Polycom® Unified Communications for Cisco Webex
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Before You Begin

Topics:

- Audience, Purpose, and Required Skills
- Getting Help

This partner deployment guide explains how to integrate Polycom products into Cisco Webex environments. It also centers around various Cisco and Polycom call control infrastructures. Polycom’s integrated suite of hardware devices and software applications allows you to integrate best of breed video and audio communications across Cisco platforms.

Audience, Purpose, and Required Skills

This guide is written for a technical audience. Polycom expects the administrator to be a mid-level IT professional experienced in system administration.

The primary audience for this guide is administrators who configure, customize, manage, and troubleshoot video endpoints that are integrated with Polycom Unified Communication for Cisco Webex. If configuring Microsoft calendars, the administrator should have Microsoft IT administrative expertise. The administrator should also be familiar with video conferencing concepts.

Getting Help

For more information about installing, configuring, and administering Polycom products, refer to the Polycom Documentation Library or Documents & Software at Polycom Support.

Related Documentation

For additional documentation resources related to Polycom’s cloud services, see https://cloudsupport.polycom.com/Services/.

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.
Polycom Unified Communications for Cisco Webex Requirements

Topics:

- Products Tested with this Release

The Cisco Webex application enables users to join scheduled meetings by leveraging calendaring integration and use one click to join the meeting. It is available for endpoints that support calendaring. When the meeting time arrives, the user can select the meeting and, join the conference with just one click.

Products Tested with this Release

Polycom products are tested extensively with a wide range of products. You can view a list of the products that have been tested for compatibility with this release.

Polycom strives to support any system that is standards-compliant and investigates reports of Polycom systems that are not interoperable with other vendor systems. Note that the following list is not a complete inventory of compatible equipment, but the products that have been tested with this release.

Note: Polycom recommends that you upgrade all of your Polycom systems with the latest software versions. Any compatibility issues may already have been addressed by software updates. Refer to Polycom Service Policies at http://support.polycom.com/content/support/service-policies.html to see the Current Polycom Interoperability Matrix.

Products Tested with this Release

<table>
<thead>
<tr>
<th>Product</th>
<th>Tested Versions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unified Communications Manager</td>
<td>11.5.1.14900</td>
</tr>
<tr>
<td>VCS-C/Expressway</td>
<td>X8.9.1</td>
</tr>
<tr>
<td>VCS-E/Expressway</td>
<td>X8.9.1</td>
</tr>
<tr>
<td>Polycom® RealPresence® DMA</td>
<td>9.0.1</td>
</tr>
<tr>
<td>Polycom® RealPresence® Access Director™</td>
<td>4.2.5.2</td>
</tr>
<tr>
<td>Webex Web Client</td>
<td>N/A</td>
</tr>
<tr>
<td>Webex Teams Client</td>
<td>Unsupported</td>
</tr>
<tr>
<td>Cisco Webex Meetings Client</td>
<td>33.4.4.5 or higher</td>
</tr>
<tr>
<td>Polycom® RealPresence® Group Series</td>
<td>6.1.5 and higher</td>
</tr>
<tr>
<td>Polycom Trio™</td>
<td>5.7.2 and higher</td>
</tr>
<tr>
<td>Product</td>
<td>Tested Versions</td>
</tr>
<tr>
<td>------------------------------</td>
<td>-----------------</td>
</tr>
<tr>
<td>Polycom® Pano™ App</td>
<td>1.1 and higher</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Endpoint</th>
<th>Registration</th>
<th>Firewall</th>
<th>Protocol</th>
<th>Audio</th>
<th>Video</th>
<th>Content</th>
<th>One Touch Dial</th>
</tr>
</thead>
<tbody>
<tr>
<td>RealPresence Group Series</td>
<td>Cisco Unified Communications Manager</td>
<td>RealPresence DMA/Access Director</td>
<td>SIP</td>
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<td>✓</td>
<td>✓</td>
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<tr>
<td>Polycom Trio Visual+</td>
<td>Cisco Unified Communications Manager</td>
<td>RealPresence DMA/Access Director</td>
<td>SIP</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>Polycom Trio Visual Pro</td>
<td>Cisco Unified Communications Manager</td>
<td>RealPresence DMA/Access Director</td>
<td>SIP</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>RealPresence Group Series</td>
<td>Cisco Unified Communications Manager</td>
<td>VCS-E</td>
<td>SIP</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>Polycom Trio Visual+</td>
<td>Cisco Unified Communications Manager</td>
<td>VCS-E</td>
<td>SIP</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>Polycom Trio Visual Pro</td>
<td>Cisco Unified Communications Manager</td>
<td>VCS-E</td>
<td>SIP</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
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<tr>
<td>RealPresence Group Series</td>
<td>VCS-C</td>
<td>VCS-E</td>
<td>SIP and H.323</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
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<tr>
<td>RealPresence Group Series</td>
<td>RealPresence DMA</td>
<td>RealPresence Access Director</td>
<td>SIP and H.323</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
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<tr>
<td>Polycom Trio Visual+</td>
<td>RealPresence DMA</td>
<td>RealPresence Access Director</td>
<td>SIP</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>Polycom Trio Visual Pro</td>
<td>RealPresence DMA</td>
<td>RealPresence Access Director</td>
<td>SIP</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
</tbody>
</table>

**Supported Deployment Models**
You can deploy Polycom products with Cisco Webex using different deployment models.
Polycom supports the following deployment models when integrating Polycom products with Cisco Webex environments:

- Direct Registration of Polycom RealPresence Systems with CUCM
- Direct Secure Registration of Polycom RealPresence Endpoints with VCS
- Direct Secure Registration of Polycom RealPresence Endpoints to RealPresence DMA
Registering RealPresence Group Series or Polycom Trio with CUCM

Topics:

- Configuring CUCM for Polycom Endpoints TCP/Unsecure Registration
- Configuring CUCM for Polycom Endpoint TLS/Secure Registration
- Configuring RealPresence Group Series for Secure CUCM Registration
- Configuring a RealPresence Group Series for One Touch Dial Registration
- Configuring a Polycom Trio for CUCM Registration
- Configuring a Polycom Trio for Secure CUCM Registration
- Configure Polycom Trio Visual Plus Codec Priorities
- Configure Polycom Trio Visual Pro Codec Priorities
- Configure a Polycom Trio Dial Plan and One Touch Dial Service

To register a RealPresence Group Series system with CUCM, you need to perform steps in both the CUCM and the RealPresence Group Series system.

For more information about the Cisco Unified Communications Manager, see the Cisco Unified Communications Manager Documentation Guide. For more information on Polycom RealPresence Group Series, see the Administrator's Guide.

Configuring CUCM for Polycom Endpoints TCP/Unsecure Registration

Configure CUCM for Polycom endpoint TCP/unsecure registration using the CUCM web administrator interface.

Before you configure CUCM, review the Cisco Unified Communications Manager Considerations.

Create a Security Profile for TCP/Unsecure Registration

Enable the Phone Security Profile with digest authentication to secure the Polycom endpoint’s connection to CUCM.

Procedure

1. Log in to the CUCM console.
2. Select System > Security Profile > Phone Security Profile.
3. Select Add New.
4. In the Phone Security Profile Type drop-down menu, select Third-party SIP Device (Advanced) and click Next.
5. On the **Phone Security Profile Information** page, complete the following fields:
   
   a. In the **Name** text box, enter a profile name for the system.
   
   b. In the **Description** field, enter a description for the security profile.
   
   c. If you want to use digest authentication (recommended), select the **Enable Digest Authentication** check box. Digest authentication requires valid login passwords to register devices.
   
   d. Select the default values for all other fields. This example uses digest authentication.
6. Click **Save**.
In the status bar near the top of the page, **Update Successful** displays.

**Add a System User for TCP/Unsecure Registration**

Create a CUCM system user for each Polycom endpoint.

**Note:** If you can't add a user, your system may be LDAP integrated. In that case, use an existing user ID (essentially associating the endpoint to an existing user) or have your LDAP administrator create a new user ID for each CUCM device.

**Procedure**

1. Select **User Management** > **End User**.

2. Click **Add New**.
   The following screen displays. This example creates two users, one for the RealPresence Group Series integration and one for Polycom Trio.
3. Complete the required fields (User ID and Last Name, at a minimum).

   a. To use digest authentication, enter the Digest Credentials (password) for the Polycom system. You configure this information in the Polycom system in a later step.

   b. In the Confirm Digest Credentials text box, enter the same value that you entered in step (a).
Note: The end user Password and PIN fields are arbitrary and aren’t used for registration.

4. Click Save.
   In the status bar near the top of the page, an Update Successful message displays.

Create a SIP Profile for TCP/Unsecure Registration
CUCM associates specific SIP parameters with an endpoint or trunk via a SIP Profile.
This configuration creates a SIP profile that is associated with all Polycom devices.

Procedure
1. Select Device > Device Settings > SIP Profile.
2. Click Find to see the list of existing SIP Profiles, and select the Standard SIP Profile (a default in CUCM).
3. Once open, select Copy.
4. Change the Name to something meaningful for your deployment, and then ensure the following configuration:
   a. Use Fully Qualified Domain Name in SIP Requests should be checked.
   b. Allow Presentation Sharing using BFCP should be checked.
   c. Early Offer support for voice and video calls should not be checked. This section shows the data as an example.
### SIP Profile Configuration

#### Status
- **Status:** Ready
- **Information:** All SIP devices using this profile must be restarted before any changes will take effect.

#### SIP Profile Information

<table>
<thead>
<tr>
<th>Name*</th>
<th>Polycom Standard SIP Profile</th>
</tr>
</thead>
<tbody>
<tr>
<td>Description</td>
<td>Default SIP Profile + BFCP</td>
</tr>
<tr>
<td>Default MTP Telephony Event Payload Type*</td>
<td>101</td>
</tr>
<tr>
<td>Resource Priority Namespace List</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Early Offer for G.Clear Calls*</td>
<td>Disabled</td>
</tr>
<tr>
<td>SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*</td>
<td>TIAS and AS</td>
</tr>
<tr>
<td>User-Agent and Server header information*</td>
<td>Send Unified CM Version Information as User-Agent</td>
</tr>
<tr>
<td>Redirect by Application</td>
<td></td>
</tr>
<tr>
<td>Disable Early Media on 180</td>
<td></td>
</tr>
<tr>
<td>Outgoing T.38 INVITE include audio mline</td>
<td></td>
</tr>
<tr>
<td>Enable ANAT</td>
<td></td>
</tr>
<tr>
<td>Require SDP Inactive Exchange for Mid-Call Media Change</td>
<td></td>
</tr>
<tr>
<td>Use Fully Qualified Domain Name in SIP Requests</td>
<td></td>
</tr>
<tr>
<td>Parameters used in Phone</td>
<td>Value</td>
</tr>
<tr>
<td>----------------------------------</td>
<td>-----------</td>
</tr>
<tr>
<td>Timer Invite Expires (seconds)*</td>
<td>180</td>
</tr>
<tr>
<td>Timer Register Delta (seconds)*</td>
<td>5</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)*</td>
<td>3600</td>
</tr>
<tr>
<td>Timer T1 (msec)*</td>
<td>500</td>
</tr>
<tr>
<td>Timer T2 (msec)*</td>
<td>4000</td>
</tr>
<tr>
<td>Retry INVITE*</td>
<td>6</td>
</tr>
<tr>
<td>Retry Non-INVITE*</td>
<td>10</td>
</tr>
<tr>
<td>Start Media Port*</td>
<td>15384</td>
</tr>
<tr>
<td>Stop Media Port*</td>
<td>32766</td>
</tr>
<tr>
<td>Call Pickup URI*</td>
<td>x-disco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI*</td>
<td>x-disco-serviceuri-opickup</td>
</tr>
<tr>
<td>Call Pickup Group URI*</td>
<td>x-disco-serviceuri-opickup</td>
</tr>
<tr>
<td>Meet Me Service URI*</td>
<td>x-disco-serviceuri-meetme</td>
</tr>
<tr>
<td>User Info*</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level*</td>
<td>Nominal</td>
</tr>
<tr>
<td>Call Hold Ring Back*</td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block*</td>
<td>Off</td>
</tr>
<tr>
<td>Caller ID Blocking*</td>
<td>Off</td>
</tr>
<tr>
<td>Do Not Disturb Control*</td>
<td>User</td>
</tr>
<tr>
<td>Telnet Level for 7940 and 7950*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Timer Keep Alive Expires (seconds)*</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)*</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)*</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirections*</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To First Digit Timer (milliseconds)*</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI*</td>
<td>x-disco-serviceuri-cfwall</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI*</td>
<td>x-disco-serviceuri-abbrdial</td>
</tr>
<tr>
<td>Conference Join Enabled</td>
<td></td>
</tr>
<tr>
<td>RFC 2543 Hold</td>
<td></td>
</tr>
<tr>
<td>Semi Attended Transfer</td>
<td></td>
</tr>
<tr>
<td>Enable VAD</td>
<td></td>
</tr>
<tr>
<td>Stutter Message Waiting</td>
<td></td>
</tr>
</tbody>
</table>
5. Click Save. In the status bar near the top of the page, an Update Successful message displays.

**Add a Device Entry for TCP/Unsecure Registration**

Create a CUCM device entry for each endpoint to allow the device to register properly with CUCM.

**Procedure**

1. Select Device > Phone.
2. Click Add New.
3. Select Third-party SIP Device (Advanced) and click Next.
4. Configure the following settings in the **Device Information** section.

a. In the **MAC Address** text box, enter a unique MAC Address for the RealPresence Group Series system. This can be any valid, unique MAC address. CUCM actually uses the RealPresence Group Series system user name to identify the RealPresence Group Series system.
Note: This field is arbitrary for third-party SIP Devices in CUCM, however Polycom recommends configuring the actual MAC address of the RealPresence Group Series system to avoid conflicts.

b. In the Description text box, enter a description (Optional).

c. From the Device Pool list, select the device pool appropriate for your Cisco Unified Communications Manager system video devices.

d. From the Phone Button Template list, select Third-party SIP Device (Advanced).

e. If your CUCM implementation uses partitions and Call Search Spaces, from the Calling Search Space list, select an appropriate calling search space for the RealPresence Group Series system (Optional).

f. If your CUCM implementation uses the CUCM Locations form of Call Admission Control (CAC), from the Location list, select an appropriate location for the RealPresence Group Series system. This location should contain video bandwidth. Before making this selection, see Telepresence Deployment Design Considerations and Cisco Unified Communications Manager Considerations.

5. Configure the following settings in the Protocol Specific Information section.

![Protocol Specific Information](image)

a. From the Device Security Profile list, select the profile created in “Create a Security Profile for TCP/Unsecure Registration.”

b. In the Digest User field, select a user created in “Add a System User for TCP/Unsecure Registration.”

c. From the SIP Profile list, select the profile created in “Add a SIP Profile for TCP/Unsecure Registration.”

d. Select Allow Presentation Sharing using BFCP.

6. Click Save.

In the status near the top of the page, an Update Successful message displays.

7. In the Association Information section, click Line [1] - Add a new DN and complete the following required fields:
In the Directory Number field, enter the phone’s extension number.

In the Route Partition field, choose the appropriate value for your CUCM deployment.

8. Click Save.
   In the status bar near the top of the page, an Update Successful message displays.

9. Reset the Polycom system in CUCM.

10. Follow the same steps to create a second third-party SIP Device (Advanced) for the Polycom Trio. For example, TPTrio2.

Configuring CUCM for Polycom Endpoint TLS/Secure Registration

Configure CUCM for Polycom endpoint TLS/secure registration using the CUCM web administrator interface.

Before you configure CUCM, review the Cisco Unified Communications Manager Considerations.

Create a Security Profile for TLS/Secure Registration

You must create a security profile to use with your Polycom endpoints. Each endpoint uses the same security profile, so you only need to create one.

If you want to create a secure profile, you can choose to enable digest authentication to secure the Polycom endpoint’s connection to CUCM.

---

**Note:** Polycom recommends using digest authentication for Polycom endpoint registration.

**Procedure**

1. Log in to the CUCM console.

2. Select System > Security Profile > Phone Security Profile.
3. Select **Add New**.

4. In the **Phone Security Profile Type** drop-down menu, select **Third-party AS-SIP Endpoint** and click **Next**.

5. On the **Phone Security Profile Information** page, complete the following fields:
   
a. In the **Name** text box, enter a profile name for the system.

b. In the **Description** field, enter a description for the security profile.

c. Set the **Device Security Mode** to **Encrypted**.

d. Set the **Transport Type** to **TLS**.

e. Enable **Digest Authentication**.

f. Set the **SIP Phone Port** to **5061**.

6. Click **Save** then **Apply Config**.
   
   In the status bar near the top of the page, **Update Successful** displays.

**Add a System User for TLS/Secure Registration**

Create a CUCM system user for each Polycom endpoint.

For each Generic Single/Multiple Screen Room System device added in CUCM, only a single system user is necessary when adding secure ITP systems.
Note: If you can't add a user, your system may be LDAP integrated. In that case, use an existing user ID (essentially associating the endpoint to an existing user) or have your LDAP administrator create a new user ID for each CUCM device.

Procedure

1. Select User Management > End User.

2. Click Add New.

   The following screen displays.

   ![User Information Screen]

3. Complete the required fields (User ID and Last Name, at a minimum).
   a. To use digest authentication, enter the Digest Credentials (password) for the Polycom system. You configure this information in the Polycom system in a later step.
   b. In the Confirm Digest Credentials text box, enter the same value that you entered in step (a).
Create a SIP Profile for TLS/Secure Registration

CUCM associates specific SIP parameters with an endpoint or trunk via a SIP Profile. This configuration creates an association with a SIP profile in CUCM and Polycom devices.

Procedure

1. Select Device > Device Settings > SIP Profile.
2. Click Find to see the list of existing SIP Profiles, and select the Standard SIP Profile (a default in CUCM).
3. When the profile opens, select Copy.

   **Note:** Leave the SIP settings at the default. However, consult a CUCM administrator for any SIP settings that may be specific to your deployment.

4. Change the Name to something meaningful for your deployment, then ensure the following configuration settings:

   a. Ensure that the Use Fully Qualified Domain Name in SIP Requests check box is selected.
   b. Ensure that the Allow Presentation Sharing using BFCP check box is selected.
   c. Ensure that the Early Offer support for voice and video calls check box is NOT selected.

   This following graphics show the data as an example.
### SIP Profile Configuration

<table>
<thead>
<tr>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Status: Ready</td>
</tr>
<tr>
<td>All SIP devices using this profile must be restarted before any changes will take affect.</td>
</tr>
</tbody>
</table>

### SIP Profile Information

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default MTP Telephony Event Payload Type*</td>
<td>Resource Priority Namespace List</td>
</tr>
<tr>
<td>Early Offer for G.Clear Calls*</td>
<td>SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*</td>
</tr>
<tr>
<td>User-Agent and Server header information*</td>
<td>Polycom Standard SIP Profile</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Option</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Redirect by Application</td>
<td></td>
</tr>
<tr>
<td>Disable Early Media on 180</td>
<td></td>
</tr>
<tr>
<td>Outgoing T.38 INVITE include audio, mline</td>
<td></td>
</tr>
<tr>
<td>Enable ANAT</td>
<td></td>
</tr>
<tr>
<td>Require SDP Inactive Exchange for Mid-Call Media Change</td>
<td></td>
</tr>
<tr>
<td>Use Fully Qualified Domain Name in SIP Requests</td>
<td></td>
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</table>

*Required fields
**Optional fields**
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invite Expires (seconds)</td>
<td>180</td>
</tr>
<tr>
<td>Timer Register Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)</td>
<td>3600</td>
</tr>
<tr>
<td>Timer T1 (msec)</td>
<td>500</td>
</tr>
<tr>
<td>Timer T2 (msec)</td>
<td>4000</td>
</tr>
<tr>
<td>Retry INVITE</td>
<td>6</td>
</tr>
<tr>
<td>Retry Non-INVITE</td>
<td>10</td>
</tr>
<tr>
<td>Start Media Port</td>
<td>15384</td>
</tr>
<tr>
<td>Stop Media Port</td>
<td>32766</td>
</tr>
<tr>
<td>Call Pickup URI</td>
<td>x-disco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI</td>
<td>x-disco-serviceuri-opickup</td>
</tr>
<tr>
<td>Call Pickup Group URI</td>
<td>x-disco-serviceuri-opickup</td>
</tr>
<tr>
<td>Meet Me Service URI</td>
<td>x-disco-serviceuri-meetme</td>
</tr>
<tr>
<td>User Info</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level</td>
<td>Nominal</td>
</tr>
<tr>
<td>Call Hold Ring Back</td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block</td>
<td>Off</td>
</tr>
<tr>
<td>Caller ID Blocking</td>
<td>Off</td>
</tr>
<tr>
<td>Do Not Disturb Control</td>
<td>User</td>
</tr>
<tr>
<td>Telnet Level for 7940 and 7950</td>
<td>Disabled</td>
</tr>
<tr>
<td>Timer Keep Alive Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirections</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To First Digit Timer (milliseconds)</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI</td>
<td>x-disco-serviceuri-cfwall</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI</td>
<td>x-disco-serviceuri-abbrdial</td>
</tr>
<tr>
<td>Conference Join Enabled</td>
<td></td>
</tr>
<tr>
<td>RFC 2543 Hold</td>
<td></td>
</tr>
<tr>
<td>Semi Attended Transfer</td>
<td></td>
</tr>
<tr>
<td>Enable VAD</td>
<td></td>
</tr>
<tr>
<td>Stutter Message Waiting</td>
<td></td>
</tr>
</tbody>
</table>
5. Click **Save**.
   In the status bar near the top of the page, an **Update Successful** message displays.

---

**Add a Device Entry for TLS/Secure Registration**

Create a CUCM device entry for each endpoint. This configuration adds a device to CUCM, which in turn enables the device to securely register with CUCM.

To securely register a RealPresence Group Series or Polycom Trio system with CUCM, add a **Third-party AS-SIP Endpoint** device for single-screen systems and a **Generic Multiple Screen Room System** (GMSRS) for multi-screen systems.

**Procedure**

1. Select **Device > Phone**.
2. Click **Add New**.
3. Select **Third party AS-SIP Endpoint** and click **Next**.
4. Configure the following settings in the **Device Information** section.

   a. In the **MAC Address** text box, enter the unique MAC Address for the RealPresence Group Series system. For secure registration, this must be the actual MAC Address of the RealPresence Group Series or ITP system.
   
   b. In the **Description** text box, enter a description (Optional).
c. From the **Device Pool** list, select the device pool appropriate for your Cisco Unified Communications Manager system video devices.

d. From the **Phone Button Template** list, select **Generic Single Screen Room System** or **Generic Multiple Screen Room System**.

e. If your CUCM implementation uses partitions and Call Search Spaces, from the **Calling Search Space** list, select an appropriate **calling search space** for the Polycom system (Optional).

f. If your CUCM implementation uses the CUCM Locations form of Call Admission Control (CAC), from the **Location** list, select an appropriate location for the RealPresence Group Series system. This location should contain video bandwidth. Before making this selection, see Telepresence Deployment Design Considerations and Cisco Unified Communications Manager Considerations.

5. Configure the following settings in the **Protocol Specific Information** section.

![Protocol Specific Information](image)

a. From the **Device Security Profile** list, select the profile created in “Create a Security Profile for TLS/Secure Registration.”

b. In the **Digest User** field, select the user created in “Add a System User for TLS/Secure Registration.”

c. From the SIP Profile list, select the profile created in “Add a SIP Profile for TLS/Secure Registration.”

d. Select **Allow Presentation Sharing using BFCP**.

6. Click **Save**.

In the status bar near the top of the page, an **Update Successful** message displays.

7. In the **Association Information** section, click **Line [1] - Add a new DN** and complete the following required fields:

   a. In the **Directory Number** field, enter the phone’s extension number.

   b. In the **Route Partition** field, choose the appropriate value for your CUCM deployment.
8. Click Save and Apply Config.
   In the status bar near the top of the page, an Update Successful message displays.

### Configuring RealPresence Group Series for Secure CUCM Registration

When you securely register a Polycom endpoint with CUCM, the endpoint can make calls to Cisco endpoints. These CUCM registered endpoints can then make encrypted media calls.

Use the RealPresence Group Series web administrator interface to perform the following tasks:

- Configure SIP Settings
- Import Certificates to RealPresence Group Series or Immersive Telepresence Systems
- Enable Encrypted Media Calls

### Configure SIP Settings

Configure the SIP settings to securely register a RealPresence Group Series or Immersive Telepresence system with CUCM.

For ITP systems, configure the center codec for secure registrations.

**Procedure**

1. Enter the Polycom RealPresence Group Series system IP address or host name in a web browser.
2. Go to Admin Settings > Network > IP Network and select SIP.
3. Configure the settings in the SIP Settings section of the IP Network screen.
<table>
<thead>
<tr>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable SIP</td>
<td>Select this check box to enable the Polycom system to receive and make SIP calls.</td>
</tr>
<tr>
<td>Registrar Server</td>
<td>Specify the IP address of the Cisco Unified Communications Manager. Use the <strong>Proxy Server</strong> if this field is blank.</td>
</tr>
<tr>
<td>Proxy Server</td>
<td>Specify the IP address of the SIP proxy server. Use the <strong>Registrar Server</strong> if this field is blank. No use of Proxy Server if both fields left blank.</td>
</tr>
<tr>
<td>Transport Protocol</td>
<td>The SIP network determines the protocol in which your Polycom system operates. Select TLS for secure registration.</td>
</tr>
<tr>
<td>Sign-in Address</td>
<td>Specify the directory number allotted to this device in CUCM.</td>
</tr>
<tr>
<td>Domain User Name</td>
<td>This should match the user name created in Task2 of “Configuring CUCM for a Secure Polycom Immersive, Room, or Desktop System.”</td>
</tr>
<tr>
<td>Password</td>
<td>When enabled, allows you to specify and confirm a new password that authenticates the system to the SIP registrar server. If using <strong>Digest Authentication</strong>, select the check box for the password and set it to the <strong>Digest Credentials</strong> password used for the Cisco Unified Communications user created for this system.</td>
</tr>
</tbody>
</table>
Setting | Description
--- | ---
Registrar Server Type | Specifies whether the SIP registrar server is a Lync Server. For Cisco environments, leave this check box unselected.

**Import CA Certificate to RealPresence Group Series or Immersive Telepresence System**

You must import a valid certificate for CUCM to securely register a RealPresence Group Series or Immersive Telepresence system.

To support the sRTP/TLS feature, Polycom endpoints support Cisco Unified Communications Manager X509v3 certificates in Privacy Enhanced Mail (PEM) format.

**Procedure**

1. Open a browser window and enter the RealPresence Group Series system IP address.
2. Go to **Admin Settings > General Settings > Security > Certificates**.
3. Click **Create** for a **Client Signing Request Client**.
4. Fill out the following fields:
   - Type = Client
   - Hash Algorithm = SHA-1
   - Common Name (CN) = This must be of the format “Polycom-SEP < MAC_Address >” where the MAC Address is the actual MAC of the RealPresence Group Series system or ITP endpoint (center codec for ITP systems).
5. Fill in the other fields as appropriate for your deployment and click **Create**.
6. Generate the CSR, download this client CSR "client_csr.pem" file, and then reboot the RealPresence Group Series system or ITP center codec.

The CSR must have a valid Certification Authority (CA) trusted by CUCM to generate a certificate. This is outside the scope of this documentation.

7. Once you sign the CSR and generate the actual Certificate, continue back to Admin Settings > Security > Certificates. and .

8. Under View and Add, select Browse and select the certificate.PEM file.
Enable Encrypted Media Calls
You can enable encrypted media calls on your RealPresence Group Series system.

Procedure

1. Enter the Polycom RealPresence Group Series system IP address or host name in a browser window.

2. Go to **Admin Settings > Security Settings > Global Security > Encryption**.

3. **Note**: Set AES Encryption to **When Available** for successful secure registration.

   Configure AES Encryption to either **When Available** or **Required for All Calls** as appropriate for your deployment.
4. Configure the Virtual Security Classification (Optional).

Configuring a RealPresence Group Series for One Touch Dial Registration

RealPresence Group Series requires registration to the Polycom One Touch Dial cloud. For more information, refer to Polycom One Touch Dial Deployment Guide.

Configuring a Polycom Trio for CUCM Registration

Use the Polycom Web Configuration Utility to configure Polycom Trio 8800 and register the system to the Cisco Unified Communications Manager (CUCM).

The Web Configuration Utility is a web interface application that is helpful when you’re working remotely. You can use the Web Configuration Utility to provision one phone at a time.

Log In to the Web Configuration Utility

You must log in as an administrator to use the Web Configuration Utility.

Procedure

1. Go to Menu > Status > Network > TCP/IP Parameters > IP to obtain the IP address of your conference phone.
2. Enter the IP address in a web browser on a computer connected to the same network as the conference phone.

3. Log in to the Web Configuration Utility as an Administrator using the following credentials:
   - User ID: xyz@abc.com
   - Password: 456

4. Click Submit.

**Configure Line Settings**

Configure the line settings for your phone using the Web Configuration Utility.

**Procedure**

1. Go to Settings > Lines.

2. Select the line you want to configure and expand the Identification menu. Line 1 is selected by default.

3. Complete the following fields:

   - **Display Name**: This example shows 2223.
   - **Address**: This example shows 2223 to match the Display Name. Address represents the extension created for the device in CUCM.
Configure SIP Server Settings

Configure the SIP server settings using the Web Configuration Utility.

Procedure

1. In the **Outbound Proxy** section, configure the following settings:
   - In the **Address** field, enter the IP address or hostname of the Cisco Unified Communications Manager. In this example, the CUCM has an IP address of 10.46.81.10.
   - Set **Port** to **5060**.
   - In the **Transport** drop-down menu, select **TCP**.

2. In the **SIP Server 1** section, configure the following settings:
   - In the **Address** field, enter the IP address or hostname of the Cisco Unified Communications Manager. In this example, the CUCM has an IP address of 10.46.81.10.
   - Set **Port** to the correct port number for your environment.
   - In the **Transport** drop-down menu, select **TCP**.

3. Click **Save** to apply the settings.
4. When you successfully register on the SIP server, you see a solid phone icon
   ▪ On a phone, a solid phone icon displays on the phone screen.
   ▪ On the Polycom Trio, a green circle with a check mark appears.

Configure Date and Time Settings
Configure the date and time settings using the Web Configuration Utility.

Procedure
1. Go to Preferences > Date & Time.

   **Date & Time**
   - **Display Format**
     - Time Format: 12 AM/PM
     - Date Format: Monday, January 1
   - **Time Synchronization**
     - SNTP Server: north-america.pool.ntp.org
     - SNTP Resync Period (s): 86400
     - Time Zone: (GMT-8:00) Pacific Time (US & Canada)
   - **Daylight Savings**
     - Daylight Savings: Enable
     - Fixed Day: Enable
     - Start Date: Sunday, March 02:00
     - End Date: Sunday, November 02:00

2. In the Display Format section, choose a Time Format and Date Format that you want the phone to display.
3. In the Time Synchronization section, choose an SNTP server in your region that the phone receives its time setting from and select a region in Time Zone.
4. In the Daylight Savings section, enable or disable Daylight Savings time changes.
   When enabled, the phone’s time settings automatically adjust to daylight savings time according to the settings you configure in Fixed Day, Start Date, and End Date.
Configuring a Polycom Trio for Secure CUCM Registration

When you securely register a Polycom endpoint is with CUCM, the endpoint can make calls to Cisco endpoints. These CUCM registered endpoints can then make encrypted media calls.

**Note:** If you register Microsoft Skype for Business on Line 1, TLS registration to CUCM gets support from Office365 and not an on-premises instance.

Use the Polycom Trio web administrator interface to perform the following tasks:
- Configure SIP Line Settings
- Generate and Import Certificates to Polycom Trio

**Configure SIP Line Settings**

Configure SIP settings to securely register a RealPresence Group Series or Immersive Telepresence system with CUCM.

For ITP systems, configure the center codec for secure registrations.

**Procedure**

1. Enter the Polycom Trio IP address or host name into a browser window.
2. Go to **Settings > Lines** and select the appropriate line.
3. Configure the Polycom Trio line settings in each of the following areas:

   **Identification**

<table>
<thead>
<tr>
<th>Settings</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Display Name</td>
<td><strong>Arbitrary name</strong> value.</td>
</tr>
<tr>
<td>Address</td>
<td>This field must match the <strong>Line ID</strong> assigned to this device in CUCM.</td>
</tr>
<tr>
<td>Label</td>
<td><strong>Arbitrary name</strong> value.</td>
</tr>
<tr>
<td>Enable SRTP</td>
<td>Select the check box.</td>
</tr>
<tr>
<td>Offer SRTP</td>
<td>Select the check box.</td>
</tr>
<tr>
<td>Require SRTP</td>
<td>Leave the check box unselected.</td>
</tr>
<tr>
<td>Server Auto Discover</td>
<td>Select the check box.</td>
</tr>
<tr>
<td>TCP Fast Failover</td>
<td>Leave the check box unselected.</td>
</tr>
</tbody>
</table>
Authentication

<table>
<thead>
<tr>
<th>Settings</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Use Login Credentials</td>
<td>Set to Disabled.</td>
</tr>
<tr>
<td>Domain</td>
<td>No Value; leave blank.</td>
</tr>
<tr>
<td>User ID</td>
<td>Specify the User ID created in CUCM for this device.</td>
</tr>
<tr>
<td>Password</td>
<td>Specify the Digest Authentication password created in CUCM for this device.</td>
</tr>
</tbody>
</table>

Outbound Proxy

<table>
<thead>
<tr>
<th>Settings</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Address</td>
<td>Specify the IP address of hostname of CUCM</td>
</tr>
<tr>
<td>Port</td>
<td>Set to 5061</td>
</tr>
<tr>
<td>Transport</td>
<td>Set to TLS</td>
</tr>
</tbody>
</table>

Server 1

<table>
<thead>
<tr>
<th>Settings</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Special Interop</td>
<td>Select Standard.</td>
</tr>
<tr>
<td>Special Interop</td>
<td>Specify the IP address of hostname of CUCM.</td>
</tr>
<tr>
<td>Port</td>
<td>Set to 5061.</td>
</tr>
<tr>
<td>Transport</td>
<td>Set to TLS.</td>
</tr>
<tr>
<td>Register</td>
<td>Select Yes.</td>
</tr>
</tbody>
</table>

Generate Certificates for CUCM

You must generate certificates for CUCM for secure registration with Polycom Trio.

To support the SRTP/TLS feature, Polycom endpoints support Cisco Unified Communications Manager X509v3 certificates.

Procedure

1. Open a browser window and enter the Polycom Trio system IP address.
2. Go to Settings > Advanced > Administrative Settings > Generate CSR.
3. Click Create for a Client Signing Request Client.
4. Fill out the following fields:
   - Common Name
   - Organization
5. Fill in the other fields as appropriate for your deployment and click **Generate**. Once you create the CSR, the private key file and CSR are uploaded to the FTP server configured on the Polycom Trio.

The CSR must have a valid Certification Authority (CA) trusted by CUCM to generate a certificate. This is outside the scope of this documentation.

**Import Root CA and Signed Device Certificates**

Once you sign the CSR and generate a certificate, you must upload the root CA and signed client certificates to a web server for download.

**Procedure**

1. Open a browser window and enter the Polycom Trio system IP address.
2. Go to **Settings > Network > TLS** and select **CA Certificates** option.
3. Select **Application CA 1 > Install** and point to the **Root CA certificate URI**.
4. Select **Install**.
   
   You receive a message box after the cert installation.

5. Select **Device Certificate** option at the top of the screen.
6. Select **Application Credential 1** and **Install**. Point to the signed client certificate.
7. Generate the next point to the private key along with the CSR on the FTP server.

8. Select **TLS Profiles > Application Profile 1**. Under **Certificate column**, ensure that the drop down list box displays **Application Credential**.

9. Select **TLS Applications > SIP**. Ensure that the **Profile Name** is **TLS Application Profile 1** and **Minimum TLS Version** is **TLS 1.2**.

10. Ensure that the MTLS SIP options are enabled.
Configure Polycom Trio Visual Plus Codec Priorities

Configure the Polycom Trio Visual Plus Codec Properties using the settings for Video Codec Priority.

Procedure

1. Go to Settings > Codec Priorities > Video Codec Priority.
2. Set the codec priority to the following order:

   - X-H264UC
   - H.264 PacketizationMode0
   - H.264 HP
   - H.264 HP PacketizationMode0
   - H.264
Configure Polycom Trio Visual Pro Codec Priorities

Configure the Polycom Trio Visual Pro Codec Properties using the settings for Video Codec Priority.

Procedure

1. Go to Settings > Codec Priorities > Video Codec Priority.
2. Set the codec priority to the following order:
   - X-UlpFecUC
   - X-H264 UC
   - H-264
   - H-264 HP
   - H-264 PackatizationMode 0
   - H-264 HP PackatizationMode 0

Configure a Polycom Trio Dial Plan and One Touch Dial Service

If CUCM on line 1 has Polycom Trio registration, use or append this information to the Polycom Trio configuration.

The parameters allow Polycom Trio to provide a Join button for calendar invites and route calls to Webex through a correct line. There’s also a mandatory config to disable Far End Camera Control. For more information, refer to the Knowledge Base Article 35545.

The dialplan.1.digitmap.mode parameter is new to Polycom Trio 5.7.2 and allows the use of regex for the dialplan.

Copy this info into a text file and save as a .cfg file:

```xml
<?xml version="1.0" encoding="UTF-8" standalone="yes"?>
<UPLOAD>
<ALL
  feature.fecc.enabled="0"
  dialplan.applyToDirectoryDial="1"
  dialplan.digitmap.lineSwitching.enable="1"
  dialplan.1.digitmap.mode="regex"
  dialplan.1.digitmap=".+@.+\.webex\.com"
  exchange.meeting.parseAllowedSipUriDomains="webex.com"
  exchange.meeting.parseEmailsAsSipUris="1"
  exchange.meeting.parseOption="All"/>
</UPLOAD>
```

If CUCM on line 2 has Polycom Trio registration, use or append this information to the Polycom Trio configuration. The parameters allow Polycom Trio to provide a Join button for calendar invites and route calls to Webex through a correct line. The dialplan.1.digitmap.mode parameter is new to Polycom Trio 5.7.2 and allows the use of regex for the dialplan.
Copy this info into a text file and save as a .cfg file:

```xml
<?xml version="1.0" encoding="UTF-8" standalone="yes"?>
<UPLOAD>
<ALL
feature.fecc.enabled="0"
dialplan.applyToDirectoryDial="1"
dialplan.digitmap.lineSwitching.enable="1"
dialplan.2.applyToDirectoryDial="1"
dialplan.2.digitmap.mode="regex"
dialplan.2.digitmap=".+@.+\.webex\.com"
exchange.meeting.parseAllowedSipUriDomains="webex.com"
exchange.meeting.parseEmailsAsSipUris="1"
exchange.meeting.parseOption="All"/>
</UPLOAD>
```