



DEPLOYMENT GUIDE

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Polycom® UC Software with Skype for Business



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Conventions Used in Polycom Guides

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

Information Elements

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

Information Elements

Name	Icon	Description
Note		The Note icon highlights information of interest or important information needed to be successful in accomplishing a procedure or to understand a concept.
Important		Important highlights information of interest or important information needed to be successful in accomplishing a procedure or to understand a concept.
Caution		The Caution icon highlights information you need to know to avoid a hazard that could potentially impact device performance, application functionality, or successful feature configuration.
Warning		The Warning icon highlights an action you must perform (or avoid) to prevent issues that may cause you to lose information or your configuration setup, and/or affect phone or network performance.
Web Info		The Web Info icon highlights supplementary information available online such as documents or downloads on support.polycom.com or other locations.

Typographic Conventions

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

Typographic Conventions

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

Getting Started

Polycom® phones offer a best-in-class communications experience with an extensive list of features. This guide shows you how to deploy Polycom phones and Unified Communications (UC) software with Skype for Business and legacy Microsoft® Lync Server. Registering Polycom phones with Skype for Business enables you to communicate with enterprise-grade high-definition (HD) voice and video using familiar Microsoft solutions.



Note: Currently, Polycom phones deployed with Skype for Business and Lync Server support one registered line.

UC Software Device Compatibility

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

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

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



Web Info: To register RealPresence Trio solution with Skype for Business, see *RealPresence Trio with Skype for Business - Deployment Guide* at [RealPresence Trio](#) on Polycom Support.

Microsoft Compatibility

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)

Prerequisites - On-Premises Deployments

Before you set up Polycom phones 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`

Microsoft-Qualified Phones

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

Feature Licenses

Polycom devices purchased and shipped with Skype for Business base profile (a Skype SKU) include a feature license to register phones with Skype for Business, Lync Server, and Office 365. If you do not purchase devices with a configured Skype for Business SKU, you must purchase a feature license from a Polycom reseller or Polycom sales representative. For information about the license, log in to [Licensing & Product Registration](#).

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.

UC Software File Formats

Polycom releases UC Software 5.x.x in two file formats:

- **Cabinet (CAB) file.** As of September 2013, 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. UC Software in CAB file format is available from [Polycom UC Software for Microsoft Deployments](#) and enables you to receive automatic software updates from Skype for Business Server.
- **sip.Id.** Polycom offers all UC Software as a combined file for all phone models or as a split file for specific phone models.

Supported Features

Features available on Polycom phones vary by software release and phone model.

The following table lists features available on all Polycom phones registered with Skype for Business. For a list of new features introduced in a UC Software version, see the release notes for your UC Software version at [Polycom UC Software Support Center](#).

Features Supported on All Polycom Phones Registered with Skype for Business

Feature	Function
Device Lock	Protect phones and access personal settings with a lock code.
Distribution List	Manage, search for, and call a distribution list or individual in a list.
Quality of Experience	Monitor audio quality and troubleshoot audio problems.
User Log Upload	Users can upload diagnostic logs from the phone or Web Configuration Utility to the Skype for Business server.
BToE static pairing	Pair the phone and computer to control audio and video calls from your VVX phone and computer.
Web Sign In	Securely log in to Skype for Business from the phone using a computer or mobile device web browser. Available for online deployments only.
Auto root certificate fetch	Available using DHCP option 43
PIN Authentication	Support for Skype for Business authentication available on all Polycom phones. This sign-in method is not currently available for Skype for Business online deployments.
Narrowband audio	G.711
Call transfer, hold, mute	Flexible user phone functions
Full-duplex echo cancellation (FDX)	
Wideband audio	G.722-1
Media encryption	SRTP, SSRTTP
Direct SIP registration to Lync Server	Microsoft SIP, TLS for SIP Signaling, SRTP, SSRTTP
Peer-to-peer audio calling	Initiate and receive two-party calls
Enterprise voice	
Message Waiting Indicator (MWI)	Illumination of MWI lamp indicates new messages
Voice mail retrieval	One-touch call to voice mail attendant
Presence publication	Indicates the status of your s
Presence state control	Choose from a menu of presence states

Features Supported on All Polycom Phones Registered with Skype for Business

Calls logs	Local call history for missed, received, and outgoing calls; nonvolatile for all platforms except VxWorks phones
Log access	Local phone access to diagnostic logging
Device updates	Centralized phone updates from an out-of-band server
VLAN assignment	LLDP-MED VLAN assignment
Device sign-in	Out-of-the-box user sign-in and sign-out
Remote worker scenarios	Edge Server registration for off-location users
Firewall traversal	A/V Edge Server support using the ICE, STUN, and TURN protocols
Federation	Connect people across organizations and domains
Provisioning	Support for in-band provisioning from Skype for Business
Monitoring	Device Inventory Reports
Reporting	
Media bypass	Bypass the Skype mediation server to send media directly to a PSTN gateway
Dial plans	Support for Skype for Business Regex normalization patterns passed via an in-band provisioning to the endpoint; limited to regular expression support; option for server-side normalization
Call forwarding to s	Forward calls to another
Call forwarding to voicemail	Forward calls directly to voicemail
Response Groups	
Team-Call	
Delegates	
Private Lines	Alternate call-forwarding identity for a Skype for Business user's secondary DID
Data Center Resiliency	Active calls remain active and new call option not be available until phone registers with secondary server.
Branch Office Resiliency	Minimum call functions during wide area network (WAN) outage.
E911	Supports in-band provisioning information for Emergency 911
Location Services	Extended Link Layer Discovery Protocol (LLDP)-MED location-based information support

Features Supported on All Polycom Phones Registered with Skype for Business

Local Directory	Display Skype for Business contacts and their current presence status
Groups	Display and expand groups in the Skype for Business user's list
Web Ticket Authentication	Used to gain access to a web service; support for web tickets obtained using NTLM, PIN, or a client certificate used as authentication credentials Skype for Business Authentication: NTLM SIP Registration: TLS-DSK User Sign In: NTLM Credentials, PIN authentication NTLMv2 Authentication
Client Certificate Provisioning	Automatic provisioning using a web ticket
TCP Media	RTP Media and ICE negotiation supported over TCP when UDP is unavailable

Skype for Business On-Premises and Online Features

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

Polycom with Skype for Business Online Feature Support

Skype for Business Feature	Polycom with Skype for Business On-Premises	Polycom with Skype for Business Online
Resiliency - Branch Office	✓	na
Resiliency - Data Center Outage	✓	na
Call Park	✓	x
PIN Authentication	✓	x
Attendant Console	✓	x
Cross Pool	✓	x
Media Bypass	✓	x
Response Groups	✓	x
Private Line	✓	x
Web Sign In	x	✓
Device Lock	✓	✓
Distribution Lists	✓	✓
Quality of Experience (QoE)	✓	✓
User Log Upload	✓	✓
BToE Manual Pairing	✓	✓
Device Update	✓	✓

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
Inband Provisioning	✓	✓
Call Handling	✓	✓
Call Forward	✓	✓
Call Transfer	✓	✓
Conference Calls	✓	✓
Local Call Logs	✓	✓
Exchange Call Logs	✓	✓
Federated Calls	✓	✓
Simultaneous Ring	✓	✓
Dual Tone Multi Frequency	✓	✓
Emergency 911	✓	✓
Call Admission Control	✓	✓
Monitoring (Device Inventory)	✓	✓
Delegates	✓	✓
Team Call	✓	✓
Message Waiting Indicator	✓	✓
Exchange Integration	✓	✓
Exchange Calendar		
Extended Presence	✓	✓
Visual Voicemail	✓	✓
Boss-Admin	✓	✓

Frequently Asked Questions

Refer to the frequently asked questions (FAQs) to help answer questions you may have about deploying Polycom phones with Skype for Business before you begin.

What is the Base Profile?

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 has two options: Generic and Skype. When set to Skype, 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 with the Base Profile, Polycom recommends using the Base Profile for deployments of fewer than 20 devices requiring only default Skype for Business settings.

What are CAB files?

You can choose to download UC Software in CAB file format. CAB file format is a Microsoft Windows archive file that supports lossless data compression and embedded digital certificates that maintain archive integrity.

Polycom offers UC Software in CAB file format so that you can deploy UC Software from Skype for Business and enable the automatic software update feature.

Get Help

For more information about installing, configuring, and administering Polycom products, refer to Documents and Downloads at [Polycom Support](#).

The Polycom Community

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

Deploying Polycom Phones with Skype for Business

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

- Set Up the Network
- Set Up Polycom UC Software
- Provision 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, complete Task 1: Set Up the Network only. If you want to customize default settings, complete all tasks.

Set Up the Network

To set up a network to connect your Polycom devices complete the following procedure.

To set up your network:

- 1 Set up or verify Domain Name System (DNS) service (SRV) records to allow the devices to discover Skype for Business server automatically. For information on creating and verifying DNS SRV records, see [Required DNS Records for Automatic Client Sign-In](#) on Microsoft TechNet.
- 2 Obtain a root certificate authority (CA) security certificate using one of the following ways:
 - 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 Lightweight Directory Access Protocol (LDAP) Domain Name System (DNS) query. Phones you register with Skype for Business are enabled with this feature by default and no additional configuration is required.
 - When provisioning phones from within an enterprise, you can use Dynamic Host Configuration Protocol (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. For a list of DHCP options and sub-options supported by RealPresence Trio solution, refer to [Set Up Polycom UC Software](#). 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. The PIN Auth menu does not display and is not available for Skype for Business Online.

- To troubleshoot for missing or misconfigured DHCP Option 43 or when using older DHCP systems with a limited field length, you can use an STS URI (Skype for Business certificate server URL) to override DHCP Option 43.
You can also set up PIN Authentication in a test environment without the need to fully deploy Skype for Business DHCP and to verify that your DHCP server is set up correctly. For more information on configuring DHCP Option 43, refer to [Set Up DHCP for Devices](#) on Microsoft TechNet.
 - If you need to install a security certificate manually on your Microsoft Edge Server, the signing CA that issued this certificate must be listed on the Polycom Trusted Certificate Authority List in the Polycom UC Software 4.1.0 Administrator's Guide. You must use Base64 format. For instructions on manually installing a certificate, refer to [Send Diagnostic Logs from the Phone](#).
- 3** (Optional) If you are using a provisioning, or boot server, configure DHCP Option 66 if available. If not available, set DHCP options using one of the following methods:
- If you are using a Polycom phone with a Skype for Business SKU, use Option 161 with the address (URL or IP address) of the provisioning server. You can set the provisioning server address or URL through the device menu.
 - If you are using a Polycom phone with an Open SIP SKU, use Option 160 with the address (URL or IP address) of the provisioning server. You can set the provisioning server address or URL through the device menu or refer to the section [Set the Base Profile Using the Web Configuration Utility](#).
- 4** Ensure that you set up each user with a Skype for Business account and credentials that can be used on the phone to sign in. Also set up PIN Authentication type if you are using any of the following devices in your deployment: VVX 201, 300/310, 301/311, 400/410, 401/411, 500/501, 600/601 business media phones, and SoundStructure VoIP Interface.

Set Up Polycom UC Software

The latest UC Software for Microsoft deployments is available at [Polycom UC Software for Microsoft 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, or you can use the template configuration files in the PartnerConfig > Microsoft directory of the UC Software download.



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

To set up Polycom UC Software:

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

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

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

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

- To use Skype for Business Server to push software to the phones, complete the steps in the section [Deploy UC Software from Skype for Business Server](#).
 - To use your own provisioning server to push software to the phones, complete the steps in the section [Deploy UC Software from a Provisioning Server](#).
- 3 Download, save, and extract UC Software to the root directory you created. Polycom provides Skype for Business-specific template configuration files in the PartnerConfig > Microsoft directory of the UC Software download.
 - If you are deploying UC Software from Skype for Business Server, download the CAB file version of Polycom UC Software.
 - If you are deploying phones from your own provisioning server, download the split or combined version of Polycom UC Software in XML format.
 - 4 After the UC Software directory is extracted, open the folder in your root directory.
 - 5 Configure a Call Park Orbit Policy. You must configure a call park orbit policy to enable the call park feature. See [Configuring Call Park](#) on the Microsoft web site.
 - 6 (Optional) To use the BToE feature, download the Polycom BToE connector application and enable BToE. For complete instructions on setting up BToE, see the latest *Polycom VVX Business Media Phones for Skype for Business - User Guide* on [Polycom UC Software for Microsoft Deployments](#).

Provisioning the Phones

Polycom provides manual per-phone provisioning methods and centralized provisioning methods. The method labeled device.set is an advanced method for users familiar with Polycom configuration files and uses centralized provisioning to set the Base Profile for multiple phones. For complete information on provisioning with Polycom UC Software, see the *Polycom UC Software Administrator Guide* on [Polycom UC Software for Microsoft Deployments](#).



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

Manual Provisioning Methods

Polycom provides five per-phone manual methods you can use to register Polycom devices with Skype for Business. All manual provisioning methods set the Base Profile of a phone to Skype. The Base Profile is a feature on each Polycom phone that, when set to Skype, automatically provisions the phone with the default parameters required to work with Skype for Business. For details on all of the default parameters and values, refer to [UC Software Parameters for Microsoft Deployments](#).

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

- [Set the Base Profile to Skype Using MKC during startup](#). Set the Base Profile to Skype using an MKC method during phone startup. This is the fastest manual provisioning method.
- [Set the Base Profile to Skype from the Setup Menu During Startup](#). Set the Base Profile to Skype during startup using the phone boot Setup menu.
- [Set the Base Profile Using MKC](#). Set the Base Profile to Skype using MKC during normal phone functioning.
- [Set the Base Profile from the Settings Menu](#). Set the Base Profile to Skype from the phone's Settings menu during normal phone functioning.
- [Set the Base Profile Using the Web Configuration Utility](#). Use the Polycom Web Configuration Utility to set the Base Profile from a web browser. This is particularly useful when working remotely. Not available when using Polycom UC Software 5.1.1.



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

Manually Reboot the Phone

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

To manually reboot the phone:

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

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

Set the Base Profile to Skype Using MKC during startup

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

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

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

- 1 Power on the phone or restart it after you have reset the phone to factory default settings.
- 2 A few seconds into the device's startup cycle, the phone displays the message 'Starting Application', press Cancel to interrupt and a Cancel soft key. Press the **Cancel** soft key.
- 3 When the phone displays three soft keys—Start, Setup, and About—press and hold the following key combinations on the phone keypad for about 3 seconds to enter the MKC for the phone model:
 - VVX 300, 310, 400, 410, 500, 600, 1500, press **1, 4, 9**

- 4 Press and hold the MKC keys to cause the Base Profile Password menu to display. Enter the password (default 456) to change the Base Profile and press **Ok**.
The **Base Profile** menu displays.
- 5 Press the **Edit** soft key, use the keypad keys to set the Base Profile to **Skype**, and press **Ok > Exit**.
- 6 Highlight **Save & Reboot** and press the **Select** soft key.
The phone reboots and displays the Sign In screen. Users can now sign in.

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

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

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

- 1 Power on the phone or restart after you have reset the phone to factory default settings.
- 2 A few seconds into the device power-up cycle, the phone displays the message 'Starting Application, press Cancel to interrupt' and a Cancel soft key. Press the **Cancel** soft key.
- 3 When the phone displays three soft keys—Start, Setup, and About—press the **Setup** soft key, enter the password (default 456), and press **Ok**.
The phone displays a diagram of keypad keys you can use to navigate the Setup menu. You will need to use these keys in the next few steps.
- 4 Press the **Setup** soft key and the Setup menu displays.
- 5 Using the keypad keys, scroll down, highlight **Base Profile**, and select the **Edit** soft key.
- 6 Using the keypad keys, set the Base Profile to **Skype**, and press **Ok > Exit**.
- 7 Highlight **Save & Reboot** and press the **Select** soft key.
- 8 The phone reboots and displays the Sign In screen. Users can now sign in.

Set the Base Profile Using MKC

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

To set the Base Profile to Skype using MKC:

- 1 Press the phone's **Home/Menu** key.
- 2 From the idle screen, press and hold the following key combinations on the phone keypad for about 3 seconds. MKC keys vary by phone.
 - VVX 300, 310, 400, 410, 500, and 600, press **1, 4, 9**
- 3 Press and hold the MKC keys to cause the Base Profile screen to display. Enter the password (default 456) and press **Enter**.
- 4 In the **Base Profile** menu, select **Skype**.

The phone automatically restarts and displays the Sign In screen. Users can now [Sign In Methods](#).



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

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

Set the Base Profile from the Settings Menu

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

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

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

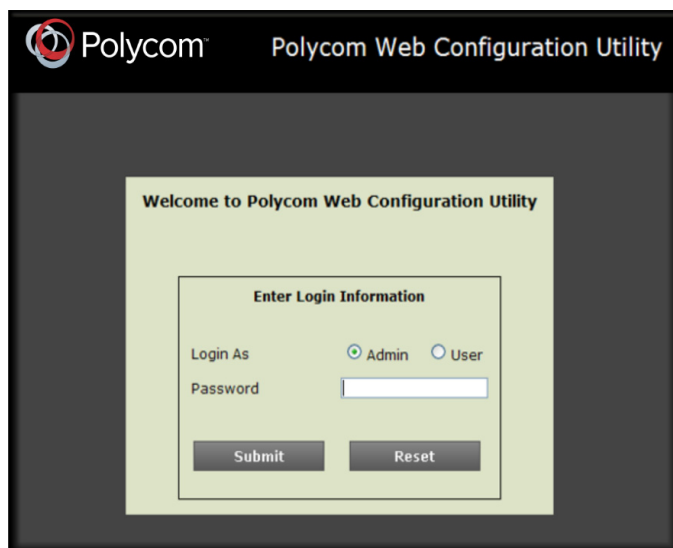
Set the Base Profile Using the Web Configuration Utility

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

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

- 1 Provide power to your phones and allow the phones to complete the power-up process.
- 2 Obtain the IP address of each phone in your deployment by going to **Settings > Status > Platform > Phone**. The IP address displays in the IP: field.

- 3 Enter the phone's IP address in the address bar of a web browser. The Web Configuration Utility login screen displays, as shown next.



- 4 Choose **Admin** to log in as an administrator, and then enter the administrator password (default 456) and click **Submit**.
- 5 In the Home page, navigate to the **Simple Setup** menu.
- 6 From the Base Profile drop-down, choose **Skype**, and click **Save** at the bottom of the page.
- 7 In the confirmation dialog, choose **Yes**. The phone automatically restarts.
Users can now sign in.

Centralized Provisioning

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

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

After you set up a provisioning server, you can provide default settings to all your devices using Microsoft-specific template configuration files in the PartnerConfig > Microsoft directory of the UC Software download. If you require further help setting up a provisioning server or using Polycom configuration files effectively, see the *Polycom UC Software Administrator Guide* on [Polycom UC Software for Microsoft Deployments](#).



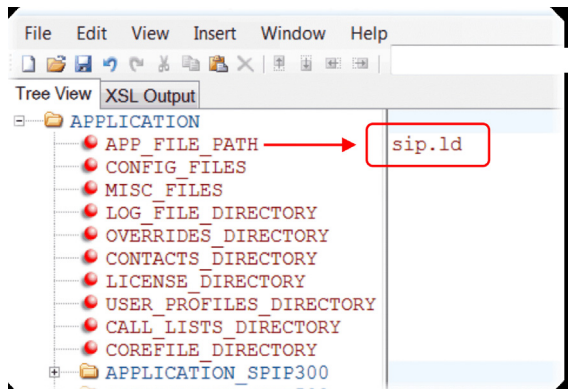
Caution: Using an existing server to deploy your provisioning server can affect performance of your Skype for Business deployment. Misconfiguration or nonstandard deployment of the Microsoft Internet Information Services (IIS) web server may affect your ability to obtain accurate Microsoft support.

Centralized Provisioning Methods

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

- [Set Up Polycom with Skype for Business Online and Microsoft® Exchange Online](#). As of UC Software 5.4.0A, you can switch between an on-premises and online deployment without interrupting service. In online deployments, several Microsoft features are available that you do not need to configure. Refer to the table Polycom with Skype for Business Online Feature Support. If you need to configure media ports for Skype for Business Online deployments, see [Skype for Business Online](#) for specific port numbers.
- [Deploy UC Software from Skype for Business Server](#). Download UC Software in CAB file format and place the software on Skype for Business Server. Default feature settings are applied to all your phones.
- [Deploy UC Software from a Provisioning Server](#). This method requires you to set up your own provisioning server. Setting up your own provisioning server enables you to customize feature settings using the template configuration files included in the UC Software download. With this method, users can sign in with their credentials from the phone's interface.

If you are deploying UC Software from Skype for Business Server and customizing features using Polycom configuration files, delete the default `sip.ld` value from the `APP_FILE_PATH` field in your master configuration file, as shown in the figure. Deleting the `sip.ld` value ensures that you do not deploy UC Software from a Microsoft Server and your own provisioning server, which sends your phones into a reboot cycle.



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

If you need to configure media ports for Skype for Business Online deployments, see [Skype for Business Online](#) for specific port numbers.

For a list of supported in-band parameters for Polycom with Skype for Business Online, refer to [In-Band Provisioning](#). When using Skype for Business Online and Microsoft Exchange Online, note the following:

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

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

To set up Exchange online:

- 1 Install and open the [Skype for Business Online, Windows Powershell Module](#).
- 2 Type the command `Import-Module SkypeOnlineConnector`.
- 3 Connect to the Skype for Business tenancy using the command
`$session=New-CsOnlineSession -Credential $cred`
- 4 When the Powershell credential request dialog displays, enter your Skype for Business user name and password.
- 5 Import the session with the command
`Import-PSSession $session -Verbose -AllowClobber`
- 6 Set policies with the command `CsIPPhonePolicies`.

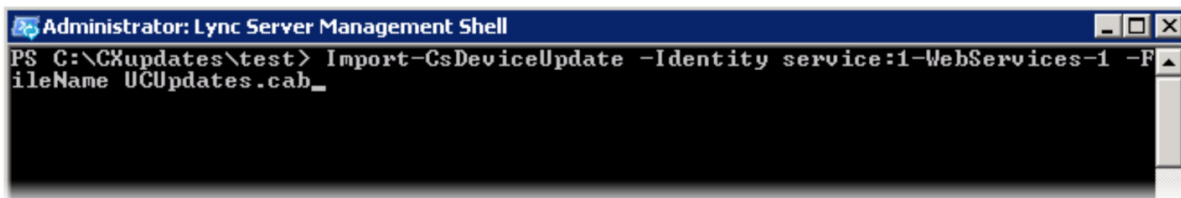
For a list of available Skype for Business Online parameters to configure, refer to [Skype for Business On-Premises and Online Features](#).

Deploy UC Software from Skype for Business Server

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

To deploy UC Software from Skype for Business Server:

- 1 Download and save UC Software in CAB file format to your computer. You can obtain all Microsoft-compatible UC Software from [UC Software for Microsoft Deployments](#).
- 2 Go to Skype for Business Server and copy the CAB file to a C: drive directory.
- 3 Use the Skype for Business Server Management Shell to go to a particular directory.
- 4 In the Skype for Business Server Management Shell, run the following import command:
`Import-CsDeviceUpdate -Identity service:1-WebServices-1 -FileName UCUpdates.cab`



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

Skype for Business Server 2015 Control Panel

Administrator | Sign out
6.0.9319.0 | Privacy statement

Home
Users
Topology
IM and Presence
Persistent Chat
Voice Routing
Voice Features
Response Groups
Conferencing
Clients
Federation and External Access
Monitoring and Archiving
Security
Network Configuration

CLIENT VERSION POLICY CLIENT VERSION CONFIGURATION DEVICE UPDATE TEST DEVICE DEVICE LOG CONFIGURATION DEVICE CONFIGURATION MOBILITY POLICY PUSH NOTIFICATION CONFIGURATION

Search

Edit Action Refresh

Device type	Model	Locale	Pool	Approved version	Pending version	Restore version	Brand
3PIP	VVX300	ENU	WebServer:pool01.polycom-...		5.4.1.9139		POLYCOM
3PIP	VVX310	ENU	WebServer:pool01.polycom-...		5.4.1.9139		POLYCOM
3PIP	VVX400	ENU	WebServer:pool01.polycom-...		5.4.1.9139		POLYCOM
3PIP	VVX410	ENU	WebServer:pool01.polycom-...		5.4.1.9139		POLYCOM
3PIP	VVX500	ENU	WebServer:pool01.polycom-...		5.4.1.9139		POLYCOM
3PIP	VVX600	ENU	WebServer:pool01.polycom-...		5.4.1.9139		POLYCOM
3PIP	VVX1500	ENU	WebServer:pool01.polycom-...		5.4.1.9139		POLYCOM
3PIP	SSTRVOIP	ENU	WebServer:pool01.polycom-...		5.4.1.9139		POLYCOM
3PIP	VVX101	ENU	WebServer:pool01.polycom-...		5.4.1.9139		POLYCOM
3PIP	VVX201	ENU	WebServer:pool01.polycom-...		5.4.1.9139		POLYCOM
3PIP	VVX301	ENU	WebServer:pool01.polycom-...		5.4.1.9139		POLYCOM
3PIP	VVX311	ENU	WebServer:pool01.polycom-...		5.4.1.9139		POLYCOM

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

You have successfully configured UC Software on Skype for Business Server.

Deploy UC Software from a Provisioning Server

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

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

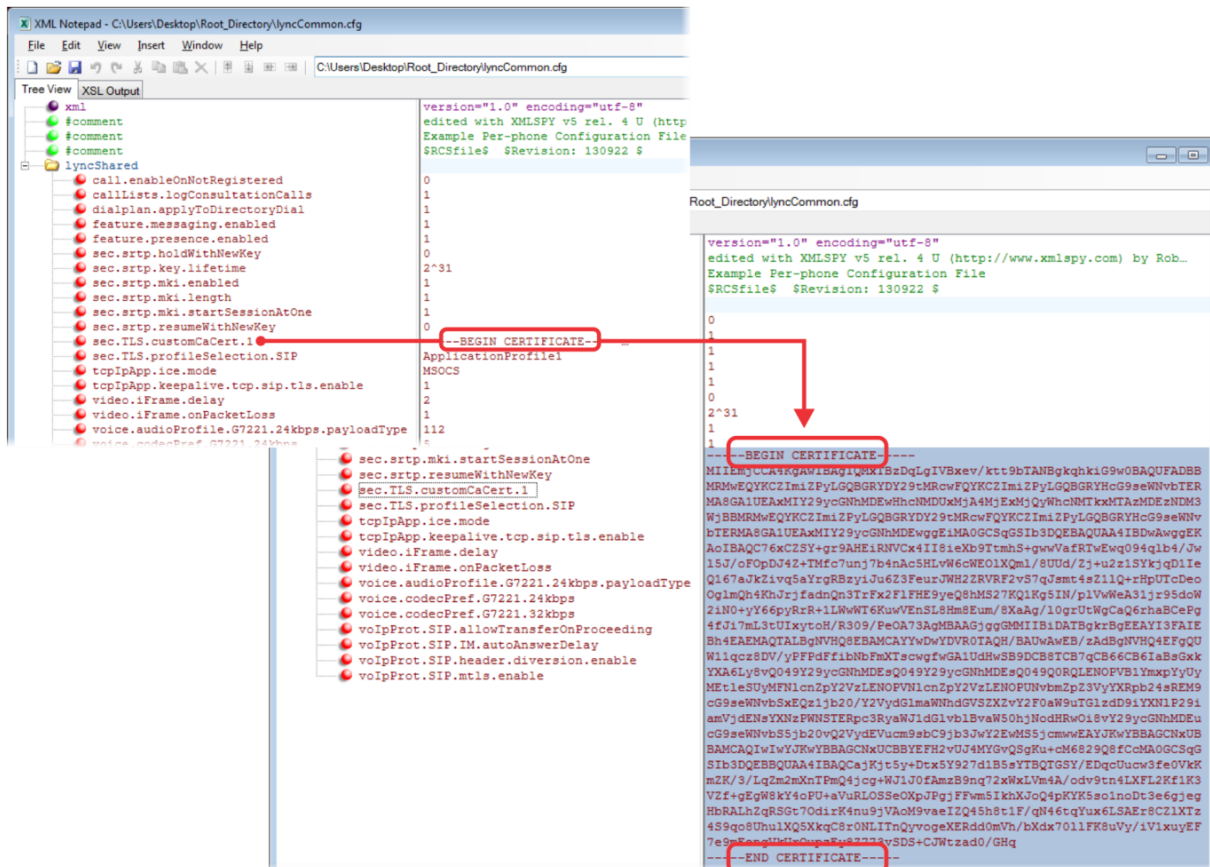
To deploy UC Software from a provisioning server:

- 1 Locate the following three Skype for Business configuration files in your UC Software download in the folder **PartnerConfig > Microsoft**:
 - `lyncSharedExample.cfg`. This file contains all of the parameters for settings that are shared by all the phones in your deployment.
 - `lyncSharedLCExample.cfg`. This is a per-phone file. Use this file to display the Sign In screen and enable users to enter sign-in credentials on the phone. Because users enter their credentials on the device, this is a secure way to provision with Skype for Business Server.
 - `000000000000.cfg`. This is the master configuration file. In the CONFIG_FILES field, enter the names of all the configuration files containing settings you want to apply to the phones.

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

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

- If you are manually installing a root CA security certificate, go to step 4. If not, go to step 5.
- Open your renamed file `lyncSharedExample.cfg` - this example uses the name `lyncCommon.cfg`. If you are manually configuring a root CA certificate, configure the following two parameters:
 - Enter the root CA certificate, in Base64 format, in `sec.TLS.customCaCert.1`.
 - Set the application profile in `sec.TLS.profileSelection.SIP`.



- Open the master configuration file `000000000000.cfg`. In the `CONFIG_FILES` field, enter the name(s) of your two Skype for Business configuration files and save.

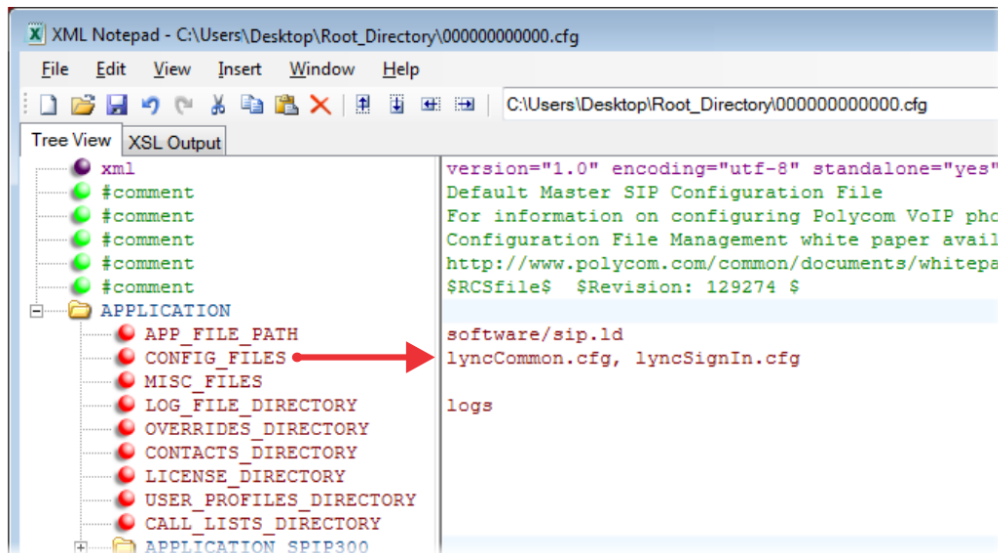
Configuration files you enter in the `CONFIG_FILES` field are read left to right. If you have configured the same setting in two configuration files, the setting listed first (left) is applied. Ensure that you do not have the same parameter in more than one configuration file.

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

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

➤ Port: `reg.1.server.1.port=<port_number>`

The following example shows `lyncCommon.cfg` and `lyncSignIn.cfg`. You must list the names of every file you want to apply to your phones in the `CONFIG_FILES` field of the master configuration file, separated by a comma, as shown next.



- 6 Power on your phones. Your phones display the Skype for Business Sign In screen and users can [Sign In Methods](#) of the phone.



Note: You can store your two Skype for Business configuration files and the master configuration file in different directories; however, you must specify the file location path of the two Skype for Business files in the `CONFIG_FILES` field of the master configuration file, for example:

- `directory/lyncCommon.cfg`
- `directory/lyncSignIn.cfg`

Set the Base Profile with `device.set` Parameters

Use a provisioning server and configuration files to set the Base Profile of multiple phones to Skype. This is a power provisioning method for administrators familiar with centralized provisioning and configuration files.

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

To set the Base Profile using `device.set` parameters:

- 1 Locate the `device.cfg` template configuration file.
- 2 Place the `device.cfg` file on your provisioning server.
- 3 Locate and change the values of the three parameters:
 - `device.baseProfile=Skype`
 - `device.set=1`

- ```
➤ device.baseProfile.set=1
```
- 4 Rename and save the file.
  - 5 Power on the phones.
  - 6 Once boot-up is complete, remove device.set from the template configuration file and save the file without device.set.
  - 7 Verify that the device Base Profile is set to **Skype** by going to **Settings > Advanced**.
  - 8 Enter the password (default 456) and press **Enter**.
- Users can now sign in.

## Configuring In-Band Provisioning Settings

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

Refer to [In-Band Provisioning](#) for more information about configuring in-band parameters and a list of all in-band parameters.

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

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

### In-band Provisioning Device Settings

| Parameter                                       | Permitted Values                                                                                                                                                                                      |
|-------------------------------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>lync.provisionDeviceParams.enabled</code> | <p>1 (default) - Enable (accept) in-band provisioning device settings sent from Skype for Business.</p> <p>0 - Disable (block) in-band provisioning device settings sent from Skype for Business.</p> |

# Configuring Features for Skype for Business

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This section details a number of features and functions available on Polycom phones registered with Skype for Business.

## Configuring Call Forwarding

This section provides details on using UC Software call forwarding call forwarding with Skype for Business.

If you are registering phones with Skype for Business using UC Software 5.4.0A or earlier, you can configure `softkey.feature.forward=1` to display or remove the Forward soft key.

If you are registering phones with Skype for Business using UC Software 5.4.0A or later, the Microsoft server automatically sends Call Forwarding functionality and the Forward soft key in-band to the phones. In this case:

- The parameter `softkey.feature.forward` is overridden by Microsoft server settings and has no effect.
- If `softkey.feature.forward=1`, the Microsoft server settings override `softkey.feature.forward`.
- If `softkey.feature.forward=0`, the phone receives the Forward soft key setting from the Microsoft server. If you want to disable the Forward soft key, you must disable it on the Microsoft server.

## 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 DHCP Option 43, and is not available for online deployments.
- **Online Web Sign In.** For online deployments only, this method enables secure sign-in from a browser on your computer or mobile device. The phone generates a unique pairing code used to sign in on a secure Office 365 website. For more information about Web Sign In, refer to [Web Sign In for Skype for Business Online](#)
- **Via PC with BToE Sign In.** 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.



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

## Configuring a Skype for Business Sign In Method and Credentials

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

### Skype for Business Sign In Method Parameters

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

## Example Sign In Configurations

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

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

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

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

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

## ***PIN Authentication***

You can sign in to Skype for Business using PIN authentication. Polycom UC Software 5.1.1 introduces PIN authentication for SoundStructure VoIP Interface registered with Skype for Business.

To use PIN authentication, you must enable the Web Configuration Utility, which is disabled by default. Refer to the section [Enabling Access to the Web Configuration Utility](#). After you enable the Web Configuration Utility, you can enable or disable PIN authentication using `reg.1.auth.usePinCredentials`.

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

## ***Disabling the Sign-In and Sign-Out Soft Keys***

If your VVX business media phones are used as shared devices in your organization, you can remove the sign-out soft key to prevent users from signing others out. Or, you can remove both the sign-in and sign-out soft keys.

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

### **Sign-In and Sign-Out Soft Key Parameters**

| Parameter Template                                                       | Permitted Values                                                                                                                                                                                                                                                                                          |
|--------------------------------------------------------------------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>feature.lync.hideSignInSignOut</code><br><code>features.cfg</code> | 0 (default) - The Sign In and Sign Out soft keys display on the Home screen and phone menus.<br>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. |

**Sign-In and Sign-Out Soft Key Parameters**

| Parameter Template                                                     | Permitted Values                                                                                                                                                                                                                                                                                 |
|------------------------------------------------------------------------|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>feature.lync.hideSignOut</code><br><code>features.cfg</code>     | 0 (default) - The Sign Out soft key displays on the Home screen and phone menus.<br>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. |
| <code>softkey.feature.simplifiedSignIn</code><br><code>lync.cfg</code> | 0 (default) - The Sign In and Sign Out soft keys are removed from the Home screen and display in the Features menu.<br>1 - The Sign In and Sign Out soft keys displays on the Home screen and phone menus.                                                                                       |

**Web Sign In for Skype for Business Online**

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

To enable this sign in option for users, set the parameter `reg.1.auth.loginCredentialType=onlineDeviceAuth`.

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

**Configuring Web Sign In for Skype for Business Online**

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

**Skype for Business Online Web Sign-In Parameters**

| Parameter Template                                                          | Permitted Values                                                                                                                                                                                                                                                                                                                                                                                                      |
|-----------------------------------------------------------------------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>reg.1.auth.loginCredentialType</code><br><code>reg-advancd.cfg</code> | Specify the credential type the user must provide to log in. You cannot log in to the phone with Microsoft credentials if <code>reg.1.auth.loginCredentialType</code> is set to the default value.<br><code>LoginCredentialNone</code> (default)<br><code>onlineDeviceAuth</code> - Enables users to sign in to the phone using Web Sign In.<br><code>usernameAndPassword</code> - Provide description of this value. |

**Sign In with Better Together over Ethernet (BToE)**

You can use this sign-in method with the Better Together over Ethernet (BToE) feature. The BToE feature enables you to place, answer, and hold audio and video calls from your Polycom VVX phone and Skype for Business client on your computer. This sign in method is available after you download the BToE connector

application and pair your computer and phone. To download the application and for detailed instructions, see the *Polycom VVX Business Media Phones - User Guide* at [Polycom UC Software for Microsoft Deployments](#).

## Polycom Manual BToE PC Pairing

This feature enables users to manually pair their VVX business media phone with their computer using the Polycom Better Together over Ethernet Connector application. When an administrator enables this feature users can select Auto or Manual pairing mode in the Web Configuration Utility or in the Administrator Settings menu on the 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.

- To use the Manual PC Pairing feature, users must update to UC Software 5.5.1 and BToE Connector application version 3.4.0.
- Users can use UC Software versions earlier than 5.5.1 with BToE Connector application version 3.4.0 but Manual PC Pairing is not available.
- BToE Connector application versions earlier than 3.4.0 work with UC Software 5.5.1 but Manual PC Pairing is not available.

### ***Enable or Disable BToE PC Pairing from the Phone Menu***

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

#### **To enable or disable BToE PC Pairing from the phone menu:**

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

### ***Enable or Disable BToE PC Pairing from the Web Configuration Utility***

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

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

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

## Skype for Business Exchange Calendar Integration

As of UC Software 5.3.0, Polycom phones supports Exchange Server autodiscover. Or, you can enable the parameter `feature.EWSAutodiscover.enabled` in configuration files or on a per-phone basis with the Web Configuration Utility. When using a UC Software release prior to 5.3.0, you can enable the exchange calendar using centralized provisioning or with the Web Configuration Utility after you [Enabling Access to the Web Configuration Utility](#).



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

### ***Enable Exchange Calendar from a Provisioning Server***

Exchange calendar autodiscovery is enabled by default. You have the option to enable Skype for Business Exchange calendar using the following parameters on your central provisioning server. Note that the parameters are not included in the template configuration files and you must enter the parameter manually to one of your existing configuration files.

#### **To enable the exchange calendar from a provisioning server:**

- 1 Add the following parameter to one of your configuration files:

- `feature.exchangeCalendar.enabled=1`
- `exchange.server.url=https://<example URL>`

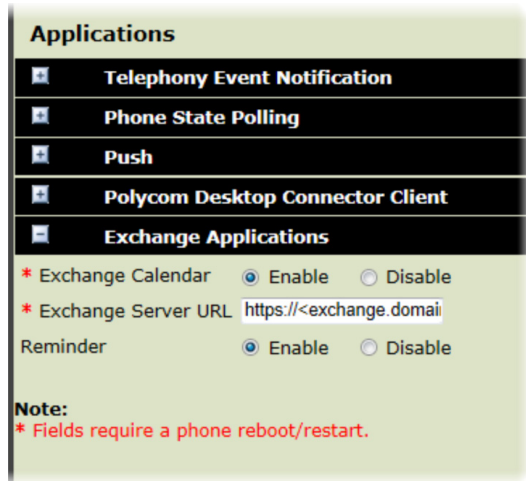
### ***(Optional) Enable the Exchange Calendar Manually***

Exchange calendar autodiscovery is enabled by default. You have the option to use the Web Configuration Utility to manually enable Skype for Business Exchange Calendar. This is useful for troubleshooting if autodiscovery is not working or misconfigured. This method applies only to a single phone at a time.

#### **To enable the exchange calendar manually:**

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





- 4 In the **Exchange Calendar** field, select **Enable**.
- 5 Enter the exchange web services URL using a Microsoft Exchange Server URL, for example `https://<mail.com>/ews/exchange.asmx`. In this example, the URL part `<mail.com>` is specific to an organization
- 6 At the bottom of the browser page, click **Save**.
- 7 When the confirmation dialog displays, click **Yes**.

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

## Skype for Business User Profiles

Administrators 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.

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

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

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

Polycom recommends that you create a single default user password for all users. You can reset a user's password by removing the password parameter from the override file. This causes the phone to use the default password in the `<user>.cfg` file.



Note: To convert a phone-based deployment to a user-based deployment, copy the `<MACAddress>-phone.cfg` file to `<user>-phone.cfg` and copy `phoneConfig<MACAddress>.cfg` to `<user>.cfg`.

To set up the user profile feature, you must:

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

## Create a User Profile Configuration File

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

### To create phone configuration file:

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

## Create a User Configuration File

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

### To create a user configuration file:

- 1 On the provisioning server, create a user configuration file for each user to log in to the phone. The name of the file is the user's ID to log in to the phone. For example, if the user's login ID is `user100`, the name of the user's configuration file is `user100.cfg`.
- 2 In each `<user>.cfg` file, you can add and set values for the user's login password (optional).
- 3 Add and set values for any user-specific parameters, such as:
  - Registration details (for example, the number of lines the profile displays and line labels).
  - Feature settings (for example, browser settings).



Note: If you add optional user-specific parameters to `<user>.cfg`, add only those parameters that will not cause the phone to restart or reboot when the parameter is updated. For information on which parameters cause the phone to restart or reboot, see the reference section Configuration Parameters.

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

- Log in to a phone to access their personal phone settings.

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

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

## Stored User Settings

If a user updates their password or other user-specific settings using the Main Menu on the phone, the updates are stored in `<user>-phone.cfg`, not `<MACaddress>-phone.cfg`.

If a user updates their contact directory while logged in to a phone, the updates are stored in `<user>-directory.xml`. Directory updates display each time the user logs in to a phone. On VVX phones, an up-to-date call lists history is defined in `<user>-calls.xml`. This list is retained each time the user logs in to their phone. Configuration parameter precedence (from first to last) for a phone that has the user profile feature enabled is:

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

## Unified Contact Store

Administrators can unify users' contacts with Microsoft Exchange Server to enable users to access and manage contacts from any application or device synchronized with the Exchange Server such as the VVX business media phone, Skype for Business client, Outlook, or Outlook Web Application from a mobile device. For example, if a user deletes a contact from a VVX business media 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 VVX business media 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](#).

## Skype for Business Enhanced Presence Status

The presence feature enables users to monitor the status of remote contacts from their phone. Users can monitor changes in the status of up to 200 remote contacts in real time when they add them as Favorites on the VVX phone and expansion module. Note that contacts can block others from monitoring their phones. For more information about the Microsoft presence feature, see the *Polycom VVX Business Media Phones for Skype for Business - User Guide* on [Polycom UC Software for Microsoft Deployments](#).

The VVX Expansion Modules with paper displays do not support Skype for Business registrations, and you cannot configure paper display expansion modules with Skype for Business features. You can only configure VVX Color expansion modules to work with Skype for Business.

## Configuring Presence Status

The following section lists parameters that configure presence settings for Microsoft servers.





When you enable the `pres.idleSoftkeys` parameter, the My Status and Contacts soft keys display on the phone's home screen. The `pres.reg` parameter uses the address of phone line 1 for the presence feature. Press the Contacts soft key to display your Contacts list.

### Presence Parameters




| Parameter Template                                                          | Permitted Values                                                                                                                                                                                                                                             |
|-----------------------------------------------------------------------------|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>feature.presence.enabled</code><br><code>lyncSharedExample.cfg</code> | 1 (default) - Enable the presence feature to manage your buddy list and display the status of your contacts.<br>0 - Disable the presence feature.                                                                                                            |
| <code>pres.idleSoftkeys</code><br><code>features.cfg</code>                 | 1 (default) - The MyStat and Buddies presence idle soft keys display.<br>0 - The MyStat and Buddies presence idle soft keys not do display.                                                                                                                  |
| <code>pres.reg</code><br><code>features.cfg</code>                          | 1 (default) - The valid line/registration number used for presence. This registration sends a SUBSCRIBE for presence. If the value is not a valid registration, this parameter is ignored.<br>1 - 34 - The valid line/registration number used for presence. |

The following table describes the presence status icons that display on phones registered with Microsoft servers.

### Presence Status Icons

| Icon                                                                                | Description                                         |
|-------------------------------------------------------------------------------------|-----------------------------------------------------|
|  | Available                                           |
|  | Busy, In a Call, In a Meeting, In a Conference Call |
|  | Away, Be Right Back, Inactive, Off Work             |
|  | Do Not Disturb, Presenting, In Presentation         |

**Presence Status Icons**

| Icon                                                                              | Description |
|-----------------------------------------------------------------------------------|-------------|
|  | Offline     |
|  | Unknown     |
|  | Blocked     |

## Enhanced Feature Line Key (EFLK)

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

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

- Registration
- 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 on the phone menu. The phone menu option to customize line keys is not available during active calls. After a user enables custom line keys on the phone menu, contacts on the phone's local contact directory are not available.

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

Note the following points when using EFLK:

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

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

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

The user .csv customization files cannot be edited manually. To apply a common customization to multiple phones, administrators can rename any user file by replacing the <MACaddress> part of the user file name

with <000000000000>-<sign-in address>.csv. You must use centralized provisioning to share custom .csv files.

## EFLK Limitations

Note the following limitations when using EFLK:

- The .csv file is always stored in the root directory and you cannot use a sub-directory.
- The phone does not load the .csv file when checking the server for updates using check sync. Configuration changes require a phone restart or reboot. Updates to the phone configuration by pressing **Settings > Basic > Update Configuration** also require a restart or reboot.
- The user cannot configure Speed Dials.
- The previous FLK feature using `lineKey.reassignment.enabled` does not work with UC Software 5.4.1 or later on phones using a Skype for Business Base Profile. The later EFLK feature requires UC Software 5.4.1 or later.

## Configuring EFLK

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

### EFLK Parameters

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

## Configuring Shared Line Appearance (SLA) for Skype for Business

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

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

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

The following features are not supported on SLA lines:

- BToE
- Conference class
- Call Park

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

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

#### SLA for Skype for Business Parameters

| Parameter Template          | Permitted Values                                                                                                                                                                                                                                                                 |
|-----------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| up.SLA.ringType<br>lync.cfg | 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 menu system, which overrides the value you set here.<br>1 (default)<br>0 - 25 |

## Polycom NoiseBlock™

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

### *Polycom NoiseBlock Parameters*

The following parameters configure the Polycom NoiseBlock feature.

#### Polycom NoiseBlock Parameters

| Parameter Template             | Permitted Values                                                                                          |
|--------------------------------|-----------------------------------------------------------------------------------------------------------|
| voice.ns.hf.blocker<br>new.cfg | 0 (default) - Polycom NoiseBlock technology is disabled.<br>1 - Polycom NoiseBlock technology is enabled. |

## Centralized Conference Control Protocol (CCCP)

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

### *Centralized Conference Control Protocol (CCCP) Parameters*

The following parameters configure CCCP.

**CCCP Parameters**

| Parameter Template                                                           | Permitted Values                  |
|------------------------------------------------------------------------------|-----------------------------------|
| <code>feature.cccp.enabled</code>                                            | 1 (enabled) - Enable use of CCCP. |
| <code>lyncSharedLCExample.cfg</code> ,<br><code>lyncSharedExample.cfg</code> | 0 - Disable use of CCCP.          |

## Enabling Skype for Business Exchange Integration

Exchange Integration is available for Skype for Business, Office 365, and Lync Server 2010 and 2013 deployments. This feature enables set up of visual voicemail, call log synchronization, Outlook contact search, and Skype for Business Address Book Service (ABS) adaptive search. Each of these features is enabled by default on Polycom phones registered with Skype for Business.



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

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

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

Note the following enhancements:

- Verify which Exchange Server services are not working on each phone by going to **Status > Diagnostics > Warnings** on the phone.
- View the status of each service in the Web Configuration Utility.
- The phone receives voicemails from Skype for Business Server and messages play on the phone. You cannot download voicemail messages to the phone.

## Setting Up Calendar Features

- Connect the phone to the Exchange Server using one of two available methods.
  - Method one. By default this is enabled. Install and run the autodiscovery service on the Microsoft Server to get an exchange server URL automatically.
  - Method two. (Optional) Configure the Exchange Server URL. Using this method, the URL takes precedence over the default autodiscovery service.
- Visual voicemail. On the server, enable unified messaging and enable messages to play on the phone for each user. If you disable `feature.exchangeVoiceMail.enabled`, the Message Center and Skype for Business Voice mail menus display the message. Skype for Business Server only plays voicemail and you cannot download voicemails or play locally on the phone.
- Call log synchronization: On the server, enable the option to save calls logs to each user's conversation history in Outlook.



- ABS adaptive search. On the server, enable the ABS service. There are three possible configurations.
  - Outlook and ABS are both enabled by default. When both are enabled, the phone displays the Skype for Business Directory.
  - If you disable Outlook and enable only ABS, the phone displays the Skype for Business Directory.
  - If you enable Outlook and disable ABS, the Outlook Contact Search displays in Directories.



Web Info: For help with Lync Server 2010, refer to Microsoft [Configure Exchange Services for the Autodiscover Service](#).

For help with Lync Server 2013, refer to Microsoft [Configuring Unified Messaging on Microsoft Exchange Server to work with Lync Server 2013](#).

## Configuring Microsoft Exchange Integration

The following table lists parameters that configure Microsoft Exchange integration.

### Exchange Integration Parameters

| Parameter Template                                                        | Permitted Values                                                                                                                                                                                                                                                                         |
|---------------------------------------------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>exchange.meeting.alert.followOfficeHours</code><br>applications.cfg | 1 - Audible alerts occur during business hours.<br>0 - Audible alerts occur at all times.                                                                                                                                                                                                |
| <code>exchange.meeting.alert.tonePattern</code><br>applications.cfg       | positiveConfirm (default) - Set the tone pattern of the reminder alerts using any tone specified by <code>se.pat.*</code> . See section Customize Audio Sound Effects in the UC Software Administrator Guide.                                                                            |
| <code>exchange.meeting.alert.toneVolume</code><br>applications.cfg        | 10 (default) - Set the volume level of reminder alert tones.<br>0 - 17                                                                                                                                                                                                                   |
| <code>exchange.meeting.parseWhen</code><br>applications.cfg               | NonSkypeMeeting (default) - disables the number-searching logic of the Calendar to look for additional numbers to dial in Skype Meeting calendar entries.<br>Always - enables the number-searching logic of the Calendar to look for additional numbers to dial even for Skype Meetings. |
| <code>exchange.meeting.phonePattern</code><br>applications.cfg            | NULL (default)<br>string<br>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).                                                                             |
| <code>exchange.meeting.reminderEnabled</code><br>applications.cfg         | 1 (default) - Meeting reminders are enabled.<br>0 - Meeting reminders are disabled.                                                                                                                                                                                                      |

**Exchange Integration Parameters**

| Parameter Template                                                      | Permitted Values                                                                                                                                                                                                                                                                                                                      |
|-------------------------------------------------------------------------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>exchange.meeting.reminderInterval</code><br>applications.cfg      | 300 seconds (default)<br>60 - 900 seconds<br>Set the interval at which phones display reminder messages.                                                                                                                                                                                                                              |
| <code>exchange.meeting.reminderSound.enabled</code><br>applications.cfg | 1 - The phone makes an alert sound when users receive reminder notifications of calendar events.<br>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 <code>exchange.meeting.reminderEnabled</code> is also enabled. |
| <code>exchange.meeting.reminderType</code><br>applications.cfg          | Customize the calendar reminder and tone.<br>2 (default) - Reminder is always audible and visual.<br>1 - The first reminder is audible and visual reminders are silent.<br>0 - All reminders are silent.                                                                                                                              |
| <code>exchange.reconnectOnError</code><br>applications.cfg              | 1 (default) - The phone attempts to reconnect to the Exchange server after an error.<br>0 - The phone does not attempt to reconnect to the Exchange server after an error.                                                                                                                                                            |
| <code>exchange.server.url</code><br>applications.cfg                    | NULL (default)<br>string<br>The Microsoft Exchange server address.                                                                                                                                                                                                                                                                    |
| <code>feature.exchangeCalendar.enabled</code><br>features.cfg           | 0 - Exchange calendar service is disabled.<br>1 - The Exchange calendar feature is enabled and users can view meeting notifications on the phone.                                                                                                                                                                                     |
| <code>feature.exchangeCallLog.enabled</code><br>features.cfg            | 1 (default)<br>0                                                                                                                                                                                                                                                                                                                      |
| <code>feature.exchangeContacts.enabled</code><br>features.cfg           | 1 (default)<br>0                                                                                                                                                                                                                                                                                                                      |
| <code>feature.EWSAutodiscover.enabled</code><br>features.cfg            | 1 - Exchange autodiscovery is enabled and the phone automatically discovers the Exchange server using the email address or SIP URI information.<br>0 - Exchange autodiscovery is disabled on the phone and you must manually configure the Exchange server address.                                                                   |
| <code>feature.exchangeVoiceMail.enabled</code><br>features.cfg          | 1 (default)<br>0                                                                                                                                                                                                                                                                                                                      |

**Exchange Integration Parameters**

| Parameter Template                                                         | Permitted Values                                                                                                                                                                                                                                                 |
|----------------------------------------------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>feature.lync.abs.enabled</code><br><code>features.cfg</code>         | 1 - Enable comprehensive contact search in the Lync Server address book service.<br>0 - Disable comprehensive contact search in the Lync Server address book service.                                                                                            |
| <code>feature.lync.abs.maxResult</code><br><code>features.cfg</code>       | 12 (default)<br>5 - 50<br>The value for this parameter defines the maximum number of contacts to display in a Lync Server address book service contact search.                                                                                                   |
| <code>up.oneTouchDirectory</code><br><code>features.cfg</code>             | 1 - The Skype for Business Search icon displays on the Home screen.<br>0 - The Skype for Business Search icon does not display on the Home screen.                                                                                                               |
| <code>up.oneTouchVoiceMail<sup>1</sup></code><br><code>features.cfg</code> | 0 - The phone displays a summary page with message counts. The user must press the Connect soft key to dial the voicemail server.<br>1 - The phone dials voicemail services directly (if available on the call server) without displaying the voicemail summary. |

<sup>1</sup> Change causes phone to restart or reboot.

## Updating Polycom UC Software

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

### Update UC Software Manually

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

#### To update UC Software manually:

- 1 Download and unzip UC Software to a directory on your provisioning server.
- 2 On the phone, go to **Settings > Advanced**, enter the password (default 456)
- 3 Go to **Network Configuration > Provisioning Server > DHCP Menu > Boot Server**.
- 4 In the Boot Server menu, choose **Static** if you are testing or provisioning a few phones, or choose **Option 66** if you are provisioning in a large environment and want phones to use a boot server defined in DHCP. If you choose Option 66, skip step 5 and go to step 6.
- 5 Go back to **Provisioning Server** and do the following:
  - Choose a server type in the **Server Type** field.

- Enter the server address, for example, `http://server.domain.com/41X` or `ftp://ftp.domain.com/41X`.
- Enter your server user name and server password, if required.
- 6** Press **Back** until you are prompted to save your settings.
- 7** Choose **Save Configuration** to save your settings. The phone reboots.
- 8** Confirm that the phone is running a Skype for Business-enabled Polycom UC Software version.
  - On the VVX 500 Business Media phone, choose **Settings > Status > Platform > Application > Main**. The UC Software version displays beside Version.



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

## Configuring UC Software Automatic Updates

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

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

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

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

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

## Configuring Automatic Software Update

The following table lists parameters that configure automatic software updates.

### Automatic Software Update Parameters

| Parameter Template                                   | Permitted Values                                                                                                                                                                                                                                                                                             |
|------------------------------------------------------|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>device.prov.lyncDeviceUpdateEnabled</code>     | 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.<br>1 (default) - The automatic device update is enabled on the phone and the phone receives software updates from the server. |
| <code>device.prov.lyncDeviceUpdateEnabled.set</code> | 0 (default) - Disable automatic device update for all devices.<br>1 - Enable automatic device update for all devices and use <code>device.prov.lyncDeviceUpdateEnabled</code> .                                                                                                                              |
| <code>lync.deviceUpdate.popUpSK.enabled</code>       | 0 (disable) - Disable the Information popup that indicates when an automatic software update is available.<br>1 - Enable the Information popup that indicates when an automatic software update is available.                                                                                                |
| <code>lync.deviceUpdate.serverPollInterval</code>    | 7200 seconds (default) - The time interval in seconds that the phone sends a software update request to the Skype for Business Server.<br>min=1800 seconds<br>max=28800 seconds                                                                                                                              |
| <code>lync.deviceUpdate.userInactivityTimeout</code> | 900 seconds [15 minutes] (default) - Sets the user inactivity timeout period after which the phone's software is automatically updated.<br>Min=300 seconds<br>Max=1800 seconds                                                                                                                               |

## Resetting the Phone to Default Settings and Factory Default

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

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

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



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

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

## ***Reset Local Phone Settings***

You can reset the phone settings users make locally from the phone.

### **To reset local phone settings:**

- 1 On your phone, go to **Settings > Advanced**.
- 2 Enter the password (default 456).
- 3 Go to **Administration Settings > Reset to Defaults > Reset Local Configuration**.
- 4 At the prompt '**Are you sure?**', tap **Yes**.

## ***Reset Web Configuration Utility Settings***

You can reset phone settings made from the phone Web Configuration Utility.

### **To reset web settings:**

- 1 On your phone, go to **Settings > Advanced**.
- 2 Enter the password (default 456).
- 3 Go to **Administration Settings > Reset to Defaults > Reset Web Configuration**.
- 4 At the prompt '**Are you sure?**', tap **Yes**.

The phone may reboot, depending on the parameters set using the Web Configuration Utility.

## ***Manually Reset the Phone to Factory Default Settings***

You can reset the phone to factory default settings.

### **To manually reset the phone to factory default settings:**

- 1 On your phone, go to **Settings > Advanced**.
- 2 Enter the password (default 456), and press **Enter**.
- 3 Go to **Administration Settings > Reset to Defaults**, and select **Reset to Factory**.
- 4 At the prompt '**Are you sure?**', tap **Yes**.

The phone reboots to factory default settings.

## Changing the Default Password

As of UC Software 5.1.0, 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 administrators 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 system
- Web Configuration Utility
- Use the parameter `reg.1.auth.password` in the template configuration file `lyncPerPhoneExample.cfg`.

## Enabling Access to the Web Configuration Utility

Polycom UC Software 5.1.1 introduces a security enhancement for VVX phones and expansion modules and the SoundStructure VoIP Interface registered with Microsoft Lync Server 2013.

As of UC Software 5.1.1, access to the Web Configuration Utility for phones registered with Lync Server or Skype for Business Server is disabled by default. Administrators must enable access to a phone's Web Configuration Utility from the phone menu system or using configuration parameters.

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.

If you set the Base Profile of a phone to Skype or 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.

After you successfully access the phone, you can enable access to the Web Configuration Utility from the phone menu system or using the parameters listed in the table Enable Web Configuration Utility. After you successfully enable the Web Configuration Utility for the SoundStructure VoIP Interface, you can use the Web Configuration Utility to change the administrator password.

### ***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 from the phone's menu system.

**To enable access to the Web Configuration Utility from the phone menu:**

- 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**.

## ***Change the Base Profile from the Phone Menu***

You can change the phone's Base Profile from the phone menu.

**To change the phone's Base Profile:**

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

## ***Enabling the Web Configuration Utility***

The security update for Skype for Business with Polycom UC Software 5.1.1 includes a new 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.

**Web Configuration Utility Parameters**

| Parameter Template                                                                                            | Permitted Values                                                                                                                    |
|---------------------------------------------------------------------------------------------------------------|-------------------------------------------------------------------------------------------------------------------------------------|
| <code>device.sec.coreDumpEncryption.enabled</code><br><code>device.cfg, site.cfg</code>                       | 0 (default)<br>1                                                                                                                    |
| <code>device.sec.coreDumpEncryption.enabled.set</code><br><code>device.cfg, site.cfg</code>                   | 0 (default)<br>1                                                                                                                    |
| <code>httpd.cfg.enabled</code><br><code>lyncSharedExample.cfg,</code><br><code>lyncSharedLCExample.cfg</code> | Base Profile = Generic<br>1 (default) - The Web Configuration Utility is enabled.<br>0 - The Web Configuration Utility is disabled. |
|                                                                                                               | Base Profile = Skype<br>0 (default) - The Web Configuration Utility is disabled.<br>1 - The Web Configuration Utility is enabled.   |



**Web Configuration Utility Parameters**

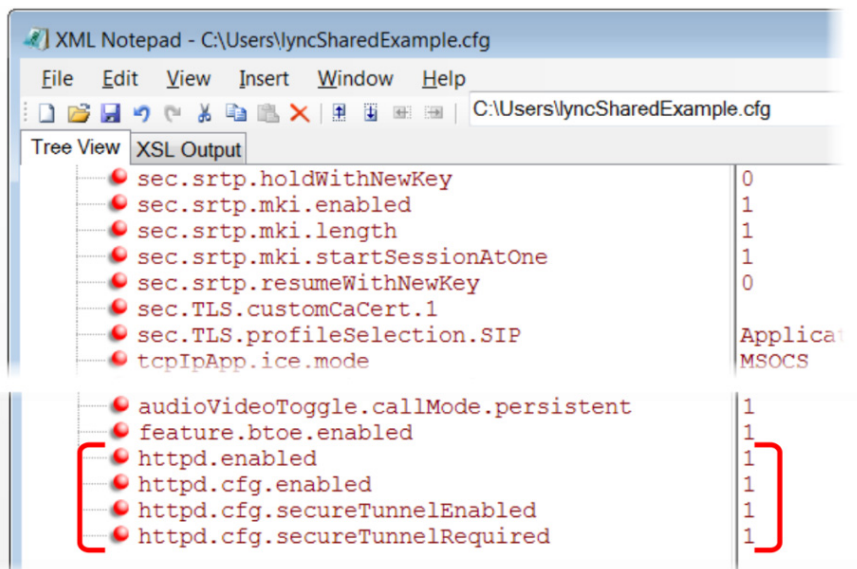
| Parameter Template                                                                  | Permitted Values                                                                                                                                                                                                                                            |
|-------------------------------------------------------------------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| httpd.cfg.secureTunnelRequired<br>lyncSharedExample.cfg,<br>lyncSharedLCExample.cfg | 1 (default) - Access to the Web Configuration Utility is allowed only over a secure tunnel (HTTPS) and non-secure (HTTP) is not allowed.<br>0 - Access to the Web Configuration Utility is allowed over both a secure tunnel (HTTPS) and non-secure (HTTP). |
| httpd.enabled<br>lyncSharedExample.cfg,<br>lyncSharedLCExample.cfg                  | Base Profile = Generic<br>1 (default) - The web server is enabled.<br>0 - The web server is disabled.<br><br>Base Profile = Skype<br>0 (default) - The web server is disabled.<br>1 - The web server is enabled.                                            |

**Example Skype for Business Security Update Configuration**

This section provides an example configuration for the Skype for Business security update. When the phone registers, the Web Configuration Utility is disabled. This example configuration illustrates how to enable access to a phone's Web Configuration Utility when phones are registered with Skype for Business.

By default, a pop-up message displays on phones registered with Skype for Business. This message prompts administrators to change the default password use to access the phone's Web Configuration Utility as an administrator.

After you change the default password, enable access to the Web Configuration Utility using the parameters shown in the following figure.



## Configuring Boss-Admin

This section explains the Boss-Admin feature for Skype for Business and Lync Server 2010.

If you are configuring Boss-Admin for Lync Server 2010, refer to [Configure Boss-Admin for Lync Server 2010](#) for additional procedures. For details about Safe Transfer, refer to [Safe Transfer for Boss-Admin](#).

For more information about this user feature, see the latest *Polycom VVX Business Media Phones for Skype for Business - User Guide* on [Polycom UC Software for Microsoft Deployments](#).

As a Boss, you can use the Skype for Business or Lync client to assign delegates to share a line so that both can place, answer, hold, transfer calls, and set ringtones on the boss line. You cannot assign yourself as a delegate to a Boss. Bosses can assign up to 25 delegates to their line. The maximum number of boss lines delegates can be assigned depends on the number of line keys on their VVX phone. If the number of boss lines exceeds the maximum number of line keys, a connected VVX Expansion Module with color display shows the remaining configured boss lines.

The following table lists the maximum number of line keys on VVX phones that can display as boss lines.

### Maximum Delegate Line Keys for Assigned Bosses

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

## Configure Boss-Admin for Lync Server 2010

You configure the Lync Boss-Admin feature from the Lync client application on a computer. However, if you are using Lync Server 2010, an administrator must complete the following procedure.

### To configure Boss-Admin for Lync Server 2010:

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

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

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

The SQL server operation is sent to the presence backend 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 backend, shown as <se.fabrikam.com> in this example.

## Safe Transfer for Boss-Admin

To enable the safe transfer feature and display the Safe Transfer soft key, you must enable `feature.lyncSafeTransfer.enabled`. A safe transfer transfers a call to another party and allows you to continue monitoring the call with the option to resume before the call goes to voicemail. If the call is answered by the other party, you are disconnected from the call.

### Configuring Safe Transfer

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

#### Configure Safe Transfer

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

## Extended Link Layer Discovery Protocol (LLDP)

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

Media Endpoint Discover (MED) capabilities include:

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

### Configuring LLDP Fast Start Count

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

**LLDP Parameters**

| Parameter Template                                    | Permitted Values                                                                                                                                                                                                                                                                                                                                                                                   |
|-------------------------------------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| device.net.lldpFastStartCount<br>device.cfg, site.cfg | Configure the fast-start LLDP packets that the phone sends when booting up or when the network comes up.<br>5 (default)<br>3 - 10<br>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. |

## International Dialing Prefix

Enter a '+' symbol before you dial an international phone numbers to identify to the switch that the phone number you are dialing is international.

### Configuring International Dialing Prefixes

The following parameters configure the international dialing prefixes.

**International Dialing Prefix Parameters**

| Parameter Template                                | Permitted Values                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                  |
|---------------------------------------------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| call.international<br>Dialing.enabled<br>site.cfg | This parameter applies to all numeric dial pads on the phone, including for example, the contact directory.<br>Changes you make to this parameter cause a restart or reboot.<br>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 "*".<br>0 - When you disable this parameter, you cannot dial "+" and you must enter the international exit code of the country you are calling from to make international calls. |
| call.international<br>Prefix.key<br>site.cfg      | The phone supports international call prefix (+) with both '0' and '*'.<br>0 (default) - Set the international prefix with *.<br>1 - Set the international prefix with 0.                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                         |

## Comfort Noise

The phone sends background noise to near-end users. When enabled, the Comfort Noise payload type is negotiated in SDP with a default of 13 for 8 KHz codecs and a configurable value between 96 and 127 for 16 KHz codecs.

## Configure Comfort Noise

The following table lists the parameters you can use to configure Comfort Noise Control.

### Comfort Noise Parameters

| Parameter Template                    | Permitted Values                                                                                                                                                                                                                                                                                                     |
|---------------------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| voice.CN16KPayload<br>sip-interop.cfg | Set the dynamic payload type to be used for Comfort Noise RTP packets.<br>122 (default)<br>96 - 127                                                                                                                                                                                                                  |
| voice.CNControl<br>sip-interop.cfg    | 0 (default) - Comfort noise payloads are not published in the SDP body of the INVITE message.<br>1 - Comfort noise payloads are published in the SDP body of the INVITE message by including the supported. Comfort noise uses payload type 13 for an 8 KHz sample rate codec and dynamic payload type 16k Hz codec. |

## Configuring Music on Hold

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

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

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

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

## Upload a Music File

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

### To upload a music file:

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

6 Click **Save**.

## Configuring Music on Hold

The following table lists parameters that configure MoH.

### Music on Hold Parameters

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

## Music on Hold Error Messages

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

### MoH Error Messages

| Failure Scenario                                                         | Error Message                                |
|--------------------------------------------------------------------------|----------------------------------------------|
| Phone fails to download the MoH file because the current file was active | 'Download failed'<br>'Current MoH is Active' |
| MoH file download failed                                                 | 'Download Failed'                            |
| MoH file size is zero                                                    | 'Download Failed'                            |
| MoH file size exceeds the maximum size of 500KB                          | 'File size exceeded 500KB'                   |

**MoH Error Messages**

| Failure Scenario                                                | Error Message                          |
|-----------------------------------------------------------------|----------------------------------------|
| An incorrect .wav file format is specified                      | 'Unsupported .wav file format'         |
| A network failure occurs while the phone downloads the MoH file | 'Download failed'<br>'Network is down' |

## Dial Plans

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

**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                                                                                                                 |

**Supported Dial Plans**

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

**Unsupported Dial Plans**

Examples of dial plans not supported include the following:

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

**Unsupported Dial Plans**

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

**Use a Master Key Identifier (MKI) to Secure Audio**

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 crypto with or without MKI.



**MKI Parameters**

| Parameter Template                                                | Permitted Values                                                                                                                                                                                                                                     |
|-------------------------------------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>sec.srtp.mki.enabled</code><br><code>sip-interop.cfg</code> | 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.<br>0 - Polycom phone offers only one non-MKI crypto in the SDP offer. |

## 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 or from the phone menu.

Device Lock is enabled 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. To recover a forgotten lock code, users can press the Forget Code soft key after which users are prompted to re-enter their sign-in password and create a new lock code. If users sign into their Skype for Business account using the Web Sign In method, the Forget Code soft key is not available - users are automatically signed out of the phone after six unsuccessful unlock attempts, must sign in again, and are prompted to create a new lock code.

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

## ***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"`.

## Configuring Device Lock for Skype for Business

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

### Device Lock Parameters

| Parameter                                                             | Permitted Values                                                                                                                                                                                                                                                                                                                                                                                                                             |
|-----------------------------------------------------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>feature.deviceLock.enable</code><br>features.cfg                | Enables or disables the Device Lock feature on the phone.<br>1 (Default) - Device Lock is enabled.<br>0 - Device Lock is disabled.                                                                                                                                                                                                                                                                                                           |
| <code>phoneLock.authorized.x.value</code><br>features.cfg             | 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, <code>phoneLock.authorized.1.value="cwi57@cohovineyard.com"</code> .                                                                                                                                                                                                                                        |
| <code>up.btoeDeviceLock.timeOut</code><br>features.cfg                | Configure a time delay after which the phone locks when the user locks the computer paired with the phone.<br>10 seconds (default)<br>0 - 40 seconds                                                                                                                                                                                                                                                                                         |
| <code>up.deviceLock.createLockTimeout</code><br>features.cfg          | Specify the timeout in minutes after which the Create Lock Code screen disappears and the user is signed out.<br>0 (default) - No timeout for the Create Lock Code prompt.<br>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.                                                                        |
| <code>up.deviceLock.signOutOnIncorrectAttempts</code><br>features.cfg | Specify whether to sign out the user from the phone after six unsuccessful attempts to unlock the phone.<br>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.<br>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. |

## 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 menu. This feature is available on all VVX business media phones registered with Skype for Business Server on-premises or online and with Microsoft Lync 2013 or 2010 Server.

Logs are uploaded to the Skype for Business or Lync Server at the following location which you can specify in the Skype for Business/Lync 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 menu or Web Configuration Utility are detailed in the latest *Polycom VVX Business Media Phones for Skype for Business - User Guide* on [Polycom Voice Support](#).

## Configure User Log Upload

The following table lists parameters that configure user log uploading.

### Configure User Log Uploading

| Parameter Template                     | Permitted Values                  |
|----------------------------------------|-----------------------------------|
| <code>feature.logUpload.enabled</code> | 1 (default) - Enable log uploads. |
| <code>features.cfg</code>              | 0 - Disable log uploads.          |

## Send Diagnostic Logs from the Phone

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

### To send diagnostic logs from the phone:

- » Go to **Settings > Basic > Diagnostic Logs > Upload Logs**. Files are uploaded as plain text.  
If the log upload is successful, the phone displays a message that the upload was successful.  
If the log upload fails, the phone displays a message that the log upload failed.

## Send Diagnostic Logs from the Web Configuration Utility

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

### To send diagnostic logs from the Web Configuration Utility:

- 1 Enter the IP address of the phone into a web browser and log in as Administrator or User.
- 2 Go to **Diagnostics > Upload Logs**. Files are uploaded as plain text.
- 3 View upload URLs at **Skype for Business Status > Skype for Business Parameters**:
  - Update Server Internal URL for on-premises deployments
  - Update Server External URL online deployments.
 If the log upload is successful, the phone displays a message that the upload was successful.  
If the log upload fails, the phone displays a message that the log upload failed.

## Setting Log Levels

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

### Set Log Levels from the Phone

You can set log levels from the phone menu.

**To set log levels from the phone:**

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

**Set Log Levels from the Web Configuration Utility**

You can set log levels from the Web Configuration Utility.

**To set log levels from the Web Configuration Utility:**

- 1 Enter the IP address of the phone into a web browser and log in as Administrator or User.
- 2 Go to **Settings > Logging**.
- 3 In **Server Log Level**, select a log level.

 ***toggling Between Audio-only or Audio-Video Calls***

When this feature is enabled on the VVX 1500, and VVX camera-enabled VVX 500/501 and 600/601 business media phones, a soft key displays to toggle calls between audio-only or audio-video.

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

When you enable this feature using `feature.audioVideoToggle.enabled`, calls are audio-only by default, and you must toggle the call to use audio-video before the call begins. After a video call has ended, the phone returns to audio-only.

If you want a call mode setting to persist until users manually change the call mode, also enable `audioVideoToggle.callMode.persistent`. Use the following table to configure audio/video toggle options.

**Audio/Video Toggle Parameters**

| Parameter Template                                                        | Permitted Values                                                                                                                                                                                                                                       |
|---------------------------------------------------------------------------|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>feature.audioVideoToggle.enabled</code><br><code>features.cf</code> | 0 (default) - The audio/video toggle feature is disabled.<br>1 - The audio/video toggle feature is enabled.                                                                                                                                            |
| <code>audioVideoToggle.callMode.persist</code><br><code>video.cfg</code>  | 0 (default) - The user setting to choose audio only or video calls by default is not retained after a phone restart or reboot.<br>1 - The user setting to choose audio only or video calls as the default is retained after a phone restart or reboot. |
| <code>video.autoStartVideoTx</code><br><code>video.cfg</code>             | 1 (default) - Automatically begin video to the far side when you start a call.<br>0 - Video to the far side does not begin.                                                                                                                            |
| <code>video.callMode.default</code><br><code>video.cfg</code>             | audio (default)<br>Select the outbound call mode. To use this parameter, you must enable <code>feature.audioVideoToggle.enabled</code> .                                                                                                               |

## Manually Installing a Certificate

If you need to set up a remote worker, you must manually enter a certificate to the phone. You can add the certificate using two parameters included in the `lyncSharedExample.cfg` and `lyncSharedLCEExample.cfg` files. 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 [Enabling Access to the Web Configuration Utility](#).

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

### ***Manually Install a Certificate Using Configuration Files***

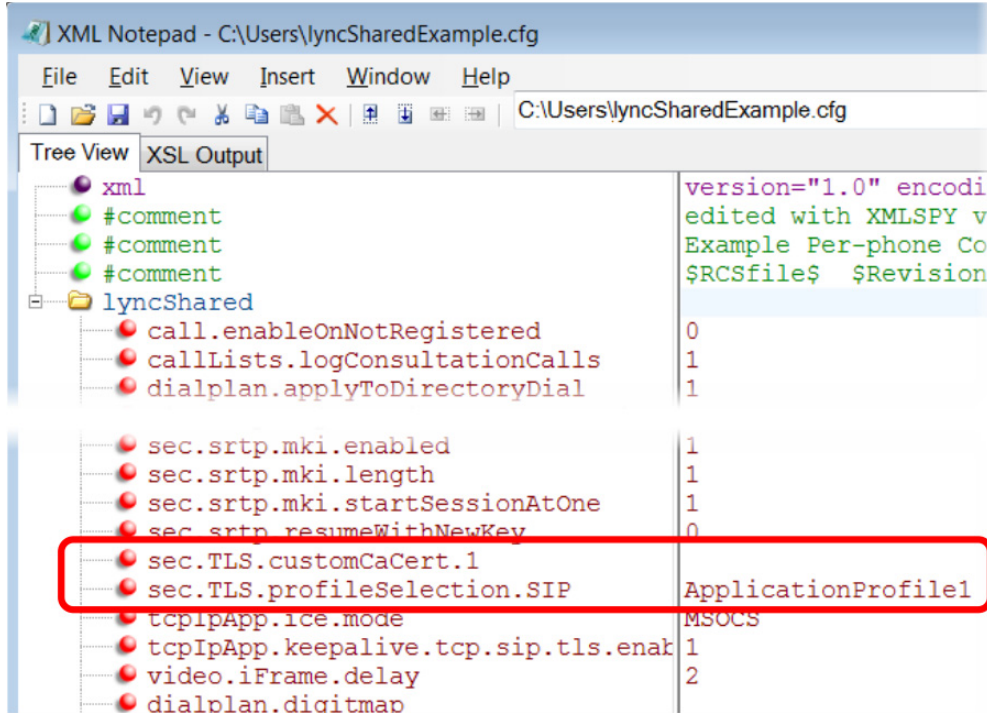
You can manually install a certificate using configuration parameters in the template files available with UC Software.

#### **To manually install a certificate using configuration files:**

- 1 Locate the `lyncSharedExample.cfg` and `lyncSharedLCEExample.cfg` configuration files in the **PartnerConfig > Microsoft** directory of the UC Software download.
- 2 Place the configuration file in a location in your Skype for Business directory.

**3** Enter the certificate and application profile to the following two parameters:

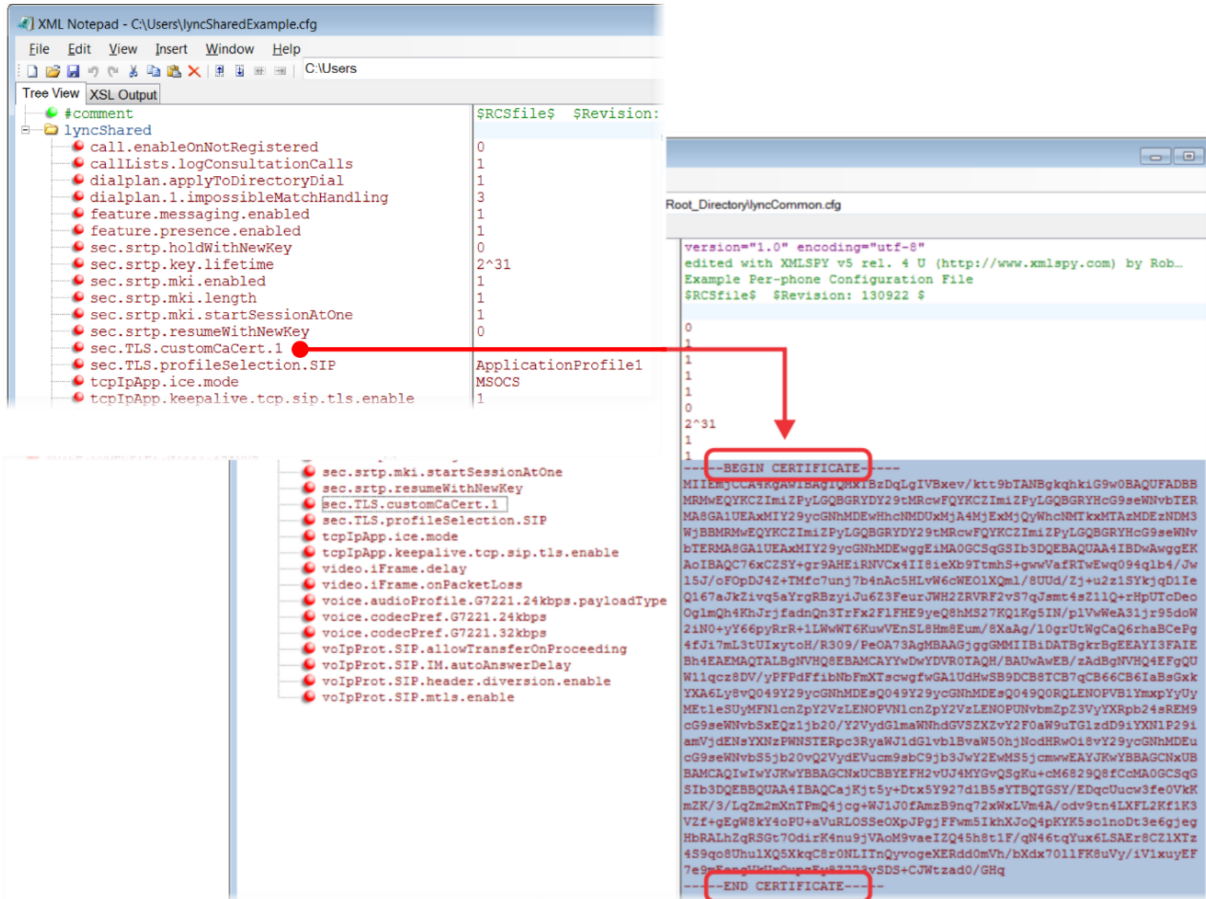
- sec.TLS.customCaCert.1=<enter the certificate>
- sec.TLS.profileSelection.SIP=<ApplicationProfile1>



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.

- 4 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.

## Manually Install a Certificate with the Web Configuration Utility

You can use the Web Configuration Utility to install a certificate manually after you [Enabling Access to the Web Configuration Utility](#).

### To install a certificate with the Web Configuration Utility:

- 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**.

## Configure Client Media Port Ranges for QoE

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

### To configure client media port ranges:

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

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

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

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

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

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

## Setting QoE Parameters on the Skype for Business Server

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

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

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

```
> Set-CsMediaConfiguration -Identity Global -EnableInCallQoS:$TRUE
-InCallQoSIntervalSeconds x (where x is a digit from 1 to 65535).
```



- `voice.qualityMonitoring.rtcp.r.enable`. Set to 1 (default) to allow the phone to collect RTCP XR metrics.

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

**QoE Parameters on Server Media Configuration**

```
PS C:\Users\administrator.COHOWINERY> Get-CsMediaConfiguration | fl
Identity : Global
EnableQoS : True
EncryptionLevel : RequireEncryption
EnableSiren : False
MaxVideoRateAllowed : UGA600K
EnableInCallQoS : True
InCallQoSIntervalSeconds : 35
EnableRtpRtcpMultiplexing : True
```

## Query QoE Status from the Web Configuration Utility

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

**To query the in-band QoE status:**

- 1 Enter the IP address of the phone into a web browser and log in as Administrator or User.
- 2 Go to **Diagnostics > Skype for Business Status > Quality of Experience**.

## Configuring QoE

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

**QoE Parameters**

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

**QoE Parameters**

| Parameter Template                                                  | Permitted Values                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                           |
|---------------------------------------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>voice.qoe.event.networkmos.threshold.bad.features.cfg</code>  | <p>Defines the threshold for Network MOS as follows:<br/>                     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.</p> <p>19 (default) - Indicates a MOS score of 1.9.<br/>                     10 - 50 - Indicates a MOS score between 1 - 5.<br/>                     networkMOS &gt; 2.9 signifies good quality<br/>                     networkMOS &gt; 2.9 &lt; 1.9 signifies poor quality<br/>                     networkMOS &lt; 1.9 signifies bad quality</p> |
| <code>voice.qoe.event.networkmos.threshold.poor.features.cfg</code> | <p>Defines the threshold for Network MOS as follows:<br/>                     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.</p> <p>29 (default) - Indicates a MOS score of 2.9.<br/>                     10 - 50 - Indicates a MOS score between 1 - 5.<br/>                     networkMOS &gt; 2.9 signifies good quality<br/>                     networkMOS &gt; 2.9 &lt; 1.9 signifies poor quality<br/>                     networkMOS &lt; 1.9 signifies bad quality</p> |

***Polycom-Supported Skype for Business QoE Elements***

This section lists the Microsoft Quality of Experience (QoE) elements supported by Polycom phones. For a list of all parameters that report QoE data, see [Microsoft \[MS-QoE\] PDF at \[MS-QoE\]: Quality of Experience Monitoring Server Protocol](#).

**Polycom-Supported Skype for Business QoE Elements**

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

**Polycom-Supported Skype for Business QoE Elements**

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

**Polycom-Supported Skype for Business QoE Elements**

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

**Polycom-Supported Skype for Business QoE Elements**

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

## Data Center Resiliency

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

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

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.

## Configuring Better Together over Ethernet (BToE) Firewall Ports

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

### BToE Firewall Ports

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

## Configuring Polycom Experience Cloud

The Polycom Experience Cloud (PEC) service is an experimental feature enabling you to view basic diagnostic and phone usage data including start and stop events, call quality information, packet statistics, call duration, and call logs.

### Experience Cloud Parameters

| Parameter Template                  | Permitted Values                                                                               |
|-------------------------------------|------------------------------------------------------------------------------------------------|
| <code>log.level.change.apps</code>  | Initial logging level for the Apps log module.<br>4 (default)<br>0 - 6                         |
| <code>log.level.change.bfcp</code>  | Initial logging level for the BFCP content log module.<br>4 (default)<br>0 - 6                 |
| <code>log.level.change.pec</code>   | Initial logging level for the Polycom Experience Cloud (PEC) log<br>4 (default)<br>0 - 6       |
| <code>log.level.change.mr</code>    | Initial logging level for the Networked Devices log module.<br>4 (default)<br>0 - 6            |
| <code>log.level.change.mraud</code> | Initial logging level for the Networked Devices Audio log module.<br>4 (default)<br>0 - 6      |
| <code>log.level.change.mrcam</code> | Initial logging level for the Networked Devices Camera log module.<br>4 (default)<br>0 - 6     |
| <code>log.level.change.mrcon</code> | Initial logging level for the Networked Devices Connection log module.<br>4 (default)<br>0 - 6 |
| <code>log.level.change.mrdis</code> | Initial logging level for the Networked Devices Display log module.<br>4 (default)<br>0 - 6    |
| <code>log.level.change.mrmgr</code> | Initial logging level for the Networked Devices Manager log module.<br>4 (default)<br>0 - 6    |
| <code>log.level.change.ppcip</code> | Initial logging level for the Polycom People+Content IP log module.<br>4 (default)<br>0 - 6    |

**Experience Cloud Parameters**

| Parameter Template                 | Permitted Values                                                                          |
|------------------------------------|-------------------------------------------------------------------------------------------|
| <code>log.level.change.prox</code> | Initial logging level for the Proximity log module.<br>4 (default)<br>0 - 6               |
| <code>log.level.change.ptp</code>  | Initial logging level for the Precision Time Protocol log module.<br>4 (default)<br>0 - 6 |



# UC Software Parameters for Microsoft Deployments

Polycom provides several template files for Microsoft deployments:

[Skype Base Profile](#)

[Skype for Business Online Parameters](#)

[Lync Shared LC Example](#)

[Lync Shared Example](#)

[Lync Per Phone Example](#)

[Lync.cfg](#)

[Base Profile device.set](#)

[In-Band Provisioning](#)

## Skype Base Profile

The following table lists the Skype Base Profile parameters.

### Skype Base Profile

| Parameter                                   | Permitted Values                                                                                                                                                                                                                                                                            |
|---------------------------------------------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>call.DefaultTransferType</code>       | blind (default) - Pressing the Transfer soft key immediately transfers the call to another party.<br>consultative - Pressing the Transfer soft key puts the call on hold while placing a new call to the other party.                                                                       |
| <code>call.enableOnNotRegistered</code>     | 0 (default) - Calls are not permitted without registration.<br>1 - Users can make calls when the phone is not registered.<br>Note: Setting this parameter to 1 enables you to use VVX 1500 phones to make calls using the H.323 protocol even though an H.323 gatekeeper is not configured. |
| <code>callLists.collapseDuplicates</code>   | 0 (default) - Duplicate entries from multiple call lists are displayed separately.<br>1 - Duplicate entries from multiple call lists collapse into a single entry indicating the total number of entries.                                                                                   |
| <code>callLists.logConsultationCalls</code> | 1 (default) - All consultation calls are logged. (Calls made to a third party—while the original party is on hold—when settings up a conference call are called consultation calls.)<br>0 - Consultation calls are not logged.                                                              |

**Skype Base Profile**

| Parameter                                       | Permitted Values                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                 |
|-------------------------------------------------|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>device.lync.timeZone</code>               | 1 (default) - Lync Time Zone Control is enabled.<br>0 - Lync Time Zone Control is disabled.                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                      |
| <code>dialplan.1.applyToForward</code>          | 1 (default) - The dial plan applies to forwarded calls.<br>0 - The dial plan does not apply to forwarded calls.                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                  |
| <code>dialplan.1.conflictMatchHandling</code>   | 1 (default) - When the digits entered are matching more than one digit map, timeout is considered before dialing the digits.<br>0 - When the digits entered match a digit map the digits are dialed immediately even though there are conflicting digit maps.                                                                                                                                                                                                                                                                                                                                                                                                                                    |
| <code>dialplan.1.digitmap.timeOut</code>        | Specify a timeout in seconds for each segment of digit map. After you press a key, the phone will wait this many seconds before matching the digits to a dial plan and dialing the call.<br>4 seconds (default)<br>string of positive integers separated by  , for example<br>3   3   3   3   3<br>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.                                                                                                                                                                                                                    |
| <code>dialplan.1.impossibleMatchHandling</code> | This parameter applies to digits you enter in dial mode, the dial mode when you pick up the handset, headset, or press the New Call key. The phone is not in dial mode when you are hot dialing, dialing, or call list dialing.<br>0 (default) - The digits entered up to and including the point an impossible match occurred are sent to the server immediately.<br>1 - Give reorder tone.<br>2 - Allow users to accumulate digits and dispatch call manually with the Send soft key. If a call orbit number begins with # or *, you need to set this parameter to 2 to retrieve the call using off-hook dialing.                                                                              |
| <code>dialplan.1.lyncDigitmap.timeOut</code>    | Use this parameter for lines registered with Lync Server or Skype for Business Server. Specify a timeout in seconds for each segment of a digit map. After you press a key, the phone will wait this many seconds before matching the digits to a dial plan and dialing the call.<br>4 seconds (default)<br>0 to 99 seconds<br>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 of three seconds is used. Changes to the value of this parameter cause the phone to restart. |

**Skype Base Profile**

| Parameter                                         | Permitted Values                                                                                                                                                                                                                                                                                                                                                                                          |
|---------------------------------------------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>dialplan.applyToDirectoryDial</code>        | <p>1 (default) - The dial plan is applied to numbers dialed from the directory or speed dial, including auto-call numbers.</p> <p>0 - The dial plan is not applied to numbers dialed from the directory or speed dial list.</p>                                                                                                                                                                           |
| <code>dialplan.digitmap</code>                    | <p>The digit map used for the dial plan. The string is limited to 2560 bytes and 100 segments of 64 bytes; a comma is also allowed; a comma will turn dial tone back on; '+' is allowed as a valid digit; extension letter 'R' is used as defined above. This parameter enables the phone to automatically initiate calls to numbers that match a digit map pattern.</p> <p>NULL (default)<br/>string</p> |
| <code>dialplan.1.digitmap</code>                  | <p>x.T (default)<br/>string</p>                                                                                                                                                                                                                                                                                                                                                                           |
| <code>dialplan.userDial.timeOut</code>            | <p>Specifies the time in seconds that the phone waits before dialing a number you enter while the phone is on hook. You can apply <code>dialplan.userDial.timeOut</code> only when its value is lower than <code>up.IdleTimeOut</code>.</p> <p>4 seconds (default)<br/>0 to 99 seconds</p>                                                                                                                |
| <code>exchange.meeting.parseOption</code>         | <p>Determine from which field in the meeting invite the VMR or meeting number should be fetched.</p> <p>Location (default)<br/>All<br/>LocationAndSubject<br/>Description<br/>Enum</p>                                                                                                                                                                                                                    |
| <code>feature.audioVideoToggle.enabled</code>     | <p>0 (default) - The audio/video toggle feature is disabled.<br/>1 - The audio/video toggle feature is enabled.</p>                                                                                                                                                                                                                                                                                       |
| <code>feature.btoe.enabled</code>                 | <p>1 (default) - BToE is enabled on the phone and the phone can pair with a computer.<br/>0 - BToE is disabled on the phone and cannot pair with a computer.</p>                                                                                                                                                                                                                                          |
| <code>feature.exchange2007.interop.enabled</code> | <p>0 (default) - Interoperability with Microsoft Lync Server 2007 is disabled.<br/>1 - Interoperability with Microsoft Lync Server 2007 is enabled.</p>                                                                                                                                                                                                                                                   |

**Skype Base Profile**

| Parameter                                      | Permitted Values                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                  |
|------------------------------------------------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>feature.exchangeCalendar.enabled</code>  | <p>1 (default) - The Exchange calendar feature is enabled on the phone and users can view meeting notifications on the phone.</p> <p>0 - Exchange calendar is disabled.</p> <p>For the VVX 300, 310, 400, 410, 500, and 600 business media phones. You must enable this parameter if you enable <code>feature.exchangeCallLog.enabled</code>. If you disable <code>feature.exchangeCalendar.enabled</code>, also disable <code>feature.exchangeCallLog.enabled</code> to ensure call log functionality.</p>                                                                                                                                                                                                                                                                                       |
| <code>feature.exchangeCallLog.enabled</code>   | <p>0 (default) - The Exchange call log feature is disabled and the user call logs history cannot be retrieved from the Exchange server. You must also enable the parameter <code>feature.exchangeCalendar.enabled</code> to use the Exchange call log feature.</p> <p>1 - 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.</p> <p>For the VVX 300, 310, 400, 410, 500, and 600 business media phones. You must enable this parameter if you enable <code>feature.exchangeCalendar.enabled</code> to use the Exchange call log feature. If you disable <code>feature.exchangeCalendar.enabled</code>, also disable <code>feature.exchangeCallLog.enabled</code> to ensure call log functionality.</p> |
| <code>feature.EWSAutodiscover.enabled</code>   | <p>1 (default) - Exchange autodiscovery is enabled and the phone automatically discovers the Exchange server using the email address or SIP URI information.</p> <p>0 - Exchange autodiscovery is disabled on the phone and you must manually configure the Exchange server address.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                          |
| <code>feature.exchanges.enabled</code>         | <p>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.</p> <p>0 - The Exchange call log feature is disabled and the user call logs history cannot be retrieved from the Exchange server. You must also enable the parameter <code>feature.exchangeCallLog.enabled</code> to use the Exchange call log feature.</p>                                                                                                                                                                                                                                                                                                                                                                        |
| <code>feature.exchangeVoiceMail.enabled</code> | <p>1 - The Exchange voicemail feature is enabled and users can retrieve voicemails stored on the Exchange server from the phone.</p> <p>0 - The Exchange voicemail feature is disabled and users cannot retrieve voicemails from Exchange Server on the phone. You must also enable <code>feature.exchangeCalendar.enabled</code> to use the Exchange feature.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                |

**Skype Base Profile**

| Parameter                                                     | Permitted Values                                                                                                                                                                                                                                                             |
|---------------------------------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>feature.lync.abs.enabled</code>                         | 1 (default) - Users can search for contacts on the phone's global address book.<br>0 - The global address book search is disabled on the phone.                                                                                                                              |
| <code>feature.LyncCCCP.enabled</code>                         | 1 (default) - Use of CCCP is enabled.<br>0 - Use of CCCP is disabled.                                                                                                                                                                                                        |
| <code>feature.LyncCCCP2010AudioWorkaround.enabled</code>      | 1 (default) - When using Lync Server 2013 with an AVMCU server version Lync 2010 or earlier, the Hold/Resume soft key response is delayed 1.5 - 2 seconds.<br>0 - The soft keys respond without a delay.                                                                     |
| <code>feature.LyncCCCPDominantSpeakerDetection.enabled</code> | 0 (default) - In CCCP calls the phone does not display a handset icon beside the active speaker.<br>1 - In CCCP calls the phone displays a handset icon beside the active speaker.                                                                                           |
| <code>feature.lyncSafeTransfer.enabled</code>                 | 1 (default) - Enable the safe transfer feature and display of the Safe Transfer soft key.<br>0 - Disable the safe transfer feature and display of the Safe Transfer soft key.                                                                                                |
| <code>feature.messaging.enabled</code>                        | 1 (default)<br>0                                                                                                                                                                                                                                                             |
| <code>feature.moh.enabled</code>                              | 1 (default) - Enable the music on hold feature.<br>0 - Disable the music on hold feature.                                                                                                                                                                                    |
| <code>feature.presence.enabled</code>                         | 1 (default) - Enable the presence feature to manage your buddy list and display the status of your s.<br>0 - Disable the presence feature to manage your buddy list and display the status of your s.                                                                        |
| <code>feature.usb.device.content</code>                       | 1 (default) - Content sharing using the People+Content IP application on a computer connected by USB to RealPresence Trio is enabled.<br>0 - Content sharing using the People+Content IP application on a computer connected by USB to RealPresence Trio is disabled.        |
| <code>httpd.cfg.enabled</code>                                | Base Profile = Generic<br>1 (default) - The Web Configuration Utility is enabled.<br>0 - The Web Configuration Utility is disabled.<br><br>Base Profile = Skype<br>0 (default) - The Web Configuration Utility is disabled.<br>1 - The Web Configuration Utility is enabled. |

**Skype Base Profile**

| Parameter                                         | Permitted Values                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                     |
|---------------------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>httpd.cfg.secureTunnelRequired</code>       | <p>1 (default) - Access to the Web Configuration Utility is allowed only over a secure tunnel (HTTPS) and non-secure (HTTP) is not allowed.</p> <p>0 - Access to the Web Configuration Utility is allowed over both a secure tunnel (HTTPS) and non-secure (HTTP).</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                               |
| <code>httpd.enabled</code>                        | <p>Base Profile = Generic</p> <p>1 (default) - The web server is enabled.</p> <p>0 - The web server is disabled.</p> <p>Base Profile = Skype</p> <p>0 (default) - The web server is disabled.</p> <p>1 - The web server is enabled.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                              |
| <code>locInfo.source</code>                       | <p>This parameter specifies the phone's source location information which you can use to locate a phone in environments that have multiple sources of location information.</p> <p>MS_E911_LIS (default) - Location information sent from Skype for Business Server is used as the current location.</p> <p>LLDP - Location information sent from the network switch is used as the current location.</p> <p>CONFIG - You can manually configure location information as the current location.</p> <p>If location information is not available from a specified default or configured source, the fallback priority is as follows:<br/> Generic profile: LLDP &gt; CONFIG &gt; MS_E911_LIS<br/> Skype profile: MS_E911_LIS &gt; CONFIG &gt; LLDP</p> |
| <code>audioVideoToggle.callMode.persistent</code> | <p>0 (default) - The user setting to choose audio only or video calls by default is not retained after a phone restart or reboot.</p> <p>1 - The user setting to choose audio only or video calls as the default is retained after a phone restart or reboot.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                    |
| <code>feature.exchangeVoiceMail.enabled</code>    | <p>0 (default) - The Exchange voicemail feature is disabled and users cannot retrieve voicemails from Exchange Server on the phone. You must also enable <code>feature.exchangeCalendar.enabled</code> to use the Exchange feature.</p> <p>1 - The Exchange voicemail feature is enabled and users can retrieve voicemails stored on the Exchange server from the phone.</p>                                                                                                                                                                                                                                                                                                                                                                         |
| <code>phoneLock.Allow.AnswerOnLock</code>         | <p>0 (default) - Users can answer incoming calls without entering a password.</p> <p>1 - Users must enter a password to answer incoming calls.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                   |
| <code>prov.quickSetup.limitServerDetails</code>   | <p>0 (default) - The Zero Touch provisioning server level details is disabled.</p> <p>1 - The Zero Touch provisioning server level details is enabled.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                           |

**Skype Base Profile**

| Parameter                                               | Permitted Values                                                                                                                                                                                                                                         |
|---------------------------------------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>reg.1.applyServerDigitMapLocally</code>           | 0 (default) - Dialplan rules are processed by Skype for Business Server.<br>1 - Dialplan normalization rules are downloaded from the Skype for Business Server and processed on the phone.                                                               |
| <code>reg.1.auth.useLoginCredentials</code>             | 0 (default) - Login credentials, BToE Sign in, and Web Sign types are not available for authentication with the server.<br>1 - Login credentials, BToE Sign in, and Web Sign types are available for authentication with the server.                     |
| <code>reg.1.auth.usePinCredentials</code>               | 0 (default) - PIN authentication sign in method is not available for authentication on the server.<br>1 - PIN authentication sign in method is available for authentication on the server.                                                               |
| <code>reg.1.offerFullCodecListUponResume</code>         | 0 (default) - Only the audio codec negotiated during call establishment is sent when a held call is resumed.<br>1 - All supported codes are sent when a held call is resumed.                                                                            |
| <code>reg.1.server.1.registerRetry.baseTimeout</code>   | The base time period to wait before a registration retry. Used in conjunction with <code>reg.x.server.y.registerRetry.maxTimeout</code> to determine how long to wait. The algorithm is defined in RFC 5626.<br>10 seconds (default)<br>10 - 120 seconds |
| <code>reg.1.server.1.registerRetry.maxTimeout</code>    | Set the maximum period of time in seconds that the phone tries to register.<br>180 seconds (default)<br>60 to 1800 seconds                                                                                                                               |
| <code>reg.1.server.1.specialInterop</code>              | Identifies the SIP signaling as Skype for Business Server and enables Skype for Business features. This parameter supports Lync Server 2010 and 2013.<br>lync2010 (default)<br>lync2010, lcs2005, ocs2007r2                                              |
| <code>reg.1.server.1.transport</code>                   | The transport method the phone uses to communicate with the SIP server.<br>TLS (default)                                                                                                                                                                 |
| <code>reg.1.serverFeatureControl.signalingMethod</code> | Controls the method used to perform call forwarding requests to the server.<br>serviceMsForward (default)                                                                                                                                                |
| <code>reg.1.useteluriAsLineLabel</code>                 | 0 (default) - The line key label displays the Skype for Business TelURI, or line address.<br>1 - The line key label displays the Skype for Business account user name.                                                                                   |

**Skype Base Profile**

| Parameter                                   | Permitted Values                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                    |
|---------------------------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>roaming_buddies.reg</code>            | Set the line index number for the registered line you want to enable Presence and Instant Messaging. For Microsoft deployments, the value is always 1.<br>1 (default)<br>0 to 34                                                                                                                                                                                                                                                                                                                                                                                                    |
| <code>sec.srtp.holdWithNewKey</code>        | 0 (default) - A new key is not provided when holding a call.<br>1 - A new key is provided when holding a call.                                                                                                                                                                                                                                                                                                                                                                                                                                                                      |
| <code>sec.srtp.key.lifetime</code>          | The lifetime of the master key used for the cryptographic parameter in SDP. The value specified is the number of SRTP packets.<br>2 <sup>31</sup> (default)<br>0 - The master key lifetime is not set. If set to a valid value (at least 1024, or a power such as 210), the master key lifetime is set. When the lifetime is set, a re-invite with a new key will be sent when the number of SRTP packets sent for an outgoing call exceeds half the value of the master key lifetime.<br>Note: Setting this parameter to a non-zero value may affect the performance of the phone. |
| <code>sec.srtp.mki.enabled</code>           | The master key identifier (MKI) is an optional parameter for the cryptographic parameter in the SDP that uniquely identifies the SRTP stream within an SRTP session. MKI is expressed as a pair of decimal numbers in the form <code> mki:mki_length </code> , where <code>mki</code> is the MKI value and <code>mki_length</code> its length in bytes.<br>1 (default) - A 4-byte MKI parameter is sent within the SDP message of the SIP INVITE / 200 OK.<br>0 - The MKI parameter is not sent.                                                                                    |
| <code>sec.srtp.mki.length</code>            | The length of the master key identifier (MKI), in bytes. Microsoft Skype for Business offers 1-byte MKIs.<br>1 (default)<br>1 - 4                                                                                                                                                                                                                                                                                                                                                                                                                                                   |
| <code>sec.srtp.mki.startSessionAtOne</code> | 1 (default) - Use an MKI value of 1 at the start of an SDP session.<br>0 - The MKI value increments for each new crypto key.                                                                                                                                                                                                                                                                                                                                                                                                                                                        |
| <code>sec.srtp.resumeWithNewKey</code>      | 0 (default) - A key is not provided when resuming a call.<br>1 - A key is provided when resuming a call.                                                                                                                                                                                                                                                                                                                                                                                                                                                                            |
| <code>sec.TLS.profileSelection.SIP</code>   | Set the TLS application profile used to store the CA certificate.<br><code>ApplicationProfile1</code> (default)                                                                                                                                                                                                                                                                                                                                                                                                                                                                     |



**Skype Base Profile**

| Parameter                                            | Permitted Values                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                         |
|------------------------------------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>softkey.feature.forward</code>                 | <p>1 (default) - The Forward soft key displays.</p> <p>0 - The Forward soft key does not display.</p> <p>If you are registering phones with Lync Server or Skype for Business using UC Software 540A or earlier, you can configure <code>softkey.feature.forward=1</code> to display or remove the Forward soft key.</p> <p>If you are registering phones with Lync Server or Skype for Business using UC Software 540A or later, the Microsoft server automatically sends Call Forwarding functionality and the Forward soft key in-band to the phones. In this case:</p> <p>The parameter <code>softkey.feature.forward</code> is overridden by Microsoft server settings and has no effect.</p> <p>If <code>softkey.feature.forward=1</code>, the Microsoft server settings override <code>softkey.feature.forward</code>.</p> <p>If <code>softkey.feature.forward=0</code>, the phone receives the Forward soft key setting from the Microsoft server. If you want to disable the Forward soft key, you must disable it on the Microsoft server.</p> |
| <code>softkey.feature.MeetNow</code>                 | <p>1 (default) - The Meet Now soft key displays on the idle screen.</p> <p>0 - The Meet Now soft key does not display on the idle screen.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                            |
| <code>softkey.feature.simplifiedSignIn</code>        | <p>1 (default) - If 1 and the value for <code>voIpProt.server.x.specialInterop</code> is <code>lync2010</code>, the Sign In soft key is displayed.</p> <p>0 - The Sign In soft key is not displayed.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                 |
| <code>tcpIpApp.ice.mode</code>                       | Specify that ICE and TURN work with Skype for Business Server. MSOCS (default)                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                           |
| <code>tcpIpApp.keepalive.tcp.sip.tls.enable</code>   | <p>1 (default) - Enable keepalive packets and keep the TLS profile from timing out.</p> <p>0 - Disable keepalive packets.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                            |
| <code>tcpIpApp.port.rtp.videoPortRange.enable</code> | <p>1 (default) - Video ports are chosen from the range specified by <code>tcpIpApp.port.rtp.videoPortRangeStart</code> and <code>tcpIpApp.port.rtp.videoPortRangeEnd</code>.</p> <p>0 - Video ports are also chosen within the range specified by <code>tcpIpApp.port.rtp.mediaPortRangeStart</code> and <code>tcpIpApp.port.rtp.mediaPortRangeEnd</code>.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                           |
| <code>tcpIpApp.sntp.address</code>                   | Specify the address of an SNTP server.<br>NULL (default)                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                 |
| <code>tone.dtmf.rfc2833Payload</code>                | 101 (default)                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                            |

**Skype Base Profile**

| Parameter                                                | Permitted Values                                                                                                                                                                                                                                                              |
|----------------------------------------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>up.numOfDisplayColumns</code>                      | Applies only to VVX 500 and 600 business media phones. Set the number of columns to display on the phone idle screen. Each column can display a maximum of four lines.<br>2 (default)<br>VVX 500=1-3<br>VVX 600=1-4                                                           |
| <code>up.oneTouchDirectory</code>                        | 1 (default) - The Skype for Business Search icon displays on the Home screen.<br>0 - The Skype for Business Search icon does not display on the Home screen.                                                                                                                  |
| <code>up.oneTouchVoiceMail</code>                        | 1 (default) - The phone dials voicemail services directly (if available on the call server) without displaying the voicemail summary.<br>0 - The phone displays a summary page with message counts. The user must press the Connect soft key to dial the voicemail server.    |
| <code>video.autoStartVideoTx</code>                      | 1 (default) - Automatically begin video to the far side when you start a call.<br>0 - Video to the far side does not begin.                                                                                                                                                   |
| <code>video.callMode.default</code>                      | Audio (default) -<br>Video -                                                                                                                                                                                                                                                  |
| <code>video.enable</code>                                | 0 (default) - Video is not enabled and all calls—both sent and received—are audio-only.<br>1 - Video is sent in outgoing calls and received in incoming calls if the other device supports video. Note: Enabling video on the VVX 500 and 600 disables G.722.1C codec.        |
| <code>video.iFrame.delay</code>                          | Configure a time delay in seconds from the start of video until the I-frame is sent. When nonzero, an extra I-frame is transmitted after video starts. Use a value of 2 seconds if you are using this parameter in a Skype for Business environment.<br>2 (default)<br>0 - 10 |
| <code>video.iFrame.onPacketLoss</code>                   | 1 (default) - An I-frame is transmitted to the far end when a received RTCP report indicates that video RTP packet loss has occurred.<br>0 - No An I-frame is transmitted to the far end when a received RTCP report indicates that video RTP packet loss has occurred.       |
| <code>voice.audioProfile.G7221.24kbps.payloadType</code> | The payload type for the G.722.1 24kbps codec.<br>112 (default)<br>0 - 127                                                                                                                                                                                                    |

**Skype Base Profile**

| Parameter                                                         | Permitted Values                                                                                                                                                                                                                                                                                                     |
|-------------------------------------------------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>voice.codecPref.G7221.24kbps</code>                         | The priority of the G.722.1 24kbps codec. If 0 or Null, the codec is disabled. A value of 1 is the highest priority.<br>5 (default)<br>0 - 27                                                                                                                                                                        |
| <code>voice.codecPref.G7221.32kbps</code>                         | The priority of the G.722.1 32kbps codec. If 0 or Null, the codec is disabled. A value of 1 is the highest priority.<br>0 (default)<br>0 - 27                                                                                                                                                                        |
| <code>voIpProt.SIP.allowTransferOnProceeding</code>               | 0 (default) A transfer is not allowed during the proceeding state of a consultation call.<br>1 - A transfer can be completed during the proceeding state of a consultation call.                                                                                                                                     |
| <code>voIpProt.SIP.failoverOn503Response</code>                   | 0 (default)<br>1                                                                                                                                                                                                                                                                                                     |
| <code>voIpProt.SIP.header.diversion.enable</code>                 | 0 (default) - The diversion header is not displayed.<br>1 - The diversion header is displayed if received.                                                                                                                                                                                                           |
| <code>voIpProt.SIP.IM.autoAnswerDelay</code>                      | The time interval in seconds from receipt of the instant message invitation to accepting the invitation automatically.<br>40 (default)<br>0 - 40                                                                                                                                                                     |
| <code>voIpProt.SIP.mtls.enable</code>                             | 0 (default) - Disable Mutual TLS.<br>1 - Disable Mutual TLS.                                                                                                                                                                                                                                                         |
| <code>voIpProt.SIP.serverFeatureControl.cf</code>                 | This parameter overrides <code>voIpProt.SIP.serverFeatureControl.dnd</code> .<br>1 (default) - Server-based DND is enabled and the call server has control of DND.<br>0 - Server-based do-not-disturb (DND) is not enabled.                                                                                          |
| <code>voIpProt.SIP.serverFeatureControl.dnd</code>                | This parameter overrides <code>voIpProt.SIP.serverFeatureControl.dnd</code> .<br>1 (default) - Server-based DND is enabled and the call server has control of DND.<br>0 - Server-based do-not-disturb (DND) is not enabled.                                                                                          |
| <code>voIpProt.SIP.serverFeatureControl.localProcessing.cf</code> | This parameter overrides <code>voIpProt.SIP.serverFeatureControl.localProcessing.cf</code> .<br>0 (default) - If 0 and <code>reg.x.serverFeatureControl.cf</code> is set to 1, the phone does not perform local Call Forward behavior.<br>1 - If 1 phone performs local Call Forward behavior on all calls received. |

**Skype Base Profile**

| Parameter                                             | Permitted Values                                                                                                                                                                                                                                                                         |
|-------------------------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| voIpProt.SIP.serverFeatureControl.localProcessing.dnd | This parameter overrides voIpProt.SIP.serverFeatureControl.localProcessing.cf.<br>0 (default) - If 0 and reg.x.serverFeatureControl.cf is set to 1, the phone does not perform local Call Forward behavior.<br>1 - The phone performs local Call Forward behavior on all calls received. |
| voIpProt.SIP.useInactiveHold                          | 0 (default)                                                                                                                                                                                                                                                                              |
| voIpProt.SIP.useSendonlyHold                          | 1 (default) - The phone sends the "a=sendonly" in the 'Hold' invite request.<br>0 - The phone does not send the "a=sendonly" in the 'Hold' invite request.                                                                                                                               |

**Skype for Business Online Parameters**

The following table lists the Polycom UC Software parameter names that correspond to the Skype for Business Online parameters you can configure.

**UC Software Parameters Supported with Skype for Business Online**

| UC Software Parameter Name                                                     | Skype for Business Online Parameter Name |
|--------------------------------------------------------------------------------|------------------------------------------|
| btoe.pairingMode                                                               | BetterTogetherOverEthernetPairingMode    |
| device.prov.lyncDeviceUpdateEnabled<br>device.prov.lyncDeviceUpdateEnabled.set | EnableDeviceUpdate                       |
| device.prov.password<br>device.prov.password.set                               | LocalProvisioningServerpassword          |
| device.prov.serverName<br>device.prov.serverName.set                           | LocalProvisioningServerAddress           |
| device.prov.serverType<br>device.prov.serverType.set                           | LocalProvisioningServerType              |
| device.prov.user<br>device.prov.user.set                                       | LocalProvisioningServerUser              |
| dialplan.userDial.timeOut                                                      | UserDialTimeoutMS                        |
| feature.btoe.enabled                                                           | EnableBetterTogetherOverEthernet         |
| feature.exchangeCalendar.enabled                                               | EnableExchangeCalendar                   |

**UC Software Parameters Supported with Skype for Business Online**

|                                                                                                                                                          |                                     |
|----------------------------------------------------------------------------------------------------------------------------------------------------------|-------------------------------------|
| One of:<br>voice.volume.persist.headset<br>voice.volume.persist.bluetooth.headset<br>voice.volume.persist.usbHeadset<br>voice.volume<br>.persist.handset | VoiceVolumePersistMode              |
| powerSaving.enable                                                                                                                                       | EnablePowerSaveMode                 |
| powerSaving.idleTimeout.offHours                                                                                                                         | PowerSavePostOfficeHoursTimeoutMS   |
| powerSaving.idleTimeout.officeHours                                                                                                                      | PowerSaveDuringOfficeHoursTimeoutMS |
| up.oneTouchVoiceMail                                                                                                                                     | EnableOneTouchVoicemail             |

**Lync Shared LC Example**

The next table lists parameters and values in the `lyncSharedLCExample.cfg` template.

**Lync Shared LC Example Parameters**

| Parameter                                         | Permitted Values                                                                                                                                                                                                                                                                                                              |
|---------------------------------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>audioVideoToggle.callMode.persistent</code> | 0 (default) - The user setting to choose audio only or video calls by default is not retained after a phone restart or reboot.<br>1 - The user setting to choose audio only or video calls as the default is retained after a phone restart or reboot.                                                                        |
| <code>btoe.PairingMode</code>                     | Auto (default) - The phone pairs with the computer automatically when the computer is connected to the phone's PC port and the BToE connector application is running.<br>Manual - The phone generates a six-digit pair code to enter to the BToE connector application when the computer is connected to the phone's PC port. |
| <code>call.enableOnNotRegistered</code>           | 0 (default) - Calls are not permitted without registration.<br>1 - Users can make calls when the phone is not registered.<br>Note: Setting this parameter to 1 enables you to use VVX 1500 phones to make calls using the H.323 protocol even though an H.323 gatekeeper is not configured.                                   |
| <code>callLists.logConsultationCalls</code>       | 1 (default) - All consultation calls are logged. (Consultation calls are calls made to a third party while the original party is on hold.)<br>0 - Consultation calls are not logged.                                                                                                                                          |
| <code>device.set</code>                           | A global parameter that allows you to install software and change device parameters.<br>1 (default)<br>0 or 1                                                                                                                                                                                                                 |

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**Lync Shared LC Example Parameters**

|                                                                      |                                                                                                                                                                                                                                                                                                                 |
|----------------------------------------------------------------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>device.prov.lyncDeviceUpdateEnabled</code>                     | <p>1 (default) - The automatic device update is enabled on the phone and the phone receives software updates from the server.</p> <p>0 - 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.</p>   |
| <code>dialplan.applyToDirectoryDial</code>                           | <p>1 (default) - The dial plan is applied to numbers dialed from the directory or speed dial, including auto-call numbers.</p> <p>0 - The dial plan is not applied to numbers dialed from the directory or speed dial list.</p>                                                                                 |
| <code>dialplan.1.applyToForward</code>                               | <p>1 (default) - The dial plan applies to forwarded calls.</p> <p>0 - The dial plan does not apply to forwarded calls.</p>                                                                                                                                                                                      |
| <code>feature.audioVideoToggle.enabled</code>                        | <p>0 (default) - The audio/video toggle feature is disabled.</p> <p>1 - The audio/video toggle feature is enabled.</p>                                                                                                                                                                                          |
| <code>feature.btoe.enabled</code>                                    | <p>1 (default) - The Better Together over Ethernet feature is enabled.</p> <p>0 - The Better Together over Ethernet feature is disabled.</p>                                                                                                                                                                    |
| <code>feature.cccp.enabled</code>                                    | <p>1 (default) - Enable CCCP.</p> <p>0 - Disable CCCP.</p>                                                                                                                                                                                                                                                      |
| <code>feature.lyncbtoe.autosignin.signoff.enabled<sup>1</sup></code> | <p>0 (default) - When the BToE application is unpaired, the credentials cached on the phone when the user signs in from the BToE application are erased.</p> <p>1 - When the BToE application is unpaired, the credentials cached on the phone when the user signs in from the BToE application are stored.</p> |
| <code>feature.presence.enabled</code>                                | <p>1 (default) - Enable the presence feature to manage your buddy list and display the status of your s.</p> <p>0 - Disable presence.</p>                                                                                                                                                                       |
| <code>httpd.enabled</code>                                           | <p>Base Profile = Generic</p> <p>1 (default) - The web server is enabled.</p> <p>0 - The web server is disabled.</p><br><p>Base Profile = Skype</p> <p>0 (default) - The web server is disabled.</p> <p>1 - The web server is enabled.</p>                                                                      |
| <code>httpd.cfg.enabled</code>                                       | <p>Base Profile = Generic</p> <p>1 (default) - The Web Configuration Utility is enabled.</p> <p>0 - The Web Configuration Utility is disabled.</p><br><p>Base Profile = Skype</p> <p>0 (default) - The Web Configuration Utility is disabled.</p> <p>1 - The Web Configuration Utility is enabled.</p>          |

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**Lync Shared LC Example Parameters**

|                                                             |                                                                                                                                                                                                                                                                                                                                       |
|-------------------------------------------------------------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>httpd.cfg.secureTunnelRequired</code>                 | <p>1 (default) - Access to the Web Configuration Utility is allowed only over a secure tunnel (HTTPS) and non-secure (HTTP) is not allowed.</p> <p>0 - Access to the Web Configuration Utility is allowed over both a secure tunnel (HTTPS) and non-secure (HTTP).</p>                                                                |
| <code>reg.1.applyServerDigitMapLocally</code>               | <p>1 (default) - Dialplan normalization rules are downloaded from the Skype for Business Server and processed on the phone.</p> <p>0 - Dialplan rules are processed by Skype for Business Server.</p>                                                                                                                                 |
| <code>reg.1.auth.useLoginCredentials</code>                 | <p>1 (default) - Enable the Sign In screen on the phone.</p> <p>0 - Disable the Sign In screen on the phone.</p>                                                                                                                                                                                                                      |
| <code>reg.1.auth.usePinCredentials</code>                   | <p>0 (default) - Disable PIN authentication sign in method.</p> <p>1 - Enable PIN Authentication method.</p>                                                                                                                                                                                                                          |
| <code>reg.1.serverFeatureControl.cf</code>                  | <p>1 (default) - Server based call forwarding is enabled for this line.</p> <p>0 - Server-based call forwarding is disabled for this line.</p>                                                                                                                                                                                        |
| <code>reg.1.serverFeatureControl.localProcessing.cf</code>  | <p>0 (default) - If 0 and <code>reg.1.serverFeatureControl.cf</code> is set to 1, the phone does not perform local Call Forward behavior.</p> <p>1 - The phone performs local Call Forward behavior on all calls received.</p>                                                                                                        |
| <code>reg.1.serverFeatureControl.dnd</code>                 | <p>This parameter overrides <code>voIpProt.SIP.serverFeatureControl.dnd</code>.</p> <p>1 (default) - Server-based DND is enabled and the call server has control of DND.</p> <p>0 - Server-based do-not-disturb (DND) is not enabled.</p>                                                                                             |
| <code>reg.1.serverFeatureControl.localProcessing.dnd</code> | <p>This parameter overrides <code>voIpProt.SIP.serverFeatureControl.localProcessing.cf</code>.</p> <p>0 (default) - If 0 and <code>reg.x.serverFeatureControl.cf</code> is set to 1, the phone does not perform local Call Forward behavior.</p> <p>1 - The phone will perform local Call Forward behavior on all calls received.</p> |
| <code>reg.1.serverFeatureControl.signalingMethod</code>     | <p>Controls the method used to perform call forwarding requests to the server.</p> <p><code>serviceMsForward</code></p>                                                                                                                                                                                                               |
| <code>reg.1.server.1.registerRetry.baseTimeout</code>       | <p>The base time period in seconds to wait before a registration retry. Used in conjunction with <code>reg.x.server.y.registerRetry.maxTimeout</code> to determine how long to wait. The algorithm is defined in RFC 5626.</p> <p>10 seconds (default)</p> <p>10 to 120 seconds</p>                                                   |

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**Lync Shared LC Example Parameters**


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|                                                      |                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                              |
|------------------------------------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>reg.1.server.1.registerRetry.maxTimeout</code> | <p>Set the maximum period of time in seconds that the phone tries to register.</p> <p>180 seconds (default)</p> <p>60 to 1800 seconds</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                    |
| <code>reg.1.server.1.specialInterop</code>           | <p>Identifies the SIP signaling as Skype for Business Server and enables Skype for Business features. This parameter supports Lync Server 2010 and 2013.</p> <p>NULL (default)</p> <p>lync2010, lcs2005, ocs2007r2</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                       |
| <code>reg.1.server.1.transport</code>                | <p>The transport method the phone uses to communicate with the SIP server.</p> <p>TLS (default)</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                          |
| <code>reg.1.useteluriAsLineLabel</code>              | <p>0 (default) - The line key label displays the Skype for Business TelURI, or line address.</p> <p>1 - The line key label displays the Skype for Business account user name.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                            |
| <code>roaming_buddies.reg</code>                     | <p>Set the line index number for the registered line you want to enable Presence and Instant Messaging.</p> <p>1 (default)</p> <p>0 to 34</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                |
| <code>sec.srtp.holdWithNewKey</code>                 | <p>0 (default) - A new key is not provided when holding a call.</p> <p>1 - A new key is provided when holding a call.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                    |
| <code>sec.srtp.key.lifetime</code>                   | <p>The lifetime of the master key used for the cryptographic parameter in SDP. The value specified is the number of SRTP packets.</p> <p><math>2^{31}</math> (default)</p> <p>0 - The master key lifetime is not set.</p> <p>positive integer minimum 1024 or power of 2 notation</p> <p>If set to a valid value (at least 1024, or a power such as <math>2^{10}</math>), the master key lifetime is set. When the lifetime is set, a re-invite with a new key will be sent when the number of SRTP packets sent for an outgoing call exceeds half the value of the master key lifetime.</p> <p>Note: Setting this parameter to a nonzero value may affect the performance of the phone.</p> |
| <code>sec.srtp.mki.enabled</code>                    | <p>The master key identifier (MKI) is an optional parameter for the cryptographic parameter in the SDP that uniquely identifies the SRTP stream within an SRTP session. MKI is expressed as a pair of decimal numbers in the form <code> mki:mki_length </code>, where <code>mki</code> is the MKI value and <code>mki_length</code> its length in bytes.</p> <p>1 (default) - A 4-byte MKI parameter is sent within the SDP message of the SIP INVITE / 200 OK.</p> <p>0 - The MKI parameter is not sent.</p>                                                                                                                                                                               |



**Lync Shared LC Example Parameters**

|                                                          |                                                                                                                                                                                                                                                                               |
|----------------------------------------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>sec.srtp.mki.length</code>                         | The length of the master key identifier (MKI), in bytes. Microsoft Skype for Business offers 1-byte MKIs.<br>1 (default)<br>1 to 4                                                                                                                                            |
| <code>sec.srtp.mki.startSessionAtOne</code>              | 1 (default) - Use an MKI value of 1 at the start of an SDP session.<br>0 - The MKI value increments for each new crypto key.                                                                                                                                                  |
| <code>sec.srtp.resumeWithNewKey</code>                   | 0 (default) - A key is not provided when resuming a call.<br>1 - A key is provided when resuming a call.                                                                                                                                                                      |
| <code>sec.TLS.customCaCert.1</code>                      | The custom certificate for TLS Application Profile.<br>NULL (default)                                                                                                                                                                                                         |
| <code>sec.TLS.profileSelection.SIP</code>                | Enter the TLS platform profile or TLS application profile.<br><code>ApplicationProfile1</code> (default)                                                                                                                                                                      |
| <code>softkey.feature.simplifiedSignIn</code>            | 1 (default) - If 1 and <code>voIpProt.server.x.specialInterop</code> is <code>lync2010</code> , the Sign In soft key is displayed.<br>0 - The Sign In soft key is not displayed.                                                                                              |
| <code>tcpIpApp.ice.mode</code>                           | Specifies that ICE and TURN work with Skype for Business Server.<br>MSOCS                                                                                                                                                                                                     |
| <code>tcpIpApp.keepalive.tcp.sip.tls.enable</code>       | 1 (default) - Enable keepalive packets and keep the TLS profile from timing out.<br>0 - Disable keepalive packets.                                                                                                                                                            |
| <code>video.iFrame.delay</code>                          | Configure a time delay in seconds from the start of video until the I-frame is sent. When nonzero, an extra I-frame is transmitted after video starts. Use a value of 2 seconds if you are using this parameter in a Skype for Business environment.<br>2 (default)<br>0 - 10 |
| <code>video.iFrame.onPacketLoss</code>                   | 1 (default) - An I-frame is transmitted to the far end when a received RTCP report indicates that video RTP packet loss has occurred.<br>0 - No An I-frame is transmitted to the far end when a received RTCP report indicates that video RTP packet loss has occurred.       |
| <code>voice.audioProfile.G7221.24kbps.payloadType</code> | The payload type for the G.722.1 24kbps codec.<br>112 (default)<br>0 - 127                                                                                                                                                                                                    |
| <code>voice.codecPref.G7221.24kbps</code>                | The priority of the G.722.1 24kbps codec. If 0 or Null, the codec is disabled. A value of 1 is the highest priority.<br>5 (default)<br>0 - 27                                                                                                                                 |

**Lync Shared LC Example Parameters**

|                                                     |                                                                                                                                                                                  |
|-----------------------------------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>voice.codecPref.G7221.32kbps</code>           | The priority of the G.722.1 32kbps codec. If 0 or Null, the codec is disabled. A value of 1 is the highest priority.<br>0 (default)<br>0 - 27                                    |
| <code>voIpProt.SIP.allowTransferOnProceeding</code> | 0 (default) A transfer is not allowed during the proceeding state of a consultation call.<br>1 - A transfer can be completed during the proceeding state of a consultation call. |
| <code>voIpProt.SIP.IM.autoAnswerDelay</code>        | The time interval in seconds from receipt of the instant message invitation to accepting the invitation automatically.<br>40 (default)<br>0 - 40                                 |
| <code>voIpProt.SIP.header.diversion.enable</code>   | 0 (default) - The diversion header is not displayed.<br>1 - The diversion header is displayed if received.                                                                       |
| <code>voIpProt.SIP.mtls.enable</code>               | 0 (default) - Disable Mutual TLS.<br>1 - Enable Mutual TLS.                                                                                                                      |

<sup>1</sup> Change causes phone to restart or reboot.

## Lync Shared Example

The next table describes parameters and values in the `lyncSharedExample.cfg` template.

**Lync Shared Example**

| Parameter                                         | Permitted Values                                                                                                                                                                                                                                                                                                              |
|---------------------------------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>audioVideoToggle.callMode.persistent</code> | 0 (default) - The user setting to choose audio only or video calls by default is not retained after a phone restart or reboot.<br>1 - The user setting to choose audio only or video calls as the default is retained after a phone restart or reboot.                                                                        |
| <code>btoe.PairingMode</code>                     | Auto (default) - The phone pairs with the computer automatically when the computer is connected to the phone's PC port and the BToE connector application is running.<br>Manual - The phone generates a six-digit pair code to enter to the BToE connector application when the computer is connected to the phone's PC port. |
| <code>call.enableOnNotRegistered</code>           | 0 (default) - Calls are not permitted without registration.<br>1 - Users can make calls when the phone is not registered.<br>Note: Setting this parameter to 1 enables you to use VVX 1500 phones to make calls using the H.323 protocol even though an H.323 gatekeeper is not configured.                                   |

**Lync Shared Example**

|                                                             |                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                              |
|-------------------------------------------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>callLists.logConsultationCalls</code>                 | <p>1 (default) - All consultation calls are logged. (Calls made to a third party—while the original party is on hold—when settings up a conference call are called consultation calls.)</p> <p>0 - Consultation calls are not logged.</p>                                                                                                                                                                                                                                                                                                                                                                                                    |
| <code>device.set</code>                                     | <p>A global parameter that allows you to install software and change device parameters.</p> <p>1 (default)</p> <p>0 or 1</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                 |
| <code>device prov.lyncDeviceUpdateEnabled.set</code>        | <p>0 (default) - Disable automatic updates for all devices.</p> <p>1 - Enable automatic device update for all devices and use of <code>device prov.lyncDeviceUpdateEnabled</code>.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                       |
| <code>device prov.lyncDeviceUpdateEnabled</code>            | <p>1 (default) - The automatic device update is enabled on the phone and the phone receives software updates from the server.</p> <p>0 - The automatic device update is disabled and the phone does not receive software updates from the server.</p> <p>Changing the value of this parameter reboots the phone.</p>                                                                                                                                                                                                                                                                                                                         |
| <code>dialplan.applyToDirectoryDial</code>                  | <p>1 (default) - The dial plan is applied to numbers dialed from the directory or speed dial, including auto-call numbers.</p> <p>0 - The dial plan is not applied to numbers dialed from the directory or speed dial list.</p>                                                                                                                                                                                                                                                                                                                                                                                                              |
| <code>dialplan.digitmap</code>                              | <p>NULL (default)</p> <p>The digit map used for the dial plan. The string is limited to 2560 bytes and 100 segments of 64 bytes; a comma is also allowed; a comma will turn dial tone back on; '+' is allowed as a valid digit; extension letter 'R' is used as defined above. This parameter enables the phone to automatically initiate calls to numbers that match a digit map pattern.</p>                                                                                                                                                                                                                                               |
| <code>dialplan.1.impossibleMatchHandling<sup>1</sup></code> | <p>This parameter applies to digits you enter in dial mode, the dial mode when you pick up the handset, headset, or press the New Call key. The phone is not in dial mode when you are hot dialing, dialing, or call list dialing.</p> <p>0 (default) - The digits entered up to and including the point an impossible match occurred are sent to the server immediately.</p> <p>1 - Give reorder tone.</p> <p>2 - Allow user to accumulate digits and dispatch call manually with the Send soft key.</p> <p>If a call orbit number begins with # or *, you need to set this parameter to 2 to retrieve the call using off-hook dialing.</p> |
| <code>feature.audioVideoToggle.enabled</code>               | <p>0 (default) - The audio/video toggle feature is disabled.</p> <p>1 - The audio/video toggle feature is enabled.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                       |
| <code>feature.btoe.enabled</code>                           | <p>1 (default) - BToE is enabled on the phone and the phone can pair with a computer.</p> <p>0 - BToE is disabled on the phone and cannot pair with a computer.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                          |

**Lync Shared Example**

|                                                                      |                                                                                                                                                                                                                                                                                              |
|----------------------------------------------------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>feature.cccp.enabled</code>                                    | <p>1 (default) - Enable CCCP.<br/>0 - Disable CCCP.</p>                                                                                                                                                                                                                                      |
| <code>feature.lyncbtoe.autosignin.signoff.enabled<sup>1</sup></code> | <p>0 (default) - When the BToE app is unpaired, the credentials cached on the phone when the user signs in from the BToE application are erased.<br/>1 - When the BToE app is unpaired, the credentials cached on the phone when the user signs in from the BToE application are stored.</p> |
| <code>feature.presence.enabled</code>                                | <p>1 (default) - Enable the presence feature to manage your buddy list and display the status of your s.<br/>0 - Disable the presence feature to manage your buddy list and display the status of your s.</p>                                                                                |
| <code>httpd.enabled</code>                                           | <p>Base Profile = Generic<br/>1 (default) - The web server is enabled.<br/>0 - The web server is disabled.</p> <p>Base Profile = Skype<br/>0 (default) - The web server is disabled.<br/>1 - The web server is enabled.</p>                                                                  |
| <code>httpd.cfg.enabled</code>                                       | <p>Base Profile = Generic<br/>1 (default) - The Web Configuration Utility is enabled.<br/>0 - The Web Configuration Utility is disabled.</p> <p>Base Profile = Skype<br/>0 (default) - The Web Configuration Utility is disabled.<br/>1 - The Web Configuration Utility is enabled.</p>      |
| <code>httpd.cfg.secureTunnelRequired</code>                          | <p>1 (default) - Access to the Web Configuration Utility is allowed only over a secure tunnel (HTTPS) and non-secure (HTTP) is not allowed.<br/>0 - Access to the Web Configuration Utility is allowed over both a secure tunnel (HTTPS) and non-secure (HTTP).</p>                          |
| <code>reg.1.offerFullCodecListUponResume</code>                      | <p>0 (default) - Only the audio codec negotiated during call establishment is sent when a held call is resumed.<br/>1 - All supported codes are sent when a held call is resumed.</p>                                                                                                        |
| <code>sec.srtplib.holdWithNewKey</code>                              | <p>0 (default) - A new key is not provided when holding a call.<br/>1 - A new key is provided when holding a call.</p>                                                                                                                                                                       |

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**Lync Shared Example**


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|                                                    |                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                            |
|----------------------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>sec.srtp.key.lifetime</code>                 | <p>The lifetime of the master key used for the cryptographic parameter in SDP. The value specified is the number of SRTP packets.</p> <p><math>2^{31}</math> (default)</p> <p>0 - The master key lifetime is not set. If set to a valid value (at least 1024, or a power such as 210), the master key lifetime is set. When the lifetime is set, a re-invite with a new key will be sent when the number or SRTP packets sent for an outgoing call exceeds half the value of the master key lifetime.</p> <p>Note: Setting this parameter to a non-zero value may affect the performance of the phone.</p> |
| <code>sec.srtp.mki.enabled</code>                  | <p>The master key identifier (MKI) is an optional parameter for the cryptographic parameter in the SDP that uniquely identifies the SRTP stream within an SRTP session. MKI is expressed as a pair of decimal numbers in the form <code> mki:mki_length </code>, where <code>mki</code> is the MKI value and <code>mki_length</code> its length in bytes.</p> <p>1 (default) - A 4-byte MKI parameter is sent within the SDP message of the SIP INVITE / 200 OK.</p> <p>0 - The MKI parameter is not sent.</p>                                                                                             |
| <code>sec.srtp.mki.length</code>                   | <p>The length of the master key identifier (MKI), in bytes. Skype for Business offers 1-byte MKIs.</p> <p>1 (default)</p> <p>1 - 4.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                    |
| <code>sec.srtp.mki.startSessionAtOne</code>        | <p>1 (default) - Use an MKI value of 1 at the start of an SDP session. 0 - The MKI value increments for each new crypto key.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                           |
| <code>sec.srtp.resumeWithNewKey</code>             | <p>0 (default) - A key is not provided when resuming a call.</p> <p>1 - A key is provided when resuming a call.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                        |
| <code>sec.TLS.customCaCert.1</code>                | <p>The custom certificate for TLS Application Profile.</p> <p>NULL (default)</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                           |
| <code>sec.TLS.profileSelection.SIP</code>          | <p>Set the TLS application profile used to store the CA certificate.</p> <p><code>ApplicationProfile1</code> (default)</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                 |
| <code>tcpIpApp.ice.mode</code>                     | <p>Specify that ICE and TURN work with Skype for Business Server.</p> <p>MSOCS (default)</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                               |
| <code>tcpIpApp.keepalive.tcp.sip.tls.enable</code> | <p>1 (default) - Enable keepalive packets and keep the TLS profile from timing out.</p> <p>0 - Disable keepalive packets.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                              |
| <code>video.iFrame.delay</code>                    | <p>Configure a time delay in seconds from the start of video until the I-frame is sent. When nonzero, an extra I-frame is transmitted after video starts. Use a value of 2 seconds if you are using this parameter in a Skype for Business environment.</p> <p>2 (default)</p> <p>0 - 10</p>                                                                                                                                                                                                                                                                                                               |

**Lync Shared Example**

|                                                          |                                                                                                                                                                                                                                                                         |
|----------------------------------------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>video.iFrame.onPacketLoss</code>                   | 1 (default) - An I-frame is transmitted to the far end when a received RTCP report indicates that video RTP packet loss has occurred.<br>0 - No An I-frame is transmitted to the far end when a received RTCP report indicates that video RTP packet loss has occurred. |
| <code>voice.audioProfile.G7221.24kbps.payloadType</code> | The payload type for the G.722.1 24kbps codec.<br>112 (default)<br>0 - 127                                                                                                                                                                                              |
| <code>voice.codecPref.G7221.24kbps</code>                | The priority of the G.722.1 24kbps codec. If 0 or Null, the codec is disabled. A value of 1 is the highest priority.<br>5 (default)<br>0 - 27                                                                                                                           |
| <code>voice.codecPref.G7221.32kbps</code>                | The priority of the G.722.1 32kbps codec. If 0 or Null, the codec is disabled. A value of 1 is the highest priority.<br>0 (default)<br>0 - 27                                                                                                                           |
| <code>voIpProt.SIP.allowTransferOnProceeding</code>      | 0 (default) A transfer is not allowed during the proceeding state of a consultation call.<br>1 - A transfer can be completed during the proceeding state of a consultation call.                                                                                        |
| <code>voIpProt.SIP.IM.autoAnswerDelay</code>             | The time interval in seconds from receipt of the instant message invitation to accepting the invitation automatically.<br>40 (default)<br>0 - 40                                                                                                                        |
| <code>voIpProt.SIP.header.diversion.enable</code>        | 0 (default) - The diversion header is not displayed.<br>1 - The diversion header is displayed if received.                                                                                                                                                              |
| <code>voIpProt.SIP.mtls.enable</code>                    | 0 (default) - Diable Mutual TLS.<br>1 - Disable Mutual TLS.                                                                                                                                                                                                             |

<sup>1</sup> Change causes phone to restart or reboot.

**Lync Per Phone Example**

The following table lists parameters in the template file `lyncPerPhoneExample.cfg`.

**Lync Per Phone Example**

| Parameter                               | Permitted Values                                                                                                                                                                                                                                                                                   |
|-----------------------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| device.set                              | A global parameter that you enable to make changes to <device> parameters. Use this parameter to change only <device> parameter values. Once you have made your changes, remove this parameter from the configuration file.<br>0 (default)<br>0 or 1                                               |
| device.prov.lyncDeviceUpdateEnabled.set | 0 (default) - Disable automatic updates for all devices.<br>1 - Enable automatic device update for all devices and use of device.prov.lyncDeviceUpdateEnabled.                                                                                                                                     |
| device.prov.lyncDeviceUpdateEnabled     | 1 (default) - The automatic device update is enabled on the phone and the phone receives software updates from the server.<br>0 - 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. |
| dialplan.1.applyToForward               | 1 (default) - The dial plan applies to forwarded calls.<br>0 - The dial plan does not apply to forwarded calls.                                                                                                                                                                                    |
| httpd.enabled                           | Base Profile = Generic<br>1 (default) - The web server is enabled.<br>0 - The web server is disabled.<br><br>Base Profile = Skype<br>0 (default) - The web server is disabled.<br>1 - The web server is enabled.                                                                                   |
| httpd.cfg.enabled                       | Base Profile = Generic<br>1 (default) - The Web Configuration Utility is enabled.<br>0 - The Web Configuration Utility is disabled.<br><br>Base Profile = Skype<br>0 (default) - The Web Configuration Utility is disabled.<br>1 - The Web Configuration Utility is enabled.                       |
| 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.<br>0 - Access to the Web Configuration Utility is allowed over both a secure tunnel (HTTPS) and non-secure (HTTP).                                        |
| reg.1.address                           | Specify the line registration. For example: user1@example.com<br>NULL (default)                                                                                                                                                                                                                    |

**Lync Per Phone Example**

|                                                       |                                                                                                                                                                                                                                                                                                                                                                                                                                                                    |
|-------------------------------------------------------|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>reg.1.applyServerDigitMapLocally</code>         | <p>0 (default) - Dialplan rules are processed by Skype for Business Server.</p> <p>1 - Dialplan normalization rules are downloaded from the Skype for Business Server and processed on the phone.</p>                                                                                                                                                                                                                                                              |
| <code>reg.1.auth.domain</code>                        | <p>The domain of the authorization server that is used to check the user names and passwords. For example, <code>example.com</code>.</p> <p>NULL (default)</p>                                                                                                                                                                                                                                                                                                     |
| <code>reg.1.auth.password</code>                      | <p>The user Sign In password for authentication challenges for this registration. Login credentials you enter to the configuration file override Active Directory login credentials and disable use of PIN authentication on the phone interface at Settings &gt; Authentication.</p> <p>NULL (default)</p>                                                                                                                                                        |
| <code>reg.1.auth.userId</code>                        | <p>User ID to be used for authentication challenges for this registration. If the User ID is non-Null, it will override the user parameter entered into the Authentication submenu on the Settings menu of the phone. Login credentials you enter to the configuration file override Active Directory login credentials and disable use of PIN authentication on the phone interface at Settings &gt; Authentication.</p> <p><code>user1</code> (default)</p>      |
| <code>reg.1.auth.usePinCredentials</code>             | <p>1 (default) - Enable the PIN authentication sign in method.</p> <p>0 - Disable the PIN authentication sign in method.</p>                                                                                                                                                                                                                                                                                                                                       |
| <code>reg.1.auth.loginCredentialType</code>           | <p>Specify the credential type the user must provide to log in. You can choose the username/password or extension/PIN.</p> <p><code>LoginCredentialNone</code> (default)</p> <p>You cannot log in to the phone with Microsoft credentials if the parameter <code>reg.1.auth.loginCredentialType</code> is set to the default value. You must manually set parameter <code>reg.1.auth.loginCredentialType</code> to the value <code>usernameAndPassword</code>.</p> |
| <code>reg.1.server.1.registerRetry.baseTimeout</code> | <p>The base time period to wait before a registration retry. Used in conjunction with <code>reg.x.server.y.registerRetry.maxTimeOut</code> to determine how long to wait. The algorithm is defined in RFC 5626.</p> <p>10 seconds (default)</p> <p>10 - 120 seconds</p>                                                                                                                                                                                            |



**Lync Per Phone Example**

|                                                             |                                                                                                                                                                                                                                                                                                                                       |
|-------------------------------------------------------------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>reg.1.server.1.registerRetry.maxTimeout</code>        | <p>Set the maximum period of time in seconds that the phone tries to register.</p> <p>180 seconds (default)</p> <p>60 to 1800 seconds</p>                                                                                                                                                                                             |
| <code>reg.1.server.1.specialInterop</code>                  | <p>Identifies the SIP signaling as Skype for Business Server and enables Skype for Business features. This parameter supports Lync Server 2010 and 2013.</p> <p>lync2010 (default)</p> <p>lync2010, lcs2005, ocs2007r2</p>                                                                                                            |
| <code>reg.1.server.1.transport</code>                       | <p>The transport method the phone uses to communicate with the SIP server.</p> <p>TLS (default)</p>                                                                                                                                                                                                                                   |
| <code>reg.1.serverFeatureControl.cf</code>                  | <p>1 (default) - Server based call forwarding is enabled for this line.</p> <p>0 - Server-based call forwarding is disabled for this line.</p>                                                                                                                                                                                        |
| <code>reg.1.serverFeatureControl.dnd</code>                 | <p>This parameter overrides <code>voIpProt.SIP.serverFeatureControl.dnd</code>.</p> <p>1 (default) - Server-based DND is enabled and the call server has control of DND.</p> <p>0 - Server-based do-not-disturb (DND) is not enabled.</p>                                                                                             |
| <code>reg.1.serverFeatureControl.localProcessing.cf</code>  | <p>0 (default) - If 0 and <code>reg.1.serverFeatureControl.cf</code> is set to 1, the phone does not perform local Call Forward behavior.</p> <p>1 - The phone performs local Call Forward behavior on all calls received.</p>                                                                                                        |
| <code>reg.1.serverFeatureControl.localProcessing.dnd</code> | <p>This parameter overrides <code>voIpProt.SIP.serverFeatureControl.localProcessing.cf</code>.</p> <p>0 (default) - If 0 and <code>reg.x.serverFeatureControl.cf</code> is set to 1, the phone does not perform local Call Forward behavior.</p> <p>1 - The phone will perform local Call Forward behavior on all calls received.</p> |
| <code>reg.1.serverFeatureControl.signalingMethod</code>     | <p>Controls the method used to perform call forwarding requests to the server.</p> <p>serviceMsForward (default)</p>                                                                                                                                                                                                                  |
| <code>reg.1.offerFullCodecListUponResume</code>             | <p>0 (default) - Only the audio codec negotiated during call establishment is sent when a held call is resumed.</p> <p>1 - All supported codes are sent when a held call is resumed.</p>                                                                                                                                              |

**Lync Per Phone Example**

|                                               |                                                                                                                                                                                                |
|-----------------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>reg.1.useteluriAsLineLabel</code>       | 0 (default) - The line key label displays the Skype for Business TelURI, or line address.<br>1 - The line key label displays the Skype for Business account user name.                         |
| <code>roaming_buddies.reg</code>              | Set the line index number for the registered line you want to enable Presence and Instant Messaging. For Microsoft deployments, the value is always 1.<br>1 (default)<br>0 to 34               |
| <code>softkey.feature.simplifiedSignIn</code> | 1 (default) - If 1 and the value for <code>voIpProt.server.x.specialInterop</code> is <code>lync2010</code> , the Sign In soft key is displayed.<br>0 - The Sign In soft key is not displayed. |

**Lync.cfg**

This template file is located in the `Config` folder of the UC Software download.

**Lync.cfg Parameters**

| Parameter                                  | Permitted Values                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                          |
|--------------------------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>BossLine.x.AdminUri</code>           |                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                           |
| <code>BossLine.x.RingType</code>           |                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                           |
| <code>btoe.PairingMode</code>              | Auto (default) - The phone pairs with the computer automatically when the computer is connected to the phone's PC port and the BToE connector application is running.<br>Manual - The phone generates a six-digit pair code to enter to the BToE connector application when the computer is connected to the phone's PC port.                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                             |
| <code>dhcp.option43.override.stsUri</code> | Use this parameter to override DHCP Option 43 and provision phones with an STS URI generated by the Skype for Business or Lync server.<br>NULL (default) - If NULL and you do not configure Option 43, the PIN Authentication menu does not display on the phone and PIN Authentication is not available in the Web Configuration Utility.<br>0 - 255 characters - Enter the STS URI automatically generated by the server to override Option 43. For example:<br><code>https://webpoolblu2a04.infra.lync.com:443/CertProv/CertProvisioningService.svc</code><br>In this example, the part "webpoolblu2a04.infra.lync.com" is unique and varies for each deployment environment.<br>When you enter a correct value for this parameter the PIN Authentication menu displays on the phone and users can sign in using PIN Auth credentials. |

**Lync.cfg Parameters**

| Parameter                                                | Permitted Values  |
|----------------------------------------------------------|-------------------|
| dir.stopTogglePage                                       | 1 (default)       |
| feature.certRenewBackOff.enabled                         | 1 (default)       |
| feature.lync.abs.enabled                                 | 0 (default)       |
| feature.lync.abs.maxResult                               | 12 (default)      |
| feature.lyncbtoe.autosignin.signoff.enabled <sup>1</sup> | 0 (default)       |
| locInfo.1.A1                                             |                   |
| locInfo.1.A3                                             |                   |
| locInfo.1.country                                        |                   |
| locInfo.1.HNO                                            |                   |
| locInfo.1.HNS                                            |                   |
| locInfo.1.label                                          |                   |
| locInfo.1.LOC                                            |                   |
| locInfo.1.NAM                                            |                   |
| locInfo.1.PC                                             |                   |
| locInfo.1.POD                                            |                   |
| locInfo.1.PRD                                            |                   |
| locInfo.1.RD                                             |                   |
| locInfo.1.STS                                            |                   |
| prov.login.lcCache.domain                                |                   |
| prov.login.lcCache.user                                  |                   |
| reg.1.dialPlanName                                       |                   |
| reg.1.lisdisclaimer                                      |                   |
| reg.1.lync.autoProvisionCertLocation                     | 6 (default)       |
| reg.1.ringType.privateLine                               | default (default) |
| reg.1.serverAutoDiscovery                                | 1 (default)       |
| softkey.feature.simplifiedSignIn                         | 0                 |
| up.SLA.ringType                                          | ringer1 (default) |

<sup>1</sup> Change causes phone to restart or reboot.

## Base Profile device.set

The following parameters are located in the `device.cfg` folder of the UC Software download.

### Base Profile device.set Parameters

| Parameter                                        | Permitted Values                                                                                                                                                                                                                                     |
|--------------------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>device.baseProfile</code> <sup>1</sup>     | This parameter sets the value for the device Base Profile. Set this parameter to Lync.<br>NULL (default)<br>Lync                                                                                                                                     |
| <code>device.baseProfile.set</code> <sup>1</sup> | This parameter sets the value for the device Base Profile. Set this parameter to Lync.<br>NULL<br>Lync                                                                                                                                               |
| <code>device.set</code> <sup>1</sup>             | A global parameter that you enable to make changes to <device> parameters. Use this parameter to change only <device> parameter values. Once you have made your changes, remove this parameter from the configuration file.<br>0 (default)<br>0 or 1 |

<sup>1</sup> Change causes phone to restart or reboot.

## In-Band Provisioning

When you register phones with Skype for Business or Lync Server, the Microsoft server automatically provisions phones with in-band provisioning parameters and settings that operate Microsoft features. You can:

- View in-band provisioning parameters from your phone or using the Web Configuration Utility.
- Disable (block) in-band provisioning device settings, recommended if using on-premises provisioning. Refer to [Configuring In-Band Provisioning Settings](#).
- View a list of all in-band provisioning parameters. Refer to [In-Band Provisioning Parameters](#).

### View In-Band Provisioning Parameters from the Phone

You can view In-band provisioning parameters from the phone menu.

#### To view in-band provisioning parameters from the phone menu:

- 1 On your phone, go to **Settings > Advanced**, enter the password (default 456), and press **Enter**.
- 2 Go to **Administration Settings > Upload Configuration**.
- 3 Scroll down and select **SIP**.
- 4 Press the **Upload** soft key.

The phone uploads `MAC-upload-CallServer.cfg` to your boot server.

- 5 Open `MAC-upload-CallServer.cfg` to view the in-band provisioning parameters.

## View In-Band Provisioning Parameters using the Web Configuration Utility

You can view in-band provisioning parameters by exporting from the phone's Web Configuration Utility.

### To view in-band provisioning parameters with the Web Configuration Utility:

- 1 Log into the Web Configuration Utility as an Administrator.
- 2 Go to **Utilities > Import & Export Configuration**, shown next.



- 3 Expand **Export Configuration**, click the **Export Configuration File** drop-down menu, choose **Lync**, and click **Export**.
- 4 Save the XML file to your computer.

## Disable (Block) Automatic In-Band Settings

You can choose to enable and accept or disable and block in-band provisioning device settings sent from the Skype for Business Server.

Disabling in-band device settings blocks the following:

- EnableDeviceUpdate
- IPPhoneAdminPasswd
- LocalProvisioningServerAddress
- LocalProvisioningServerUser
- LocalProvisioningServerPassword

**Disable In-Band Device Settings**

| Parameter                                       | Permitted Values                                                                                                                                                                           |
|-------------------------------------------------|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>lync.provisionDeviceParams.enabled</code> | 1 (default) - Enable (accept) in-band provisioning device settings sent from Skype for Business.<br>0 - Disable (block) in-band provisioning device settings sent from Skype for Business. |

***In-Band Provisioning Parameters***

The following table lists parameters that the Microsoft server automatically uses to provision phones.

**In-Band Provisioning Parameters**

| Parameter                                                | Permitted Values                                                                                                                                                                                                               |
|----------------------------------------------------------|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>dialplan.1.e911dialmask</code>                     | 112<br>100                                                                                                                                                                                                                     |
| <code>dialplan.1.e911dialstring</code>                   | 911                                                                                                                                                                                                                            |
| <code>dialplan.1.originaldigitmap</code>                 | Default value depends on how the Skype for Business Server is configured.                                                                                                                                                      |
| <code>dialplan.routing.emergency.1.value</code>          | 911                                                                                                                                                                                                                            |
| <code>msg.mwi.1.callBack</code>                          | Enter the contact to call when retrieving messages for this registration if <code>msg.mwi.x.callBackMode</code> is set to <code>contact</code> .<br>The default value depends on how the Lync Server is configured.            |
| <code>msg.mwi.1.callBackMode</code>                      | The message retrieval mode and notification for registration x. The value <code>contact</code> indicates that a call is placed to the contact specified by <code>msg.mwi.x.callback</code> .<br><code>contact</code> (default) |
| <code>reg.1.ice.turn.callAdmissionControl.enabled</code> | 1 (default)                                                                                                                                                                                                                    |
| <code>reg.1.lisdisclaimer</code>                         | Default value depends on how the Skype for Business Server is configured.                                                                                                                                                      |
| <code>reg.x.srtp.enable</code>                           | 1 (default) - SRTP for incoming SIP calls is enabled for a given line x.<br>0 - SRTP offered SIP calls are declined.                                                                                                           |

**In-Band Provisioning Parameters**

|                                                             |                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                     |
|-------------------------------------------------------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <code>reg.1.srtp.offer</code>                               | <p>This parameter applies to the registration initiating (offering) a phone call.</p> <p>1 (default) - The registration includes a secure media stream description along with the usual non-secure media description in the SDP of a SIP INVITE.</p> <p>0 - No secure media stream is included in SDP of a SIP invite.</p>                                                                                                                                                                                                                          |
| <code>reg.1.srtp.require</code>                             | <p>1 (default) - The registration is only allowed to use secure media streams. Any offered SIP INVITEs must include a secure media description in the SDP or the call will be rejected. For outgoing calls, only a secure media stream description is included in the SDP of the SIP INVITE, meaning that the non-secure media description is not included. If this parameter set to 1, <code>reg.x.srtp.offer</code> is also be set to 1, regardless of the value in the configuration file.</p> <p>0 - Secure media streams are not required.</p> |
| <code>tcpIpApp.ice.turn.callAdmissionControl.enabled</code> | 1 (default)                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                         |
| <code>tcpIpApp.ice.username</code>                          | Default value depends on how the Skype for Business Server is configured. This unique value is created for each registration and changes every eight minutes.                                                                                                                                                                                                                                                                                                                                                                                       |
| <code>tcpIpApp.ice.password</code>                          | Default value depends on how the Skype for Business Server is configured                                                                                                                                                                                                                                                                                                                                                                                                                                                                            |
| <code>tcpIpApp.ice.turn.server</code>                       | Default value depends on how the Skype for Business Server is configured                                                                                                                                                                                                                                                                                                                                                                                                                                                                            |
| <code>tcpIpApp.ice.turn.tcpPort</code>                      | 443 (default)                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                       |
| <code>tcpIpApp.ice.turn.udpPort</code>                      | 3478 (default)                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                      |