels from the Phone

You can set log levels from the phone.

Procedure

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

Related Topics

Setting Log Levels

Set Log Levels from the Web Configuration Utility

You can set log levels from the Web Configuration Utility.

Procedure

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.

Related Topics

Setting Log Levels

Hardware and Accessories

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

Polycom Manual BToE PC Pairing on VVX Phones

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

When you enable this feature users can select Auto or Manual pairing mode in the Web Configuration Utility or in the Features 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.

Note	You can pair and unpair the VVX 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.

Table 1. Supported UC Software Version for Manual Pairing

UC Software Version	BToE Application Version	Manual Pairing	Automatic Pairing
UC Software version 5.8.0	BToE version 3.8.0	Yes	Yes
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	BToE Application Version	Manual Pairing	Automatic Pairing
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

Related Topics

Hardware and Accessories

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.

Related Topics

Polycom Manual BToE PC Pairing on VVX Phones

BToE Widget Parameters

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

Table 1. 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.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		0 - Does not display the BToE widget on the phone's home screen.	

Related Topics

BToE Widget

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.

Procedure

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**.

Related Topics

Polycom Manual BToE PC Pairing on VVX Phones

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.

Procedure

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**.

Related Topics

Polycom Manual BToE PC Pairing on VVX Phones

Directories and Contacts

You can configure phones with a local contact directory and link contacts to speed dial buttons.

Additionally, call logs stored in the Missed Calls, Received Calls, and Placed Calls call lists let you view user phone events like remote party identification, time and date of call, and call duration. This section provides information on contact directory, speed dial, and call log parameters you can configure on your Polycom phone.

Common Area Phone (CAP) for VVX Phones

You can configure your phone with Common Area Phone (CAP) Mode to restrict user's access to configuration settings on phones deployed in common areas, typically lobbies, employee lounges, and conference rooms.

You enable CAP Mode on a per-phone basis and CAP Mode is independent of any other configuration you make on the Skype server or apply to the Skype user account.

Note	Polycom recommends that you do not enable Boss-Admin or Shared Line Appearance while CAP is enabled.

Use of CAP requires UC Software 5.7.0 or later. After you enable this feature using `feature.CAP.enable=1`, CAP Mode and CAP Admin Mode are available on the phone. By default, CAP Mode is enabled and CAP Admin Mode is disabled.

While a phone is running in CAP Mode, users can access only basic settings and features. You can make more features available by enabling parameters for the corresponding feature, listed below.

Table 1. Features Available in CAP Mode

Soft Key / Menu	CAP Mode Default	Parameters to Enable
Status/DND	Disabled	`feature.doNotDisturb.enable` `softkey.feature.mystatus`
Call Forward	Disabled	`feature.forward.enable`
Device Lock	Disabled	`feature.deviceLock.enable`

Soft Key / Menu	CAP Mode Default	Parameters to Enable

Exchange Call Logs	Disabled	`Local logs:` `feature.callList.enabled`
		
		`Exchange call logs:` `feature.callList.enabled` `feature.exchangeCallLog.enabled` `feature.EWSAutodiscover.enabled`

Local Contact Directory	Disabled	`feature.directory.enabled`
Exchange Calendar	Disabled	`feature.EWSAutodiscover.enabled` `feature.exchangeCalendar.enabled` `homeScreen.calendar.enable`
Exchange Contacts	Disabled	`feature.EWSAutodiscover.enabled` `feature.exchangeContacts.enabled`
Exchange Voicemail/ Messages	Disabled	`feature.voicemail.enabled` `feature.EWSAutodiscover.enabled` `feature.exchangeVoiceMail.enabled` `feature.exchangeSipVMPlay.enabled`
Redial	Disabled	`homeScreen.redial.enable`

You can use the phone's administrator password to enable CAP Admin Mode. CAP Admin Mode provides access to all phone settings available from the phone interface. In addition, in CAP Admin Mode, the phone displays Sign In / Sign Out soft keys that allow you to sign users in or out of the phone. Alternatively, you can sign into a phone in CAP Mode without enabling CAP Admin Mode from the Common Area Phone provisioning portal at `https://aka.ms/skypecap` .

Any CAP-enabled phone that is not signed in with a Skype account and is left idle for three minutes displays a notice that the phone is not in use.

The following settings are available in CAP Admin Mode.

- Basic Settings
- Sign In/Sign Out

- My Status (under **Features > Presence > My Status**)

Related Topics

Directories and Contacts

Related Topics

Shared Phones with Skype for Business

Disable CAP Admin Mode on VVX Phones

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

Procedure

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

Related Topics

Common Area Phone (CAP) for VVX Phones

CAP Web Sign In on VVX Phones

After you enable the CAP feature and the phone is in CAP Mode, you can generate a code on the phone that you use to log into the Common Area Phone Provisioning Portal, a Microsoft web service that enables you to sign in multiple phones using any tenant account without the need to authenticate as a user on each phone.

You can log into the Common Area Provisioning Portal at https://aka.ms/skypecap using any account having administrator rights to the Microsoft tenant. Note that your Skype deployment must use Modern Authentication to access CAP web sign-in. For more information, see Skype for Business topologies supported with Modern Authentication on Microsoft Technet.

Note	Sign in using accounts that are designated only for the Common Area locations. The CAP portal is designed only for Common Area Phone accounts. Provisioning a CAP phone from the Provisioning Portal changes that phone's Active Directory user account password to a random string generated by Microsoft. For this reason, do not use the Provisioning Portal to sign in to a phone on behalf of an end user.

Related Topics

Common Area Phone (CAP) for VVX Phones

Sign In to a CAP-Enabled VVX Phone

You can sign out of a CAP-enabled phone using a code sent to the phone by the Common Area Provisioning Portal.

Procedure

1. While signed out of the phone, select Web Sign-in (CAP).

   The phone displays a code.
2. In the provisioning portal, enter the code in the field beside the user name and press Provision.

   The user's password is reset to a random string and the phone is signed in.

Related Topics

Common Area Phone (CAP) for VVX Phones

Common Area Phone Parameters

The following table lists parameters that configure the Common Area Phone (CAP) feature.

Use of CAP requires UC Software 5.7.0 or later.

Table 1. 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

Related Topics

Common Area Phone (CAP) for VVX Phones

Hot Desking for VVX Phones

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, users do not have permission to enable or disable Hot Desking.

Hot Desking Sign-In Methods

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

Host Desking Feature Limitations

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

Automatic Sign-Out Scenarios

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.

Related Topics

Directories and Contacts

Related Topics

Shared Phones with Skype for Business

Hot Desking Parameters

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

Table 1. 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

Related Topics

Hot Desking for VVX Phones

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 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.

Related Topics

Directories and Contacts

Configuring Contacts

The following table includes parameter to configure the Contact Directories.

Table 1. Configuring Contacts

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`up.queryContactInfo`	Enable or disable the parameter to retrieve the details of a specific contact from the Active Directory. 0 (default) 1 Make sure to enable `feature.lync.abs` enabled parameter.	No

Related Topics

Directories and Contacts

Call Logs

The phone records and maintains user phone events to a call log, which contains call information such as remote party identification, time and date of the call, and call duration.

The log is stored on the provisioning server as an XML file named <MACaddress>-calls.xml. If you want to route the call logs to another server, use the `CALL_LISTS_DIRECTORY` field in the master configuration file. All call logs are enabled by default.

The phones automatically maintain the call log in three separate call lists that users can access: Missed Calls, Received Calls, and Placed Calls. Users can clear lists manually on their phones, or delete individual records or all records in a group (for example, all missed calls).

Related Topics

Directories and Contacts

Call Log Parameters

Use the parameters in the following table to configure call logs

Table 1. Call Log Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg, features.cfg`	`callLists.collapseDuplicates`	Lync Base Profile – 0 (default) Generic Base Profile – 1 (default) 1 – Consecutive incomplete calls to/from the same party and in the same direction are collapsed into one record in the calls list. The collapsed entry displays the number of consecutive calls. 0 – Each call is logged individually in the calls list.	No
`site.cfg, features.cfg`	`callLists.logConsultationCalls`	ync Base Profile – 1 (default) Generic Base Profile – 0 (default) 0 – Consultation calls not joined into a conference call are not logged as separate calls in the calls list. 1 – Each consultation calls is logged individually in the calls list.	No
`features.cfg`	`feature.callList.enabled`	1 (default) - Allows you to enable the missed, placed, and received call lists on all phone	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		menus including the Home screen and dial pad. 0 - Disables all call lists. Hiding call lists from the Home screen and dial pad requires UCS 5.4.2 RevAA or higher.	
`features.cfg`	`feature.callListMissed.enabled`	0 (Default) - The missed call list is disabled 1 - The missed call list is enabled. To enable the missed, placed, or received call lists, `feature.callList.enabled` must be enabled.	No
`features.cfg`	`feature.callListPlaced.enabled`	0 (Default) - The placed call list is disabled 1 - The placed call list is enabled. To enable the missed, placed, or received call lists, `feature.callList.enabled` must be enabled.	No
`features.cfg`	`feature.callListReceived.enabled`	0 (Default) - The received call list is disabled 1 - The received call list is enabled. To enable the missed, placed, or received call lists, `feature.callList.enabled` must be enabled.	

Template	Parameter	Permitted Values	Change Causes Restart or Reboot

features.cfg	feature.exchangeCallLog.enabled	If Base Profile is:
		
		Generic - 0 (default)
		
		Skype for Business - 1 (default)
		
		1 - The Exchange call log feature is enabled, user call logs are synchronized with the server, and the user call log history of Missed, Received, and outgoing calls can be retrieved on the phone.
		
		You must also enable the parameter `feature.callList.enabled` to use the Exchange call log feature.
		
		• The value of the configuration parameter callLists.collapseDuplicates that collapses call lists has no effect in a Skype for Business environment.
		• The local call logs are not generated when the following parameters are disabled:
		  ○ feature.callListMissed.enabled
		  ○ feature.callListPlaced.enabled
		  ○ feature.callListReceived.enabled
		
		0 - The Exchange call log feature is disabled, the user call logs history cannot be retrieved from the Exchange server, and	

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		the phone generates call logs locally.	

Related Topics

Call Logs

Call Controls

This section shows you how to configure call control features.

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"` .

Related Topics

Call Controls

Enhanced Feature Line Key (EFLK) for VVX Phones

This feature enables users with Microsoft-registered VVX phones to assign contacts to specific line keys on a VVX phone or an expansion module connected to a VVX business media phone.

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 is not supported on VVX 201 business media phones.

VVX phones display 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.

Related Topics

Call Controls

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.

Related Topics

Enhanced Feature Line Key (EFLK) for VVX Phones

Configuring EFLK

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

Table 1. EFLK Parameters

Template	**Parameter**	**Permitted Values**	**Change Causes Restart or Reboot**
`features.cfg`	`feature.flexibleLineKey.enable`	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**.	

Related Topics

Enhanced Feature Line Key (EFLK) for VVX Phones

Configuring Boss-Admin for VVX Phones

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 a VVX business media phone, the phones can support up to 18 bosses.

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

You can 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 with Skype for Business - User Guide* on Polycom Voice Support.

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

Table 1. Maximum Delegate Line Keys for Assigned Bosses

Phone Model	Maximum Bosses Assigned
VVX 201 business media phone	1
VVX 250 business media phone	3
VVX 300/301/310/311 business media phone	5
VVX 350 business media phone	5
VVX 400/401/410/411 business media phone	11
VVX 450 business media phone	11
VVX 500/501 business media phone	11
VVX 600/601 business media phone	15

Related Topics

Call Controls

Configuring Boss-Admin Parameters

Use the following parameter to configure access for maximum number of Boss line keys for 500 and 600 series VVX business media phones.

Table 1. Boss-Admin Parameters

Template	Parameter	Parameter Template	Change Causes Reboot or Restart
`feature.cfg`	`up.numOfDisplayColumns`	Specify the maximum number of columns displayed on VVX 500 and 600 series business media phones. 2 (default) 1 – 4 Note that, the maximum number of columns supported on VVX 500 and 600 series phones are 3 and 4 respectively.	Yes

Related Topics

Configuring Boss-Admin for VVX Phones

Configure Boss-Admin on VVX Phones with Lync Server 2010

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

Procedure

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'dialogInf
   ```

   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.

Related Topics

Configuring Boss-Admin for VVX Phones

Safe Transfer for Boss-Admin on VVX Phones

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.

Related Topics

Call Controls

Configuring Safe Transfer on VVX Phones

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

Table 1. Configure Safe Transfer

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.lyncSafeTransfer.enabled`	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.	No

Related Topics

Safe Transfer for Boss-Admin on VVX Phones

Busy Options to Manage Incoming Calls on VVX Phones

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.

Related Topics

Call Controls

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 business media phones.

SLA Limitations

The following features are not supported on SLA lines:

- BToE
- Conference class
- Call Park

SLA Configuration for Skype for Business

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.

Table 1. SLA for Skype for Business Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
	`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. 0 - 25	

Related Topics

Call Controls

Related Topics

Shared Phones with Skype for Business

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.

Related Topics

Call Controls

Centralized Conference Control Protocol (CCCP) Parameters

The following parameters configure CCCP.

Table 1. CCCP Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
	`feature.cccp.enabled`	1 (enabled) - Enable use of CCCP. 0 - Disable use of CCCP.	

Related Topics

Centralized Conference Control Protocol (CCCP)

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.

Related Topics

Call Controls

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.

Related Topics

Dial Plans

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.

Related Topics

Dial Plans

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.

Table 1. Dial Plan, Dial Plan Normalization, and Digit Map Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`dialplan.1.digitmap`	x.T	No
`site.cfg`	`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)	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		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.	
`site.cfg`	`dialplan.1.lyncDigitmap.timeOut`	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.	No
`site.cfg`	`dialplan.TranslationInAutoComp`	1 (default) - The translated string displays in the auto-complete list.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		0 - The translated string does not display in the auto-complete list.	
`site.cfg`	`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`	Skype Base Profile: 1(default) Generic Base Profile: 0 (default) 1 - Enable dial plan normalization. Dial plan normalization rules are downloaded from the Microsoft Server and processed on the phone. 0 - Disable dial plan normalization. Dial plan rules are processed by the Microsoft Server.	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)	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		0 to 65535 seconds	

Related Topics

Dial Plans

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

Table 1. 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(+)	
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)	(1|2|3) or [1|2|3]	Match either 1, 2, or 3.

Number	Element	Meaning	Example	Description of Example
		character when all characters are surrounded by brackets or square brackets		
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)

Related Topics

Dial Plans

Hybrid Line Registration

VVX phones support the hybrid line registration feature that enables you to register a Skype for Business server on one line and an OpenSIP server on other lines.

Make sure to register Skype for Business server on line 1 and other lines with any Open SIP server. Using `reg.limit` parameter you can configure the hybrid line registration feature.

Note	When enabling hybrid line registration feature, set the value for `call.stickyAutoLineSeize` and `call.enableOnNotRegistered` parameter to 1.

In addition, you can choose the number of lines to configure by setting the appropriate value in the parameter. You can configure and register a maximum of three different servers using this parameter.

Note	Polycom VVX 101 business media phones and VVX 150 business IP phones do not support Hybrid Line Registration.

Skype for Business features behave the same way if the phone is hybrid mode enabled or not. However, when you enable the `up.oneTouchVoicemail` parameter, you can call voicemail services directly from the phone. Otherwise, the phone displays a summary page with message counts.

Hybrid Line Registration does not support Hot Desking and advanced OpenSIP features such as Shared Line Appearance, Busy Lamp field, and Automatic Call Distribution.

Note	When VVX phones are hybrid mode enabled, the H.323 protocol is not supported.

You can view the hybrid registration status along with the registered line specific information in the Web Configuration Utility. When you enable the hybrid line registration feature on your VVX phone, existing feature behaviors change to the following:

- **Do Not Disturb**
  The Do Not Disturb (DND) feature now provides an **All** option. When the user selects this option, DND is enabled on all registered lines. If only one line is registered to the server, users can set DND directly by navigating to **Settings** > **Features** > **DND**. The server-based DND behaves the same way in hybrid registration mode.
- **Recent Calls**
  Users can view all the registered line's information and call logs for the corresponding registered line under the **Recent Calls** menu. However, when a Skype for Business user signs out of the phone, the server-based call logs and the registered line's information for the Skype for Business server are not available.
- **Call Forwarding**
  Users can enable call forwarding on a per-registered line in a hybrid registration mode that allows them to forward an incoming call to a contact, voicemail, or delegates. The server-based call forwarding feature behaves the same way in hybrid registration mode.
- **Dial Plan**
  You can configure dial plans that enable the phone to select the line automatically for an outgoing call. Every line must have a unique dial plan. When a dial plan is not configured for a line, the value defined in the global parameter for a dial plan takes priority.
  Additionally, you can configure the line switching feature based on the dial plan when the phone is on-hook. The line switching feature enables the dialed number to switch to the corresponding line. For example, when you place a call from the phone and the number corresponds to an Open SIP line, the line switching feature enables the dialed number to switch to the corresponding line. Using the `reg.1.mergeServerDigitMapLocally` parameter, you can merge the dial plans received from the server with the dial plans configured for line 1 on your provisioning server. When a user dials a number directly from the call list or directory list, the value set in the corresponding parameter decides whether the dial plan must be parsed or not. You can also configure the dial plan for SIP URL dialing. For SIP URL dialing, the value defined in the per-registration dial plan parameter takes priority over the general dial plan parameter.
- **Conference Calls**
  Hybrid registration mode enables users to make Skype for Business and OpenSIP conference calls in parallel. Users can switch between OpenSIP and Skype for Business conference calls. However, you can't make conference calls between phones registered to Skype for Business and OpenSIP servers.
- **Securing Phone Calls with SRTP**
  When hybrid line registration is enabled, the Secure Real-Time Transport Protocol (SRTP) parameters for enable, require, and offer supports on per-line basis.
- **Enhanced 911**
  In hybrid registration mode, the Enhanced 911 service is supported only on the line registered with the Skype for Business server.
- **Presence Status**
  When hybrid line registration is enabled, users can set the presence status for the line registered to the Skype for Business server only. Users can view the device presence only on the line registered to the Skype for Business server.
- **Contact Directory**
  Users can access contact directories of all the registered lines. When user selects a contact from any directory and presses **Dial**, the call generates automatically based on the dial plan for the corresponding line without the need for the user to select the line. However, the home menu icon and presence status aren't available for the BroadSoft UC-One directory. You must set the `homeScreen.UCOne.enable` parameter value to `0`.
- **Call Park**

In hybrid registration mode, users can only park and unpark an active call to the corresponding registered line. For example, when an active call exists on a Skype for Business line and a user chooses to park the call, the call park applies only on the Skype for Business line.

- **Music on Hold**
  When a user places a call on hold using the registered line, the user hears only the call hold tone for the corresponding line. For example, when a Skype for Business call is put on hold, the user hears the Skype for Business hold tone.
- **Flexible Line Key**
  When the phone is in hybrid registration mode and the Flexible Line Key (FLK) feature is enabled, you can assign FLK to lines that are not hybrid line registered. However, when hybrid registration is enabled after FLK assignment, the feature replaces lines 2 and 3 configured with FLKs with hybrid registered lines.

Related Topics

Call Controls

Hybrid Line Registration Parameters

The following table includes parameters to configure the Hybrid Line Registration feature.

Table 1. Hybrid Line Registration Parameter

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg.cfg`	`reg.limit`	Specify the maximum number of lines to use for registration. 1 (default) 1 to 3 For VVX 201 business media phones, the maximum number of lines to use for registration is limited to 2.	Yes
`sip-interop.cfg`	`reg.1.mergeServerDigitMapLocally`	1 (default) - Allows the dial plans from `dialplan.1.digitmap` to append on top of the dial plans received from the server.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		0 - Does not allow the dial plans from `dialplan.1.digitmap` to append on top of the dial plans received from the server.	
`site.cfg`	`dialplan.digitmap.lineSwitching.enable`	0 (default) - Disables the line switching in dial plan to switch the call to the dial plan matched line. 1 - Enables the line switching in dial plan to switch the call to the dial plan matched line.	No
`features.cfg`	`reg.1.urlDialing.enabled`	0 (default) – Disables URL dialing. 1 – Enables URL dialing.	No
`site.cfg`	`tcpIpApp.port.rtp.lync.audioPortRangeStart`	Specifies the audio port range of start port for Lync/Skype for Business. 5350 (default) 1024 to 65436	No
`site.cfg`	`tcpIpApp.port.rtp.lync.videoPortRangeStart`	Specifies the video port range of start port for Lync/Skype for Business. 5406 (default) 1024 to 65486	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`tcpIpApp.port.rtp.lync.audioPortRangeEnd`	Specifies the audio port range of end port for Lync/Skype for Business. 5402 (default) 1024 to 65485	No
`site.cfg`	`tcpIpApp.port.rtp.lync.videoPortRangeEnd`	Specifies the video port range of end port for Lync/Skype for Business. 5458 (default) 1024 to 65535	No

Related Topics

Hybrid Line Registration

Support for PSTN Gateway on Failover

When a VVX phone becomes unregistered due to an outage and can't reach the Skype for Business server for a specified time interval, the phone fails over to an alternate PSTN gateway server.

You can view the PSTN failover details in the Web Configuration Utility.

When you enable this feature, calls switch to the configured PSTN gateway in the event of an outage. However, if the phone fails over, only basic call-related functions and soft keys are available.

Make sure the value of `call.enableOnNotRegistered` and `reg.x.srtp.simplifiedBestEffort` parameter is set to 1.

Note	The failover feature does not work if you enable the hybrid line registration feature.

Ensure the Direct Inward Dialing number registered on the Skype for Business server and the number used for the PSTN gateway are same.

Related Topics

Call Controls

PSTN Gateway Failover Parameters

The following table includes parameters to configure phones to fail over to an alternate PSTN gateway in the event of an outage or if the phones can't reach the Skype for Business server.

Table 1. PSTN Gateway Failover Parameters

Template	Parameter	Pemitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.sfbPstnFailover.enabled`	Enable or disable for phones to fail over to a PSTN gateway during an outage. 0 (default) 1	Yes
`site.cfg`	`reg.x.server.y.address`	If this parameter is set, it takes precedence even if the DHCP server is available. Null (default) - SIP server does not accepts registrations. IP address or hostname - SIP server that accepts registrations. This parameter is only applicable during a failover to PSTN gateway in Skype for Business deployments.	No
`reg-basic.cfg`	`reg.x.server.y.pstnServerAuth.userId`	Specify the user identification for the PSTN gateway. Null (default) String (maximum of 255 characters)	No

Template	Parameter	Pemitted Values	Change Causes Restart or Reboot
reg-basic.cfg	reg.x.server.y.pstnServerAuth.password	Specify the PSTN user's password. Null (default) String (maximum of 255 characters)	No

Related Topics

Support for PSTN Gateway on Failover

Presence Status

You can enable users to monitor the status of other remote users and phones.

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

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

Related Topics

Call Controls

Presence Status Parameters

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

Table 1. Presence Status Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.presence.enabled	0 (default) - Disable the presence feature—including	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		buddy managements and user status. 1 - Enable the presence feature with the buddy and status options.	
features.cfg	pres.idleSoftkeys	1 (default) - The MyStat and Buddies presence idle soft keys display. 0 - The MyStat and Buddies presence idle soft keys do not display.	No
features.cfg	pres.reg	The valid line/registration number that is used for presence. This registration sends a SUBSCIRBE for presence. If the value is not a valid registration, this parameter is ignored. 1 (default) 1 - 34	No

Related Topics

Presence Status

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.

Note	Federal, state, and/or local laws may legally require that you notify some or all of the call parties when a call recording is in progress.

This feature is available on the following devices:

- VVX 401, 411 business media phones
- VVX 5xx and 6xx series business media phones
- VVX 250, 350, and 450 business IP phones
- SoundStructure VoIP Interface

Related Topics

Call Controls

Local Call Recording Parameters

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

Local Call Recording Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.callRecording.enabled`	0 (default) - Disable audio call recording. 1 - Enable audio call recording.	Yes

Related Topics

Local Call Recording

Local Digit Map

The local digit map feature allows the phone to automatically call a dialed number when configured.

Dial plans apply on-hook when no Skype for Business line is registered or when line switching is enabled and at least one line has a non-empty dial plan.

Digit maps are defined by a single string or a list of strings. If a dialed number matches any string of a digit map, the call is automatically placed. If a dialed number matches no string—an impossible match—you can specify the phone's behavior. If a number ends with #, you can specify the phone's behavior, called trailing # behavior. You can also specify the digit map timeout, the period of time after you dial a number that the call is placed. The configuration syntax of the digit map is based on recommendations in section 2.1.5 of RFC 3435.

Note	For instructions on how to modify the local digit map, see *Technical Bulletin 11572: Changes to Local Digit Maps on SoundPoint IP, SoundStation IP, and Polycom VVX 1500 Phones* at Polycom Engineering Advisories and Technical Notifications.

Related Topics

Call Controls

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.

Table 1. Configure the Local Digit Map

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`dialplan.applyToCallListDial`	Choose whether the dial plan applies to numbers dialed from the received call list or missed call list, including sub-menus. 1 (default) 0	Yes
`site.cfg`	`dialplan.applyToDirectoryDial`	Lync Base Profile – 1 (default) Generic Base Profile – 0 (default) 0— The dial plan is not applied to numbers dialed from the directory or speed dial, including auto-call contact numbers. 1—The dial plan is applied to numbers dialed from the directory	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		or speed dial, including auto-call contact numbers.	
`site.cfg`	`dialplan.applyToForward`	Lync Base Profile – 1 (default) Generic Base Profile – 0 (default) 0—The dial plan does not apply to forwarded calls. 1—The dial plan applies to forwarded calls.	Yes
`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)	

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.	Yes
`site.cfg`	`dialplan.digitmap`	Specify the digit map used for the dial plan using a string compatible with the digit map feature of MGCP described in 2.1.5 of RFC 3435. This parameter enables the phone to automatically initiate calls to numbers that match a digit map pattern. Generic Base Profile (default) – `[2-9]11|0T|+011xxx.T|0[2-9]xxxxxxxxx|+1[2-9]xxxxxxxx|[2-9]xxxxxxxxx|[2-9]xxxT` Lync Base Profile (default) – NULL `[2-9]11|0T|+011xxx.T|0[2-9]xxxxxxxxx|+1[2-9]xxxxxxxx|[2-9]xxxxxxxxx|[2-9]xxxT` (default) The string is limited to 2560 bytes and 100 segments of 64 bytes,	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		and the following characters are allowed in the digit map • A comma (,), which turns dial tone back on. • A plus sign (+) is allowed as a valid digit. • The extension letter 'R' indicates replaced string. • The extension letter 'Pn' indicates precedence, where 'n' range is 1-9. 1—Low precedence 9—High precedence	
`debug.cfg`	`dialplan.filterNonDigitUriUsers`	Determine whether to filter out (+) from the dial plan. 0 (default) 1	Yes
`site.cfg`	`dialplan.impossibleMatchHandling`	0 (default)—The digits entered up to and including the point an impossible match occurred are sent to the server immediately. 1—The phone gives a reorder tone. 2—Users can accumulate digits and dispatch the call manually by pressing Send. If a call orbit number begins with pound (#) or asterisk (*), you need to set the value to 2 to retrieve the call using off-hook dialing.	Yes
`site.cfg`	`dialplan.removeEndOfDial`	Sets if the trailing # is stripped from the digits sent out.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		1 (default) 0	
`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
`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).	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		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.	
`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
`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	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		dialplan.routing.server.x.address . x=15: 911, others: Null (default) SIP URL (single entry) x is the index of the emergency entry description where x must use sequential numbering starting at 15.	
site.cfg	dialplan.routing.server.x.address	Set the IP address or hostname of a SIP server to use for routing calls. Multiple servers can be listed starting with x=1 to 3 for fault tolerance. Null (default) IP address hostname Blind transfer for 911 or other emergency calls may not work if registration and emergency servers are different entities.	Yes
site.cfg	dialplan.routing.server.x.port	Set the port of a SIP server to use for routing calls. 5060 (default) 1 to 65535	Yes
site.cfg	dialplan.routing.server.x.transport	Set the DNS lookup of the first server to use and dialed if there is a conflict with other servers. DNSnaptr (default) TCPpreferred UDPOnly	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		TLS TCPOnly For example, if `dialplan.routing.server.1.transport` = "UDPOnly" and `dialplan.routing.server.2.transport` = "TLS", then UDPOnly is used.	
`site.cfg`	`dialplan.userDial.timeOut`	Specify the time in seconds that the phone waits before dialing a number entered while the phone is on hook. Generic Base Profile (default) – 0 Lync Base Profile (default) – 4 0-99 seconds You can apply `dialplan.userDial.timeOut` only when its value is lower than `up.IdleTimeOut` .	No

Related Topics

Local Digit Map

Open SIP Digit Map

If you are using a list of strings, each string in the list can be specified as a set of digits or timers, or as an expression which the gateway uses to find the shortest possible match.

In addition, the digit map feature allows SIP URI dialing to match the URIs based on dial plan.

The following is a list of digit map string rules for open SIP environments.

- The following letters are case sensitive: x, T, R, S, and H.
- You must use only *, #, +, or 0-9 between the second and third R.
- If a digit map does not comply, it is not included in the digit plan as a valid map. That is, no match is made.
- There is no limit to the number of R triplet sets in a digit map. However, a digit map that contains less than a full number of triplet sets (for example, a total of 2 Rs or 5 Rs) is considered an invalid digit map.
- Digit map extension letter R indicates that certain matched strings are replaced. Using an RRR syntax, you can replace the digits between the first two Rs with the digits between the last two Rs. For example, *R555R604R* would replace 555 with 604. Digit map timer letter T indicates a timer expiry. Digit map protocol letters S and H indicate the protocol to use when placing a call.
- If you use T in the left part of RRR's syntax, the digit map will not work. For example, R0TR322R will not work.

The following examples illustrate the semantics of the syntax:

- R9R604Rxxxxxxx-Replaces 9 with 604
- xxR601R600Rxx-When applied to 1160122 gives 1160022
- R9RRxxxxxxx-Remove 9 at the beginning of the dialed number (replace 9 with nothing)
    - For example, if you dial 914539400, the first 9 is removed when the call is placed.
- RR604Rxxxxxxx-Prepend 604 to all seven-digit numbers (replace nothing with 604)
    - For example, if you dial 4539400, 604 is added to the front of the number, so a call to 6044539400 is placed.
- xR60xR600Rxxxxxxx-Replace any 60x with 600 in the middle of the dialed number that matches.
  For example, if you dial 16092345678, a call is placed to 16002345678.
- 911xxx.T-A period (.) that matches an arbitrary number, including zero, of occurrences of the preceding construct. For example:
- 911123 with waiting time to comply with T is a match
- 9111234 with waiting time to comply with T is a match
- 91112345 with waiting time to comply with T is a match and the number can grow indefinitely given that pressing the next digit takes less than T.
- `sip\:764xxxxxRR@registrar.polycomcsn.comR` - appends `@registrar.polycomcsn.com` to any URI calls matching with "764xxxxx".
  For example, if you make a SIP URI call with 76412345 then `@registrar.polycomcsn.com` is appended to the string such that the SIP URI call INVITE becomes `sip::76412345@vc.polycom.com` . Here, `@domain` string is required only for SIP URI calls from unregistered lines.
- `sip\:xxxx\@registrar\.polycomcsn\.com` - This will match with any four digit URI calls having the domain `@registrar.polycomcsn.com` .
  For example, if you configure three lines and has dial plan based line switching enabled. Now, if the third line's dial plan has `sip\:xxxx\@registrar\.polycomcsn\.com` then call will be initiated from the third line if user dial `1234@registrar.polycomcsn.com` because it matches with the third line's dial plan.

- `0xxxS|33xxH` —All four digit numbers starting with a 0 are placed using the SIP protocol, whereas all four digit numbers starting with 33 are placed using the H.323 protocol.

Note	Only VVX 500/510, 600/611, and 1500 phones support the H. On all other phones, the H is ignored and users need to perform the Send operation to complete

Related Topics

Local Digit Map

Generating Secondary Dial Tone with Digit Maps

You can regenerate a dial tone by adding a comma "," to the digit map.

You can dial seven-digit numbers after dialing "8" as shown next in the example rule `8,[2-9]xxxxxxT` :

```
[2-9]11|0T|011xxx.T|[0-1][2-9]xxxxxxxxx|8,[2-9]xxxxxxT|[2-9]xx.T
```

By adding the digit "8", the dial tone plays again, and users can complete the remaining seven-digit number. In this example, if users also have a 4-digit extension that begins with "8", then users will hear dial tone after the first "8" was dialed because "8" matches the "8" in the digit map.

If you want to generate dial tone without the need to send the "8", replace one string with another using the special character "R" as shown next in the rule *R8RR*. In the following example, replace "8" with an empty string to dial the seven-digit number:

```
[2-9]11|0T|011xxx.T|[0-1][2-9]xxxxxxxxx|R8RR,[2-9]xxxxxxT|[2-9]xx.T
```

Related Topics

Local Digit Map

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.

Related Topics

Call Controls

International Dialing Prefix Parameters

The following parameters configure the international dialing prefixes.

Table 1. International Dialing Prefix Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	`call.internationalDialing.enabled`	This parameter applies to all numeric dial pads on the phone, including for example, the contact directory. Changes you make to this parameter cause a restart or reboot. 1 (default) - Disable the key tap timer that converts a double tap of the asterisk "*" symbol to the "+" symbol to indicate an international call. By default, this parameter is enabled so that a quick double tap of "*" converts immediately to "+". To enter a double asterisk "**", tap "*" once and wait for the key tap timer to expire to enter a second "*". 0 - When you disable this parameter, you cannot dial"+" and you must enter the international exit code of the country you are calling from to make international calls.	No
site.cfg	`call.internationalPrefix.key`	The phone supports international call prefix (+) with both '0' and '*'.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		0 (default) - Set the international prefix with *. 1 - Set the international prefix with 0.	

Related Topics

International Dialing Prefix

Enhanced 911 (E.911)

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

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

Configuring the source of location information allows the phone to share its location details in the invite sent when a 911 call is made to ensure the 911 operator dispatches emergency services to the correct address.

Related Topics

Call Controls

Enhanced 911 (E.911) Parameters

Use the following parameters to configure E.911.

Table 1. E.911 Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`feature.E911.HELD.server`	NULL (default)	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		Set the IP address or hostname of the Location Information Server (LIS) address. For example, host.domain.com or https:// xxx.xxx.xxx.xxx.	
site.cfg	feature.E911.HELD.username	NULL (default) Set the user name used to authenticate to the Location Information Server.	No
site.cfg	feature.E911.HELD.password	NULL (default) Set the password used to authenticate to the Location Information Server.	No
site.cfg	feature.E911.HELD.identity	Set the vendor-specific element to include in a location request message. For example, ‘companyID’. NULL (default) String 255 character max	No
site.cfg	feature.E911.HELD.identityValue	Set the value for the vendor-specific element to include in a location request message. NULL (default) String 255 character max	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	feature.E911.locationRetryTimer	Specify the retry timeout value in seconds for the location request sent to the Location Information Server (LIS). The phone does not retry after receiving location information received through the LIS. 60 seconds (default) 60 - 86400 seconds	No
site.cfg	feature.E911.HELD.nai.enable	You can include or omit the Network Access Identifier (NAI) containing the SIP user information used to subscribe to the Location Information Server (LIS). 0 (default) – The NAI is omitted as a device identity in the location request sent to the LIS. 1 - The NAI is included as a device identity in the location request sent to the LIS.	No
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	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		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	

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	feature.E911.enabled	0 (default) – Disable the E.911 feature. The INVITE sent for emergency calls from the phone does not include the geolocation header, geolocation option in supported header, geolocation-routing header, or the GEOPRIV location object. 1 – Enable the E.911 feature. The INVITE sent for emergency calls from the phone includes the geolocation header defined in RFC 6442 and PIDF presence element as specified in RFC3863 with a GEOPRIV location object specified in RFC4119 for in Open SIP environments. This parameter is mutually exclusive of the GENBAND E.911 feature and if this parameter and feature.genband.E911.enabled are enabled, this parameter takes precedence.	No
site.cfg	feature.E911.HELD.requestType	Any (default) - Send a request to the Location Information Server (LIS) to return either 'Location by Reference' or 'Location by Value'. Note this is not the	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		‘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.	
`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	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		https:// xxx.xxx.xxx.xxx. NULL (default) Dotted-decimal IP address Hostname Fully-qualified domain name (FQDN)	
site.cfg	feature.E911.HELD.secondary.username	Set a user name to authenticate to the secondary Location information Server (LIS). NULL (default) String	No
site.cfg	feature.E911.HELD.secondary.password	Set a password to authenticate to the secondary LIS. NULL (default) String	No
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

Related Topics

Enhanced 911 (E.911)

Shared Phones with Skype for Business

Polycom phones registered with Skype for Business offer several ways to share phones and phone lines among users.

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.

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.

Related Topics

Shared Phones with Skype for Business

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.

Procedure

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.

Related Topics

Skype for Business User Profiles

Create a User Configuration File

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

Procedure

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.

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.

Related Topics

Create a User Profile Configuration File

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. 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

Related Topics

Create a User Profile Configuration File

Hot Desking for VVX Phones

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, users do not have permission to enable or disable Hot Desking.

Hot Desking Sign-In Methods

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

Host Desking Feature Limitations

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

Automatic Sign-Out Scenarios

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.

Related Topics

Directories and Contacts

Related Topics

Shared Phones with Skype for Business

Hot Desking Parameters

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

Table 1. 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

Related Topics

Hot Desking for VVX Phones

Common Area Phone (CAP) for VVX Phones

You can configure your phone with Common Area Phone (CAP) Mode to restrict user's access to configuration settings on phones deployed in common areas, typically lobbies, employee lounges, and conference rooms.

You enable CAP Mode on a per-phone basis and CAP Mode is independent of any other configuration you make on the Skype server or apply to the Skype user account.

Note	Polycom recommends that you do not enable Boss-Admin or Shared Line Appearance while CAP is enabled.

Use of CAP requires UC Software 5.7.0 or later. After you enable this feature using `feature.CAP.enable=1`, CAP Mode and CAP Admin Mode are available on the phone. By default, CAP Mode is enabled and CAP Admin Mode is disabled.

While a phone is running in CAP Mode, users can access only basic settings and features. You can make more features available by enabling parameters for the corresponding feature, listed below.

Table 1. Features Available in CAP Mode

Soft Key / Menu	CAP Mode Default	Parameters to Enable
Status/DND	Disabled	feature.doNotDisturb.enable softkey.feature.mystatus
Call Forward	Disabled	feature.forward.enable
Device Lock	Disabled	feature.deviceLock.enable
Exchange Call Logs	Disabled	Local logs: feature.callList.enabled Exchange call logs: feature.callList.enabled feature.exchangeCallLog.enabled feature.EWSAutodiscover.enabled
Local Contact Directory	Disabled	feature.directory.enabled
Exchange Calendar	Disabled	feature.EWSAutodiscover.enabled feature.exchangeCalendar.enabled homeScreen.calendar.enable
Exchange Contacts	Disabled	feature.EWSAutodiscover.enabled feature.exchangeContacts.enabled
Exchange Voicemail/ Messages	Disabled	feature.voicemail.enabled feature.EWSAutodiscover.enabled feature.exchangeVoiceMail.enabled feature.exchangeSipVMPlay.enabled
Redial	Disabled	homeScreen.redial.enable

You can use the phone's administrator password to enable CAP Admin Mode. CAP Admin Mode provides access to all phone settings available from the phone interface. In addition, in CAP Admin Mode, the phone displays Sign In / Sign Out soft keys that allow you to sign users in or out of the phone. Alternatively, you can sign into a phone in CAP Mode without enabling CAP Admin Mode from the Common Area Phone provisioning portal at `https://aka.ms/skypecap` .

Any CAP-enabled phone that is not signed in with a Skype account and is left idle for three minutes displays a notice that the phone is not in use.

The following settings are available in CAP Admin Mode.

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

Related Topics

Directories and Contacts

Related Topics

Shared Phones with Skype for Business

Disable CAP Admin Mode on VVX Phones

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

Procedure

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

Related Topics

Common Area Phone (CAP) for VVX Phones

CAP Web Sign In on VVX Phones

After you enable the CAP feature and the phone is in CAP Mode, you can generate a code on the phone that you use to log into the Common Area Phone Provisioning Portal, a Microsoft web service that enables you to sign in multiple phones using any tenant account without the need to authenticate as a user on each phone.

You can log into the Common Area Provisioning Portal at https://aka.ms/skypecap using any account having administrator rights to the Microsoft tenant. Note that your Skype deployment must use Modern

Authentication to access CAP web sign-in. For more information, see Skype for Business topologies supported with Modern Authentication on Microsoft Technet.

Note	Sign in using accounts that are designated only for the Common Area locations. The CAP portal is designed only for Common Area Phone accounts. Provisioning a CAP phone from the Provisioning Portal changes that phone's Active Directory user account password to a random string generated by Microsoft. For this reason, do not use the Provisioning Portal to sign in to a phone on behalf of an end user.

Related Topics

Common Area Phone (CAP) for VVX Phones

Sign In to a CAP-Enabled VVX Phone

You can sign out of a CAP-enabled phone using a code sent to the phone by the Common Area Provisioning Portal.

Procedure

1. While signed out of the phone, select Web Sign-in (CAP).

   The phone displays a code.
2. In the provisioning portal, enter the code in the field beside the user name and press Provision.

   The user's password is reset to a random string and the phone is signed in.

Related Topics

Common Area Phone (CAP) for VVX Phones

Common Area Phone Parameters

The following table lists parameters that configure the Common Area Phone (CAP) feature.

Use of CAP requires UC Software 5.7.0 or later.

Table 1. 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

Related Topics

Common Area Phone (CAP) for VVX Phones

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 business media phones.

SLA Limitations

The following features are not supported on SLA lines:

- BToE
- Conference class
- Call Park

SLA Configuration for Skype for Business

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.

Table 1. SLA for Skype for Business Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
	`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. 0 - 25	

Related Topics

Call Controls

Related Topics

Shared Phones with Skype for Business

Network

Polycom's Open SIP 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

Related Topics

Network

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.

Table 1. LLDP Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`device.cfg, site.cfg`	`device.net.lldpFastStartCount`	Configure the fast-start LLDP packets that the phone sends when booting up or when the network comes up. 5 (default)	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		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.	

Related Topics

Extended Link Layer Discovery Protocol (LLDP)

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`

Supported HTTP/HTTPS Services

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

Related Topics

Network

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

Procedure

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**.

Related Topics

Web Proxy Auto Discovery (WPAD)

WPAD Configuration Parameters

The following parameters configure the Web Proxy Auto Discovery (WPAD) feature.

Table 1. 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 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

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

Related Topics

Web Proxy Auto Discovery (WPAD)

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.

The following phones support Data Center Resiliency:

- VVX 201, 300/310, 301/311, 400/410, 401/411, 500/501, and 600/601 business media phones
- VVX 250, 350, and 450 business IP 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.

Related Topics

Network

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)

Table 1. STUN / TURN / ICE Parameters

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

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		443 (default) 65535	
`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`	Default (default) Default<*VVXxxx*>, where <*xxx*> is the VVX phone model number. For example, If you are using VVX 201 phone model, the value is set to DefaultVVX201 by default. 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
`firewall-nat.cfg`	`tcpIpApp.ice.NetworkMode`	The information about TCP and UDP ICE candidates. 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

<table>
<tr><th>Template</th><th>Parameter</th><th>Permitted Values</th><th>Change Causes Restart or Reboot</th></tr>
<tr><td>firewall-nat.cfg</td><td>tcpIpApp.ice.MaxCandidateGatheringInParallel</td><td>The number of ICE candidates gathering threads run in parallel in the maximum time span of 2 seconds for simultaneous incoming calls only.
3 (default for VVX 201 business media phone)
5 (default for all other VVX platforms)
2 – 24</td><td>No</td></tr>
<tr><td>firewall-nat.cfg</td><td>tcpIpApp.ice.MaxConnectivityChecksInParallel</td><td>The number of ICE connectivity checks threads that 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. The following table lists the default value set when using a particular VVX phone model.

Table 2. Default Values for VVX phones
<table>
<tr><th>VVX Phone Model</th><th>Default Value</th></tr>
<tr><td>VVX 201, 300, 310, 400</td><td>1</td></tr>
<tr><td>VVX 301, 311, 401, 411, 500</td><td>5</td></tr>
<tr><td>VVX 410</td><td>2</td></tr>
<tr><td>VVX 501, 601</td><td>7</td></tr>
<tr><td>VVX 600</td><td>3</td></tr>
</table>
1 - 24</td><td>No</td></tr>
<tr><td>firewall-nat.cfg</td><td>tcpIpApp.ice.ConnCheckInetvalPairs</td><td>Time interval in milliseconds to serialize first attempt of connectivity check of identified ICE candidate pairs per call.
25 (default)
25 - 100</td><td>No</td></tr>
<tr><td>firewall-nat.cfg</td><td>tcpIpApp.ice.ConnCheckInetvalRetries</td><td>Time interval in milliseconds to serialize the retry attempts of</td><td>No</td></tr>
</table>

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		connectivity check for identified pairs per call. 100 (default) 25 - 100	
`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 failure. 5 (default) 2 - 25	No

Related Topics

Network

Skype for Business Device and Software Support

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

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.

To help deploy QoE, you can enable client media ports and configure unique port ranges on the Skype for Business Server. For details, see Configuring Port Ranges for your Microsoft Lync Clients in Lync Server 2013. Note that VVX phones use only the Audio port and range.

Related Topics

Skype for Business Device and Software Support

Set QoE Parameters on the Skype for Business Server

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

Procedure

- The following table lists and describes each QoE parameter you need to set on the Skype for Business server.

Table 1. Skype for Business QoE Parameters

Skype Parameter	Description
`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` '.
`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.

Figure 1. 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
```

Related Topics

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

Enable In-Call QoE within your Skype Environment

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 -InCallQoSIntervalSeco
```

Related Topics

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

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.

Procedure

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**.

Related Topics

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

QoE Parameters

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

Table 1. QoE Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`voice.qoe.event.lossrate.threshold.bad`	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.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		0 to 100	
`features.cfg`	`voice.qoe.event.lossrate.threshold.poor`	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	No
`features.cfg`	`voice.qoe.event.networkmos.threshold.bad`	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	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		networkMOS < 1.9 signifies bad quality	
`features.cfg`	`voice.qoe.event.networkmos.threshold.poor`	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	No

Related Topics

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

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.

Table 1. 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

Parent Element	Child Elements/Attributes
	LocalContactURI
	RemoteContactURI
	LocalUserAgent
	RemoteUserAgent
	LocalPAI
	RemotePAI
	ConfURI
	v2:CallPriority
	v2:MediationServerBypassFlag
	v2:TrunkingPeer
	v2:RegisteredInside
	CallID
	FromTag
	ToTag
	Start

Parent Element	Child Elements/Attributes
	End
MediaLine	
	Description
	InboundStream
	OutboundStream
Description	
	Connectivity
	Security
	Transport
	LocalAddr
	RemoteAddr
	v3:ReflexiveLocalIPAddress
	v3:MidCallReport
LocalAddr, RemoteAddr, RelayAddr	
	IPAddr

Parent Element	Child Elements/Attributes
	Port
	SubnetMask
	v2:MACAddr
Connectivity	
	Ice
	IceWarningFlags (Five flags supported)
	RelayAddress
InboundStream, OutboundStream	
	Network
	Payload
	QualityEstimates
Network	
	Jitter
	PacketLoss
	BurstGapLoss

Parent Element	Child Elements/Attributes
	Delay
	Utilization
Jitter	
	InterArrival
	InterArrivalMax
Packetloss	
	LossRate
	LossRateMax
BurstGapLoss	
	BurstDensity
	BurstDuration
	GapDensity
	GapDuration
Delay	
	RoundTrip

Parent Element	Child Elements/Attributes
	RoundTripMax
Utilization	Packets
Payload	Audio
Payload.Audio	
	PayloadType
	PayloadDescription
	SampleRate
	v4:JitterBufferSizeAvg
	v4:JitterBufferSizeMax
	v4:JitterBufferSizeMin
	v4:NetworkJitterAvg
	v4:NetworkJitterMax
	v4:NetworkJitterMin
Signal	
	SignalLevel

Parent Element	Child Elements/Attributes
	NoiseLevel
	InitialSignalLevelRMS
	RecvSignalLevelCh1
	RecvNoiseLevelCh1
	RenderSignalLevel
	RenderNoiseLevel
	RenderLoopbackSignalLevel
	VsEntryCauses
	EchoEventCauses
	EchoPercentMicIn
	EchoPercentSend
	SendSignalLevelCh1
	SendNoiseLevelCh1
QualityEstimates.Audio	
	RecvListenMOS

Parent Element	Child Elements/Attributes
	RecvListenMOSMin
	NetworkMOS
NetworkMOS	
	OverallAvg
	OverallMin

Related Topics

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

Quality of Service for Audio Calls on VVX Phones

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.

Related Topics

Skype for Business Device and Software Support

Polycom UC Software Update

You can update UC Software on a per-phone basis from the phone menu or Web Configuration Utility when using UC Software 4.1.x or UC Software 5.x.x.

Related Topics

Skype for Business Device and Software Support

Update UC Software Manually

You can use an USB flash drive to update the software and configure the phone.

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

Procedure

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.

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.

Related Topics

Polycom UC Software Update

Automatic UC Software Updates

When you register 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.

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*.

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.

Related Topics

Polycom UC Software Update

Configuring Automatic Software Update

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

Table 1. Automatic Software Update Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg, site.cfg	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	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		phone and the phone receives software updates from the server.	
device.cfg, site.cfg	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 .	Yes
reg-advanced.cfg	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.	Yes
reg-advanced.cfg	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	Yes
reg-advanced.cfg	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	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	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.	No
site.cfg	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.	No
site.cfg	prov.polling.period	The polling period in seconds. 86400 (default)	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
		integer > 3600 The polling period is rounded up to the nearest number of days in absolute and random mode you set in . 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.	
`site.cfg`	`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	No
`site.cfg`	`prov.polling.timeRandomEnd`	The polling stop time when the polling mode is set to random. NULL (default) hh:mm	No

Related Topics

Polycom UC Software Update

Troubleshooting

Use the following section as a guide to resolving issues, problems, or common difficulties you may encounter while using Microsoft-enabled Polycom UC Software.

The phone fails to register

The most common issue with a failure to register is basic connectivity to the phone.

You can check basic connectivity in a number of ways:

- Obtain the host IP by looking at the phone registration status, configuration file, DNS, and Lync Computer Client Configuration Information Screen.
- Make sure the phone can communicate with the server by performing a diagnostic ping.
- From a computer connected on the same network as the phone, perform a telnet to the Lync server SIP TCP port 5061 or 443.
- Check for a DNS issue.
- Check if Lync Services is temporarily out of service, for example, a firewall or routing problem with the network.

Check that the phone is reading the configuration files. On the phone, go to Status > Platform > Configuration. The phone displays the current configuration and files. If the phone is not reading the correct configuration files, redo the provisioning procedures. If the phone is reading the configuration files, go to the next troubleshooting tip.

If the phone still cannot register, check autodiscover:

- Ensure the SRV Record exist and points to a valid A record.
- Ensure that the A record points to a valid host IP.
- Use the shell command `dnsCacheShow` to display a cached DNS entry. If an entry has a negative cache, the phone is trying to perform a lookup and is failing to resolve.

If you get a TLS error, you may have an untrusted, corrupted, or expired certificate. Check if a root CA is installed on the phone by going to Settings > Advanced > Administration Settings > TLS Security > Custom CA Certificate. If you need to troubleshoot TLS, use `log.level.change.tls=0` and `log.level.change.sip=0` to log for TLS problems.

Check for invalid user credentials. Use `log.level.change.tls=0` , `log.level.change.sip=0` , and `log.level.change.dns=0` to troubleshoot authentication failures.

Log into a computer Lync client with a user's credentials and ensure that the user account logs in. Use a simple password for testing purposes.

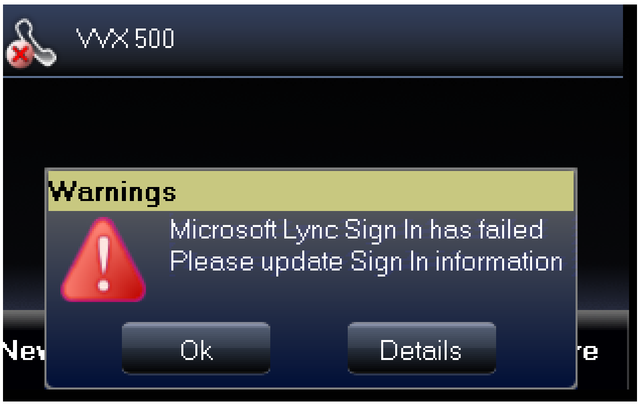
Related Topics

Troubleshooting

I cannot sign in; I'm getting a sign in failure message

PIN authentication can fail for several reasons, most commonly an invalid extension or invalid PIN.

When PIN authentication fails, a warning message displays:



Press Ok to open the PIN Authentication screen to sign in again. Any one of the following messages might display:

- Lync Sign In has failed System Administrator. This message indicates that something is wrong with the network. When you receive this message, speak to your administrator.
- Lync Sign In has failed Invalid login credentials. This message indicates that the user credentials you entered are incorrect. Try entering your credentials again and if sign in still fails, speak to your administrator.
- Lync Sign In has failed Please update Sign In Information. This message is rarely expected, and indicates a problem with the generation of certificate signing request (CSR) publishing the certificate.

Related Topics

Troubleshooting