# Cisco TelePresence Conductor Clustering with Unified CM

## Deployment Guide

TelePresence Conductor XC2.4
Unified CM 10.x

D15000.07

September 2014

# Contents

# Introduction

## About this document

This document assumes that a standalone Cisco TelePresence Conductor integration with Cisco Unified Communications Manager (Unified CM) ad hoc and rendezvous calls has been set up according to the *Cisco TelePresence Conductor with Cisco Unified Communications Manager Deployment Guide*. This guide provides details on how to:

- Extend the TelePresence Conductor integration with Unified CM to a cluster of TelePresence Conductors for ad hoc and rendezvous calls.
- Back up a TelePresence Conductor cluster.
- Remove a TelePresence Conductor peer from Unified CM for ad hoc and rendezvous calls.
- Upgrade a TelePresence Conductor cluster.

## Related documentation

For details on how to integrate a TelePresence Conductor cluster with Cisco VCS see either *Cisco TelePresence Conductor Clustering with Cisco VCS (Policy Server) Deployment Guide* or *Cisco TelePresence Conductor Clustering with Cisco VCS (B2BUA) Deployment Guide* depending on the type of Cisco VCS deployment used.

For more details on Unified CM not covered in this deployment guide, including how to implement a Unified CM or Unified CM cluster please reference the documentation on Cisco.com under the Cisco Unified Communications Manager, http://www.cisco.com/en/US/products/sw/voicesw/ps556/index.html.

For details on how to deployUnified CM, TelePresence Conductor, and the Conference bridges in an end-to-end secure network see *Cisco TelePresence Conductor with Cisco Unified Communications Manager Deployment Guide*.

## About TelePresence Conductor clustering

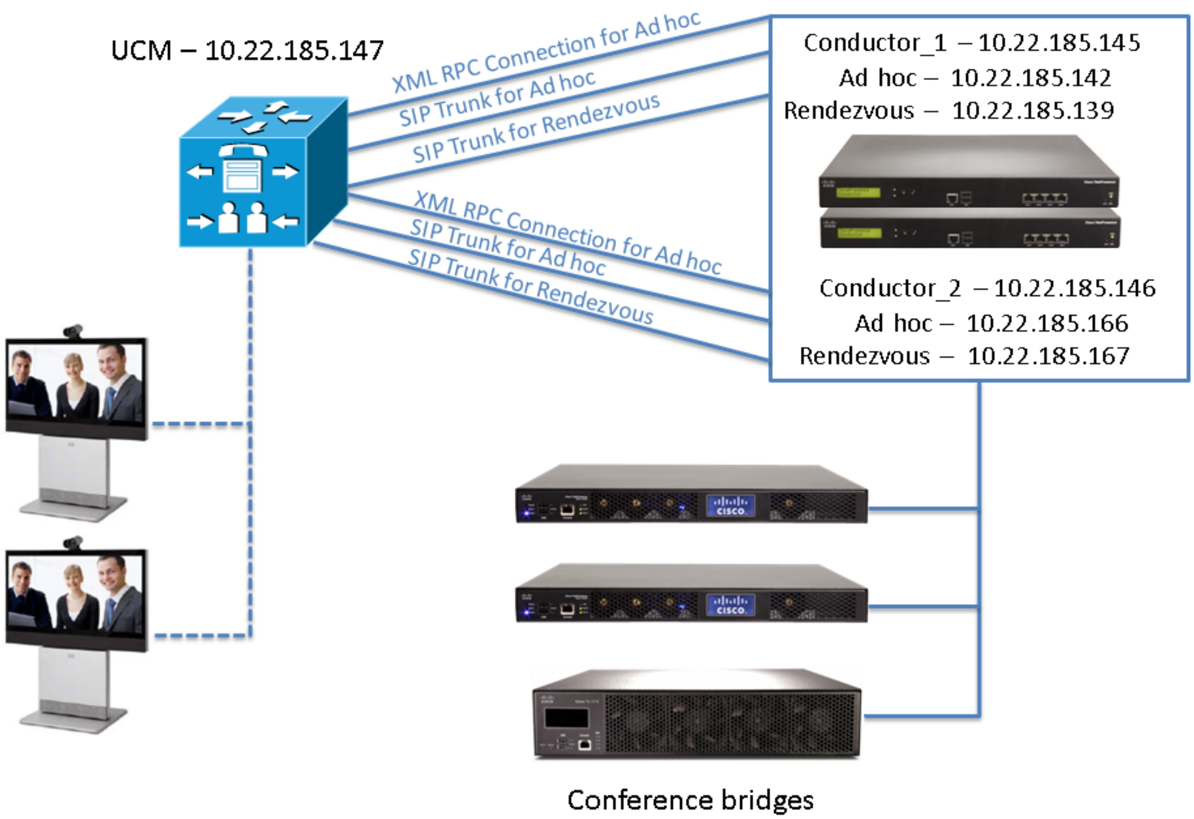
Clusters of TelePresence Conductors are used to provide redundancy in the rare case of the failure of an individual TelePresence Conductor (for example, due to a network or power outage). Each TelePresence Conductor is a peer of the other TelePresence Conductors in the cluster. Each peer knows about all conferences. It can add callers to conferences created by other peers and it can create conferences that it or other peers can add calls to.

The process to integrate a cluster of TelePresence Conductors depends upon whether the TelePresence Conductor cluster is communicating with a Cisco Video Communication Server (Cisco VCS) or a Cisco Unified Communications Manager (Unified CM). This document explains the process of creating and integrating a cluster of TelePresence Conductor peers with Unified CM.

To handle a cluster of TelePresence Conductor peers the Unified CM will be configured to have direct links to all the TelePresence Conductors in the cluster. If one TelePresence Conductor fails, Unified CM will then route the call to a different TelePresence Conductor for call completion. This process is transparent to the user and offers virtually no interruption in service.

# Example network deployment

This document uses the example network shown in the diagrams below as the basis for the deployment configuration described. During configuration, refer back to these diagrams to see the relationship between a Unified CM cluster and a redundant set of TelePresence Conductors.




## Cisco TelePresence network elements

### Unified CM

The Unified CM acts as a call processor for routing voice and video device calls. It works with other infrastructure devices in the network to process call requests.

### Conference bridges

Conference bridges are network devices that enable multiple video calls to come together in a multipoint video conference. This version of the TelePresence Conductor supports the conference bridge types TelePresence MCU and TelePresence Server.

### Endpoints

Endpoints are devices that receive and make video calls. They can be software clients on PCs and Macs such as Cisco Jabber Video for TelePresence, desktop endpoints such as the 9971 and EX90, or room systems such as the MX300.

# Creating a TelePresence Conductor cluster

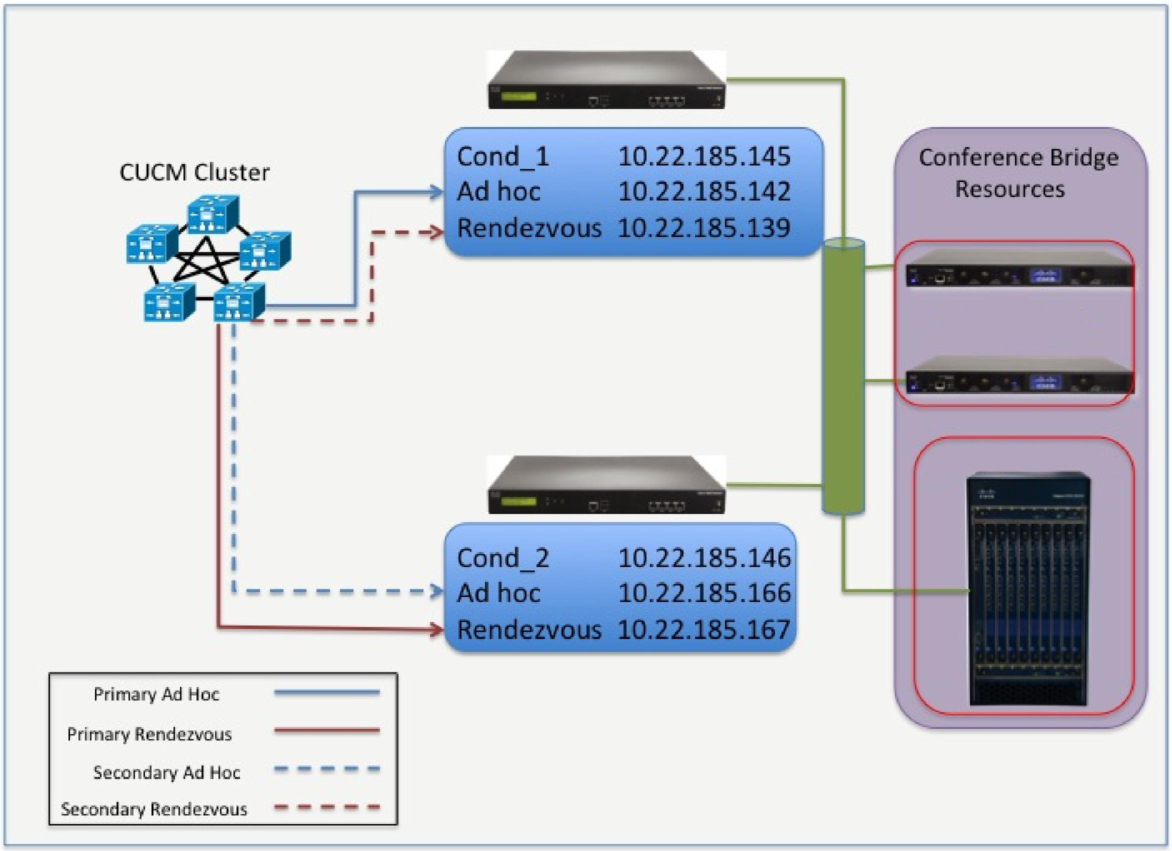
## Prerequisites

Before starting the configuration, ensure you have met the following criteria:

- A standalone TelePresence Conductor has been configured to work with a Unified CM and at least one conference bridge according to the *Cisco TelePresence Conductor with Cisco Unified Communications Manager Deployment Guide*.
- Every TelePresence Conductor to be used in the cluster must be running the same version of XC software. TelePresence Conductor clustering with Unified CM is supported in version XC2.0 and later.
- If using full capacity TelePresence Conductors, up to three peers can be clustered and all peers must be full capacity versions.
- If using TelePresence Conductor Select, up to two peers can be clustered and both peers must be a TelePresence Conductor Select.
- The Unified CM must be running version 8.6.2 or later (version 10.x or later is highly recommended).
- Enough unique IP addresses are available to configure each TelePresence Conductor peer with addresses to fulfill the requirements for ad hoc and rendezvous type call configuration.
  Each cluster peer will need, at minimum, an IP address for management plus an IP address for ad hoc conferences and another for rendezvous conferences.Additional IP addresses for ad hoc and rendezvous conferences will be required if multiple locations are handled.
- All TelePresence Conductor cluster peers must be configured to use either the same NTP servers, or NTP servers that are very closely synchronized. The NTP servers can be viewed and configured on the **Time** page (**System > Time**).
- All TelePresence Conductor cluster peers must be located closely enough so that there is a maximum round trip time of 30 milliseconds between any pair of cluster peers.
- Every conference bridge in use by TelePresence Conductor must be reachable by every TelePresence Conductor peer over HTTP/HTTPS and SIP TLS.
- For information on the ports that must be open between the TelePresence Conductor peers see Appendix 3: IP ports and protocols [p.44].
- We highly recommend that you take a backup on the initial cluster peer before adding it to the cluster.

## Integration overview

As part of a solid network design, implementation of redundancy within the system is critical.  This can be achieved for a Unified CM and TelePresence Conductor integration using additional TelePresence Conductors configured as additional options for Unified CM to use to place ad hoc and rendezvous calls. The diagram below depicts a resilient scenario in a single site design. We recommend that when configuring the Unified CM and TelePresence Conductor integration, to ensure that the primary TelePresence Conductor for ad hoc calls, **Conductor_1**, is the secondary TelePresence Conductor for rendezvous calls and the opposite configuration for **Conductor_2**, where it is the primary TelePresence Conductor for rendezvous calls and secondary for the ad hoc calls, or that ad hoc and rendezvous calls use round robin so that calls are load balanced across the TelePresence Conductor peers.




In a design where a single Unified CM cluster or multiple Unified CM clusters support multiple CAC locations, TelePresence Conductor must be configured with separate locations for each Unified CM CAC location. In addition, TelePresence Conductor must be configured to use conference bridge resources that are in the relevant Unified CM location; otherwise if this design is not followed the Unified CM CAC model will be broken.

**Note:** For ad hoc conferences the conference bridges to use are indirectly configured by the template that is configured on the TelePresence Conductor's **Locations** page (Conference template > Service Preference > Conference bridge pools > Conference bridges). The conference bridges to use for rendezvous conferences are defined by the alias dialed (Conference alias > Conference template > Service Preference > Conference bridge pools > Conference bridges) – therefore for rendezvous conferences the prefix must be location-specific.

# Configuring TelePresence Conductor

## Task 1: Checking the configuration of the initial peer

1. Decide which TelePresence Conductor is to be the initial peer. For the purposes of this example, we shall refer to this peer as **Conductor_Initial**.
   **Note:** The configuration of this system will be shared with all other peers as they are added to the cluster, unless the configuration is peer-specific. For information on which configuration is peer-specific see Peer-specific configuration [p.36].

2. Verify that no other TelePresence Conductor already has **Conductor_1**'s IP address in their clustering peers list. To do this verification:
   a. Log into every TelePresence Conductor as a user with administrator rights.
   b. Go to **System > Clustering**.
   c. Ensure that all **Peer X IP address** fields (X = 1, 2, and 3) on this page do not have **Conductor_1**'s IP address.
      If they do:
      i. Delete that Peer IP address.
      ii. Click **Save**.
      iii. Go to **Maintenance > Restart options**.
      iv. Click **Restart**.
3. Log into **Conductor_1** as a user with administrator rights.
4. Ensure that **Conductor_1** has a valid and working NTP server configured:
   a. Go to **System > Time**.
   b. In the **Status** section at the bottom of the page, the **State** should be *Synchronized*:

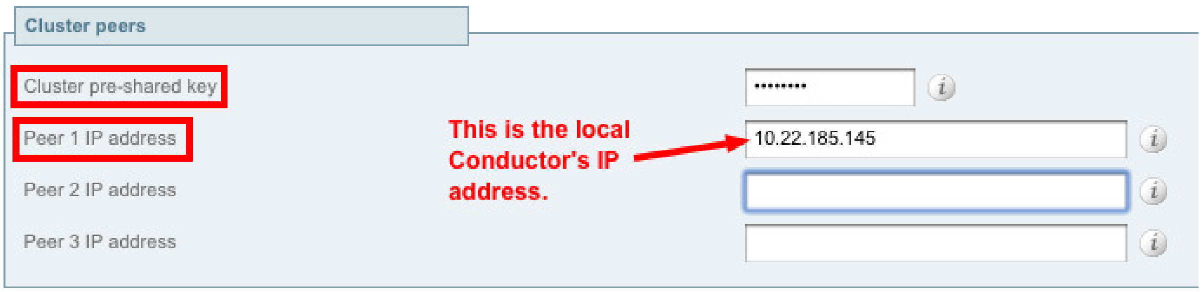

5. Ensure that **Conductor_1** has the correct DNS settings configured:
   a. Go to **System > DNS**.
   b. Ensure that **Conductor_1** has at least one valid DNS server configured.
   c. Ensure that **Conductor_1** has the correct **Domain name** and **System host name** configured:
      <System host name>.<domain name> = FQDN of this TelePresence Conductor.
6. Ensure that **Conductor_1** has the correct Clustering settings applied:
   a. Go to **System > Clustering**.
   b. Ensure that all **Peer X IP address** fields (X = 1, 2, and 3) on this page are blank. If not:
      i. Delete any entries.
      ii. Click **Save**.
   c. Ensure that **Conductor_1** has no **Cluster pre-shared key** configured. If there is a value in the **Cluster pre-shared key** field:
      i. Delete the entry.
      ii. Click **Save**.
      iii. Go to **Maintenance > Restart options**.
      iv. Click **Restart**.

## Task 2: Creating a cluster of one peer

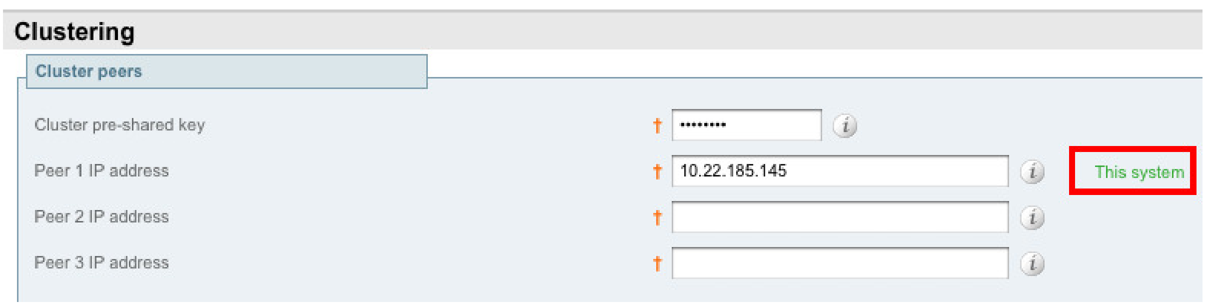
1. On **Conductor_1**, go to **System > Clustering**.
2. Enter the following values in the relevant fields:

| | |
|---|---|
| **Cluster pre-shared key** | Enter a password (this will be the same for all peers). |
| **Peer 1 IP address** | Enter the IP address of this TelePresence Conductor peer, **Conductor_1** (this is the initial peer in the cluster from which the initial configuration will be replicated from to all other peers in the cluster). |

| | |
|---|---|
| **Peer 2 IP address** | Leave blank at this point in the configuration. |
| **Peer 3 IP address** | Leave blank at this point in the configuration. |



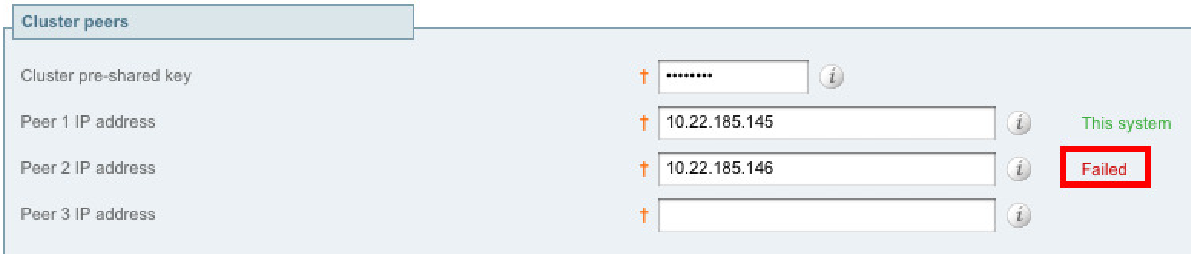

3. Click **Save**.
4. Go to **Maintenance > Restart options**.
5. Click **Restart**.
6. Log into **Conductor_1** as a user with administrator rights.
7. Go to **System > Clustering**.
8. Verify the status of this peer. It should have **This System** in green next to the IP address.




## Task 3: Configuring the cluster to accept the new peer

**Note:** These instructions specify how to add a second peer to the cluster. A third peer can be added in a similar manner using **Peer 3 IP address**, and configuring both peer 1 and peer 2 before configuring peer 3.

1. Log into the initial TelePresence Conductor, **Conductor_1**, as a user with administrator rights.
2. Go to **System > Clustering**.
3. In the **Peer 2 IP address** field, enter the new peer's IP address. For the purposes of this example we shall refer to this peer as **Conductor_2**.
4. Click **Save**.
5. Notice the peer's **Status** is *Failed*. This is normal for this stage of the configuration process.




6. Go to **Maintenance > Restart options**.
7. Click **Restart**.

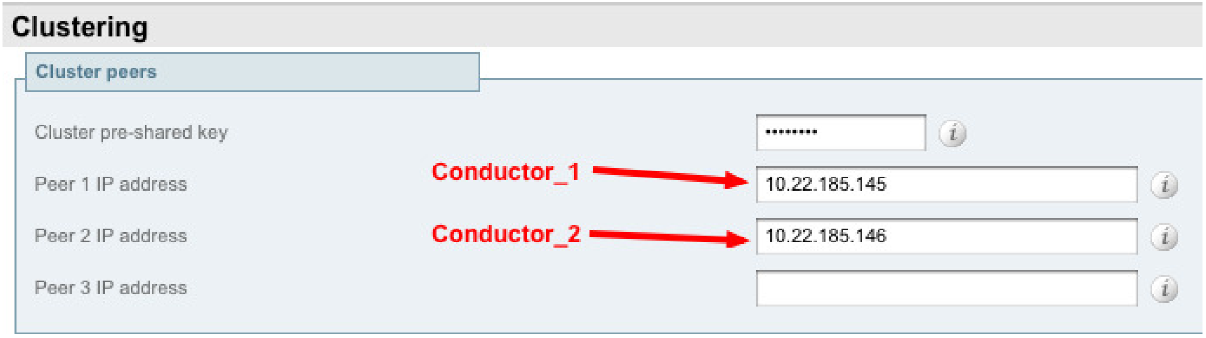
## Task 4: Checking the configuration of the second peer

1. Log into the new peer, **Conductor_2**, as a user with administrator rights.
2. Ensure that **Conductor_2** has a valid and working NTP server configured:
   a. Go to **System > Time**.
   b. In the **Status** section at the bottom of the page, the **State** should be *Synchronized*:


3. Ensure that **Conductor_2** has the correct DNS settings configured:
   a. Go to **System > DNS**.
   b. Ensure that **Conductor_2** has at least one valid DNS server configured.
   c. Ensure that **Conductor_2** has the correct **Domain name** and **System host name** configured: <System host name>.<domain name> = FQDN of this TelePresence Conductor.
4. Ensure that **Conductor_2** has the correct Clustering settings applied:
   a. Go to **System > Clustering**.
   b. Ensure that all **Peer X IP address** fields (X = 1, 2, and 3) on this page are blank. If not:
      i. Delete any entries.
      ii. Click **Save**.
   c. Ensure that **Conductor_2** has no **Cluster pre-shared key** configured. If there is a value in the **Cluster pre-shared key** field:
      i. Delete the entry.
      ii. Click **Save**.
      iii. Go to **Maintenance > Restart options**.
      iv. Click **Restart**.

## Task 5: Configuring the second peer to join the cluster

1. On **Conductor_2**, go to **System > Clustering**.
2. In the **Cluster pre-shared key** field, enter the same password that was used for the initial peer, **Conductor_1**.
3. In the **Peer 1 IP address** field, enter the IP address of the initial peer, **Conductor_1**.
4. In the **Peer 2 IP address** field, enter the IP address of the local TelePresence Conductor, **Conductor_2**.

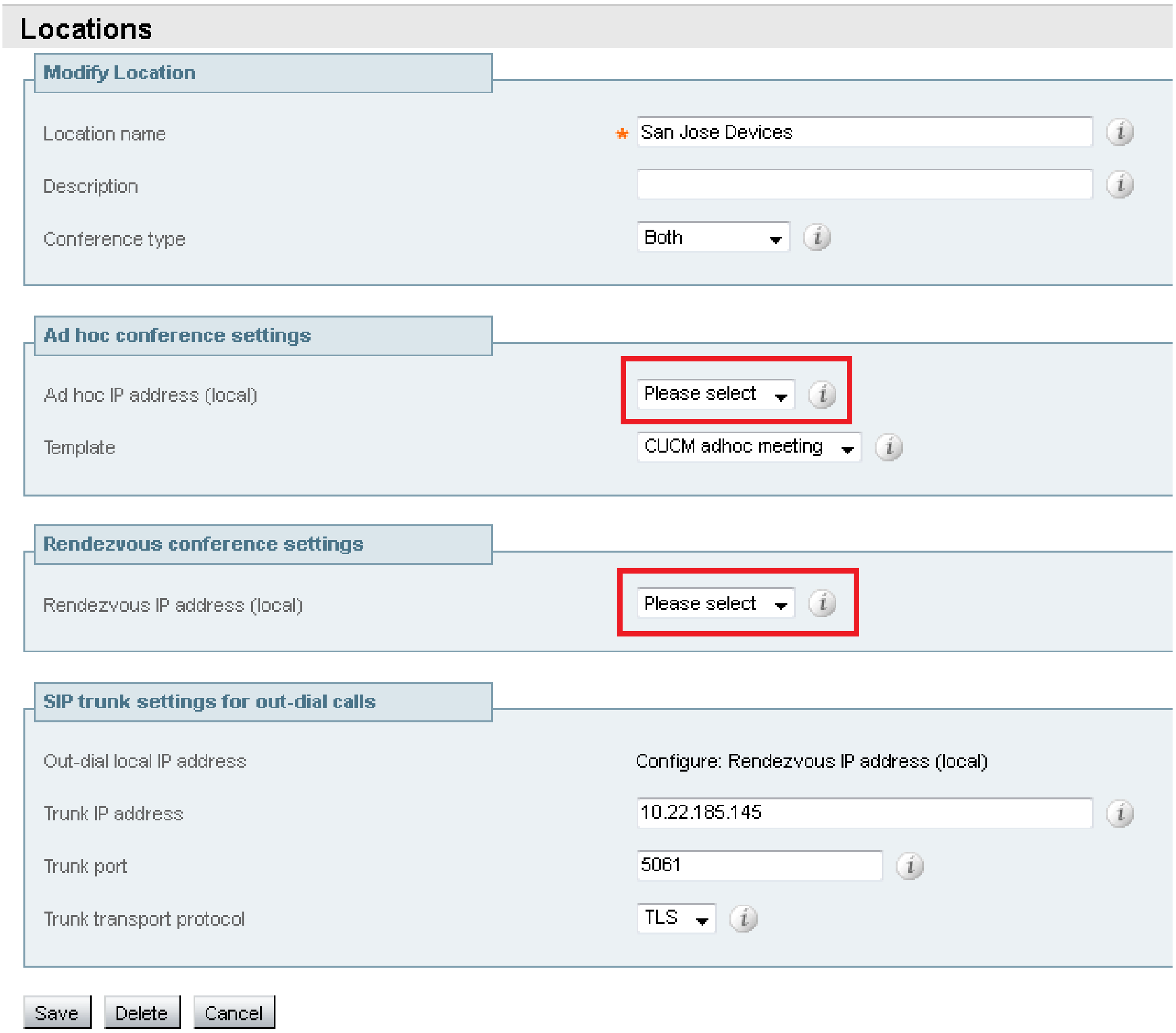



5. Click **Save**.
   **Note:** Ensure that the initial peer is accessible via the web and is not still restarting. If the second peer is restarted while the initial peer is restarting, the wrong peer may be selected as the initial peer and configuration may be lost.
6. Go to **Maintenance > Restart options**.
7. Click **Restart**.
8. Log back into **Conductor_2** as a user with administrator rights.
9. Go to **System > Clustering**.
10. Verify the **Status** of each peer. It should have **This system** in green next to this system's IP address and show **Active** for the other peer.

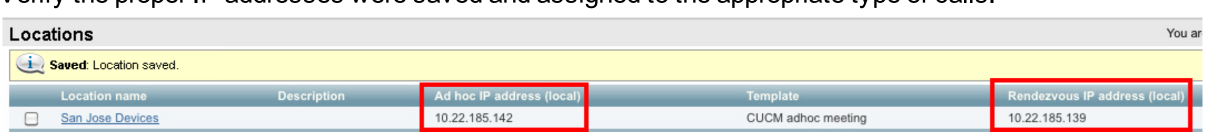

## Task 6: Updating the Location settings on the second peer.

As a part of the clustering process the configuration of Locations, conference aliases, conference templates, Service Preferences and conference bridges are replicated. The Locations' IP addresses, however, need to be configured on **Conductor_2**.

1. Log into the new peer, **Conductor_2**, as a user with administrator rights.
2. Go to **Conference configuration > Locations**.
3. Click **View/Edit** next to the existing Location name.
4. For an ad hoc Location select the appropriate ad hoc IP address from the drop-down list (under the **Ad hoc** section).
5. For a rendezvous Location select the appropriate rendezvous IP address from the drop-down list (under the **Rendezvous** section).
6. For a rendezvous Location ensure that the **Trunk port** and **Trunk transport protocol** match (typically *5061* for TLS and *5060* for TCP).




7. Click **Save**.
8. Verify the proper IP addresses were saved and assigned to the appropriate type of calls.




9. Repeat for each Location configured.

# Ensuring that Unified CM trusts TelePresence Conductor's server certificate and vice versa

For Unified CM and TelePresence Conductor to establish a TLS connection with each other:

- TelePresence Conductor and Unified CM must both have valid server certificates loaded (you must replace the TelePresence Conductor's default server certificate with a valid server certificate)
- TelePresence Conductor must trust Unified CM’s server certificate (the root CA of the Unified CM server certificate must be loaded onto TelePresence Conductor)

- Unified CM must trust TelePresence Conductor's server certificate (the root CA of the TelePresence Conductor server certificate must be loaded onto Unified CM)

See Appendix 4: Ensuring that Unified CM trusts TelePresence Conductor's server certificate and vice versa [p.45] in this document for more information on how to ensure that Unified CM trusts the TelePresence Conductor server certificate.

See *Cisco TelePresence Conductor Certificate Deployment Guide* for full details about loading certificates and how to generate CSRs on TelePresence Conductor to acquire certificates from a Certificate Authority (CA).

**Note:** In a clustered environment, you must install CA and server certificates on each peer/node individually.

We strongly recommend that you do not use self-signed certificates in a production environment.

# Updating the secure SIP trunk security profile

On the Unified CM go to **System > Security > SIP Trunk Security Profile** and select the SIP Trunk Security Profile for the TelePresence Conductor:

1. In the **X.509 Subject Name** field add the subject name or an alternate subject name provided by the secondary TelePresence Conductor peer in its certificate. (Multiple X.509 names can be separate by a space, comma, semicolon or colon.)
2. Click **Save**.
3. Repeat for the third TelePresence Conductor cluster peer if required.

# Configuring Unified CM for ad hoc conferences

**Note:** The phone/endpoint used to initiate an ad hoc conference must have a conference button. Phones/endpoints that do not have a conference button may still be participants in an ad hoc conference, but they must be added to the conference by a phone/endpoint that has a conference button.

## Task 7: Adding a SIP trunk to the secondary TelePresence Conductor for ad hoc conferences

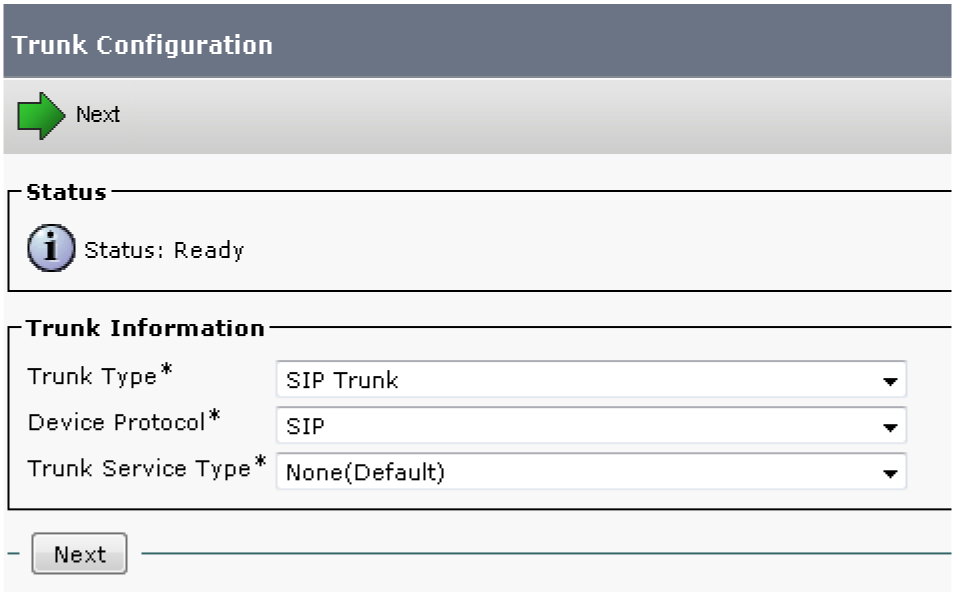
From Unified CM version 10.x onwards a SIP trunk between Unified CM and TelePresence Conductor must be explicitly configured for ad hoc conferences. The task is not required when running an earlier version of Unified CM.

Separate SIP trunks are required for rendezvous and ad hoc conferences.

To configure a SIP trunk to the TelePresence Conductor for ad hoc conferences:

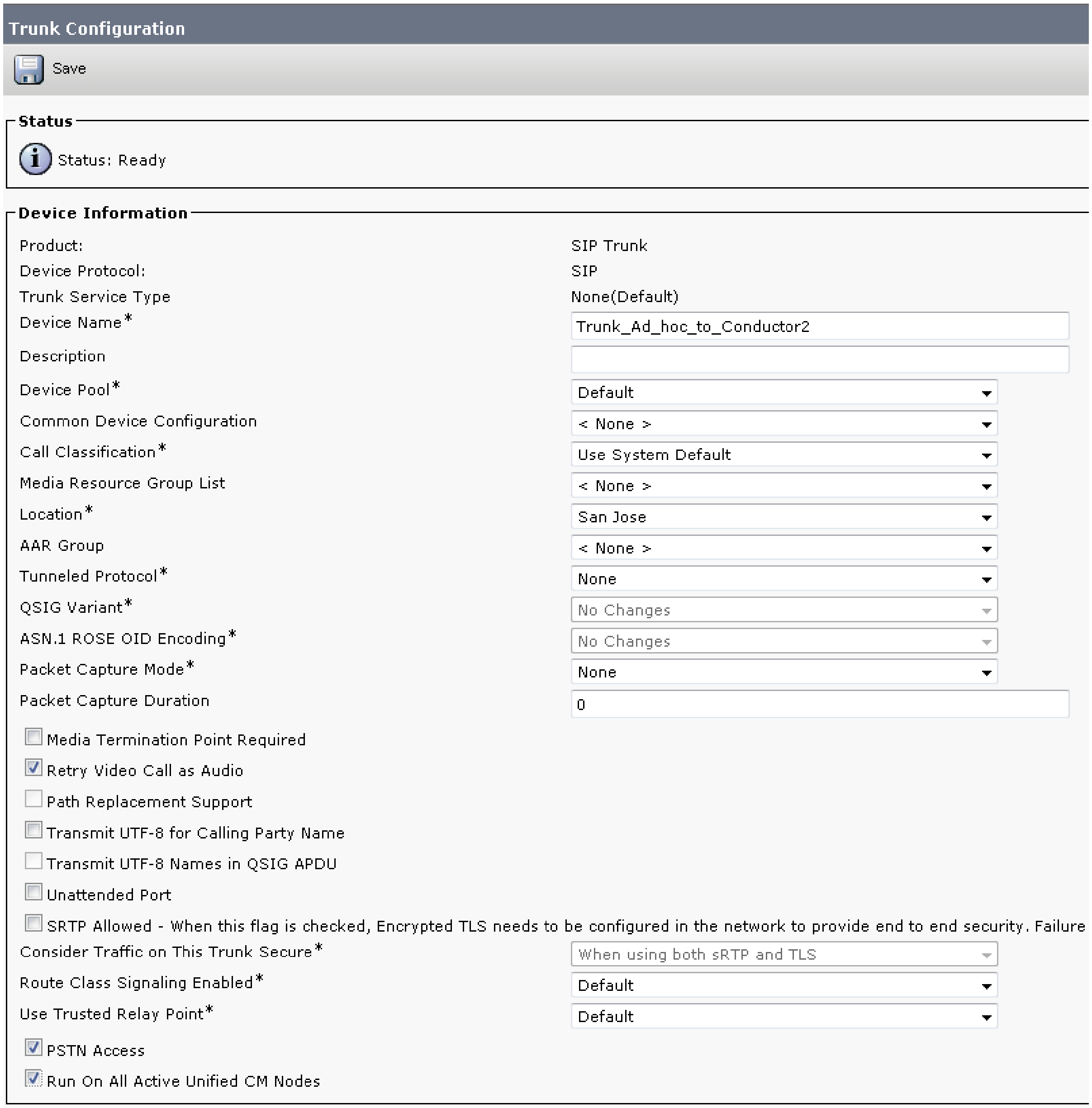
1. Go to **Device > Trunk**.
2. Click **Add New** to create a new SIP trunk.
3. Enter the following into the relevant fields:

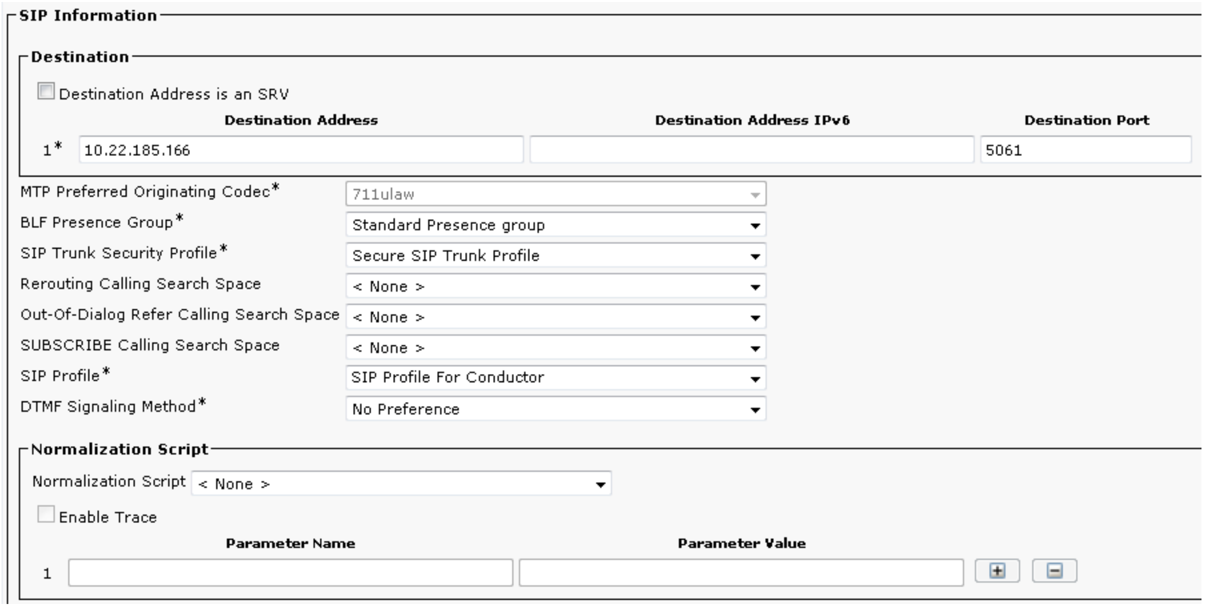
| | |
|---|---|
| **Trunk Type** | Select *SIP Trunk*. |
| **Device Protocol** | Leave as default: *SIP*. |
| **Trunk Service Type** | Leave as: *None(Default)*. |




4. Click **Next**.
5. Enter the following into the relevant fields, leave other fields as their default values:

| | |
|---|---|
| **Device Name** | Enter a trunk name. |
| **Device Pool** | Select the appropriate Device Pool. |
| **Location** | Select the same Location that was used for **Conductor_1**. |
| **Run On All Active Unified CM Nodes** | Tick this setting. |
| **Destination Address** | Enter **Conductor_2**’s Location-specific ad hoc IP address. This IP address is the **Ad hoc IP address (local)** configured in the **Ad hoc conference settings** section on the TelePresence Conductor’s **Location** page (**Conference configuration > Locations**). |
| **SIP Trunk Security Profile** | Select the *Secure SIP Trunk Profile* from the drop-down list. |
| **SIP Profile** | Select the same SIP Profile that was used for **Conductor_1**. |
| **Normalization Script** | If you specified a normalization script on the Trunk to **Conductor_1**, select the same Normalization script here. |


Trunk Configuration
Save
Status
Status: Ready
Device Information
Product:
SIP Trunk
Device Protocol:
SIP
Trunk Service Type
None(Default)
Device Name*
Trunk_Ad_hoc_to_Conductor2
Description
Device Pool*
Default
Common Device Configuration
< None >
Call Classification*
Use System Default
Media Resource Group List
< None >
Location*
San Jose
AAR Group
< None >
Tunneled Protocol*
None
QSIG Variant*
No Changes
ASN.1 ROSE OID Encoding*
No Changes
Packet Capture Mode*
None
Packet Capture Duration
0
Media Termination Point Required
Retry Video Call as Audio
Path Replacement Support
Transmit UTF-8 for Calling Party Name
Transmit UTF-8 Names in QSIG APDU
Unattended Port
SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure
Consider Traffic on This Trunk Secure*
When using both sRTP and TLS
Route Class Signaling Enabled*
Default
Use Trusted Relay Point*
Default
PSTN Access
Run On All Active Unified CM Nodes




6. Click **Save**.
7. Click **Reset**.

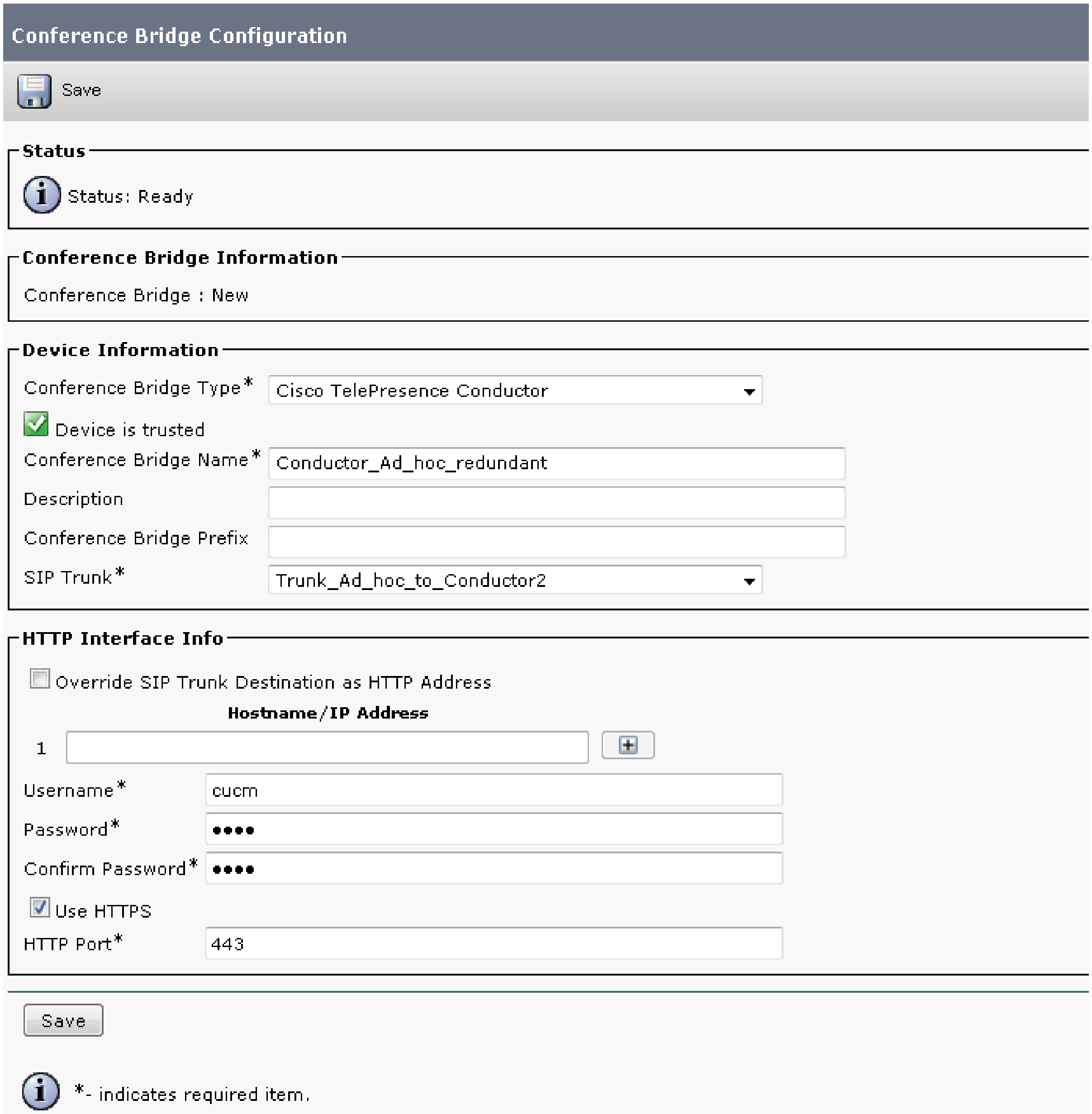
## Task 8: Adding the secondary TelePresence Conductor as a Conference Bridge

**Note:** The instructions in this step are for Unified CM version 10.0 or later. For version 8.6.2 go to Appendix 1: Unified CM version 8.6.2 configuration [p.39] and for version 9.x go to Appendix 2: Unified CM version 9.x configuration [p.41].

To configure Unified CM version 10.0 or later with TelePresence Conductor:

1. Go to **Media Resources > Conference Bridge**.
2. Click **Add New** to create a new Conference Bridge.
3. Enter the following into the relevant fields, leave other fields as their default values:

| | |
|---|---|
| **Conference Bridge Type** | Select *Cisco TelePresence Conductor* |
| **Conference Bridge Name** | Enter the TelePresence Conductor's name |
| **SIP Trunk** | Select the SIP trunk you created in Task 7: Adding a SIP trunk to the secondary TelePresence Conductor for ad hoc conferences [p.13] |
| **Username** | Enter the username of the TelePresence Conductor administration user. This appears on the TelePresence Conductor's **Administrator accounts** page (**Users > Administrator accounts**) |
| **Password** | Enter the password of the TelePresence Conductor administration user |
| **Use HTTPS** | We recommend that you tick this box. |
| **HTTP Port** | Enter ‘443’. |



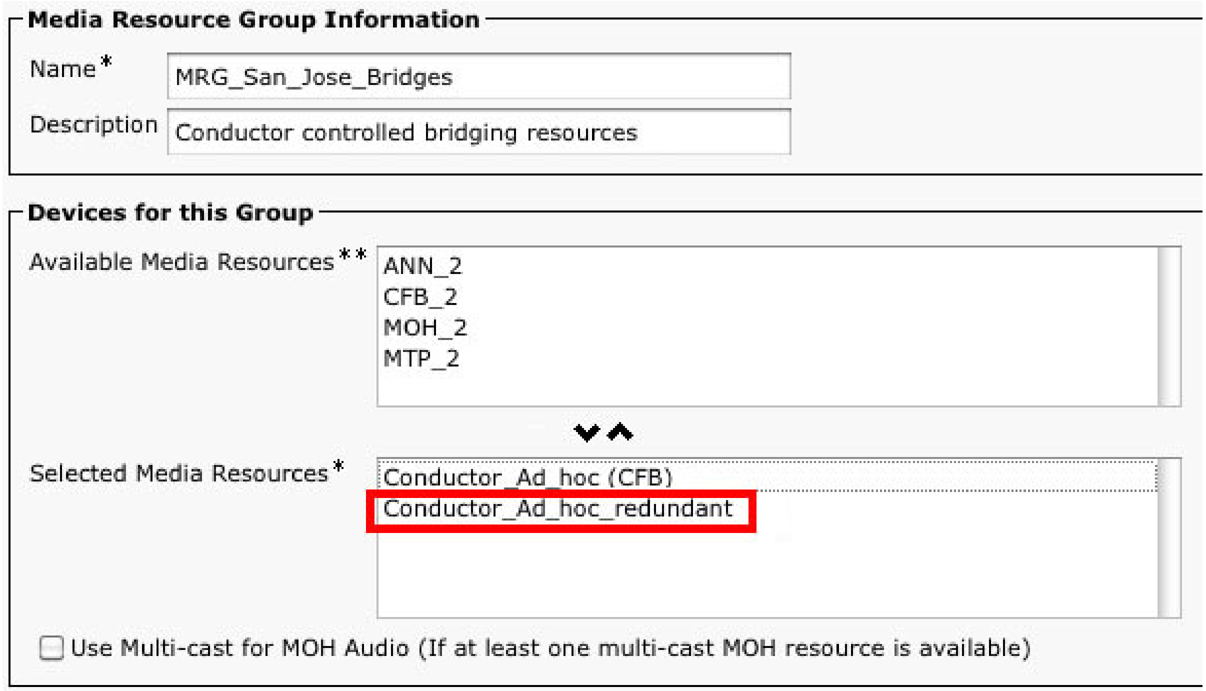

Click **Save**.
Click **Reset**.

## Task 9: Adding the secondary TelePresence Conductor to an MRG and MRGL

To configure the Unified CM with the secondary TelePresence Conductor in a Media Resource Group (MRG):

1. Go to **Media Resources > Media Resource Group**.
2. Click **Find** to list the Media Resource Groups.
3. Click on **MRG_San_Jose_Bridges**.
4. Move the TelePresence Conductor media bridge (the conference bridge configured in Task 8: Adding the secondary TelePresence Conductor as a Conference Bridge [p.16]) down to the Selected Media

Resources box. Make sure this conference bridge is the last bridge in the list as it is the redundant TelePresence Conductor.



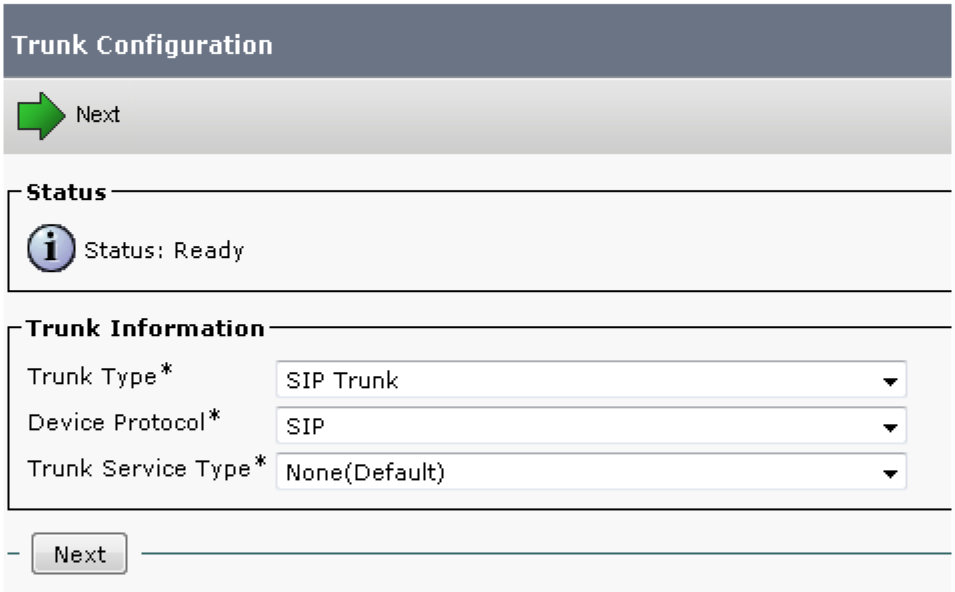

5. Click **Save**.

# Configuring Unified CM for rendezvous conferences

## Task 10: Adding a SIP trunk to the secondary TelePresence Conductor for rendezvous conferences

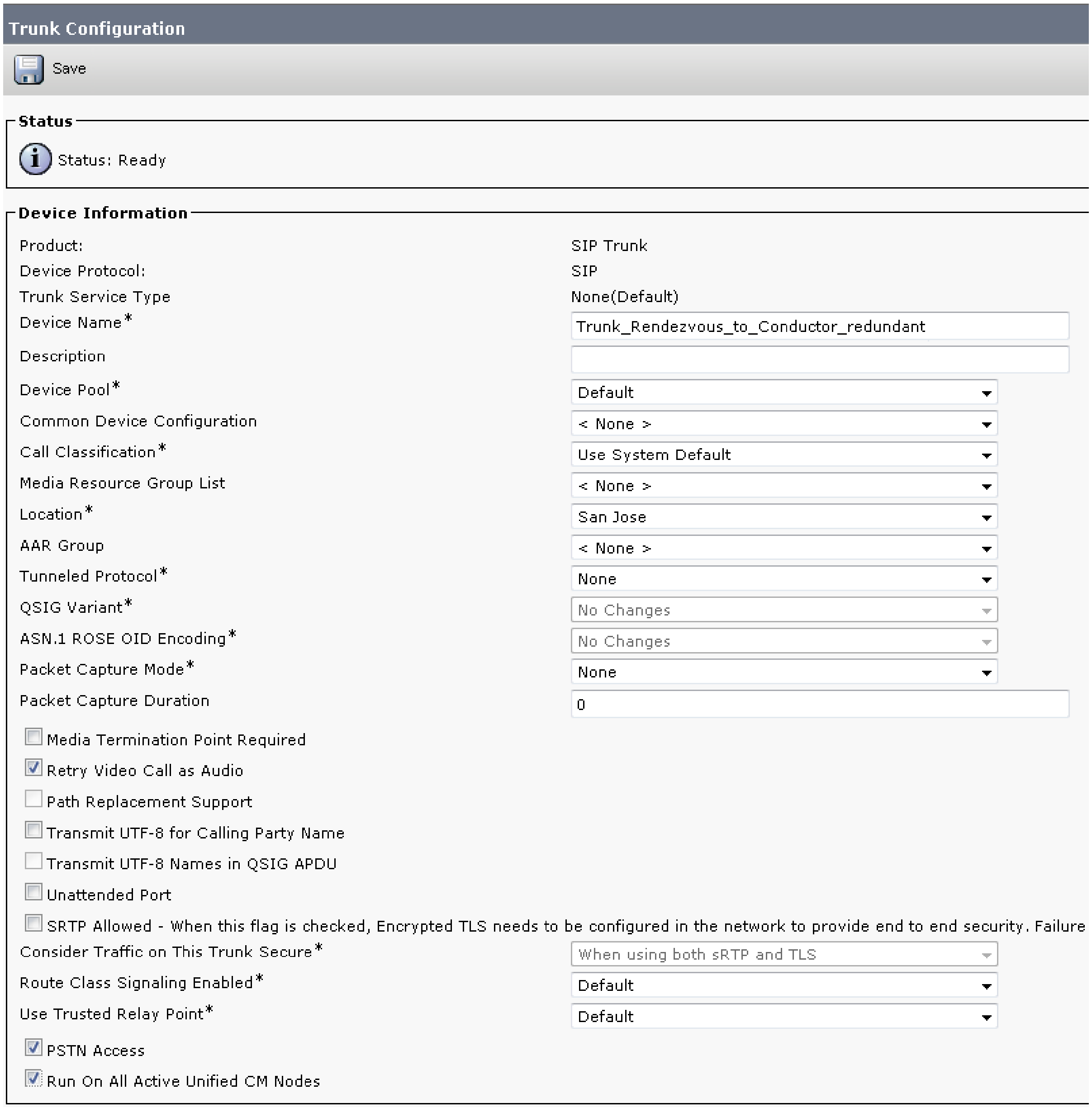
1. Go to **Device > Trunk**.
2. Click **Add New** to create a new SIP trunk.
3. Enter the following into the relevant fields, leave other fields as their default values:

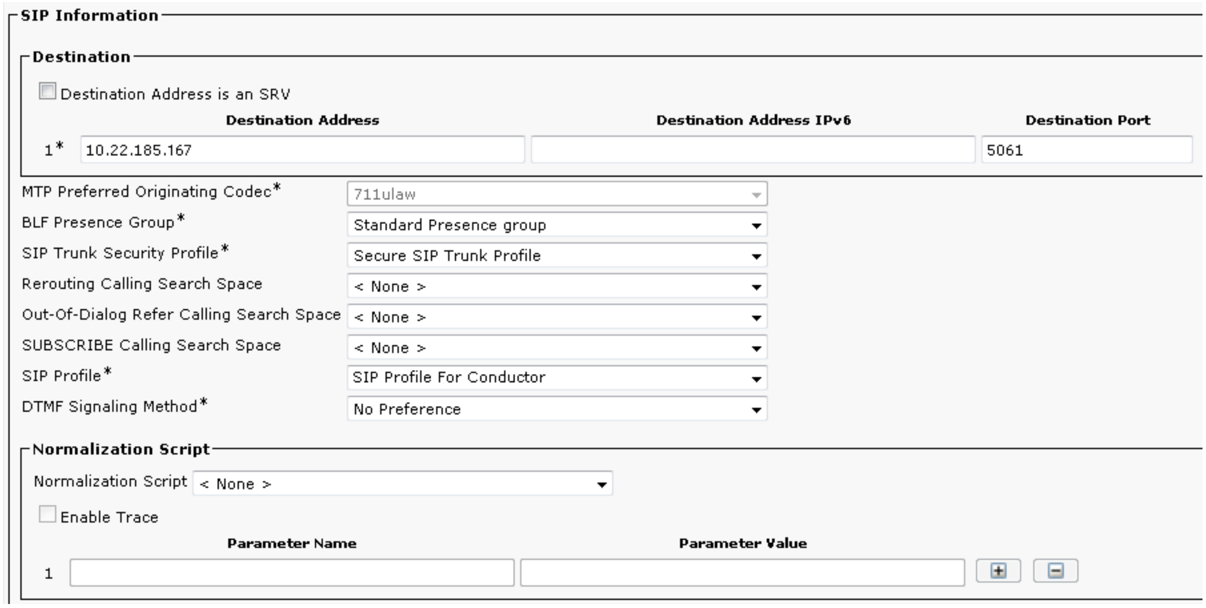
| | |
|---|---|
| **Trunk Type** | Select *SIP Trunk*. |
| **Device Protocol** | Leave as default: *SIP*. |
| **Trunk Service Type** | Leave as: *None(Default)*. |




4. Click **Next**.
5. Enter the following into the relevant fields, leave other fields as their default values:

| | |
|---|---|
| **Device Name** | Enter a trunk name |
| **Location** | Select the same Location that was used for **Conductor_1**. |
| **Device Pool** | Select the appropriate Device Pool |
| **Run On All Active Unified CM Nodes** | Tick this setting. |
| **Destination Address** | Enter **Conductor_2**'s Location-specific rendezvous IP address. This IP address is the **Rendezvous IP address (local)** configured in the **Rendezvous conference settings** section on the TelePresence Conductor's **Location** page (**Conference configuration > Locations**). |
| **SIP Trunk Security Profile** | Select the *Secure SIP Trunk Profile* from the drop-down list |
| **SIP Profile** | Select the same SIP Profile that was used for **Conductor_1**. |
| **Normalization Script** | If you specified a normalization script on the Trunk to **Conductor_1**, select the same Normalization script here. |


Trunk Configuration
Save
Status
Status: Ready
Device Information
Product: SIP Trunk
Device Protocol: SIP
Trunk Service Type None(Default)
Device Name* Trunk_Rendezvous_to_Conductor_redundant
Description
Device Pool* Default
Common Device Configuration < None >
Call Classification* Use System Default
Media Resource Group List < None >
Location* San Jose
AAR Group < None >
Tunneled Protocol* None
QSIG Variant* No Changes
ASN.1 ROSE OID Encoding* No Changes
Packet Capture Mode* None
Packet Capture Duration 0
Media Termination Point Required
Retry Video Call as Audio
Path Replacement Support
Transmit UTF-8 for Calling Party Name
Transmit UTF-8 Names in QSIG APDU
Unattended Port
SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure
Consider Traffic on This Trunk Secure* When using both sRTP and TLS
Route Class Signaling Enabled* Default
Use Trusted Relay Point* Default
PSTN Access
Run On All Active Unified CM Nodes



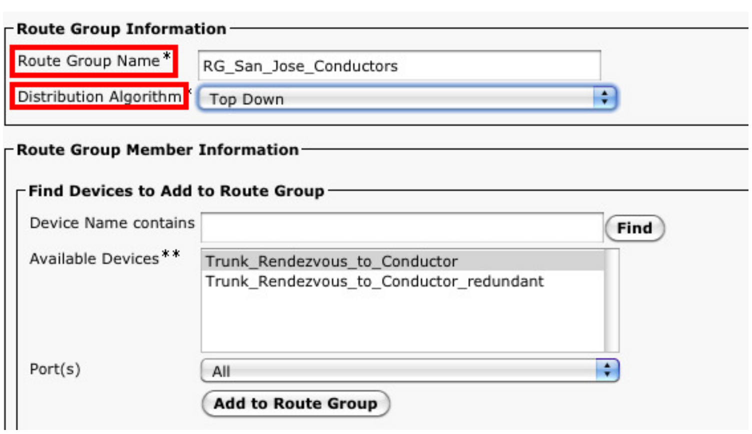

6. Click **Save**.
7. Click **Reset**.

## Task 11: Adding a route group for the SIP trunks

To configure a route group to use the SIP trunks to the TelePresence Conductor for rendezvous calls:

1. Go to **Call Routing > Route/Hunt > Route Group**.
2. Click **Add New** to create a new route pattern.
3. Enter the following into the relevant fields, leave other fields as their default values:

| | |
|---|---|
| **Route Group Name** | Enter a route group name |
| **Distribution Algorithm** | Select *Top Down* |




4. Under the Route Group Member section, highlight **Trunk_Rendezvous_to_Conductor** and click **Add to Route Group**.
5. Under the Route Group Member section, highlight **Trunk_Rendezvous_to_Conductor_redundant** and click **Add to Route Group**.

6. Once both are added, they will appear in the **Current Route Group Members** section.


7. For load balancing rendezvous calls to the opposite TelePresence Conductor to the one used for ad hoc calls, ensure that **Trunk_Rendezvous_to_Conductor_redundant** is moved to the top of the list.
8. Click **Save**.

**Note:** If the original SIP trunk, set up while following the Cisco TelePresence Conductor with Unified CM Deployment Guide (**Trunk_Rendezvous_to_Conductor**), is not listed, it may be in use elsewhere. To work around this:

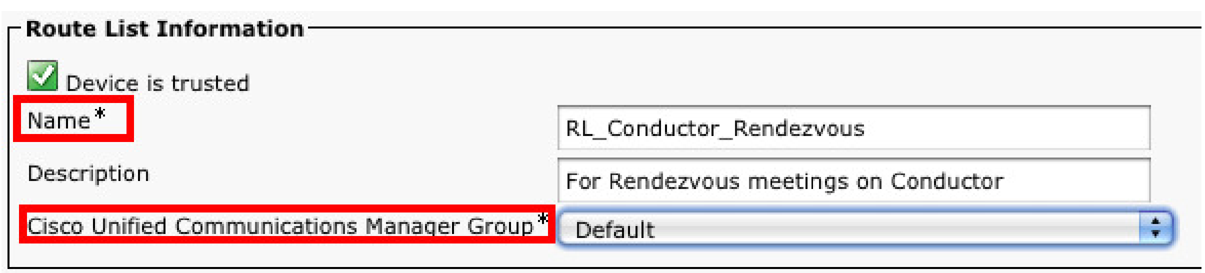
1. Create the Route Group with only the new SIP trunk (**Trunk_Rendezvous_to_Conductor_redundant**).
2. Modify the route pattern in Task 13: Editing the route pattern that matches the SIP trunk to TelePresence Conductor [p.23]
3. Return to Task 11: Adding a route group for the SIP trunks [p.21] to add the SIP trunk **Trunk_ Rendezvous_to_Conductor** to the route group.

## Task 12: Adding a route list for the route group

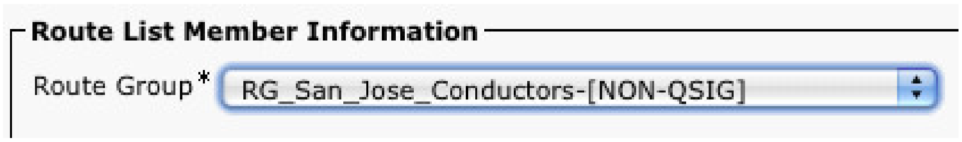
To configure a route list to use the route group that contains the SIP trunks to the TelePresence Conductor for rendezvous calls:

1. Go to **Call Routing > Route/Hunt > Route List**.
2. Click **Add New** to create a new route pattern.
3. Enter the following into the relevant fields, leave other fields as their default values:

| | |
|---|---|
| **Name** | Enter a route list name |
| **Cisco Unified Communications Manager Group** | Select the appropriate group from the drop-down list |




4. Click **Save**.
5. Click **Add Route Group**.
6. Next to the Route Group field select the route group created in Task 11: Adding a route group for the SIP trunks [p.21].

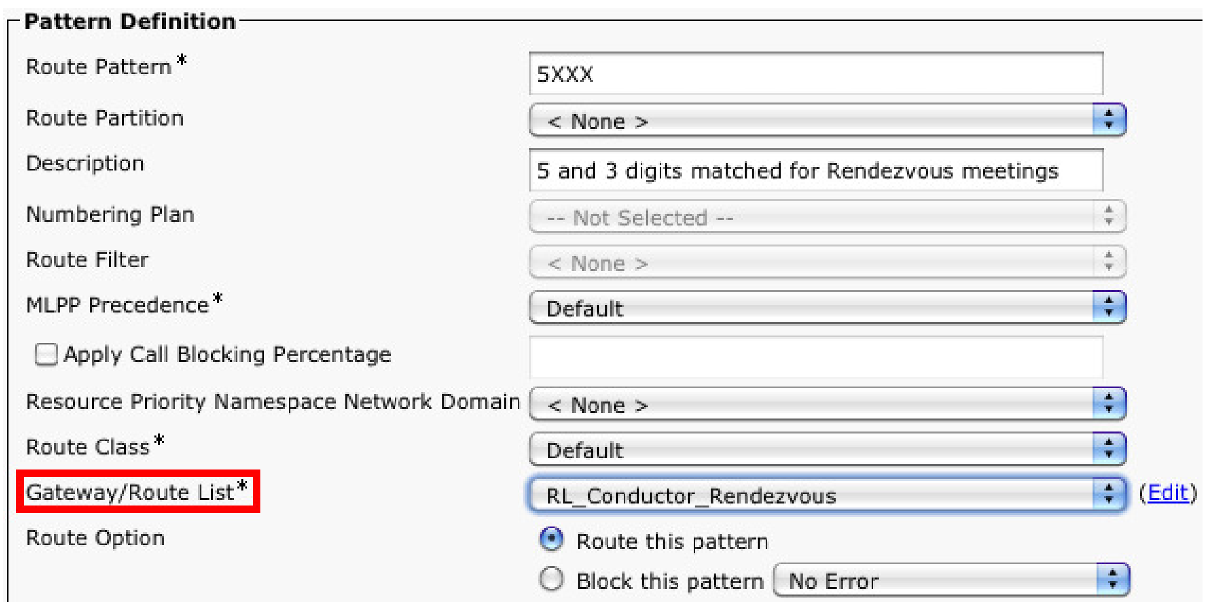



7. Click **Save**.
8. Click **Reset**.

## Task 13: Editing the route pattern that matches the SIP trunk to TelePresence Conductor

1. Go to **Call Routing > Route/Hunt > Route Pattern**.
2. Click **Find** and then select the relevant route pattern.
3. Enter the following into the relevant fields, leave other fields as their default values:

| | |
|---|---|
| **Route Pattern** | Enter a route pattern to match against the destination string |
| **Gateway/Route List** | Select the route list used in Task 12: Adding a route list for the route group [p.22] |




4. Click **Save**.

# Creating a system backup

To create a backup of TelePresence Conductor system data:

1. Go to **Maintenance > Backup and restore**.
2. Optionally, enter an **Encryption password** with which to encrypt the backup file.
   If a password is specified, the same password will be required to restore the file.
3. Click **Create system backup file**.
4. After the backup file has been prepared, a pop-up window appears and prompts you to save the file (the exact wording depends on your browser). The default name is in the format:
   **<software version>_<hardware serial number>_<date>_<time>_backup.tar.gz**.
   (The file extension is normally **.tar.gz.enc** if an encryption password is specified. However, if you use Internet Explorer to create an encrypted backup file, the filename extension will be **.tar.gz.gz** by default. These different filename extensions have no operational impact; you can create and restore encrypted backup files using any supported browser.)
   The preparation of the system backup file may take several minutes. Do not navigate away from this page while the file is being prepared.
5. Save the file to a designated location.

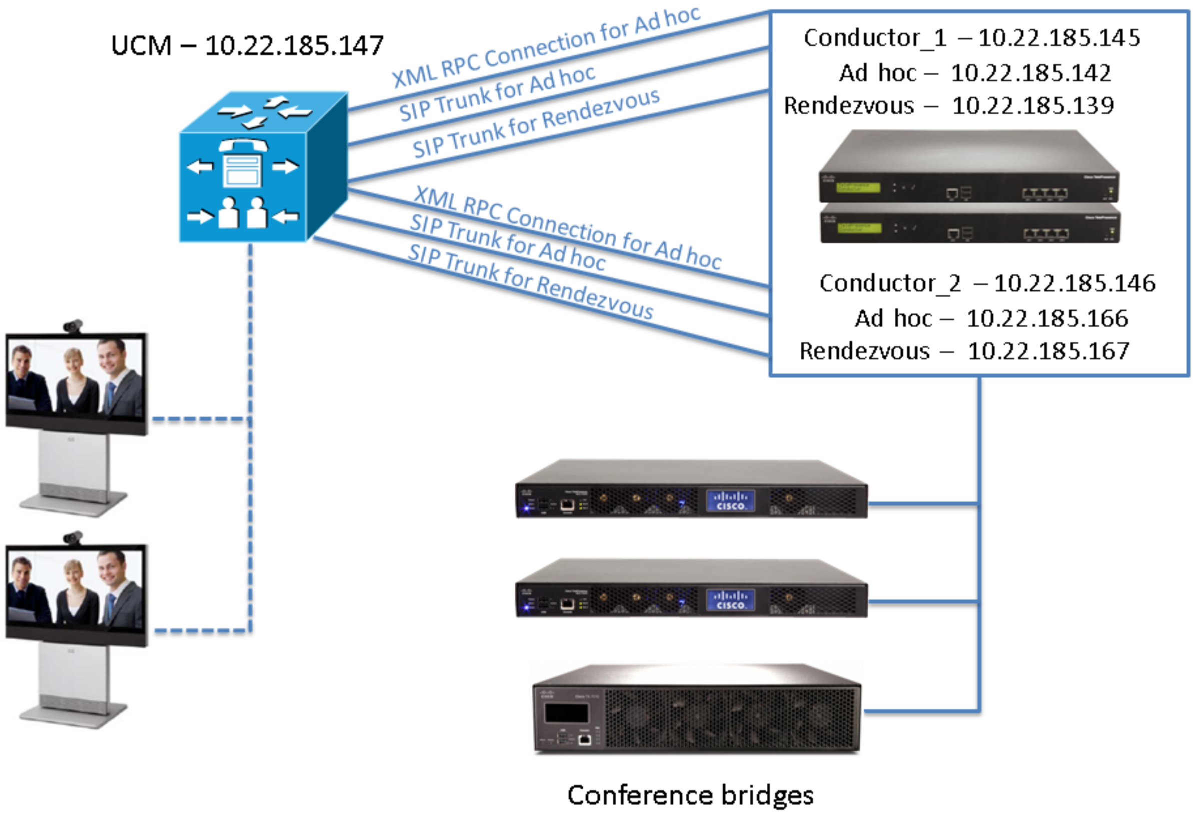
Log files are not included in the system backup file.

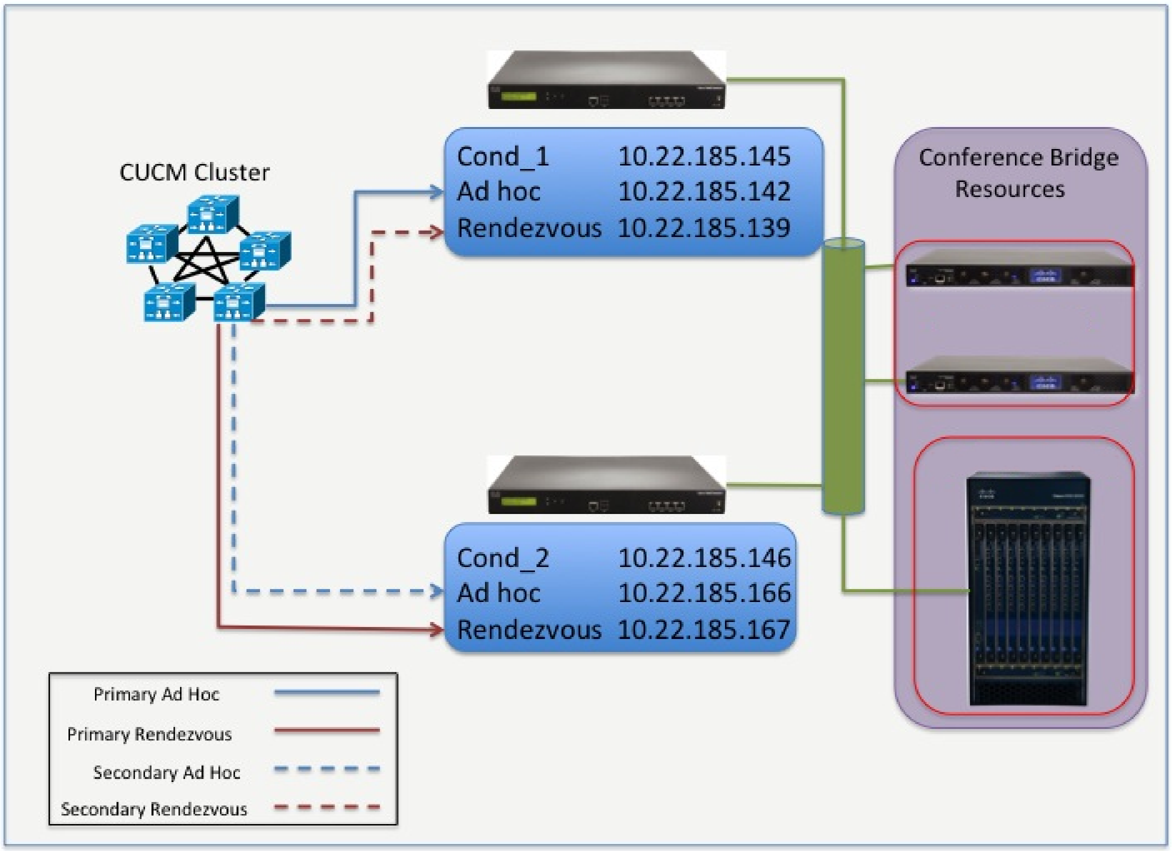
**Note:** A system backup can only be restored to the peer from which the backup was taken.

For more information see *Cisco TelePresence Conductor Administrator Guide* or the TelePresence Conductor's online help.

# Testing system configuration

Once you have completed the configuration described in the previous sections, you should test that the system is working correctly as follows. The diagrams below are references for the testing steps:





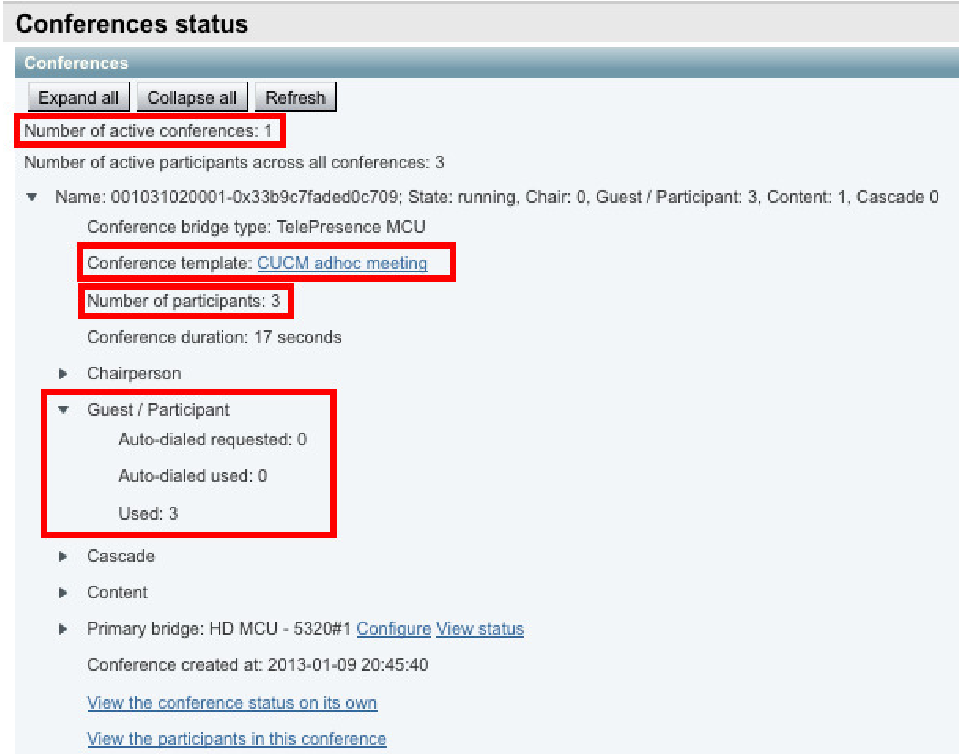

## Creating an ad hoc conference

Perform the following test with both TelePresence Conductors operational, then with one switched on and the other off, then the first one off and second on.

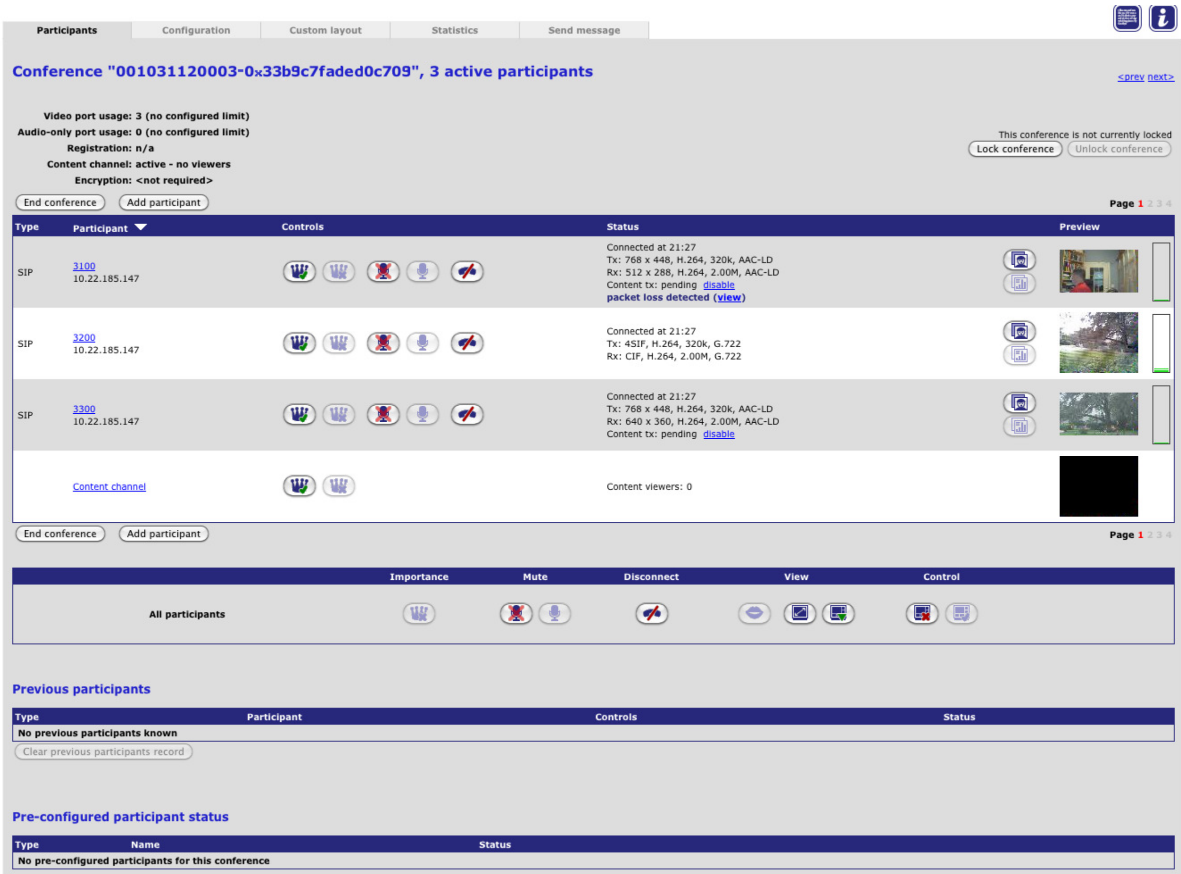
To test that three Unified CM registered endpoints can join an ad hoc conference:

1. From the 9971 dial **3100**. Verify a video and audio session is established between the 9971 and the second C20.
2. From the 9971, press the conference button and dial **3300**. Verify a video and audio session is established between the 9971 and the second C20. The call between the 9971 and second C20 has been put on hold.
   **Note:** At this point the TelePresence Conductor is not involved.
3. From the 9971 press the **Conference** tab on the screen to join the participants and move the call to a conference bridge.
   The call is now established on the MCU via **Cond_1**'s back-to-back user agent (B2BUA).

4. To verify the established call on the TelePresence Conductor, **Cond_1**, go to **Status > Conferences**.




5. To verify the established call on the TelePresence MCU, go to the **Conference Status** page (**Conferences** on the main tab



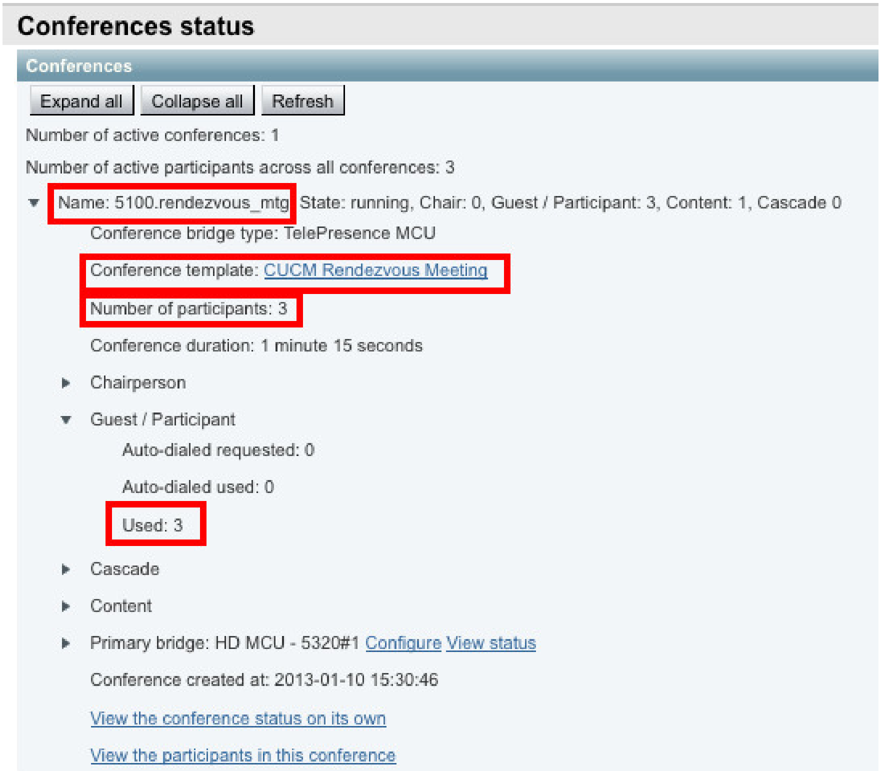

# Creating a rendezvous conference

Perform the following test with both TelePresence Conductors operational, then with one switched on and the other off, then the first one off and second on.

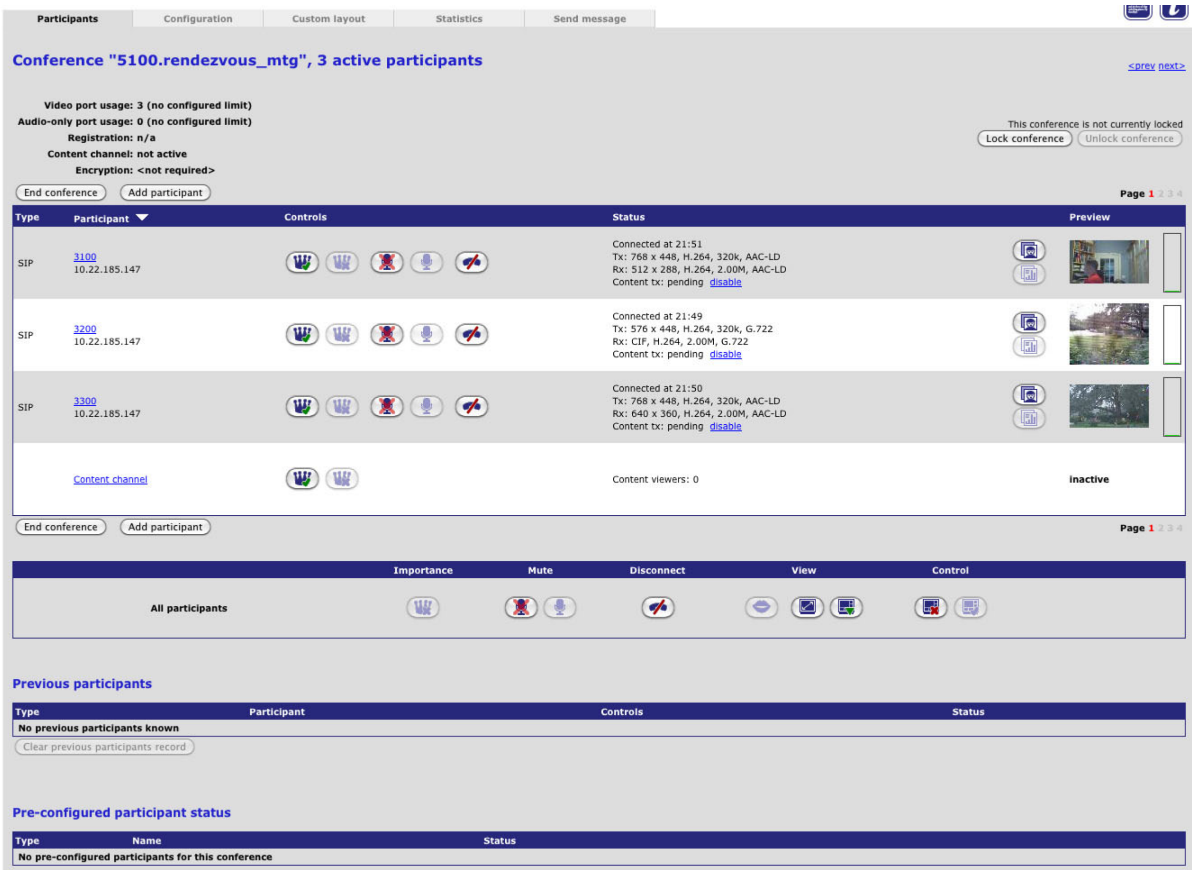
To test that two or more Unified CM registered endpoints can join a rendezvous conference:

1. From the 9971 dial **5100**. This will match the route pattern 5XXX that is associated with the SIP trunk to the TelePresence Conductor. Verify a video and audio session is established with the TelePresence MCU. An audio response of “You are the first participant to join” will be heard.
2. From the first C20 dial **5100**. Verify a video and audio session is established between the first C20 and the TelePresence MCU.
3. From the second C20 dial **5100**. Verify a video and audio session is established between the second C20 and the TelePresence MCU.
4. Each participant should be seeing video of the other participants’ camera and hearing audio from the other endpoints.

5. To verify on the TelePresence Conductor, **Cond_2**, that the call has been passed through the B2BUA, go to **Status > Conferences**.




6. To verify the establisehd call on the TelePresence MCU, go to the **Conference Status** page (**Conferences** on the main tab).



# Removing a TelePresence Conductor peer

To remove a TelePresence Conductor peer from a cluster, you must first remove the TelePresence Conductor from the Unified CM and then remove the TelePresence Conductor peer from the cluster.

## Removing a TelePresence Conductor from Unified CM

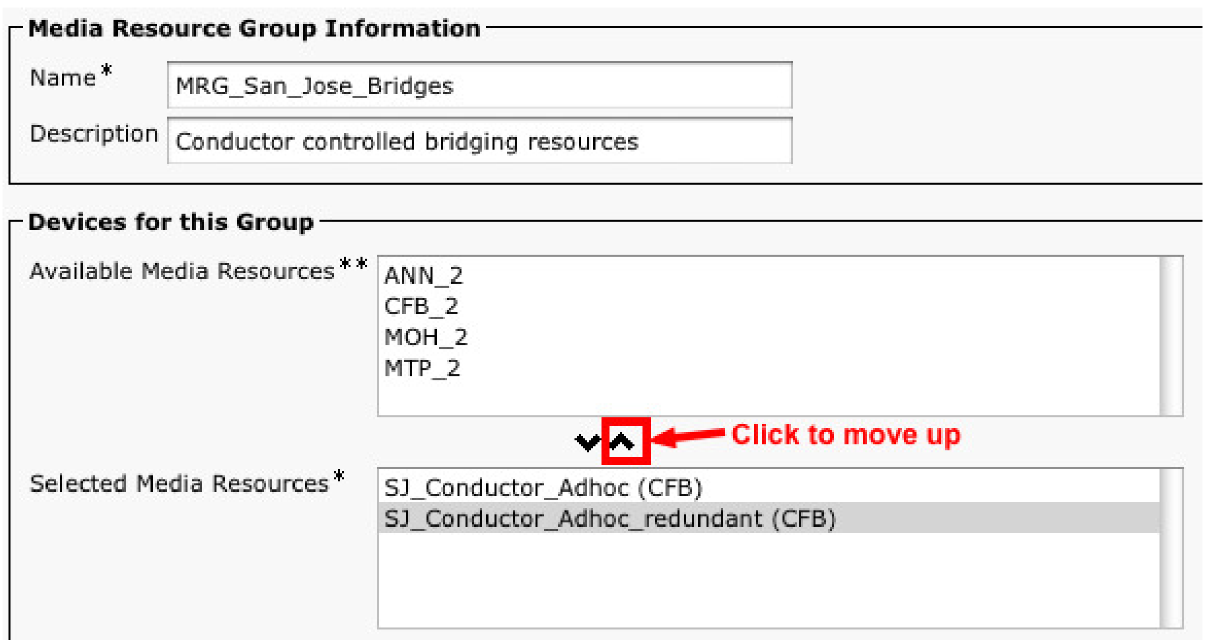
To remove a TelePresence Conductor from ad hoc calls you must remove the TelePresence Conductor from the Media Resource Group (MRG), and optionally delete the TelePresence Conductor from the Unified CM Conference bridges.

To remove a TelePresence Conductor from rendezvous calls you must remove the SIP trunk from the Unified CM to the TelePresence Conductor.

### Removing the TelePresence Conductor from the Media Resource Group

(This step is only applicable for ad hoc conferences.)

1. Go to the Unified CM web interface and log in as an admin user.
2. Go to **Media Resources > Media Resource Groups**.
3. Click **Find** to list the Media Resource Groups.
4. Click on **MRG_San_Jose_Bridges**.
5. Highlight the TelePresence Conductor that you want to remove from the group and click on the ^ to move it to the *Available Media Resources* box.

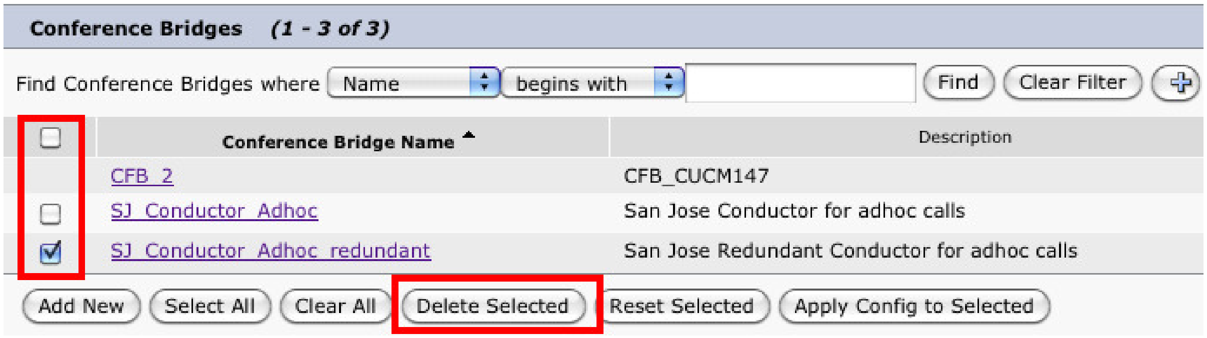



6. Click **Save**.

### (Optional) Removing the TelePresence Conductor as a conference bridge

(This step is only applicable for ad hoc conferences.)

1. Go to the Unified CM web interface and log in as an admin user.
2. Go to **Media Resources > Conference Bridges**.
3. Click **Find** to list the Conference Bridges.
4. Select the box next to the conference bridge and click **Delete Selected**.




## Removing the SIP trunk to the TelePresence Conductor used for rendezvous conferences

(This step is only applicable for rendezvous conferences.)

**Note:** Before removing the SIP trunk to the TelePresence Conductor we recommend that you note down the details, in case you want to re-instate the SIP trunk after an upgrade.

1. Go to **Device > Trunk**.
2. Click **Find** to show the configured trunks.
3. Select the trunk that is used for the TelePresence Conductor being removed.
4. At the top of the page select the Cross (**Delete**).
5. Confirm the deletion by pressing **OK**.

# Removing a peer from an existing cluster

## Placing the peer in standalone mode

Before removing a live peer from a cluster, you must place the peer in standalone mode so that it no longer communicates with other peers in the cluster. If the peer is out of service and can no longer be accessed, you do not need to place it in standalone mode. However, you must still follow the instructions to remove it from the cluster in the next section: Updating all other peers in the cluster [p.33].

To place a peer into standalone mode:

1. Log in to the peer to be removed from the cluster as a user with administrator privileges.
2. Go to **System > Clustering**.
3. Delete the **Cluster pre-shared key** value.
4. Delete all entries from the **Peer IP address** fields.
5. Click **Save**.
6. Go to **Maintenance > Restart options**.
7. Click **Restart**. When the TelePresence Conductor has restarted, it will be in standalone mode.
8. Optional: Delete the configuration or reconfigure the TelePresence Conductor.

## Updating all other peers in the cluster

After the peer to be removed has been placed in standalone mode (or if the peer is out of service and cannot be contacted), you must update all other peers in the cluster so they no longer consider the removed peer to be part of their cluster.

To do this, on each remaining peer in the TelePresence Conductor cluster:

1. Go to **System > Clustering**.
2. From the relevant **Peer x IP address** field (x = 1, 2, or 3), delete the IP address of the peer that has been removed from the cluster.
3. Click **Save**.

Repeat these steps on each remaining peer.

# Upgrading a cluster of TelePresence Conductors

The process described here is essentially disbanding, upgrading and then reclustering a cluster of TelePresence Conductors. In order to prevent downtime, one peer in the cluster is upgraded separately to the others, so that there is always at least one peer active and able to service conference requests from the Unified CMs until all peers have been upgraded and re-clustered.

## Task 1: Removing a peer from the cluster

Follow the steps in Removing a TelePresence Conductor peer [p.31] to remove one peer from the TelePresence Conductor cluster.

## Task 2: Upgrading the peer that has been removed from the cluster

On the TelePresence Conductor that has been removed from the cluster:

1. Go to the web interface and log in as a user with administrator privileges.
2. Go to **Maintenance > Upgrade**.
3. Click **Browse** and select the TelePresence Conductor software image.
4. Click **Upgrade**.
5. Follow the onscreen prompts.

## Task 3: Configuring the upgraded peer to be a cluster of one peer

Follow the steps in Task 1: Checking the configuration of the initial peer [p.7] and Task 2: Creating a cluster of one peer [p.8] to create a new cluster of one peer with the upgraded TelePresence Conductor.

## Task 4: Configuring Unified CM to use the upgraded peer

1. Follow the tasks in Configuring Unified CM for ad hoc conferences to add the upgraded TelePresence Conductor as a Conference Bridge to the Unified CM and to add it to an MRG and MRGL.
2. Follow the tasks in Configuring Unified CM for rendezvous conferences to configure Unified CM to use the upgraded TelePresence Conductor for rendezvous conferences.

## Task 5: Removing the other peers from the original cluster

Follow the tasks in Removing a peer from an existing cluster [p.32] to remove the remaining TelePresence Conductors that have not yet been upgraded from the original cluster.

## Task 6: Upgrading the other peers

Follow the steps in Task 2: Upgrading the peer that has been removed from the cluster [p.34] above to upgrade the remaining TelePresence Conductors.

# Task 7: Adding the remaining peers into the new cluster

Follow the tasks in Configuring TelePresence Conductor from Task 3: Configuring the cluster to accept the new peer [p.9] onwards to add the remaining TelePresence Conductor peers to the cluster.

# Task 8: Configuring Unified CM to use the upgraded peer(s)

1. Follow the steps in Configuring Unified CM for ad hoc conferences to add the remaining TelePresence Conductor peers as Conference Bridges to the Unified CM and to add them to an MRG and MRGL.
2. Follow the steps in Configuring Unified CM for rendezvous conferences to configure Unified CM to use the remaining TelePresence Conductor peers for rendezvous conferences.

# Task 9: Testing the system with calls

Follow the steps in Testing system configuration [p.25] to make sure that the new cluster works properly with calls.

# Peer-specific configuration

Most items of configuration are applied to all peers in a cluster. However, the following items must be specified separately on each cluster peer.

## Cluster configuration

The list of Peer IP addresses (including the peer's own IP address) that make up the cluster has to be specified on each peer and they **must** be identical on each peer (the order in which they appear is not important).

The cluster pre-shared key has to be specified on each peer and **must** be identical for all peers.

## Ethernet

The Ethernet speed is specific to each peer. Each peer may have slightly different requirements for the connection to their Ethernet switch.

## IP

**Note:** Never change the Primary LAN 1 IP address of a TelePresence Conductor that is part of a cluster. The only IP settings that can be changed when the system is part of a cluster are the additional IPv4 addresses.

The IPv4 address is specific to each peer. It **must** be different for each peer in the cluster.

The IPv4 subnet mask is specific to each peer. It can be different for each peer in the cluster.

The IPv4 gateway is specific to each peer. Each peer can use a different gateway.

Any additional IPv4 addresses added for use with Unified CM must be different for each peer in the cluster.

## System host name and domain

The system host name is specific to each peer. We recommend that it is different for each peer in the cluster so that you can easily identify each system.

The DNS domain name is specific to each peer.

## DNS servers

DNS servers are specific to each peer. Each peer can use a different set of DNS servers.

## Time

The NTP servers are specific to each peer. Each peer may use one or more different NTP servers.

The time zone is specific to each peer. Each peer may have a different local time.

# SNMP

SNMP settings are specific to each peer. They can be different for each peer.

# Logging

The **Event Log** and **Configuration Log** on each peer will only report activity for the local TelePresence Conductor.

The list of remote syslog servers is specific to each peer. We recommend that you set up a remote syslog server to which the logs of all peers can be sent. This will allow you to have a global view of activity across all peers in the cluster.

# Security certificates

The Trusted CA Certificate and Server Certificate used by the TelePresence Conductor are specific to each peer. They must be uploaded individually on each peer.

# Administration access

The SSH service and LCD panel settings are specific to each peer. They can be different for each peer.

# Root account password

The password for the root account is specific to each peer. Each peer may have a different password, and for security reasons we recommend that they do.

**Note:** The username and password for the administrator account is shared across peers.

# Locations

All ad hoc or rendezvous IP addresses assigned to Locations must be different for each peer in the cluster.

# Troubleshooting

## Unable to cluster the TelePresence Conductor

When running a TelePresence Conductor without a valid release key (as TelePresence Conductor Essentials) clustering is not supported. Contact your Cisco account representative to obtain release key and option keys.

# Appendix 1: Unified CM version 8.6.2 configuration

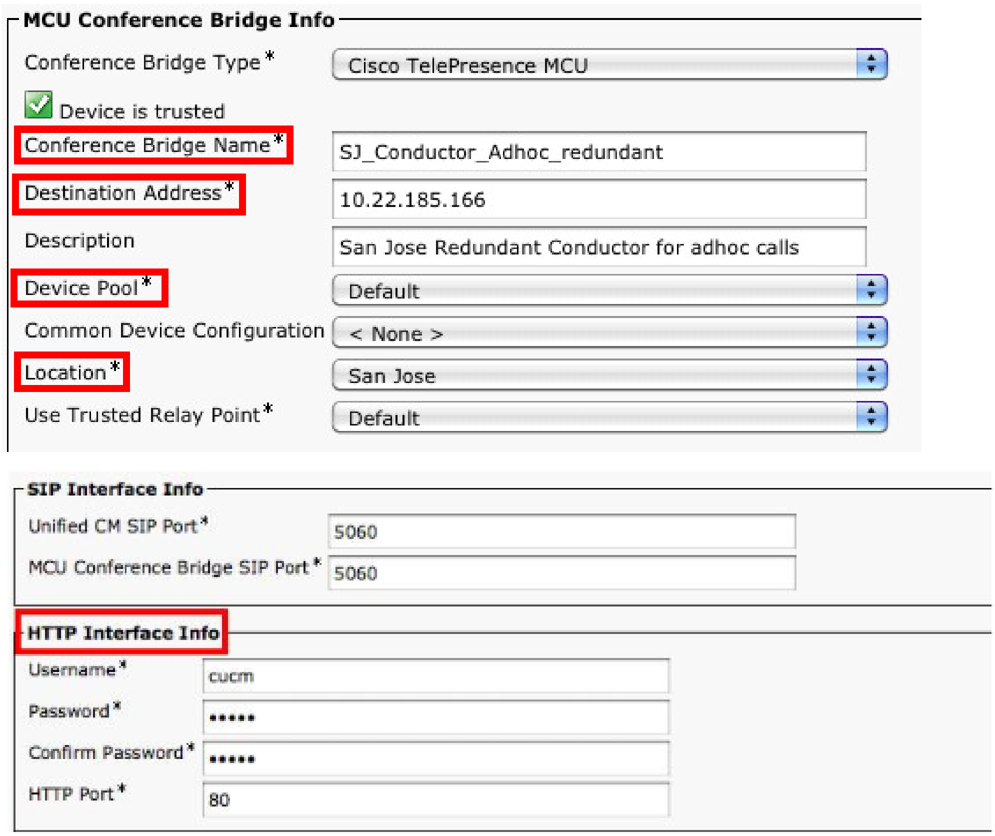
This section covers the differences between version 8.6.2 and the current version of Unified CM when configuring it for use with the TelePresence Conductor. The steps in this guide are from a version 10.0 Unified CM and should be replaced with the relevant steps from this appendix for version 8.6.2 Unified CMconfiguration.

## Adding the secondary TelePresence Conductor to Unified CM for ad hoc conferences

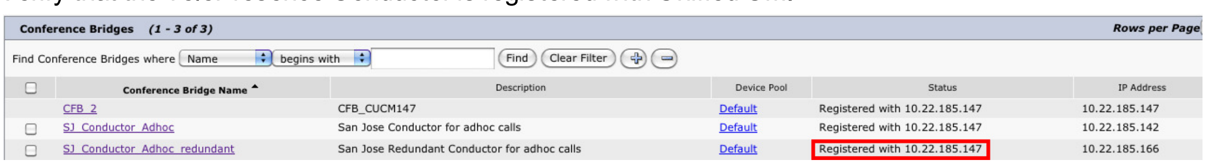
For Unified CM version 8.6.2, replace Task 8: Adding the secondary TelePresence Conductor as a Conference Bridge [p.16] with the following:

1. Go to **Media Resources > Conference Bridges**.
2. Click **Add New** to create a new conference bridge.
3. Enter the following into the relevant fields, leave other fields as their default values:

| | |
|---|---|
| **Conference Bridge Type** | Select Cisco TelePresence TelePresence MCU |
| **Conference Bridge Name** | Enter the TelePresence Conductor's Name |
| **Destination Address** | Enter the TelePresence Conductor's IP address |
| **Device Pool** | Select the appropriate pool |
| **Location** | Select the appropriate location |
| **Username** | Enter the username of the TelePresence Conductor administration user. This appears on the TelePresence Conductor's **Administrator accounts** page (**Users > Administrator accounts**) |
| **Password** | Enter the password of the TelePresence Conductor administration user |
| **HTTP Port** | Enter '80' |




4. Click **Save**.
5. Click **Reset** for the changes to take effect.
6. At the top right corner of the screen in the **Related Links:** field, select *Back to Find/List* and click **Go**. You will be taken back to the to the **Conference Bridges** page.
7. Verify that the TelePresence Conductor is registered with Unified CM.



Conference Bridges (1 - 3 of 3) Rows per Page

Find Conference Bridges where Name begins with Find Clear Filter

| | Conference Bridge Name | Description | Device Pool | Status | IP Address |
|---|---|---|---|---|---|
| | CFB_2 | CFB_CUCM147 | Default | Registered with 10.22.185.147 | 10.22.185.147 |
| | SJ_Conductor_Adhoc | San Jose Conductor for adhoc calls | Default | Registered with 10.22.185.147 | 10.22.185.142 |
| | SJ_Conductor_Adhoc_redundant | San Jose Redundant Conductor for adhoc calls | Default | Registered with 10.22.185.147 | 10.22.185.166 |

# Appendix 2: Unified CM version 9.x configuration

This section covers the differences between version 9.x and the current version of Unified CM when configuring it for use with the TelePresence Conductor. The steps in this guide are from Unified CM version 10.0 and should be replaced with the relevant steps from this appendix for Unified CM version 9.x configuration.

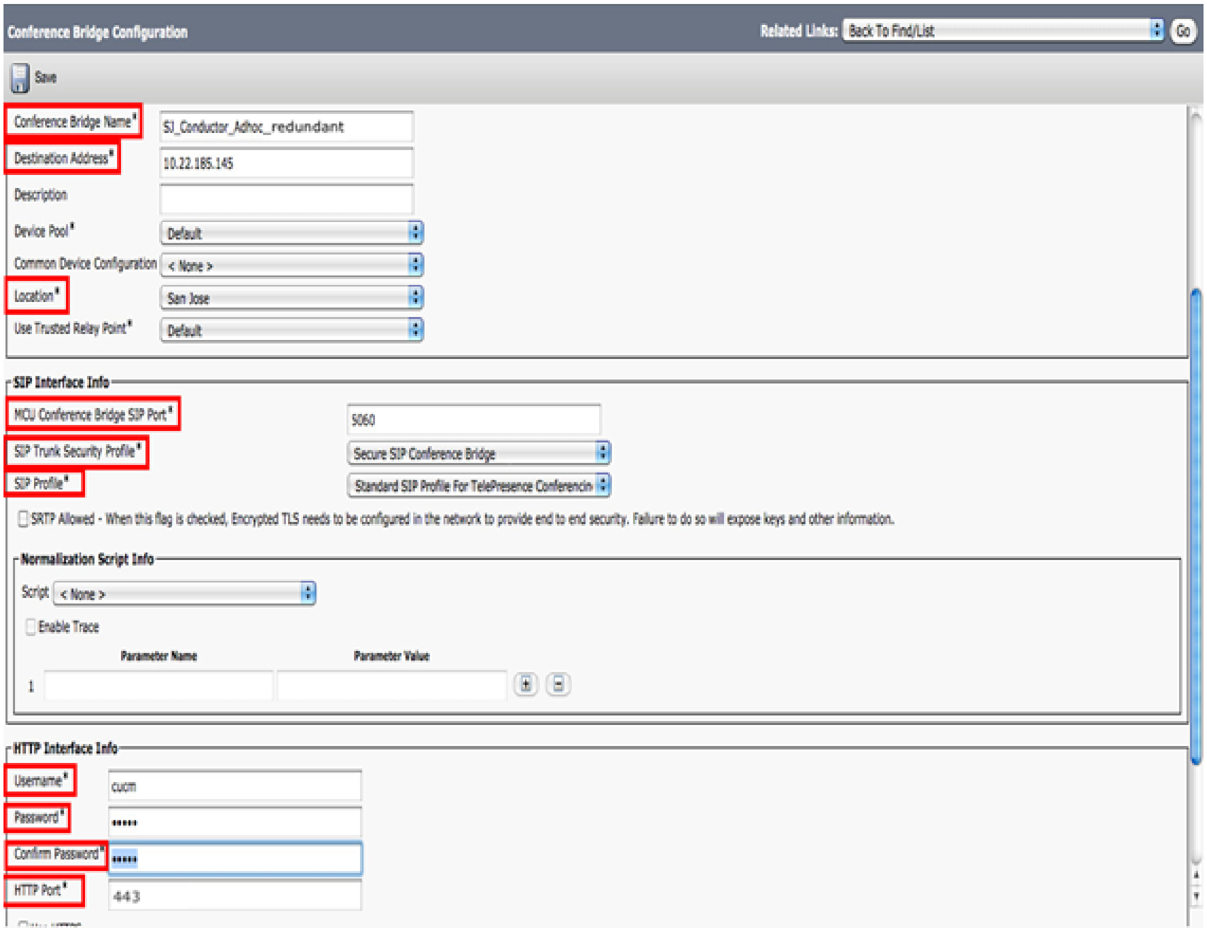
## Adding the secondary TelePresence Conductor to Unified CM for ad hoc conferences

For Unified CM version 9.x, replace Task 8: Adding the secondary TelePresence Conductor as a Conference Bridge [p.16] with the following:

1. Go to **Media Resources > Conference Bridge**.
2. Click **Add New** to create a new conference bridge.

3. Enter the following into the relevant fields, leave other fields as their default values:

| | |
|---|---|
| **Conference Bridge Type** | Select *Cisco TelePresence MCU* |
| **Conference Bridge Name** | Enter the TelePresence Conductor's Name |
| **Destination Address** | Enter the TelePresence Conductor's IP address |
| **Device Pool** | Select the appropriate pool |
| **MCU Conference bridge SIP port** | Modify the SIP listening port, if appropriate for your design, otherwise leave the default. |
| **SIP Trunk Security Profile** | Select *Secure SIP Conference Bridge* |
| **SIP Profile** | Select *Standard SIP Profile for TelePresence Conferencing* |
| **Location** | Select the appropriate location |
| **Username** | Enter the username of the TelePresence Conductor administration user. This appears on the TelePresence Conductor's **Administrator accounts** page (**Users > Administrator accounts**) |
| **Password** | Enter the password of the TelePresence Conductor administration user |
| **HTTP Port** | Enter ‘443’. |


Conference Bridge Configuration
Related Links: Back To Find/List
Go
Save
Conference Bridge Name*
SJ_Conductor_Adhoc_redundant
Destination Address*
10.22.185.145
Description
Device Pool*
Default
Common Device Configuration
< None >
Location*
San Jose
Use Trusted Relay Point*
Default
SIP Interface Info
MCU Conference Bridge SIP Port*
5060
SIP Trunk Security Profile*
Secure SIP Conference Bridge
SIP Profile*
Standard SIP Profile For TelePresence Conferencin
SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.
Normalization Script Info
Script
< None >
Enable Trace
Parameter Name
Parameter Value
1
HTTP Interface Info
Username*
cucm
Password*
Confirm Password*
HTTP Port*
443

# Appendix 3: IP ports and protocols

It is unusual to have any sort of firewall between cluster peers, but if there is, the IP protocols and ports that must be open between each and every TelePresence Conductor peer in the cluster are listed below.

For cluster communications between TelePresence Conductor peers:

- UDP port 500 (ISAKMP) is used for PKI (Public Key Infrastructure) key exchange
- Standard SIP and H.323 signaling ports are used for calls
- UDP port 1719 is used for bandwidth updates between TelePresence Conductor peers
- IP protocol 51 (IPSec AH) is used for database synchronization

If you are using the TelePresence Conductor's built-in **Firewall rules** feature then you must ensure that it is not configured to drop or reject traffic sent to UDP ports 4369 – 4380.

## IPSec communications

For IPSec between TelePresence Conductor cluster peers:

- AES256 is used for encryption, SHA256 (4096 bit key length) is used for authentication; peers are identified by their IP address and are authenticated using a pre-shared key
- Main mode is used during the IKE exchange
- diffie-hellman group ‘modp4096’ is used

# Appendix 4: Ensuring that Unified CM trusts TelePresence Conductor's server certificate and vice versa

For Unified CM and TelePresence Conductor to establish a TLS connection with each other, the following tasks are required.

## Loading server and trust certificates on TelePresence Conductor

### TelePresence Conductor server certificate

TelePresence Conductor has only one server certificate. By default, this is a certificate signed by a temporary certificate authority. We recommend that it is replaced by a certificate generated by a trusted certificate authority.

For information on how to request a certificate see *Cisco TelePresence Conductor Certificate Deployment Guide*.

To upload a server certificate:

1. Go to **Maintenance > Security certificates > Server certificate**.
2. Use the **Browse** button in the **Upload new certificate** section to select and upload the **server certificate** PEM file.
3. If you used an external system to generate the Certificate Signing Request (CSR) you must also upload the **server private key** PEM file that was used to encrypt the server certificate. (The private key file will have been automatically generated and stored earlier if the TelePresence Conductor was used to produce the CSR for this server certificate.)
   - The **server private key** PEM file must not be password protected.
   - You cannot upload a server private key if a certificate signing request is in progress.
4. Click **Upload server certificate data**.

### TelePresence Conductor trusted CA certificate

The **Trusted CA certificate** page (**Maintenance > Security certificates > Trusted CA certificate**) allows you to manage the list of certificates for the Certificate Authorities (CAs) trusted by this TelePresence Conductor. When a TLS connection to TelePresence Conductor mandates certificate verification, the certificate presented to the TelePresence Conductor must be signed by a trusted CA in this list and there must be a full chain of trust (intermediate CAs) to the root CA.

The root CA of the Unified CM server certificate must be loaded into the TelePresence Conductor's trusted CA certificate list.

To upload a new file containing one or more CA certificates, **Browse** to the required PEM file and click **Append CA certificate**. This will append any new certificates to the existing list of CA certificates. If you are replacing existing certificates for a particular issuer and subject, you have to manually delete the previous certificates.

Repeat this process on every TelePresence Conductor that will communicate with this Unified CM.

# Loading server and trust certificates on Unified CM

Certificate management for Unified CM is performed in the **Cisco Unified OS Administration** application.

All existing certificates are listed under **Security > Certificate Management**. Server certificates are of type *certs* and trusted CA certificates are of type *trust-certs*.

### Unified CM server certificate

By default, Unified CM has a self-signed server certificate **CallManager.pem** installed. We recommend that this is replaced with a certificate generated from a trusted certificate authority.

### Unified CM trusted CA certificate

To load the root CA certificate of the authority that issued the TelePresence Conductor certificate (if it is not already loaded):

1. Click **Upload Certificate/Certificate chain**.
2. Select a **Certificate Name** of *CallManager-trust*.
3. Click **Browse** and select the file containing the root CA certificate of the authority that issued the TelePresence Conductor certificate.
4. Click **Upload File**.

Repeat this process on every Unified CM server that will communicate with TelePresence Conductor. Typically this is every node that is running the CallManager service.

# Document revision history

The following table summarizes the changes that have been applied to this document.

| Revision | Date | Description |
|---|---|---|
| D15000.07 | September 2014 | Updated for release XC2.4 |
| D15000.06 | April 2014 | Updated for release XC2.3 |
| D15000.05 | December 2013 | Updated IP ports and protocols section |
| D15000.04 | October 2013 | Updated the Prerequisites section with changes introduced in XC2.2.1 |
| D15000.03 | August 2013 | Updated for release XC2.2 |
| D15000.02 | May 2013 | Updated for release XC2.1 |
| D15000.01 | December 2012 | Initial release |