



# Cisco TelePresence Conductor Deployment Guide

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# Introduction

## About the Cisco TelePresence Conductor

The Cisco TelePresence Conductor integrates tightly with the Cisco TelePresence Video Communication Server (Cisco VCS) and the Cisco TelePresence MCU products. It enables endpoints with sufficient privileges to seamlessly create and enter a conference by dialing a single number or URI.

Conference bridge resource management is performed automatically by the TelePresence Conductor and calls are routed to appropriate conference bridges by the Cisco VCS under instructions from the TelePresence Conductor. If the size of the conference grows beyond the capacity of a single conference bridge the conference is cascaded to additional conference bridges. From the perspective of the endpoint users this process occurs seamlessly.

TelePresence Conductor is capable of preferentially selecting conference bridges for conferences based on their properties. For example one could select conference bridges based on geographic location or video quality.

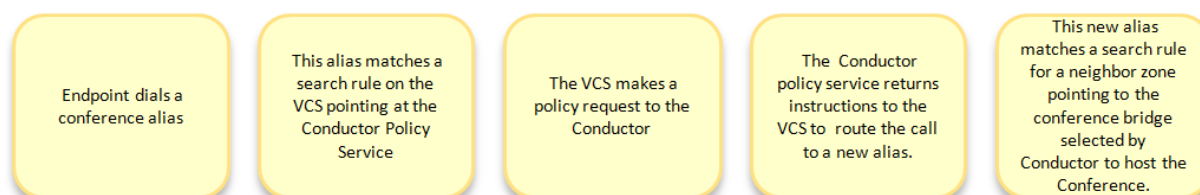
The TelePresence Conductor can be formed into a cluster of up to three for added resilience. When creating a TelePresence Conductor cluster, you nominate one peer as the initial peer, from which all other peers receive their configuration as they are added to the cluster. After the cluster has been created, changes to the configuration of any peer are updated to all other peers in the cluster. The TelePresence Conductor supports the Cisco VCS in standalone and clustered modes.

It is possible to configure up to 5 TelePresence Conductors or TelePresence Conductor Clusters per VCS or VCS cluster using a suitable non overlapping dial plan.

Endpoints registered with a CUCM 8.6(2) or later are capable of accessing TelePresence Conductor conferences through an appropriately configured SIP trunk between the CUCM and a VCS. For configuration details please refer to the deployment guide for your version of VCS.

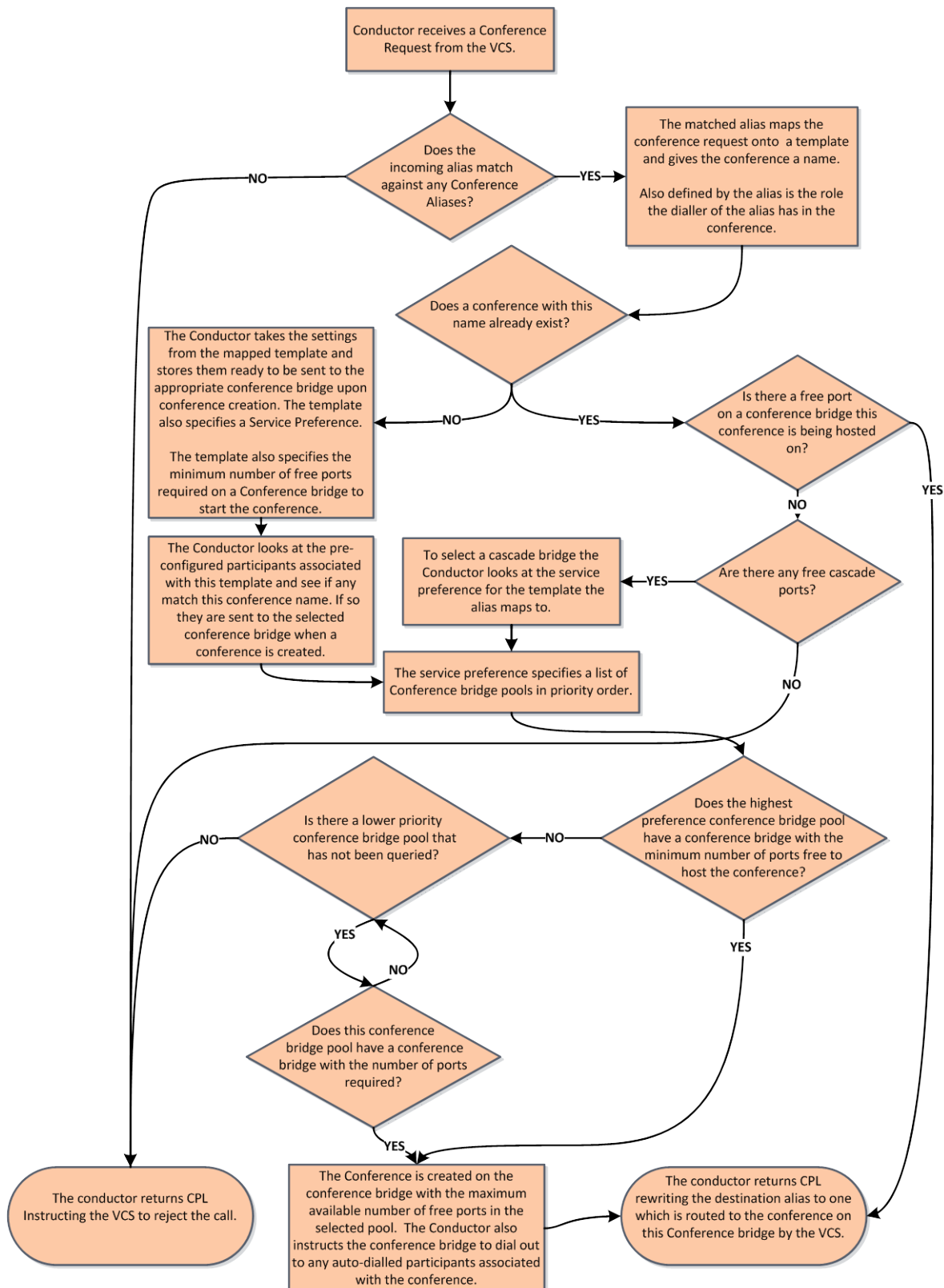
## Call flow with the Cisco TelePresence Conductor

To better understand the configuration steps taken in this document it is useful to understand how the call flows through the different parts of the video network:

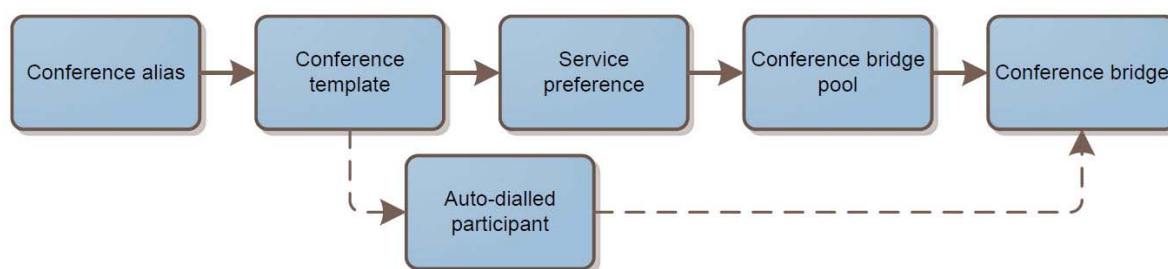


Once these parts of the call flow are complete then the call is set up and media flows between the endpoint and the conference bridge.

## Cisco TelePresence Conductor conference bridge selection process



In a simplified format the set of steps for a conference to be created once the TelePresence Conductor receives an individual valid conference request is:





The dotted line indicates an optional step that occurs concurrently with the normal conference request processing.

## About this document

This document describes how to configure a Cisco VCS (or Cisco VCS cluster), a TelePresence Conductor (or TelePresence Conductor cluster) and the pool of conference bridges that are used by the system. Following the steps in this deployment guide will allow you to configure the above devices to allow the following functionality:

- An endpoint user can call the alias **meet.<meeting name>.HD@vcs.domain**. If they are the first person to call this alias, a new conference is created by TelePresence Conductor and they are routed to it. The conference is created preferentially on a conference bridge with high definition ports, if there are no ports available on the HD conference bridge then the conference will be created on the SD conference bridge. Alternatively the conference already exists then the alias is routed to it.
- An endpoint user can call the alias **meet.<meeting name>.SD@vcs.domain** If they are the first person to call this alias a new conference is created by TelePresence Conductor and they are routed to it. The conference is created preferentially on a conference bridge with standard definition ports, if there are no ports available on this conference bridge then the call is rejected. If the conference already exists then they are routed to it.
- An endpoint user can dial the conference **meet.boss@vcs.domain** and arrive at a conference and have the endpoint **boss@vcs.domain** dialed into the conference by the conference bridge.
- An endpoint user can call the alias **teach.<lecture\_name>@vcs.domain** and create or join a lecture-type conference as a chairperson on a conference bridge with SD ports or, if no SD ports are available, a conference on the HD conference bridge.
- An endpoint user can call the alias **student.<lecture name>@vcs.domain** and create or join a lecture-type conference as a chairperson on a conference bridge with SD ports or, if no SD ports are available, a conference on the HD conference bridge.
- If the size of a **meet.<meeting name>.HD@<domain>** conference or a **teach.<lecture name>@vcs.domain** conference grows to a point where the resources required exceed those available on the conference bridge on which it is being hosted and ports are available on a second conference bridge, the TelePresence Conductor will direct new conference participants to the second conference bridge and set up a cascade between the conference bridges, provided there are available resources there.

This document takes you through the steps required to configure the Cisco VCSs, TelePresence Conductor and conference bridges to perform the actions above. It also describes how to check that the system is working as expected.

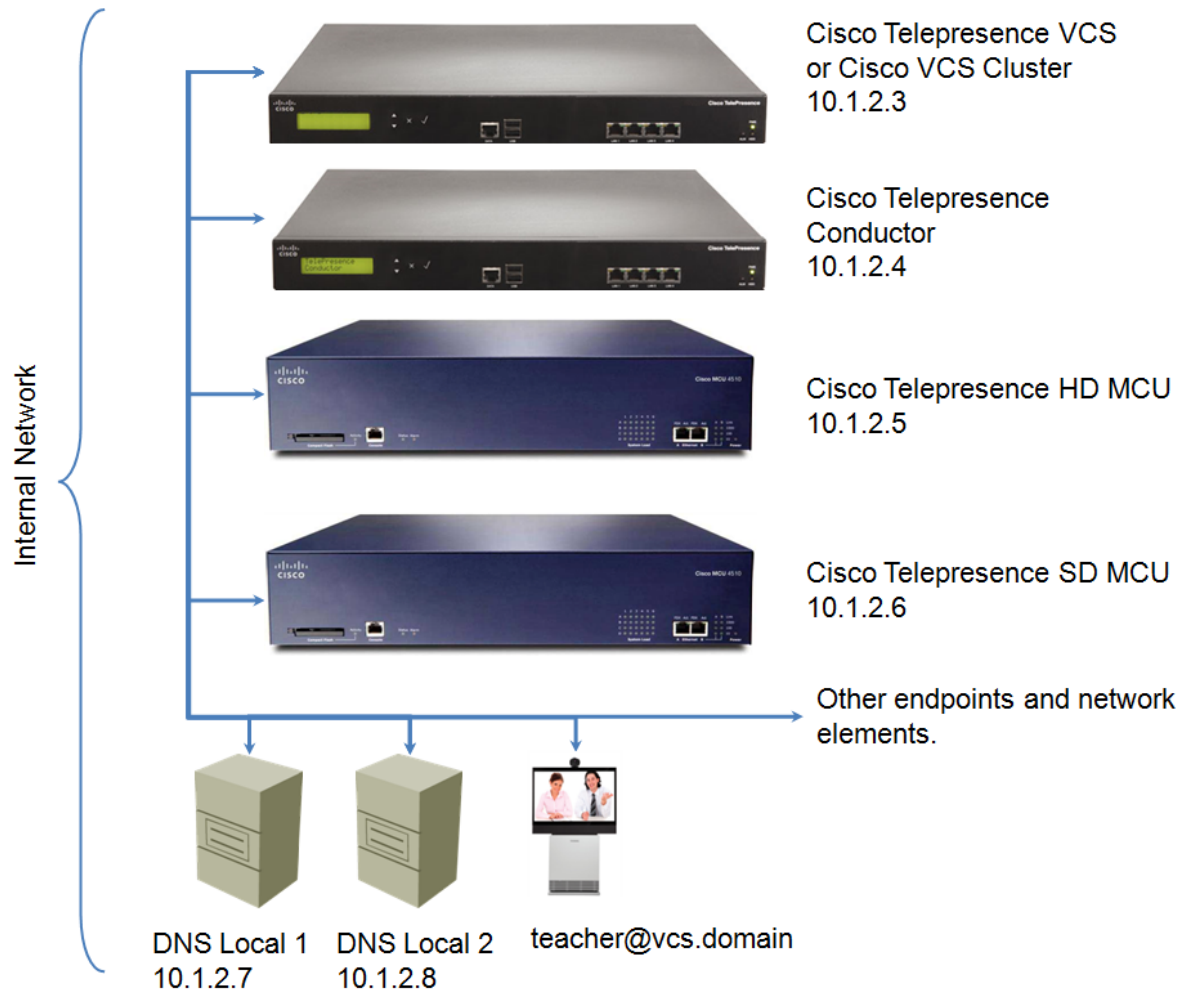
Descriptions of system configuration parameters for the Cisco VCS, TelePresence Conductor and conference bridges can be found in the Administrator Guides and online help for each product. Both the Cisco VCS and the TelePresence Conductor web interfaces offer field help (accessed by clicking the  icon next to each input field) and a context-sensitive help system (accessed by clicking the  icon in the top right corner of each page).

## Out of scope

This document does not describe how to deploy a cluster of TelePresence Conductors. For more details on this feature please see *Cisco TelePresence Conductor Cluster Creation and Maintenance Deployment Guide (D14828)*.

## Example network deployment

This document uses the example network shown in the diagram below as the basis for the deployment configuration described.



**Note:** elements on the internal network have an internal network domain name. This internal network domain name is not resolvable by a public DNS.

For example, the Cisco VCS is configured with an internally resolvable name of **vcsc.internal-domain.net** (which resolves to an IP address of 10.1.2.5 by the internal DNS servers).

## Cisco TelePresence network elements

### VCS

The Cisco VCS acts as a SIP registrar and proxy and H.323 gatekeeper for devices which are located on the internal network.



## Conference bridges

Conference bridges allow multipoint conferences for endpoints without Multiway capabilities by decoding and re-encoding the streams from the different endpoints and sending a single stream to each endpoint.

## Endpoints

These are devices which receive and make video calls. They can be software clients on PCs and Macs such as Movi, desktop endpoints such as the EX90, or room systems such as the CTS-3000

## Prerequisites

Before starting the system configuration, ensure you have access to:

- A Cisco VCS (or Cisco VCS cluster) running version X6 or later. This must already be configured to act as a H.323 gatekeeper, SIP registrar and proxy. Ensure that the system has been tested by registering at least three endpoints to it and ensuring that they are all capable of calling each other. For more information, see *VCS Administrator Guide* (D14049).
- A TelePresence Conductor unit that is powered on and accessible over the network. For assistance in reaching this stage please see *Cisco TelePresence Conductor Getting Started Guide* (D14829).
- One or more conference bridges powered on and accessible over the network. Basic configuration for the conference bridge should be completed as described in the relevant Getting Started Guide. Cisco TelePresence MCU software must be version 4.2 or later. The following Cisco TelePresence MCUs are supported by the TelePresence Conductor:
  - 4200 Series
  - 4500 Series
  - 8420 Media Blade
  - 8510 Media2 Blade

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**Note:** this guide assumes the conference bridges are connected to the network on port A.

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- A web browser with access to the web interfaces of the Cisco VCS, TelePresence Conductor and conference bridges that are being configured.

# Summary of the deployment process

The process of deploying a Cisco TelePresence Conductor consists of the following steps. Each step is described in a separate section:

## Configuring the conference bridges

- Step 1: Configuring SIP
- Step 2: Configuring H.323
- Step 3: Creating a user
- Step 4: Miscellaneous configuration

## Configuring the Cisco VCS

- Step 1: Adding the TelePresence Conductor as a policy service
- Step 2: Adding each conference bridge as a neighbor zone
- Step 3: Configuring a search rule with the TelePresence Conductor policy service as the target
- Step 4: Configuring a VCS search rule for each conference bridge

## Configuring the TelePresence Conductor

- Step 1: Changing the administrator password
- Step 2: Changing the root password
- Step 3: Initial configuration
- Step 4a: Configuring an HD conference bridge pool
- Step 4b: Adding a conference bridge to the HD conference bridge pool
- Step 5a: Configuring an SD conference bridge pool
- Step 5b: Adding a conference bridge to the SD conference bridge pool
- Step 6: Adding an HD Service Preference
- Step 7: Adding an SD Service Preference
- Step 8: Creating a conference template for the 'SD Meeting' template.
- Step 9: Creating a conference template for the 'HD Meeting' template.
- Step 10: Creating a template for a conference type of 'Lecture'
- Step 11: Creating an auto-dialed participant for the 'Meeting' template
- Step 12: Creating a conference alias for the 'SD Meeting' template with a role of 'Participant'
- Step 13: Creating a conference alias for the 'HD Meeting' template with a role of 'Participant'
- Step 14: Creating a conference alias for the 'Lecture' template with a role of 'Chairperson'
- Step 15: Creating a conference alias for the 'Lecture' template with a role of 'Guest'

## Designing a dial plan

A dial plan defines all the aliases and call routes within your network.

Before you add the Cisco TelePresence Conductor to your network, you will need to consider as part of your dial plan:

- the types of conferences required (see *Cisco TelePresence Conductor Administrator Guide* (D14826) for more information)
- the form of the conference aliases that users will dial in order to create or join conferences
- the prefixes that you will use to route calls from the Cisco VCS to the conference bridges in the TelePresence Conductor's conference bridge pool (the VCS is neighbored to each conference bridge). Each conference bridge has a unique prefix.

If you are integrating the TelePresence Conductor into an existing deployment it is important that the elements of your dial plan that are used by the TelePresence Conductor are complementary to, and do not conflict with, those elements that are already in use by your deployment. This ensures that the dial plan is easy for an endpoint user to understand, and for administrators to manage.

Cisco TelePresence Conductor is compatible with the dial plan specified in the reference design, once additional aliases have been included for users to dial to reach conferences.

It is a good idea in a large video network to distribute the registrations of conference bridges across different VCSs/VCS clusters this increases the resiliency of conference bridge dialout calls (to endpoints or during cascading) against VCS failure.

In this deployment guide we will be using the following dial plan elements, and configuring the TelePresence Conductor and Cisco VCS accordingly:

| Element   | Format                               |
|---|--------------------------------------|
| Conference aliases for lecture chairpersons                     | lecture.<name of lecture>@vcs.domain |
| Conference aliases for lecture guests                           | guest.<name of lecture>@vcs.domain   |
| Conference aliases for high definition meeting participants     | meet.<meeting name>.HD@vcs.domain    |
| Conference aliases for standard definition meeting participants | meet.<meeting name>.SD@vcs.domain    |
| Conference bridge prefixes for 10.1.2.5 and 10.1.2.6            | HDMCU, SDMCU                         |

# Configuring the conference bridges

## Step 1: Configuring SIP

1. Go to the web interface of the first conference bridge you want to configure and log in as an administrator.
2. Go to the **SIP Settings** page (**Settings > SIP**).
3. Input the following into the relevant fields, leave other fields as their default values:

| Field  | Input                        |
|--|------------------------------|
| SIP registrar usage  | Select <i>Enabled</i>        |
| SIP registrar domain   | Enter the VCS's SIP domain   |
| Username   | Enter 'sdmdu' for example    |
| Allow numeric ID registration for conferences                    | Uncheck                      |
| SIP proxy address  | Enter the VCS's IP address   |
| Maximum bit rate from Microsoft OCS/LCS clients                  | Select <i>limit disabled</i> |
| Outgoing transport   | Select <i>TLS</i>            |
| Use local certificate for outgoing connections and registrations | Uncheck                      |

The screenshot shows the SIP configuration page with the following settings:

- SIP registrar usage:** Enabled
- SIP registrar domain:** vcs.domain
- SIP registrar type:** Standard SIP
- Username:** sdmdu
- Password:** (empty)
- Allow numeric ID registration for conferences:**
- SIP call settings:**
  - SIP proxy address:** 10.1.2.3
  - Maximum bit rate from Microsoft OCS/LCS clients:** <limit disabled>
  - Outgoing transport:**  UDP  TCP  TLS
  - Use local certificate for outgoing connections and registrations:**

4. Click **Apply changes**.
5. Go to the web interface of the second conference bridge you want to configure and log in as an administrator.
6. Go to the **SIP settings** page (**Settings > SIP**)

7. Input the following into the relevant fields, leave other fields as their default values:

| Field  | Input  |
|--|--|
| SIP registrar usage  | Select <i>Enabled</i>  |
| SIP registrar domain   | Enter the VCS's SIP domain   |
| Username   | Enter 'hdmcu' for example  |
| Allow numeric ID registration for conferences                    | Uncheck  |
| SIP proxy address  | Enter the VCS's IP address   |
| Maximum bit rate from Microsoft OCS/LCS clients                  | Select <i>&lt;limit disabled&gt;</i>   |
| Outgoing transport   | Select <i>TLS</i> if possible.<br>Ensure that the SIP transport protocol matches the protocol selected for the neighbor zone on the VCS. |
| Use local certificate for outgoing connections and registrations | Uncheck  |

**SIP**

SIP registrar usage Enabled ▾

SIP registrar domain

SIP registrar type Standard SIP ▾

Username

Password

Allow numeric ID registration for conferences

---

**SIP call settings**

SIP proxy address

Maximum bit rate from Microsoft OCS/LCS clients <limit disabled> ▾

Outgoing transport  UDP  TCP  TLS

Use local certificate for outgoing connections and registrations

8. Click **Apply changes**.

## Step 2: Configuring H.323

1. Go to the **H.323 settings** page (**Settings > H.323**) on the first conference bridge
2. Input the following into the relevant fields, leave other fields as their default values:

| Field   | Input   |
|---|---|
| H.323 gatekeeper usage                        | Select <i>Enabled</i>                           |
| H.323 gatekeeper address                      | Enter the VCS's IP address                      |
| Gatekeeper registration type                  | Select <i>MCU (standard)</i>                    |
| Ethernet port association                     | Check <i>Port A IPv4</i>                        |
| Mandatory H.323 ID to register                | Enter <code>hdmcu@&lt;VCS SIP domain&gt;</code> |
| Use Password                                  | Uncheck   |
| Prefix for MCU registrations                  | Make blank                                      |
| MCU service prefix                            | Make blank                                      |
| Allow numeric ID registration for conferences | Uncheck   |
| Send resource availability indications        | Uncheck   |

The screenshot shows the H.323 settings page with the following configurations:

- H.323 gatekeeper usage:** Enabled
- H.323 gatekeeper address:** 10.1.2.3
- Gatekeeper registration type:** MCU (standard)
- Ethernet port association:**  Port A IPv4,  Port A IPv6,  Port B IPv4,  Port B IPv6
- (Mandatory) H.323 ID to register:** hdmcu@vcs.domain
- Use password:**  Password: [ ]
- Prefix for MCU registrations:** [ ]
- MCU service prefix:** [ ] (optional)
- Allow numeric ID registration for conferences:**
- Send resource availability indications:**  Thresholds: [ ] conferences [ ] video ports

3. Click **Apply changes**.
4. Go to the **H.323 settings** page (**Settings > H.323**) on the second conference bridge.
5. Input the following into the relevant fields, leave other fields as their default values:

| Field                          | Input   |
|--------------------------------|---|
| H.323 Gatekeeper usage         | Select <i>Enabled</i>                           |
| H.323 Gatekeeper address       | Enter the VCS's IP address                      |
| Gatekeeper registration type   | Select <i>MCU (standard)</i>                    |
| Ethernet port association      | Check <i>Port A IPv4</i>                        |
| Mandatory H.323 ID to register | Enter <code>hdmcu@&lt;VCS SIP domain&gt;</code> |
| Use Password                   | Uncheck   |

|   |            |
|---|------------|
| Prefix for MCU registrations                  | Make blank |
| MCU service prefix                            | Make blank |
| Allow numeric ID registration for conferences | Uncheck    |
| Send resource availability indications        | Uncheck    |

H.323

H.323 gatekeeper usage Enabled ▾

H.323 gatekeeper address 10.1.2.3

Gatekeeper registration type MCU (standard) ▾

Ethernet port association  Port A IPv4  Port A IPv6  Port B IPv4  Port B IPv6

(Mandatory) H.323 ID to register sdmcu@vcs.domain

Use password  Password:  

Prefix for MCU registrations  

MCU service prefix   (optional)

Allow numeric ID registration for conferences

Send resource availability indications  Thresholds:   conferences   video ports

6. Select **Apply changes**.



## Step 3: Creating a user

The TelePresence Conductor requires a user with administrator rights to access the conference bridge. It is recommended you create a second admin user for this task.

1. For the first conference bridge go to **Users** and click **Add new user**
2. Enter the following in the relevant fields:

| Field                                       | Input  |
|---|--|
| User ID                                     | Enter a username for TelePresence Conductor to use |
| Name  | Enter a name for this user                         |
| Password                                    | Enter a password for TelePresence Conductor to use |
| Force user to change password on next login | Ensure this is unchecked                           |
| Privilege level                             | Select <i>administrator</i>                        |

**User information**

User ID:

Name:

Password:

Re-enter password:

Disable user account:

Lock password:

Force user to change password on next login:

Privilege level:

E.164 phone number:

Associated video endpoint:

3. Click **Add user**.
4. On the second conference bridge go to **Users** and click **Add new user**.
5. Enter the following in the relevant fields:

| Field                                       | Input  |
|---|--|
| User ID                                     | Enter a username for TelePresence Conductor to use |
| Name  | Enter a name for this user                         |
| Password                                    | Enter a password for TelePresence Conductor to use |
| Force user to change password on next login | Ensure this is unchecked                           |
| Privilege level                             | Select <i>administrator</i>                        |

The screenshot shows a web form titled "User information" with a dark blue header. The form contains the following fields and options:

- User ID:
- Name:
- Password:
- Re-enter password:
- Disable user account:
- Lock password:
- Force user to change password on next login:
- Privilege level:  (dropdown menu)
- E.164 phone number:
- Associated video endpoint:  (dropdown menu)
- At the bottom right is an "Add user" button.

6. Click **Add user**.

## Step 4: Miscellaneous configuration

On both conference bridges:

1. Go to **Network > Services**.
2. Ensure that the ports are open (note that Encrypted SIP is not enabled by default.)

| TCP service                      | Port A                                   |
|----------------------------------|--|
|                                  | IPv4                                     |
| Web                              | <input checked="" type="checkbox"/> 80   |
| Secure web                       | <input checked="" type="checkbox"/> 443  |
| Incoming H.323                   | <input checked="" type="checkbox"/> 1720 |
| SIP (TCP)                        | <input checked="" type="checkbox"/> 5060 |
| Encrypted SIP (TLS)              | <input checked="" type="checkbox"/> 5061 |
| Streaming (Windows Media Player) | <input type="checkbox"/> 1755            |
| Streaming (other)                | <input type="checkbox"/> 554             |
| FTP                              | <input type="checkbox"/> 21              |

3. Click **Apply changes**.
4. Go to **Settings > Conferences** (this is the default selection after selecting **Settings**).
5. Under **Conference Settings** (this is a box on the page) ensure **Media port reservation** is set to *Disabled*.

| Conference settings                       |                                    |
|---|------------------------------------|
| Maximum video size                        | Receive 4CIF, transmit 4CIF ▾      |
| Motion / sharpness tradeoff               | Favor sharpness ▾                  |
| Transmitted video resolutions             | Allow all resolutions ▾            |
| Default bandwidth from MCU                | 4.00 Mbit/s ▾                      |
| Default bandwidth to MCU                  | <same as transmit> ▾               |
| Default view family                       | 1 focused pane, many small panes ▾ |
| Use full screen view for two participants | Enabled ▾                          |
| Active speaker display                    | None ▾                             |
| Media port reservation                    | Disabled ▾                         |

6. Click **Apply changes**.
7. Go to **Gatekeeper > Built in Gatekeeper**.
8. Under Configuration ensure **Status** is set to *Disabled*.

| Configuration |            |
|---------------|------------|
| Status        | Disabled ▾ |

9. Click **Apply changes**.

# Configuring the Cisco VCS

## Step 1: Adding the TelePresence Conductor as a policy service

A policy service is in essence a location to which HTTP or HTTPS requests containing various details about a call can be sent. CPL (Call Policy Language) is returned by the call policy service and governs what should be done with that call. The TelePresence Conductor's policy service either rejects calls or routes them to the appropriate conference bridge.

To configure the VCS with the TelePresence Conductor as a call policy service:

1. Go to the Cisco VCS web interface and log in as an admin user.
2. Go to the **Policy services** page (**VCS configuration > Dial plan > Policy services**).
3. Click **New** to create a new policy service pointing at the TelePresence Conductor.
4. Input the following into the relevant fields, leave other fields as their default values:

| Field  | Input   |
|--|---|
| Name   | Enter 'Conductor Policy Service'  |
| Protocol                                       | Select <i>HTTPS</i>   |
| Certificate verification mode                  | If you have configured the Cisco VCS with a Root CA that is valid for the Certificate on the TelePresence Conductor you can select <i>On</i> . Otherwise select <i>Off</i> . Certificates can be loaded onto the TelePresence Conductor through the web UI at <b>Maintenance &gt; Security certificates &gt; Server certificate</b><br><br>Note: Setting HTTPS certificate verification mode makes HTTPS communication highly insecure and is not recommended for production systems. |
| HTTPS certificate revocation list CRL checking | Select <i>Off</i>   |
| Server 1 address                               | Enter the TelePresence Conductor's IP address   |
| Path   | Enter<br><code>api/conference_controller/conference/conference_factory.cpl</code><br><br>Note: If you are using a printed copy of this document to copy and paste the above into the field go to the TelePresence Conductor online help and navigate to: <b>Before you start &gt; Configuring a VCS for use with the TelePresence Conductor</b>   |
| Username                                       | Enter the username of the Cisco TelePresence Conductor administration user. This appears on the Cisco Conductor's <b>Administrator accounts</b> page ( <b>Users &gt; Administrator accounts</b> )   |
| Password                                       | Enter the password of the TelePresence Conductor administration user  |
| Default CPL                                    | Enter <code>&lt;reject status='504' reason='Conductor policy service unavailable' /&gt;</code>  |

## Create policy service

Yo

Configuration

|  |   |
|--|---|
| Name   | ★ <input style="border: 1px solid #ccc;" type="text" value="Conductor Policy Service"/> <span style="float: right; font-size: 0.8em; color: #ccc;">i</span>                   |
| Description                                      | <input style="border: 1px solid #ccc;" type="text"/> <span style="float: right; font-size: 0.8em; color: #ccc;">i</span>  |
| Protocol   | HTTPS ▾ <span style="float: right; font-size: 0.8em; color: #ccc;">i</span>   |
| Certificate verification mode                    | On ▾ <span style="float: right; font-size: 0.8em; color: #ccc;">i</span>  |
| HTTPS certificate revocation list (CRL) checking | Off ▾ <span style="float: right; font-size: 0.8em; color: #ccc;">i</span>   |
| Server 1 address                                 | ★ <input style="border: 1px solid #ccc;" type="text" value="10.1.2.4"/> <span style="float: right; font-size: 0.8em; color: #ccc;">i</span>                                   |
| Server 2 address                                 | <input style="border: 1px solid #ccc;" type="text"/> <span style="float: right; font-size: 0.8em; color: #ccc;">i</span>  |
| Server 3 address                                 | <input style="border: 1px solid #ccc;" type="text"/> <span style="float: right; font-size: 0.8em; color: #ccc;">i</span>  |
| Path   | <input style="border: 1px solid #ccc;" type="text"/> <span style="float: right; font-size: 0.8em; color: #ccc;">i</span>  |
| Status path                                      | <input style="border: 1px solid #ccc;" type="text" value="status"/> <span style="float: right; font-size: 0.8em; color: #ccc;">i</span>                                       |
| Username   | <input style="border: 1px solid #ccc;" type="text" value="admin"/> <span style="float: right; font-size: 0.8em; color: #ccc;">i</span>  |
| Password   | <input style="border: 1px solid #ccc;" type="password"/> <span style="float: right; font-size: 0.8em; color: #ccc;">i</span>  |
| Default CPL                                      | <input style="border: 1px solid #ccc;" type="text" value="&lt;reject status='504' reason='Conductor pi"/> <span style="float: right; font-size: 0.8em; color: #ccc;">i</span> |

Create policy service
Cancel

5. Click **Create policy service**.

Note: Until the VCS updates its TelePresence Conductor status the status of the TelePresence Conductor policy service under **VCS configuration > Dial plan > Policy services** will list as active. Once the VCS queries the TelePresence Conductor for status this will change to inactive. This is expected behavior. The TelePresence Conductor policy service will only list itself as active when the following criteria are met:

- The TelePresence Conductor has its root and admin passwords changed from their default values. This is a security feature.
- The TelePresence Conductor has at least one conference bridge configured and with a 'usable' status. This is to ensure no requests are sent to members of a TelePresence Conductor cluster that have lost connectivity with the conference bridges.

## Step 2: Adding each conference bridge as a neighbor zone

To configure the VCS with **Neighbor zones** for both conference bridges:

1. Go to the **Zones** page (**VCS configuration > Zones**).
2. Click **Create new zone**.
3. Input the following into the relevant fields, leave other fields as their default values:

| Field          | Input  |
|----------------|--|
| Name           | Enter 'HD MCU' for example   |
| Type           | Select <i>Neighbor</i>   |
| Sip transport  | Select <i>TLS</i> if your conference bridge has the encryption option key<br>Select <i>TCP</i> otherwise.<br><br>Note: ensure that the SIP transport protocol matches the protocol selected for SIP registration on the conference bridge.<br><br>Note: Changing the transport method from TLS from TCP or vice versa <i>does not</i> change the port from 5061 to 5060. This must be done manually. |
| Peer 1 address | Enter the HD conference bridge's IP address  |
| Zone profile   | If the VCS is running 7.0.x or later select <i>Infrastructure Device</i><br>If the VCS is running 6.x select <i>Non-registering Device</i><br><br>Note: These Zone profiles perform no aliveness checking. As a result an 'Active' status given by this zone cannot be relied upon to indicate VCS to conference bridge communication is possible.   |

**Create zone** You are here: [VCS configuration](#) ▶

**Configuration**

Name \*  ⓘ

Type \*  ⓘ

Hop count \*  ⓘ

**H.323**

Mode  ⓘ

Port \*  ⓘ

**SIP**

Mode  ⓘ

Port \*  ⓘ

Transport  ⓘ

TLS verify mode  ⓘ

Accept proxied registrations  ⓘ

**Authentication**

Authentication policy  ⓘ

SIP authentication trust mode  ⓘ

**Location**

Peer 1 address  ⓘ

Peer 2 address  ⓘ

Peer 3 address  ⓘ

Peer 4 address  ⓘ

Peer 5 address  ⓘ

Peer 6 address  ⓘ

**Advanced**

Zone profile  ⓘ

H.323 call signaling port \*  ⓘ

4. Click **Create zone**.

5. Click **Create new zone**.
6. Input the following into the relevant fields, leave other fields as their default values:

| Field          | Input  |
|----------------|--|
| Name           | Enter 'SD MCU' for example   |
| Type           | Select <i>Neighbor</i>   |
| Sip transport  | Select <i>TLS</i> if your conference bridge has the encryption option key<br>Select <i>TCP</i> otherwise.<br><br>Note: ensure that the SIP transport protocol matches the protocol selected for SIP registration on the conference bridge.<br><br>Note: Changing the transport method from TLS from TCP or vice versa <i>does not</i> change the port from 5061 to 5060. This must be done manually. |
| Peer 1 address | Enter the IP address of the SD conference bridge   |
| Zone profile   | If the VCS is running 7.0.x or later Select <i>Infrastructure Device</i><br>If the VCS is running 6.x select <i>Non-registering Device</i><br><br>Note: These Zone profiles perform no aliveness checking. As a result an 'Active' status given by this zone cannot be relied upon to indicate VCS to conference bridge communication is possible.   |

**Create zone** You are here: [VCS configuration](#) > [Zones](#) > Create zone

**Configuration**

Name \*  i

Type \* Neighbor i

Hop count \*  i

**H.323**

Mode On i

Port \*  i

**SIP**

Mode On i

Port \*  i

Transport TLS i

TLS verify mode Off i

Accept proxied registrations Allow i

**Authentication**







Authentication policy Do not check credentials i

SIP authentication trust mode Off i




---

**Location**

|                |                                       |   |
|----------------|---------------------------------------|---|
| Peer 1 address | <input type="text" value="10.1.2.6"/> |  |
| Peer 2 address | <input type="text"/>                  |  |
| Peer 3 address | <input type="text"/>                  |  |
| Peer 4 address | <input type="text"/>                  |  |
| Peer 5 address | <input type="text"/>                  |  |
| Peer 6 address | <input type="text"/>                  |  |

**Advanced**

|                           |  |  |
|---------------------------|--|--|
| Zone profile              | <input type="text" value="Infrastructure device"/> |  |
| H.323 call signaling port | <input type="text" value="1720"/>                  |   |

7. Click **Create zone**.

---

**Note:** if you have selected a SIP transport of **TLS**, the conference bridge to which this zone is pointing needs to have the TLS encryption option key enabled. If you are using **UDP** or **TCP**, the port must **manually** be changed to 5060 on both the Cisco VCS and enabled on the conference bridge.

---

## Step 3: Configuring a search rule with the TelePresence Conductor policy service as the target













Search rules define where the VCS routes each call. In this case we want calls matching the format of our conference aliases to be sent to the TelePresence Conductor.

To configure the **Search rule**:

1. Go to the **Search rules** page (**VCS configuration > Dial plans > Search rules**).
2. Click **New**.
3. Input the following into the relevant fields, leave other fields as their default values:

| Field                         | Values  |
|-------------------------------|---|
| Rule name                     | Enter 'To Conductor Policy Service' for example   |
| Priority                      | Enter '10' for example  |
| Source                        | Select <i>Any</i>   |
| Request must be authenticated | Select <i>No</i>  |
| Mode                          | Select <i>Alias pattern match</i>   |
| Pattern type                  | Select <i>Regex</i>   |
| Pattern string                | Enter <code>(meet teach student)\.*@&lt;SIP domain&gt;</code><br>Note: Replace <SIP domain> with the appropriate sip domain for your network. |
| Pattern behavior              | Select <i>Leave</i>   |
| On successful match           | Select <i>Stop</i>  |
| Target                        | Select <i>Conductor Policy Service</i>  |
| State                         | Select <i>Enabled</i>   |

## Create search rule

| Configuration                 |  |
|-------------------------------|--|
| Rule name                     | * To Conductor Policy Service           |
| Description                   | <input type="text"/>                    |
| Priority                      | * 10                                    |
| Source                        | Any                                     |
| Request must be authenticated | No                                       |
| Mode                          | Alias pattern match                     |
| Pattern type                  | Prefix                                  |
| Pattern string                | * (meet teach student)\. *@vcs\.domain  |
| Pattern behavior              | Leave                                   |
| On successful match           | Stop                                    |
| Target                        | * Conductor Policy Service              |
| State                         | Enabled                                |

4. Click **Create search rule**.

## Step 4: Configuring a VCS search rule for each conference bridge

To configure the Search rule:

1. Go to the **Search rules** page (**VCS configuration > Dial plans > Search rules**).
2. Click **New**.
3. Input the following into the relevant fields, leave other fields as their default values:

| Field               | Input                             |
|---------------------|-----------------------------------|
| Rule name           | Enter 'To HD MCU' for example     |
| Priority            | Enter '15' for example            |
| Mode                | Select <i>Alias Pattern Match</i> |
| Pattern type        | Select <i>Prefix</i>              |
| Pattern string      | Enter 'HDMCU'                     |
| Pattern behavior    | Select <i>Strip</i>               |
| On successful match | Select <i>Stop</i>                |
| Target              | Select <i>HD MCU</i>              |

### Create search rule

Configuration

|                               |   |  |          |
|-------------------------------|---|--|----------|
| Rule name                     | * | <input type="text" value="To HD MCU"/>           | <i>i</i> |
| Description                   |   | <input type="text"/>                             | <i>i</i> |
| Priority                      | * | <input type="text" value="15"/>                  | <i>i</i> |
| Source                        |   | <input type="text" value="Any"/>                 | <i>i</i> |
| Request must be authenticated |   | <input type="text" value="No"/>                  | <i>i</i> |
| Mode                          |   | <input type="text" value="Alias pattern match"/> | <i>i</i> |
| Pattern type                  |   | <input type="text" value="Prefix"/>              | <i>i</i> |
| Pattern string                | * | <input type="text" value="HDMCU"/>               | <i>i</i> |
| Pattern behavior              |   | <input type="text" value="Strip"/>               | <i>i</i> |
| On successful match           |   | <input type="text" value="Stop"/>                | <i>i</i> |
| Target                        | * | <input type="text" value="HD MCU"/>              | <i>i</i> |
| State                         |   | <input type="text" value="Enabled"/>             | <i>i</i> |

4. Click **Create search rule**.
5. Click **New**.
6. Input the following into the relevant fields, leave other fields as their default values:

| Field               | Input                             |
|---------------------|-----------------------------------|
| Rule name           | Enter 'To SD MCU' for example     |
| Priority            | Enter '20' for example            |
| Mode                | Select <i>Alias pattern match</i> |
| Pattern type        | Select <i>Prefix</i>              |
| Pattern string      | Enter 'SDMCU'                     |
| Pattern behavior    | Select <i>Strip</i>               |
| On successful match | Select <i>Stop</i>                |
| Target              | Select <i>SD MCU</i>              |

### Create search rule

**Configuration**

|                               |  |
|-------------------------------|--|
| Rule name                     | <input type="text" value="To SD MCU"/>             |
| Description                   | <input type="text"/>                               |
| Priority                      | <input type="text" value="20"/> ⓘ                  |
| Source                        | <input type="text" value="Any"/> ⓘ                 |
| Request must be authenticated | <input type="text" value="No"/> ⓘ                  |
| Mode                          | <input type="text" value="Alias pattern match"/> ⓘ |
| Pattern type                  | <input type="text" value="Prefix"/> ⓘ              |
| Pattern string                | <input type="text" value="SDMCU"/>                 |
| Pattern behavior              | <input type="text" value="Strip"/> ⓘ               |
| On successful match           | <input type="text" value="Stop"/> ⓘ                |
| Target                        | <input type="text" value="SD MCU"/> ⓘ              |
| State                         | <input type="text" value="Enabled"/> ⓘ             |

7. Click **Create search rule**.

# Configuring the TelePresence Conductor

This section of the guide assumes that the Conductor is reachable over the network. For assistance in reaching this stage please see *Cisco TelePresence Conductor Getting Started Guide (D14829)*.

The TelePresence Conductor policy service will only list itself as active when the following criteria are met:

- The TelePresence Conductor has its root and admin passwords changed from their default values. This is a security feature.
- The TelePresence Conductor has at least one conference bridge configured and with a 'usable' status. This is to ensure no requests are sent to members of a TelePresence Conductor cluster that have lost connectivity with the conference bridges.

## Step 1: Changing the administrator password

1. Log into the TelePresence Conductor as the user 'admin' and the default password 'TANDBERG'.
2. Go to the **Administrator accounts** page (**Users > Administrator accounts**).
3. Click **View/Edit** for the 'admin' user
4. Enter a new password
5. Click **Save**.

## Step 2: Changing the root password

1. Log in to the TelePresence Conductor as **root** using the default password 'TANDBERG'. By default you can only do this using a serial connection or SSH.
2. Type 'passwd'.
3. Enter the new password, and when prompted, retype the new password.
4. You will receive the message:  
`passwd: password updated successfully`
5. Type 'exit' to log out of the **root** account.

## Step 3: Initial configuration

1. Log into the TelePresence Conductor as a user with administrator rights.
2. Go to the **DNS** page (**System > DNS**).
3. Input the following values into the relevant fields:

| Field            | Input   |
|------------------|---|
| System host name | Enter the hostname of your TelePresence Conductor |
| Domain name      | Enter the domain for your TelePresence Conductor  |
| Address 1        | Enter the IP address of the DNS server            |
| Address 2        | Enter the IP address of your backup DNS server    |

### DNS

#### DNS settings

System host name

Domain name

DNS requests port \*

range start

DNS requests port \*

range end

#### Default DNS servers

Address 1

Address 2

Address 3

#### Per-domain DNS servers

Address 1  Dom

Address 2  Dom

**Note:** the FQDN of the TelePresence Conductor will be <System host name>.<Domain name>

4. Click **Save**.
5. Go to the **Time** page (**System > Time**) if the default servers are unreachable then it may be necessary to enter alternate NTP servers.

### Time You a

#### NTP servers

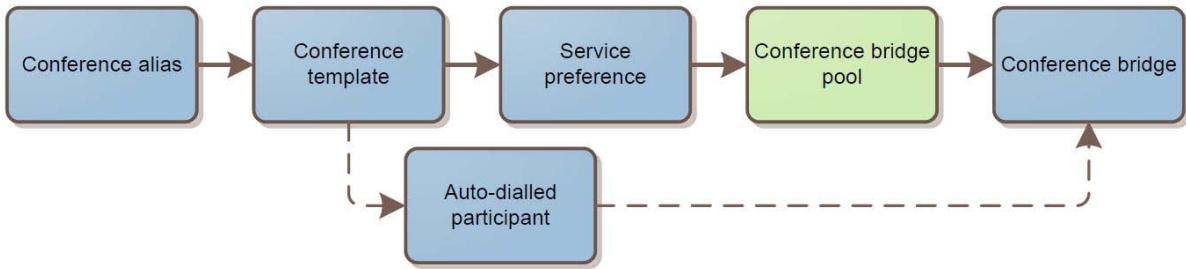
|                      |   |  |  |
|----------------------|---|--|--|
| NTP server details 1 | Address <input style="width: 150px;" type="text" value="0.ntp.tandberg.com"/> |  | Authentication <input type="text" value="Disabled"/> |
| NTP server details 2 | Address <input style="width: 150px;" type="text" value="1.ntp.tandberg.com"/> |  | Authentication <input type="text" value="Disabled"/> |
| NTP server details 3 | Address <input style="width: 150px;" type="text" value="2.ntp.tandberg.com"/> |  | Authentication <input type="text" value="Disabled"/> |
| NTP server details 4 | Address <input style="width: 150px;" type="text"/>                            |  | Authentication <input type="text" value="Disabled"/> |
| NTP server details 5 | Address <input style="width: 150px;" type="text"/>                            |  | Authentication <input type="text" value="Disabled"/> |

#### Time zone

Time zone

6. Ensure that under the **Status** section the **State** is **Synchronized**. This can take a couple of minutes.

## Step 4a: Configuring an HD conference bridge pool



1. Log into the TelePresence Conductor as a user with administrator rights.
2. Go to the **Conference bridge pools** page (**Conference Configuration > Conference bridges > Conference bridge pools**).
3. Click **New**.
4. In the **Pool name** field enter 'High Definition'.

**Conference bridge pools** You are here: [Conference configuration](#) > [Conference bridges](#) > [C](#)

**Configuration**

Pool name \*  i

Description  i

Conference bridge type  i

Raise conference bridge resource alarm  Threshold (%)  i

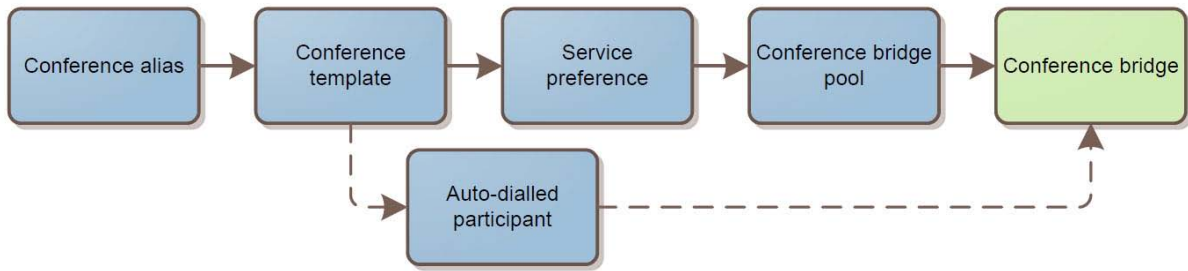
**Conference bridges in this pool**

There are no conference bridges in this pool.

5. Click **Create pool**.



## Step 4b: Adding a conference bridge to the HD conference bridge pool















1. Click **Add conference bridge**.
2. Input the following values into the relevant fields:


| Field                      | Input  |
|----------------------------|--|
| Name                       | Enter 'HD MCU' for example   |
| IP address or FQDN         | Enter the HD conference bridge's IP address  |
| Port                       | Enter '80' if using HTTP or '443' if using HTTPS to communicate with the conference bridge .   |
| Protocol                   | Select <i>HTTP</i> or <i>HTTPS</i> for secure communication.<br><br>Note: HTTP mode is highly insecure and is not recommended for production systems.  |
| Conference bridge username | Enter the conference bridge admin username (created in Step 3: Creating a user, within Configuring the conference bridges): 'conductormcu1'  |
| Conference bridge Password | Enter the conference bridge admin password   |
| Dial plan prefix           | Enter 'HDMCU'  |
| Dedicated content ports    | Enter the appropriate value for your TelePresence MCU. To discover if an MCU has any dedicated content ports follow the steps given in Appendix 5: Identifying Dedicated Content Ports on a Cisco TelePresence MCU |

## Add conference bridge

You are here: [Conference configuration](#) > [Conference bridges](#) > [Confere](#)

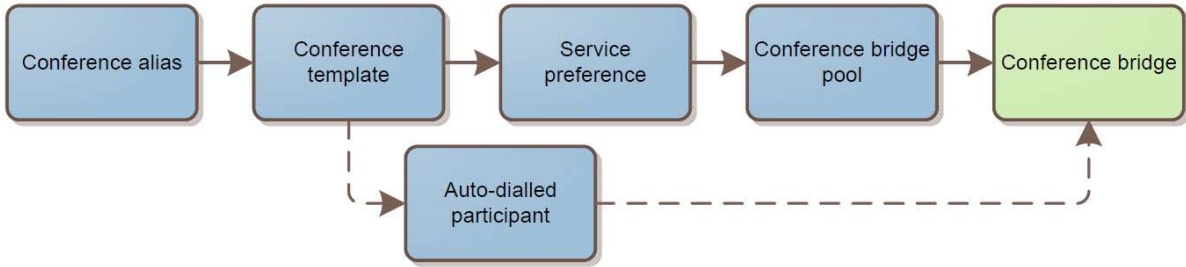
Configuration

|                            |   |   |   |
|----------------------------|---|---|---|
| Name                       | * | <input type="text" value="HD MCU"/>           |    |
| Description                |   | <input type="text"/>                          |    |
| State                      |   | <input type="text" value="Enabled"/>          |     |
| IP address or FQDN         | * | <input type="text" value="10.1.2.6"/>         |    |
| Protocol                   |   | <input type="text" value="HTTPS"/>            |      |
| Port                       | * | <input type="text" value="443"/>              |    |
| Conference bridge username | * | <input type="text" value="conductormcu1"/>    |    |
| Conference bridge password |   | <input type="password" value="....."/>        |    |
| Dial plan prefix           | * | <input type="text" value="HDMCU"/>            |    |
| Conference bridge type     | * | <input type="text" value="TelePresence MCU"/> |    |
| Conference bridge pool     | * | <input type="text" value="High Definition"/>  |  |
| Dedicated content ports    | * | <input type="text" value="0"/>                |  |



3. Click **Create conference bridge**.
4. Ensure that under the **Status** header under conference bridges in this pool the conference bridge is listed as *Active*.

## Step 5a: Configuring an SD conference bridge pool



1. Go to the **Conference bridge pool** page ([Conference Configuration > Conference bridges > Conference bridge pool](#)).
2. Click **New**.
3. In the **Pool name** field enter 'Standard Definition'.

**Conference bridge pools** You are here: [Conference configuration](#) > [Conference bridges](#) > [Conference bridge pool](#)

**Configuration**

Pool name \*  i

Description  i

Conference bridge type  i

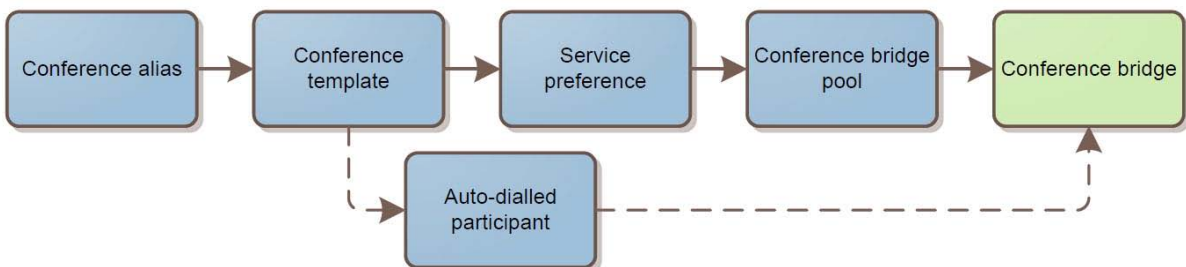
Raise conference bridge resource alarm  Threshold (%)  i

**Conference bridges in this pool**

There are no conference bridges in this pool.

4. Click **Create pool**

## Step 5b: Adding a conference bridge to the SD conference bridge pool



1. Click **Add conference bridge**

2. Input the following values into the relevant fields:

| Field                      | Input  |
|----------------------------|--|
| Name                       | Enter 'SD MCU' for example   |
| IP address or FQDN         | Enter the SD conference bridge's IP address  |
| Port                       | Enter '80' if using HTTP or '443' if using HTTPS to communicate with the conference bridge.  |
| Protocol                   | Select <i>HTTP</i> or <i>HTTPS</i> for secure communication.<br><br>Note: HTTP mode is highly insecure and is not recommended for production systems.  |
| Conference bridge username | Enter the conference bridge admin username (created in Step 3: Creating a user, within Configuring the conference bridges): 'conductormcu2'  |
| Conference bridge password | Enter the conference bridge admin password   |
| Dial plan prefix           | Enter 'SDMCU'  |
| Dedicated content ports    | Enter the appropriate value for your TelePresence MCU. To discover if an MCU has any dedicated content ports follow the steps given in Appendix 5: Identifying Dedicated Content Ports on a Cisco TelePresence MCU |

**Add conference bridge** You are here: [Conference configuration](#) > [Conference bridges](#) > [Confere](#)

**Configuration**

Name \*  i

Description  i

State  i

IP address or FQDN \*  i

Protocol  i

Port \*  i

Conference bridge username \*  i

Conference bridge password  i

Dial plan prefix \*  i

Conference bridge type \*  i

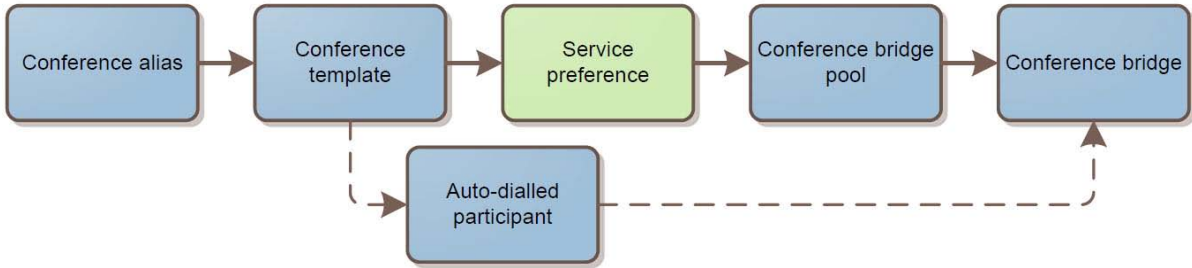
Conference bridge pool \*  i

Dedicated content ports \*  i



3. Click **Create conference bridge**.
4. Ensure that under the **Status** header under conference bridges in this pool the conference bridge is listed as *Active*.

## Step 6: Adding an HD Service Preference



1. Go to **Conference configuration > Conference bridges > Conference bridge Service Preferences**
2. Click **New**
3. In the **Service Preference name** field enter 'Prefer HD with SD fallback'
4. In the **Pools** section of the page under Pool name select *High Definition*
5. Click **Add selected pool**
6. In the **Pools** section of the page under Pool name select *Standard Definition*
7. Click **Add selected pool**

**Conference bridge Service Preferences** You are here: [Conference configuration](#) > [Conference bridge Service Preferences](#)

**Conference bridge Service Preference**

Service Preference name \*  i

Description  i

Conference bridge type \*  i

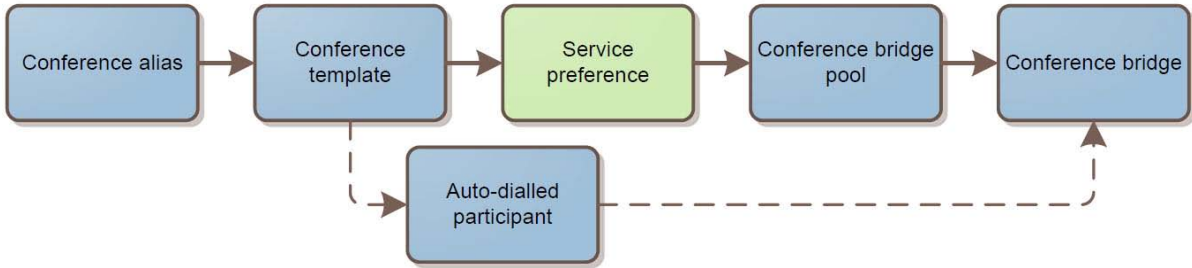
---

**Pools**

| Priority                   | Pool name                           | Change |
|----------------------------|-------------------------------------|--------|
| <input type="checkbox"/> 1 | <a href="#">High Definition</a>     | ↓      |
| <input type="checkbox"/> 2 | <a href="#">Standard Definition</a> | ↑      |
| Please select              |                                     | ▼      |

8. Click **Save**

## Step 7: Adding an SD Service Preference



1. Go to **Conference configuration > Conference bridges > Conference bridge Service Preferences**
2. Click **New**
3. In the **Service Preference name** field enter 'Prefer SD with HD fallback'
4. In the **Pools** section of the page under **Pool name** select *Standard Definition*
5. Click **Add selected pool**
6. In the **Pools** section of the page under **Pool name** select *High Definition*
7. Click **Add selected pool**

**Conference bridge Service Preferences** You are here: [Conference configuration](#) > [Conference bridge](#)

**Conference bridge Service Preference**

Service Preference name \*  i

Description  i

Conference bridge type \*  i

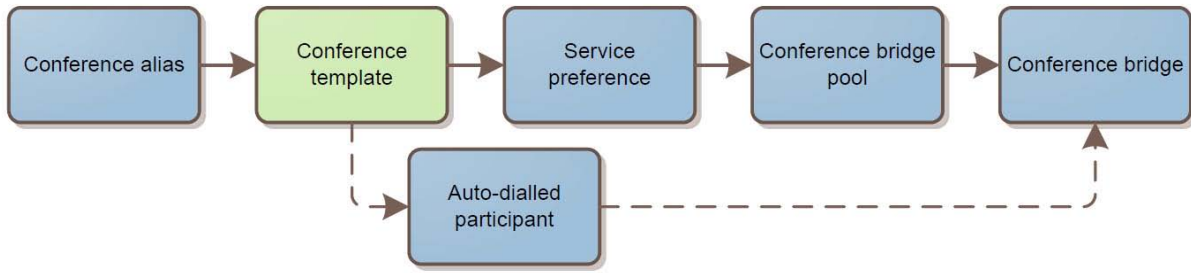
---

**Pools**

| Priority                   | Pool name                           | Change |
|----------------------------|-------------------------------------|--------|
| <input type="checkbox"/> 1 | <a href="#">Standard Definition</a> | ↓      |
| <input type="checkbox"/> 2 | <a href="#">High Definition</a>     | ↑      |
| Please select              |                                     | ▼      |

8. Click **Save**

## Step 8: Creating a conference template for the ‘SD Meeting’ template.



1. Go to the **Conference templates** page (**Conference configuration > Conference templates**).
2. Click **New**
3. Input the following into the relevant fields, leave other fields as their default values:

| Field                                | Input                                    |
|--------------------------------------|--|
| Name                                 | Enter 'SD meeting' for example           |
| Conference type                      | Select <i>Meeting</i>                    |
| No. of cascade ports to reserve      | Enter '1'                                |
| Conference bridge Service Preference | Select <i>Prefer SD with HD fallback</i> |




### Conference templates

**Modify conference template**

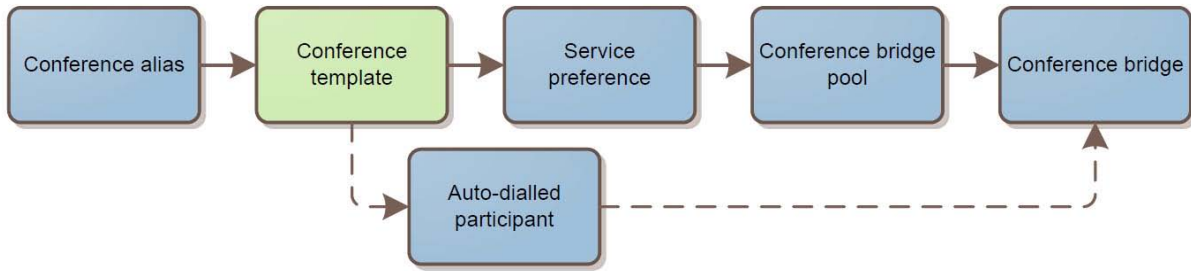
|   |   |
|---|---|
| Name                                    | * SD meeting  |
| Description                             | <input type="text"/>  |
| Conference type                         | Meeting <input type="button" value="i"/>                        |
| Call Policy mode                        | Off <input type="button" value="i"/>                            |
| Conference bridge Service Preference    | * Prefer SD with HD fallback <input type="button" value="i"/>   |
| No. of cascade ports to reserve         | * 1 <input type="button" value="i"/>                            |
| Limit number of participants            | <input type="checkbox"/> Maximum <input type="text" value="0"/> |
| Limit the conference duration (minutes) | <input type="checkbox"/> Maximum <input type="text" value="0"/> |

**Advanced**

|                                      |   |
|--------------------------------------|---|
| Conference layout                    |  <input type="button" value="Choose layout"/> <input type="button" value="i"/> |
| Participant PIN                      | <input type="text"/> <input type="button" value="i"/>   |
| Parameters to pass on to primary MCU | <a href="#">New</a>   |
| Parameters to pass on to cascade MCU | <a href="#">New</a>   |
| Allow content                        | Yes <input type="button" value="i"/>  |

4. Click **Create conference template**.

## Step 9: Creating a conference template for the 'HD Meeting' template.



1. Go to the **Conference templates** page (**Conference configuration > Conference templates**).
2. Click **New**
3. Input the following into the relevant fields, leave other fields as their default values:

| Field                                | Input                                    |
|--------------------------------------|--|
| Name                                 | Enter 'HD meeting' for example           |
| Conference type                      | Select <i>Meeting</i>                    |
| No. of cascade ports to reserve      | Enter '1'                                |
| Conference bridge Service Preference | Select <i>Prefer HD with SD fallback</i> |

### Conference templates

**Modify conference template**

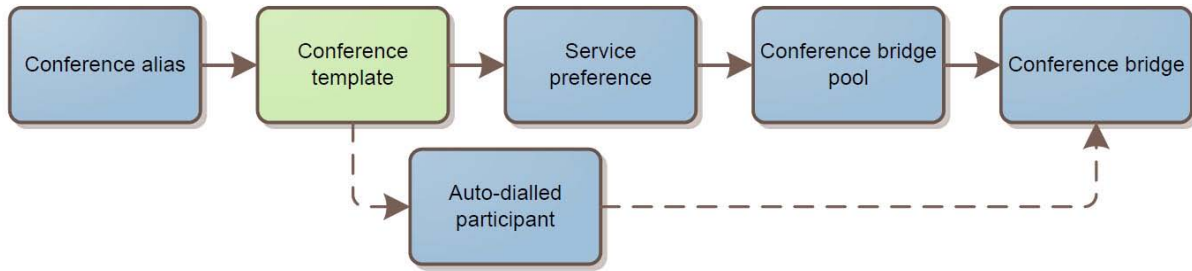
|   |   |
|---|---|
| Name                                    | * HD Meeting  |
| Description                             | <input type="text"/>  |
| Conference type                         | Meeting <input type="button" value="i"/>                        |
| Call Policy mode                        | Off <input type="button" value="i"/>                            |
| Conference bridge Service Preference    | * Prefer HD with SD fallback <input type="button" value="i"/>   |
| No. of cascade ports to reserve         | * 1 <input type="button" value="i"/>                            |
| Limit number of participants            | <input type="checkbox"/> Maximum <input type="text" value="0"/> |
| Limit the conference duration (minutes) | <input type="checkbox"/> Maximum <input type="text" value="0"/> |

**Advanced**

|                                      |   |
|--------------------------------------|---|
| Conference layout                    | <input type="button" value="Choose layout"/> <input type="button" value="i"/> |
| Participant PIN                      | <input type="text"/> <input type="button" value="i"/>                         |
| Parameters to pass on to primary MCU | <a href="#">New</a>   |
| Parameters to pass on to cascade MCU | <a href="#">New</a>   |
| Allow content                        | Yes <input type="button" value="i"/>  |

4. Click **Create conference template**.

## Step 10: Creating a template for a conference type of 'Lecture'



1. Go to the **Conference templates** page (**Conference configuration > Conference templates**).
2. Click **New**
3. Input the following into the relevant fields, leave other fields as their default values:


| Field                                | Input   |
|--------------------------------------|---|
| Name                                 | Enter 'Lecture' for example   |
| Conference type                      | Select <i>Lecture</i>   |
| No. of cascade ports to reserve      | Enter '1'   |
| No. of chairperson ports to reserve  | Enter '2'   |
| Call Policy mode                     | Select <i>Off</i>   |
| Conference bridge Service Preference | Select <i>Prefer SD with HD fallback</i>  |
| Chair PIN                            | Enter a PIN for the chair to use when entering the conference.<br><br><b>Note:</b> for TelePresence MCU software versions lower than 4.3 a Guest PIN must be specified if a Chair PIN is specified. |

### Conference templates

**Modify conference template**

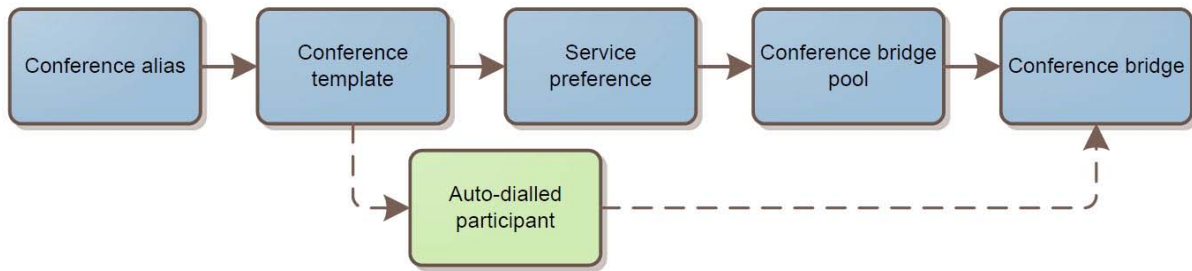
|   |   |
|---|---|
| Name                                    | * Lecture   |
| Description                             | <input type="text"/>  |
| Conference type                         | Lecture <input type="button" value="i"/>                        |
| No. of chairperson ports to reserve     | * 1 <input type="button" value="i"/>                            |
| Call Policy mode                        | Off <input type="button" value="i"/>                            |
| Conference bridge Service Preference    | * Prefer SD with HD fallback <input type="button" value="i"/>   |
| No. of cascade ports to reserve         | * 1 <input type="button" value="i"/>                            |
| Limit number of participants            | <input type="checkbox"/> Maximum <input type="text" value="0"/> |
| Limit the conference duration (minutes) | <input type="checkbox"/> Maximum <input type="text" value="0"/> |

**Advanced**

|                                      |  |
|--------------------------------------|--|
| Conference layout                    |  <input type="button" value="Choose layout"/> <input type="button" value="i"/> |
| Chair PIN                            | 55105 <input type="button" value="i"/>   |
| Guest PIN                            | <input type="text"/> <input type="button" value="i"/>  |
| Parameters to pass on to primary MCU | <input type="text" value="New"/>   |
| Parameters to pass on to cascade MCU | <input type="text" value="New"/>   |
| Allow content                        | Yes <input type="button" value="i"/>   |

4. Click **Create conference template**.

## Step 11: Creating an auto-dialed participant for the 'Meeting' template













1. Go to the **Auto-dialed participants** page (**Conference configuration > Auto-dialed participants**)
2. Click **New**
3. Input the following into the relevant fields, leave other fields as their default values:

| Field                 | Input  |
|-----------------------|--|
| Name                  | Enter 'Invite boss to meeting' for example                         |
| Conference template   | Select <i>HD Meeting</i>   |
| Conference name match | Enter <code>meet \.boss \. (HD   SD)</code>                        |
| Address               | Enter <code>boss@&lt;SIP domain&gt;</code>                         |
| Protocol              | Select a protocol supported by the endpoint and the video network. |
| Role type             | Select <i>Participant</i>  |
| Keep conference alive | Select <i>Yes</i>  |

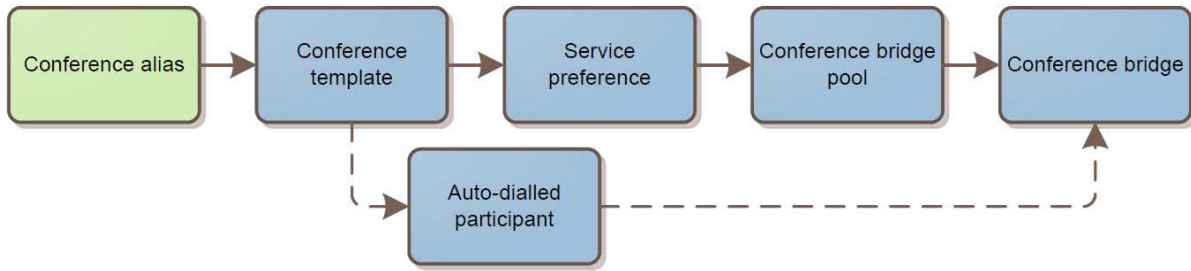
## Auto-dialed participants

**Modify participant**

|  |   |
|--|---|
| Name                                   | <input type="text" value="Invite boss to meeting"/>  |
| Description                            | <input type="text"/>                                 |
| Conference template                    | <input type="text" value="HD Meeting"/>              |
| Conference name match (must use regex) | <input type="text" value="meetl.boss\(HD SD)"/>      |
| Participant address                    | <input type="text" value="boss@vcs.domain"/>         |
| Protocol                               | <input type="text" value="H.323"/>                    |
| Role type                              | <input type="text" value="Participant"/>             |
| DTMF sequence                          | <input type="text"/>                                 |
| Keep conference alive                  | <input type="text" value="Yes"/>                       |
| Additional parameters                  | <input type="text" value="New"/>  |
| State                                  | <input type="text" value="Enabled"/>               |

4. Click **Create participant**.

## Step 12: Creating a conference alias for the 'SD Meeting' template with a role of 'Participant'



1. Go to the **Conference aliases** page (**Conference configuration > Conference aliases**).
2. Click **New**.
3. Input the following into the relevant fields, leave other fields as their default values:

| Field               | Input                             |
|---------------------|-----------------------------------|
| Name                | Enter 'SD meeting'                |
| Incoming alias      | Enter (meet\..*\SD)@< SIP domain> |
| Conference name     | Enter \1                          |
| Priority            | Enter '40'                        |
| Conference template | Select <i>SD Meeting</i>          |
| Role Name           | Select <i>Participant</i>         |

**Conference aliases** You are here: [Conference c](#)

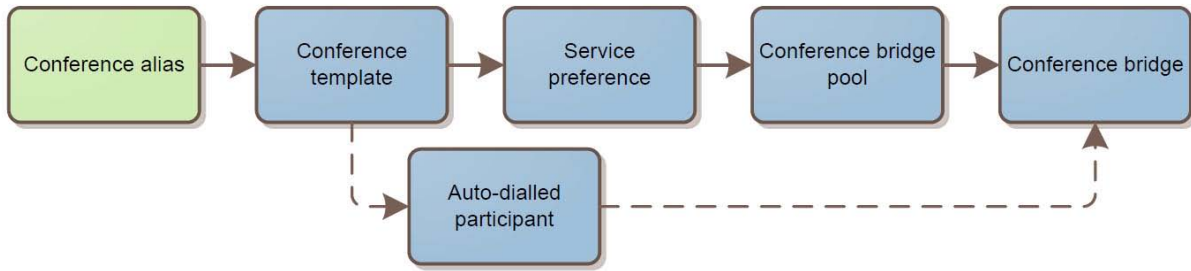
**Modify conference alias**

|   |                            |                   |
|---|----------------------------|-------------------|
| Name  | ★ SD meeting               | <a href="#">i</a> |
| Description                                     |                            | <a href="#">i</a> |
| Incoming alias (must use regex)                 | ★ (meet\..*\SD)@vcs\domain | <a href="#">i</a> |
| Conference name (must use regex replace string) | ★ \1                       | <a href="#">i</a> |
| Priority  | ★ 40                       | <a href="#">i</a> |
| Conference template                             | ★ SD meeting               | <a href="#">i</a> |
| Role name                                       | Participant                | <a href="#">i</a> |

4. Click **Create conference alias**.



## Step 13: Creating a conference alias for the 'HD Meeting' template with a role of 'Participant'



1. Go to the **Conference aliases** page (**Conference configuration > Conference aliases**).
2. Click **New**.
3. Input the following into the relevant fields, leave other fields as their default values:

| Field               | Input   |
|---------------------|---|
| Name                | Enter 'HD meeting' for example                      |
| Incoming alias      | Enter <code>(meet\..*\HD)@&lt;SIP domain&gt;</code> |
| Conference name     | Enter <code>\1</code>                               |
| Priority            | Enter '25' for example                              |
| Conference template | Select <i>HD Meeting</i>                            |
| Role Name           | Select <i>Participant</i>                           |

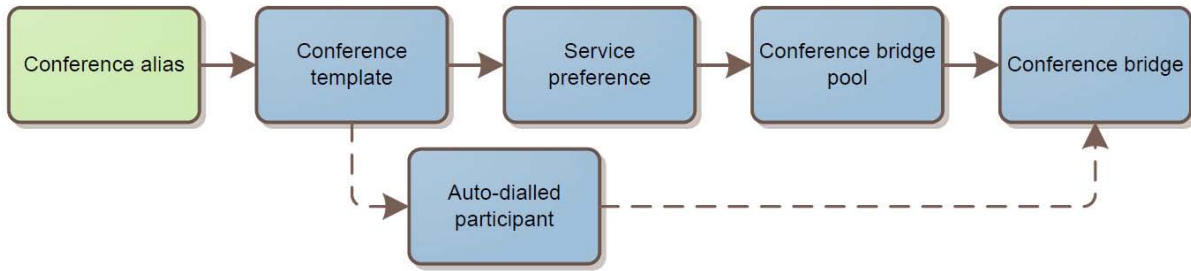
**Conference aliases** You are here: [Conference c](#)

**Modify conference alias**

|   |   |                          |   |
|---|---|--------------------------|---|
| Name  | * | HD meeting               | i |
| Description                                     |   |                          | i |
| Incoming alias (must use regex)                 | * | (meet\..*\HD)@vcs\domain | i |
| Conference name (must use regex replace string) | * | \1                       | i |
| Priority  | * | 25                       | i |
| Conference template                             | * | HD Meeting               | i |
| Role name                                       |   | Participant              | i |

4. Click **Create conference alias**.

## Step 14: Creating a conference alias for the ‘Lecture’ template with a role of ‘Chairperson’



1. Go to the **Conference aliases** page (**Conference configuration > Conference aliases**).
2. Click **New**
3. Input the following into the relevant fields, leave other fields as their default values:

| Field               | Input   |
|---------------------|---|
| Name                | Enter 'teacher' for example                       |
| Incoming alias      | Enter <code>teach\.(.*)@&lt;SIP domain&gt;</code> |
| Conference name     | Enter <code>\1</code>                             |
| Priority            | Enter '10' for example                            |
| Conference template | Select <i>Lecture</i>                             |
| Role name           | Select <i>Chairperson</i>                         |

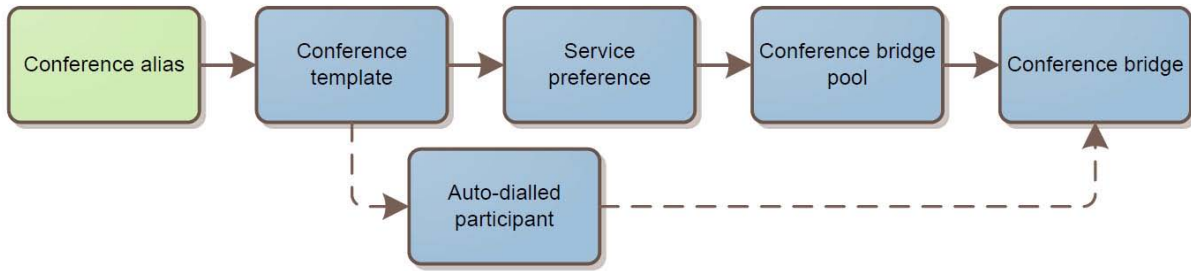
**Conference aliases** You are here: [Conference aliases](#)

**Modify conference alias**

|   |   |
|---|---|
| Name  | <input type="text" value="teacher"/> <i>i</i>                 |
| Description                                     | <input type="text"/> <i>i</i>                                 |
| Incoming alias (must use regex)                 | <input type="text" value="teach\.(.*)@vcs\.domain"/> <i>i</i> |
| Conference name (must use regex replace string) | <input type="text" value="\1"/> <i>i</i>                      |
| Priority  | <input type="text" value="10"/> <i>i</i>                      |
| Conference template                             | <input type="text" value="Lecture"/> <i>i</i>                 |
| Role name                                       | <input type="text" value="Chairperson"/> <i>i</i>             |

4. Click **Create conference alias**.

## Step 15: Creating a conference alias for the 'Lecture' template with a role of 'Guest'



1. Make sure you are on **Conference aliases** page (**Conference configuration > Conference aliases**).
2. Click **New**.
3. Input the following into the relevant fields, leave other fields as their default values:

| Field               | Input   |
|---------------------|---|
| Name                | Enter 'student' for example                         |
| Incoming alias      | Enter <code>student\.(.*)@&lt;SIP domain&gt;</code> |
| Conference name     | Enter <code>\1</code>                               |
| Priority            | Enter '15' for example                              |
| Conference template | Select <i>Lecture</i>                               |
| Role Name           | Select <i>Guest</i>                                 |

**Conference aliases** You are here: [Conference con](#)

**Modify conference alias**

|   |  |          |
|---|--|----------|
| Name  | <input type="text" value="student"/>                   | <i>i</i> |
| Description                                     | <input type="text"/>                                   | <i>i</i> |
| Incoming alias (must use regex)                 | <input type="text" value="student\.(.*)@vcs\.domain"/> | <i>i</i> |
| Conference name (must use regex replace string) | <input type="text" value="\1"/>                        | <i>i</i> |
| Priority  | <input type="text" value="15"/>                        | <i>i</i> |
| Conference template                             | <input type="text" value="Lecture"/>                   | <i>i</i> |
| Role name                                       | <input type="text" value="Guest"/>                     | <i>i</i> |

4. Click **Create conference alias**.

**Note:** setting **Call Policy mode** to **On** for a template allows control over who is able to create conferences based on that template, but only operates correctly if used in conjunction with call policy on the Cisco VCS. For more information, see *Cisco VCS Administrator Guide*.

# 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.

## Creating a meeting

To test that two or more endpoints can join a HD conference that is based on a template with a type of **meeting**, dial **meet.test.HD@<SIP domain>** from each endpoint. Both endpoints should be taken to the same conference.

## Adding an auto-dialed participant

To test that auto-dialed participants are called when a conference based on an appropriate template is created, dial **meet.boss.HD@<SIP domain>** from an endpoint. The auto-dialed participant **boss@<SIP domain>** should receive a call from the conference bridge.

## Creating a lecture

To test that two or more endpoints can use different aliases to join the same conference based on a template with a type of **lecture**, dial **teach.test@vcs.domain** from one endpoint and **student.test@vcs.domain** with the others. All endpoints should be taken to the same conference. The endpoints that dialed **student.test@vcs.domain** will see a blank screen until the endpoint that dialed **teach.test@vcs.domain** enters the conference.

## Cascading

To check that cascading is working properly it is necessary to occupy all the ports on the first conference bridge so that the TelePresence Conductor cascades the conference to the second conference bridge. If there are enough endpoints registered to the VCS you can test this by adding callers to the conference until it is cascaded.

Alternatively, you can increase the number of chairperson ports to be reserved by a lecture type template to a level that fills the primary conference bridge. This will cause the conference to be cascaded when guests dial in to a conference that is based on that template.

# Troubleshooting

## Tracking a call from VCS to TelePresence Conductor

To see the events associated with a particular call on both VCS and TelePresence Conductor look at the search history on the VCS (**Status > Search** history, then click **View** for a particular call). Searching for the Tag associated with that call in the event log on the TelePresence Conductor yields the events associated with that call:

- for calls which create conferences this tag is then associated with all future events associated with this conference (for example, conference destruction and auto-dialed participant requests to the conference bridge)
- for calls which are joining existing conferences the tag is associated with their conference join request.

A full explanation of all the terms in the event log can be found in the Cisco TelePresence Conductor Administrator Guide (D14826).

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**Note:** the call tag is specific to a call across multiple VCSs.

---

## Tracking a conference on the TelePresence Conductor

To see all events associated with a particular conference alias (i.e. across multiple individual conferences) filter by `Conference_alias_UUID` in the event log either by copying it to the filter box from the event log or by clicking on the hyperlink.

## Specific issues

### Call does not connect

If a call fails to connect:

1. On the Cisco VCS, look at the **Search details** for the call (go to **Status > Search history** and click **View** on the relevant call).
  - Check that the TelePresence Conductor search rule is being applied, under **Search details** the name of the search rule pointing at TelePresence Conductor should look like:  
**SearchRule (1)**  
**Name:** To Conference Policy Service
  - If the search rule is not used go to **VCS configuration > Dial plan > Search rules** look under **State** and check the pattern is active. Open a separate tab at **Maintenance > Tools > Check pattern**. This tool checks pattern matches. Under **Pattern type** select regex and copy the relevant **Pattern string** and **Replace string** from the **Search rules** page as well as the destination alias from the **Search history** page.
2. On the Cisco VCS look under **Status > Search history** to see if the ARQ message under **Status** lists as **TelePresence Conductor policy service unavailable**. This is the default reply provided by the VCS, and indicates that the TelePresence Conductor was unavailable.
  - On the **Cisco VCS**, check the connectivity with the TelePresence Conductor by going to **VCS configuration > Dial plan > Policy services** and click **View/Edit** for the TelePresence Conductor policy service. In the **Status** section at the bottom of the page, it should show the **State** as *Active*. If it shows *Inactive*, further details are shown in the top section next to the **Server 1 address** field.

- On the TelePresence Conductor, check the connectivity with the conference bridges by going to **Conference configuration > Conference bridges > Conference bridge pool**. If the **Status** column shows any of the conference bridges as *Unusable* then check the connectivity to the conference bridges and the authentication used.
3. If the SETUP message has status of **Forbidden**, check that:
    - The conference bridge pool has sufficient ports free to connect the call with the number of ports requested by the template.
    - The number of ports reserved for cascading is sufficient.
    - The number of ports reserved for chairpersons is not too high.

## Auto-dialed participant not dialed

If the auto-dialed participant does not get called:

1. Go to **Status > Search history** on the VCS and see what alias the conference bridge called. If no alias was called go to 2. If the alias is incorrect, rectify the **Address** field on TelePresence Conductor for the auto-dialed participant under **Conference configuration > Auto-dialed participants** on the TelePresence Conductor.
2. If **no call is made** check the Conference name match for the auto-dialed participant under **Conference configuration > Auto-dialed participants** on the TelePresence Conductor. Additionally check that all conference bridges which you expect to be registered to the Cisco VCS are actually registered, and that they are registering the expected aliases (on the VCS, go to **Status > Registrations > By alias**). This is essential if outbound calls from the conference bridge to auto-dialed participants are to be routed correctly.

## Conference bridges not registering with VCS

If the conference bridges are not registering with the VCS using either H.323 or SIP:

1. Check whether there are any registration restriction policies in place on the Cisco VCS (go to **VCS configuration > Registration > Configuration**). If there are, either:
  - Ensure that the policies are set up in such a way to allow the conference bridges to register.
  - Change the URIs registered by the conference bridges to a format that is compatible with the registration restriction policy.
2. Ensure that the conference bridge is configured exactly as described in the section *Configuring the conference bridges*.

If the conference bridges are not registering using SIP:

1. Review the SIP domain configuration on the Cisco VCS (**VCS configuration > Protocols > SIP > Domains**). Ensure the SIP domain of the conference bridge(s) that are trying to register is present. If not, either:
  - Change the SIP domain of the conference bridge(s) to be compatible.
  - Create a new SIP domain on the VCS.
2. Ensure that the conference bridge is configured exactly as described in the section *Configuring the conference bridges*.

If the conference bridges are not registering using H.323:

Ensure that the conference bridge is configured exactly as described in the section *Configuring the conference bridges*.

# Maintenance routine

## System backup

To create a system backup:

1. Go to the **Backup and restore** page (**Maintenance > Backup and restore**).
2. Click **Create system backup file**.
3. Wait for the file download dialog to appear.
4. Click **Save** and save the backup file to an appropriate location.

For more information see *Cisco TelePresence Conductor Administrator Guide (D14826)* or the TelePresence Conductor's online help.

# Appendix 1: Example call flows

## H.323 call flow

The following diagram shows a breakdown of the H.323 call flow:

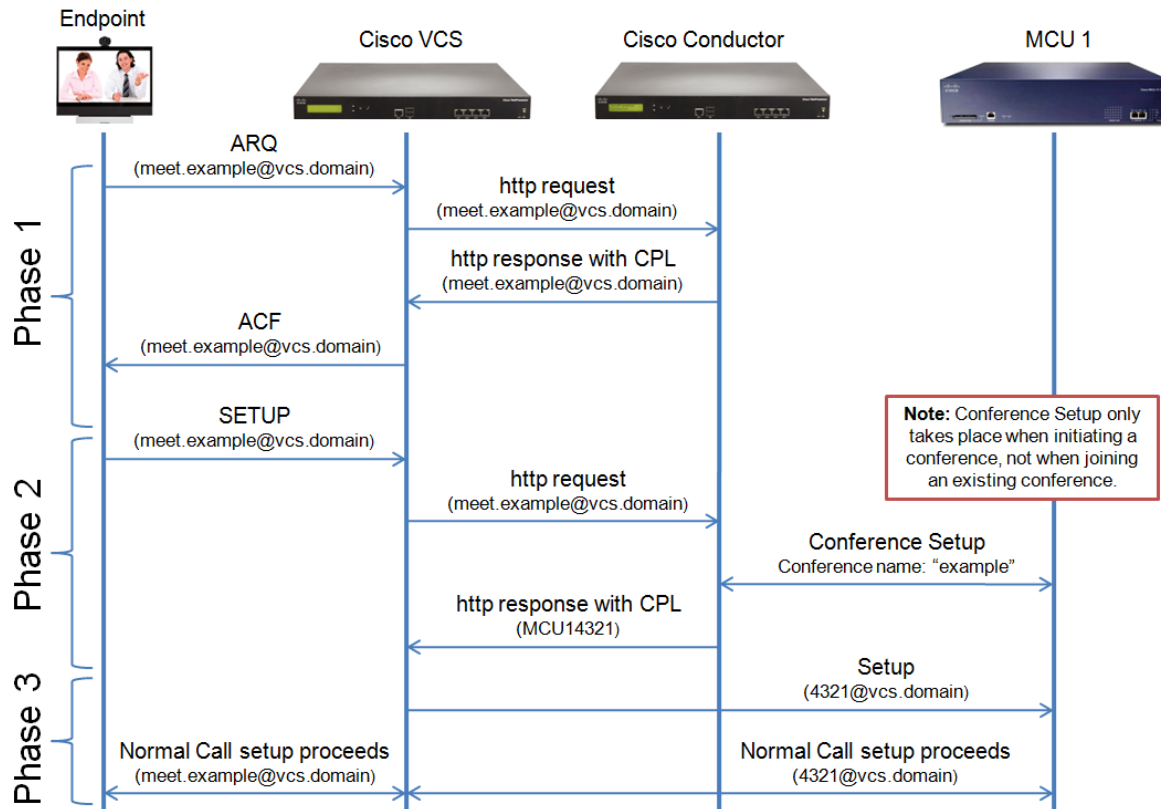


Figure 1: H.323 call flow

### Phase 1

The endpoint sends an ARQ to the Cisco VCS which matches the search rule for the TelePresence Conductor policy service. The Cisco VCS sends an HTTP request to the TelePresence Conductor. The HTTP response from the TelePresence Conductor contains CPL which tells the VCS to act as though the endpoint has been located and to proceed with call setup with the endpoint by sending an ACF to it.

### Phase 2

The VCS receives the SETUP message from the endpoint which again matches the TelePresence Conductor policy service search rule. It sends a second HTTP request to the TelePresence Conductor. If the conference is new, a fresh conference is set up on the conference bridge by the TelePresence Conductor. TelePresence Conductor sends another piece of CPL to the VCS telling it to forward the call to **MCU14321**.

### Phase 3

The VCS matches the alias **MCU14321** to its search rule **To MCU 1**. The prefix **MCU1** is stripped and the message is sent to the conference bridge neighbor zone. MCU1 picks up the call and normal H.323 call setup now proceeds.



Note: Phase 2 and 3 occur in the same way when a SIP INVITE is received by TelePresence Conductor rather than a H.323 SETUP. Phase 1 is H.323 specific.

## Cascade creation call flow

The following diagram shows a breakdown of the call flow when a cascade is created. This diagram excludes the ARQ request/response shown in the previous diagram (Figure 1: H.323 call flow) and proceeds from the point where the endpoint sends the SETUP message:

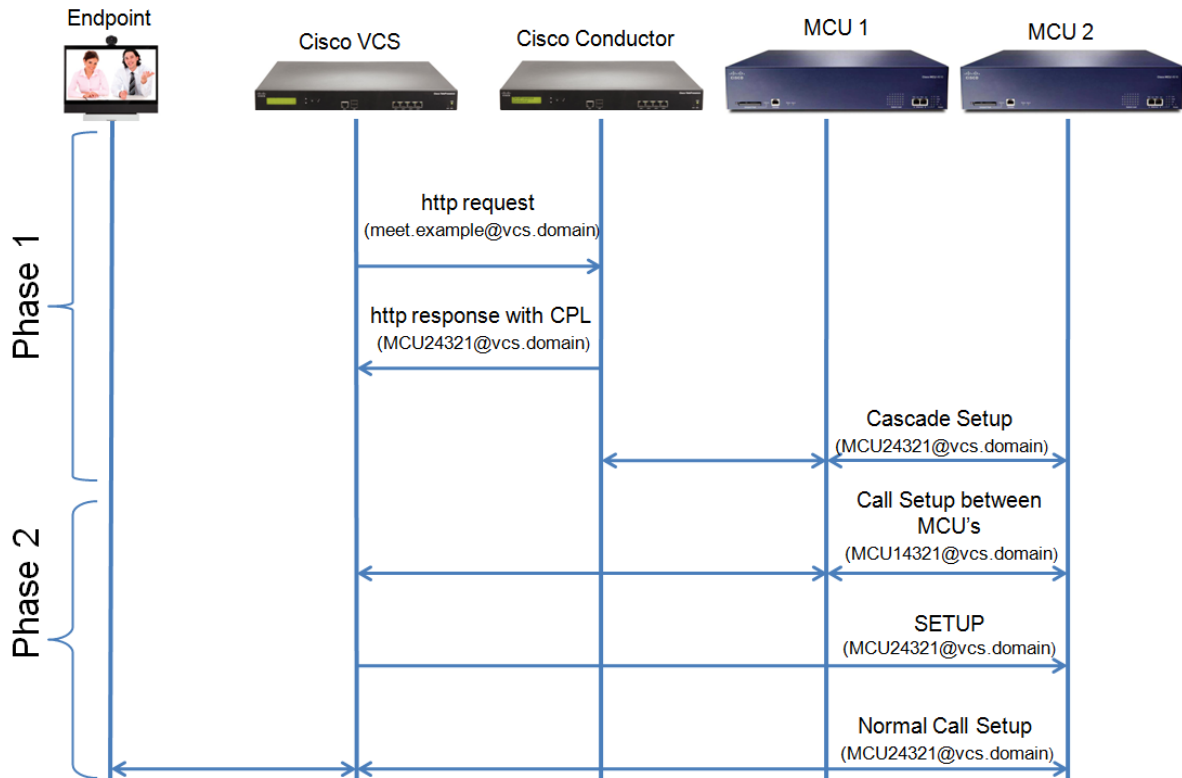


Figure 2: Cascade creation call flow

### Phase 1

The VCS receives the SETUP message from the endpoint which matches the TelePresence Conductor policy service search rule. It sends an HTTP request to the TelePresence Conductor. The TelePresence Conductor knows that there are not enough available ports on MCU 1 and that a cascade should be created. The VCS receives another piece of CPL from the TelePresence Conductor telling it to forward the call to **MCU24321** (the primary conference on MCU 1). The TelePresence Conductor contacts the conference bridges and instructs them to set up the cascade.

### Phase 2

Under instructions from the TelePresence Conductor, MCU 2 sets up an H.323 call to the alias **MCU14321**. The call setup travels through the VCS to MCU 1. Concurrently the SETUP message from the endpoint is connected to MCU 2 and normal H.323 call setup between the two then takes place.

## Appendix 2: Using TelePresence Conductor and TMS

As of 13.2 the features supported by TMS with TelePresence Conductor are limited to conference monitoring and control through the Conference Control Centre on TMS. As of TelePresence Conductor 1.1 and TMS 13.2 no scheduling support is included.

TMS versions earlier than 13.2 do not include TelePresence Conductor support.

The following sections describe the steps necessary to add TelePresence Conductor to TMS and see conferences initiated using TelePresence Conductor in the Conference Control Centre.

### Step 1: Configuring TelePresence Conductor to allow TMS connectivity

1. Log into the TelePresence Conductor as a user with admin rights.
2. Go to **System > SNMP**
3. Input the following into the relevant fields, leave other fields as their default values:

| Field          | Input   |
|----------------|---|
| SNMP mode      | Select <i>v3 plus TMS support</i>   |
| Community name | Enter a valid community name for your TMS, the default for both products is public valid community names can be checked on the TMS under <a href="#">Administrative Tools &gt; Network Settings</a> |
| Location       | Enter a description of the physical location of the TelePresence Conductor.   |

#### SNMP

##### Configuration

|                |  |
|----------------|--|
| SNMP mode      | v3 plus TMS support <span style="float: right;">i</span> |
| Community name | public <span style="float: right;">i</span>              |
| System contact | Administrator <span style="float: right;">i</span>       |
| Location       | Server Room 1  |
| Username       | public <span style="float: right;">i</span>              |

##### v3 Authentication

|                     |  |
|---------------------|--|
| Authentication mode | Off <span style="float: right;">i</span> |
|---------------------|--|

4. Click **Save**.

## Step 2: Configuring conference bridges used by TelePresence Conductor to allow TMS connectivity

For each conference bridge that is used by TelePresence Conductor to host Conferences:

1. In **Network > Services** ensure the SNMP port is enabled and set to port 161.
2. In **Network > SNMP** ensure the RO, RW and Trap community are set to public, private and public respectively.
3. In **Network > Port A**, configure a host name for your conference bridge. (If Cisco TMS is to manage your conference bridge using port B, then configure the host name in **Network > Port B**)

## Step 3: Adding TelePresence Conductor to the TMS

1. Log into the TMS as a user with administrator rights.
2. Go to **Systems > Navigator**
3. Click **Add Systems**
4. Enter the IP address or DNS name of the TelePresence Conductor
5. Click the **Advanced settings** tab to open it.
6. In the **Username** and **Password** (*not the "Admin Password" field*) fields enter the username and password of a user with admin rights on the TelePresence Conductor.
7. Click **Add Systems**

## Step 4: Adding the conference bridges to TMS

For each conference bridge to be added to TMS

1. In Cisco TMS go to **Systems > Navigator** and click **Add systems**.
2. Enter the IP address or DNS name of the Cisco TelePresence MCU.
3. Click the **Advanced settings** tab to open it.
4. In the **Username** and **Password** (*not the "Admin Password" field*) fields enter the username and password of a user with admin rights on the conference bridges.
5. Click **Add Systems**.
6. You should be returned to a screen indicating that your system has been added. Click **Finish adding systems**

## Step 5: Additional TMS configuration

1. In Cisco TMS go to **Administrative Tools > Conference Settings** and ensure that under TMS services **Enable Ad Hoc Conference Discovery** is set to yes.
2. On the same page under **TMS services** next to **Enforce Management Settings On Systems** click **Enforce Now**

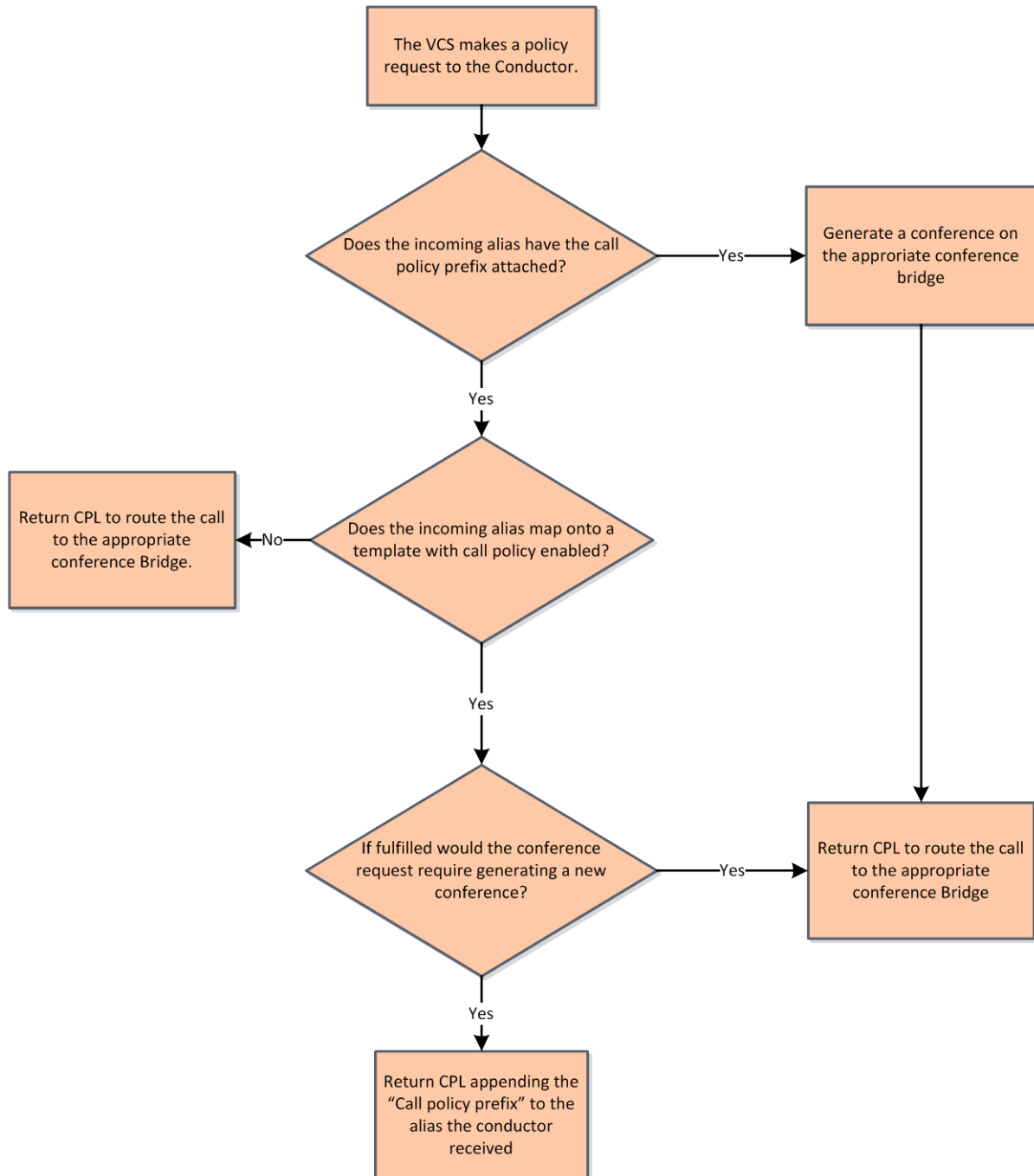
Cisco TMS should now be configured to manage TelePresence Conductor created conferences in Conference Control Centre.

## Appendix 3: Call policy mode

Call policy mode is activated on a per-template basis on the Cisco TelePresence Conductor when active. It allows the VCS to make call policy decisions about which endpoints are allowed to create a conference.

When a call policy mode enabled TelePresence Conductor receives a call that will generate a new conference TelePresence Conductor returns call policy which attaches the **Call policy prefix** to the dialed alias. This allows policy on the VCS to act on calls with the **Call policy prefix**. By allowing or denying these calls to be routed back to the TelePresence Conductor the VCS can control which users are allowed to create conferences.

TelePresence Conductor's behavior with call policy enabled on one or more templates is detailed below:



There are three main ways a VCS can filter these calls. The first two, Search Rules and Call Policy Rules, are detailed below.

The third method is by writing an external policy server. For help in doing so please see *External policy server deployment guide* D14854.01. External policy servers used in conjunction with the Cisco VCS offer powerful and fine grained methods for controlling call routing.

## Configuring call policy on the TelePresence Conductor

### Step 1: Configuring call policy for an existing template

In all cases it is first necessary to (on the TelePresence Conductor):

1. Go to **Conference Configuration > Conference templates** then:
2. Click on the conference templates for which you wish to enable Call Policy mode
3. On the drop down menu for “Call Policy mode” select **On**
4. Click **Save**

## Step 2: Configuring the call policy prefix

The call policy prefix is configurable under **Conference Configuration > Call Policy**. The default is “**create.**”, which is what will be used in the examples that follow. To change the call policy prefix:

1. Go to **Conference Configuration > Call policy prefix** then:
2. In the field marked “Call policy prefix” enter the desired call policy prefix.
3. Click **Save**.

## Using search rules to limit the ability to create conferences to authenticated users

Limiting the ability to create conferences to authenticated users only requires two search rules on the VCS pointing at the TelePresence Conductor policy service. The first has already been created in Step 2 of the VCS configuration. The second matches requests with the call policy prefix attached. To configure this:

1. Go to **VCS configuration > Dial plan > Search rules**
2. Click **Create new search rule**
3. Add “**(create\.)?**” to the start of the Pattern String. (The question mark and parentheses make the create\ part of the match optional.) Go to the **Search rules** page (**VCS configuration > Dial Plans > Search Rules**).
4. Click **New**.
5. Input the following into the relevant fields, leave other fields as their default values:

| Field                         | Values   |
|-------------------------------|--|
| Rule Name                     | Enter 'Authenticated users to Conductor Policy Service'  |
| Description                   | Enter 'This search rule only matches authenticated users dialing aliases with the call policy prefix attached' |
| Priority                      | Enter '120'  |
| Source                        | Select <i>Any</i>  |
| Request must be authenticated | Select <i>Yes</i>  |
| Mode                          | Select <i>Alias pattern match</i>  |
| Pattern type                  | Select <i>Regex</i>  |
| Pattern string                | Enter <code>create\.(meet teach student)\.\.*@&lt;SIP domain&gt;</code>  |
| Pattern Behavior              | Select <i>Leave</i>  |
| On Successful Match           | Select <i>Stop</i>   |
| Target                        | Select <i>Conductor Policy Service</i>   |
| State                         | Select <i>Enabled</i>  |

### Create search rule

Configuration

|                               |   |   |   |
|-------------------------------|---|---|---|
| Rule name                     | * | Authenticated users to Conductor Policy Service   | i |
| Description                   |   | This search rule only matches authenticated users | i |
| Priority                      | * | 120   | i |
| Source                        |   | Any   | i |
| Request must be authenticated |   | Yes   | i |
| Mode                          |   | Alias pattern match                               | i |
| Pattern type                  |   | Regex   | i |
| Pattern string                | * | create\.(meet teach student)\.*@vcs\.domain       | i |
| Pattern behavior              |   | Leave   | i |
| On successful match           |   | Stop  | i |
| Target                        | * | Conductor Policy Service                          | i |
| State                         |   | Enabled   | i |

Create search rule
Cancel

6. Click **Create search rule**

## Using call policy rules on the VCS to limit the ability to create conferences to a range of aliases

### Step 1: Configuring a call policy mode search rule on the VCS

When using a call policy mode the VCS needs to send both the first request for a conference to the TelePresence Conductor and also the second request with the call policy prefix attached. The call policy prefix in use is the “**create.**” call policy prefix. To achieve this:

1. Go to **VCS configuration > Dial plan > Search rules**
2. Click on the search rule named **To Conductor Policy Service** created in Step 3: Configuring a search rule with the TelePresence Conductor policy service as the target.
3. Add “**(create\.)?**” to the start of the Pattern String. (The question mark and parentheses make the create\ part of the match optional.)

### Edit search rule

**Configuration**

|                               |   |   |   |
|-------------------------------|---|---|---|
| Rule name                     | * | <input type="text" value="To Conductor Policy Service"/>                  | i |
| Description                   |   | <input type="text"/>  | i |
| Priority                      | * | <input type="text" value="110"/>  | i |
| Source                        |   | <input type="text" value="Any"/>  | i |
| Request must be authenticated |   | <input type="text" value="No"/>   | i |
| Mode                          |   | <input type="text" value="Alias pattern match"/>                          | i |
| Pattern type                  |   | <input type="text" value="Regex"/>  | i |
| Pattern string                | * | <input type="text" value="(create\.)?(meet teach guest)\..*@vcs.domain"/> | i |
| Pattern behavior              |   | <input type="text" value="Leave"/>  | i |
| On successful match           |   | <input type="text" value="Stop"/>   | i |
| Target                        | * | <input type="text" value="Conductor Policy Service"/>                     | i |
| State                         |   | <input type="text" value="Enabled"/>                                      | i |

#### 4. Click **Save**

---

**Note:** It is not possible to use call policy rules in conjunction with local CPL. Call policy rules are a way of generating local CPL without having to write scripts. If using an uploaded local CPL script is imperative but source alias call filtering is necessary then either extend the existing CPL script or consider using an external policy server.

---

The following set of instructions will guide you through the configuration necessary to allow only users registered with the domain "**vcs.domain**" to create conferences.

To use call policy rules log into the VCS then:

1. Go to **VCS configuration > Call Policy > Configuration**.
2. If it is not already selected, select for "Call Policy mode" select **Local CPL**
3. Click **Save**
4. If the button is present click **Delete uploaded file**
5. Go to **VCS configuration > Call Policy > Rules**.
6. Click **New**



7. Input the following into the relevant fields, leave other fields as their default values:

| Field               | Input  |
|---------------------|--|
| Source pattern      | Enter <code>.*@&lt;SIP domain&gt;</code>                             |
| Destination pattern | Enter <code>create\.(meet teach guest)\..*@&lt;SIP domain&gt;</code> |
| Action              | Select <i>Allow</i>  |

### Add Call Policy rule

**Add Call Policy rule**

Source pattern  i

Destination pattern \*  i

Action Allow i

**Tips**

Tips: Setting the source pattern to an empty string v

To match all addresses, use the pattern '.\*' (w

Add
Delete
Cancel

8. Click **Add**

---

**Note:** Call policy rules implicitly allow calls. The next steps are necessary to create rule to deny calls which do not match the `.*@<SIP domain>` pattern.

---

9. Click **New**

10. Input the following into the relevant fields, leave other fields as their default values:

| Field               | Input  |
|---------------------|--|
| Source pattern      | Enter <code>.*@.*</code>   |
| Destination pattern | Enter <code>create\.(meet teach guest)\..*@&lt;SIP domain&gt;</code> |
| Action              | Select <i>Reject</i>   |

### Add Call Policy rule

**Add Call Policy rule**

|                     |  |  |
|---------------------|--|--|
| Source pattern      | <input type="text" value=".*@.*"/>                         |  |
| Destination pattern | <input type="text" value="*teach(guest)...\*@vcs.domain"/> |  |
| Action              | <input type="text" value="Reject"/>                        |  |

**Tips**

Tips:

- Setting the source pattern to an empty string will match all addresses.
- To match all addresses, use the pattern `.*@.*`.

11. Click **Add**

## Appendix 4: Using TelePresence Conductor and Multiway

It is possible to use conference bridges managed by Cisco TelePresence Conductor to host Multiway conferences. However, due to the way Multiway is implemented, when a user adds participants to an existing conference from their endpoint by selecting “join”, TelePresence Conductor is not consulted as to whether that endpoint can enter the conference. This leads to the following issues:

- Members of a conference can exceed the “max participant limit” setting on the conference by being joined using Multiway.
- In the case of a conference bridge with all ports either in use by a participant or reserved in TelePresence Conductor for content or cascading a participant joining would take a port from the ports that TelePresence Conductor has reserved. This would prevent that port from being used by the conference it was allocated to, potentially preventing that conference from cascading or being used to share content. In the case of Content ports this can be prevented on conference bridges by under **Settings > Conference settings** setting media port reservation to *Enabled*. This will however reserve a port for content for all Conferences on that conference bridge. In this case the templates directing conferences towards such conference bridges must all have **Allow content** set to yes.

To use TelePresence Conductor and Multiway together on the VCS it is necessary for the Multiway template to be incorporated in the dial plan in such a way that the aliases generated by it are sent to TelePresence Conductor.

In the case of the dial plan used in this document and the Multiway alias **multiway@vcs.domain** one would configure the VCS (under **Applications > Conference Factory**) like so:

**Configuration**

|                    |   |  |          |
|--------------------|---|--|----------|
| Mode               |   | <input type="text" value="On"/>                    | <i>i</i> |
| Alias              |   | <input type="text" value="multiway@vcs.domain"/>   | <i>i</i> |
| Template           |   | <input type="text" value="meet.%%.SD@vcs.domain"/> | <i>i</i> |
| Number Range Start | * | <input type="text" value="1"/>                     | <i>i</i> |
| Number Range End   | * | <input type="text" value="65535"/>                 | <i>i</i> |

## Appendix 5: Identifying Dedicated Content Ports on a Cisco TelePresence MCU

This information is available on the spec sheet for the TelePresence MCU, but it is also available through the web interface, the steps below describe how to locate and use this information.

1. Navigate to the TelePresence MCU in a browser.
2. Log in as administrator
3. Navigate to **Status > Conferences** and look at the line marked “Streaming and content ports in use **0 (0)/##**” Where ## is the number of dedicated content ports of this TelePresence MCU.

| Conference status                  |             |
|------------------------------------|-------------|
| Active conferences                 | 0           |
| Active auto attendants             | 0           |
| Completed conferences              | 9           |
| Completed auto attendants          | 0           |
| Active conference participants     | 0           |
| Previous conference participants   | 58          |
| Active streaming viewers           | 0 (0) / 24  |
| TCP streaming viewers              | 0 (0) / 24  |
| ConferenceMe users connected       | 0 (0) / 12  |
| Video ports in use                 | 0 (11) / 12 |
| Audio-only ports in use            | 0 (1) / 12  |
| Streaming and content ports in use | 0 (2) / 12  |

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