



Cisco TelePresence Conductor with Cisco VCS (Policy Service)

Deployment Guide

XC2.2
X7.0 and later

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Introduction

About the Cisco TelePresence Conductor

Cisco TelePresence Conductor manages video conference bridge resources, providing resiliency and increased capacity across your video conferencing network. A video providing a high level overview of the TelePresence Conductor can be found at <http://www.youtube.com/watch?v=4-C7F2fTEYE>. This video focuses on the capabilities of the product up to version XC1.2. Within this video you can see that the TelePresence Conductor integrates tightly with the Cisco TelePresence Video Communication Server (Cisco VCS) and the Cisco TelePresence MCU products. TelePresence Conductor versions XC2.0 and later extend the supported conference bridges to include the Cisco TelePresence Server and the supported call control devices to include the Cisco Unified Communications Manager.

The TelePresence Conductor enables endpoints with sufficient privileges to create and enter a conference by dialing a single number or URI (known as rendezvous conferences). It also supports Multiway conferences, which are initiated when two endpoints already in a call together add another endpoint.

The TelePresence Conductor performs conference bridge resource management and calls are routed to appropriate conference bridges by the Cisco VCS under instructions from the TelePresence Conductor. If the conference is hosted on a TelePresence MCU and the size of the conference grows beyond the capacity of a single conference bridge, the conference is cascaded to additional TelePresence MCU conference bridges. (Cascading with TelePresence Server is not supported in the XC2.2 release.)

The TelePresence Conductor is capable of preferentially selecting conference bridges for conferences based on their properties. For example, conference bridges could be selected based on geographic location or on video quality (such as HD or SD services).

This version of the TelePresence Conductor supports the Cisco VCS in the following two types of deployments:

- Using the Cisco VCS's external policy service interface
This method may be discontinued in future versions of the TelePresence Conductor software.
- Using the TelePresence Conductor's back-to-back user agent (B2BUA)
This method requires a SIP trunk between the Cisco VCS and the TelePresence Conductor. It is the preferred method to use.

This document describes the deployment method using the Cisco VCS's external policy server interface. For more information on the deployment using the TelePresence Conductor's B2BUA, see [Cisco TelePresence Conductor with Cisco VCS \(B2BUA\) Deployment Guide](#).

The TelePresence Conductor supports the Cisco VCS in standalone and clustered modes.

You can configure up to 20 TelePresence Conductors or TelePresence Conductor clusters per Cisco VCS or Cisco VCS cluster using a suitable non-overlapping dial plan.

About this document

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

- An endpoint user can call the rendezvous conference alias `meet.<meeting name>.HD@vcs.domain`. If they are the first person to call this alias, TelePresence Conductor creates a new conference and they are

routed to it. The conference is created preferentially on a conference bridge with high definition ports, if there are not any ports available on the HD conference bridge then the conference will be created on the SD conference bridge. Alternatively, if the conference already exists then the alias is routed to it.

- An endpoint user can call the rendezvous conference 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 not any 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` automatically dialed into the conference.
- 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 there are no SD ports available, a conference on the HD conference bridge.
- An endpoint user can call the alias `learn.<lecture name>@vcs.domain` and create or join a lecture-type conference as a chairperson on a conference bridge with SD ports or, if there are no SD ports 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, then 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 also describes how to check that the system is working as expected.

Detailed 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).

Further reading

This document focuses on the use of a single TelePresence Conductor. For more details on how to deploy a cluster of TelePresence Conductors see [Cisco TelePresence Conductor Clustering with Cisco VCS \(Policy Service\) Deployment Guide \(D14828\)](#).

For details on how to deploy TelePresence Conductor with Unified CM see [Cisco TelePresence Conductor with Cisco Unified Communications Manager Deployment Guide \(D14998\)](#).

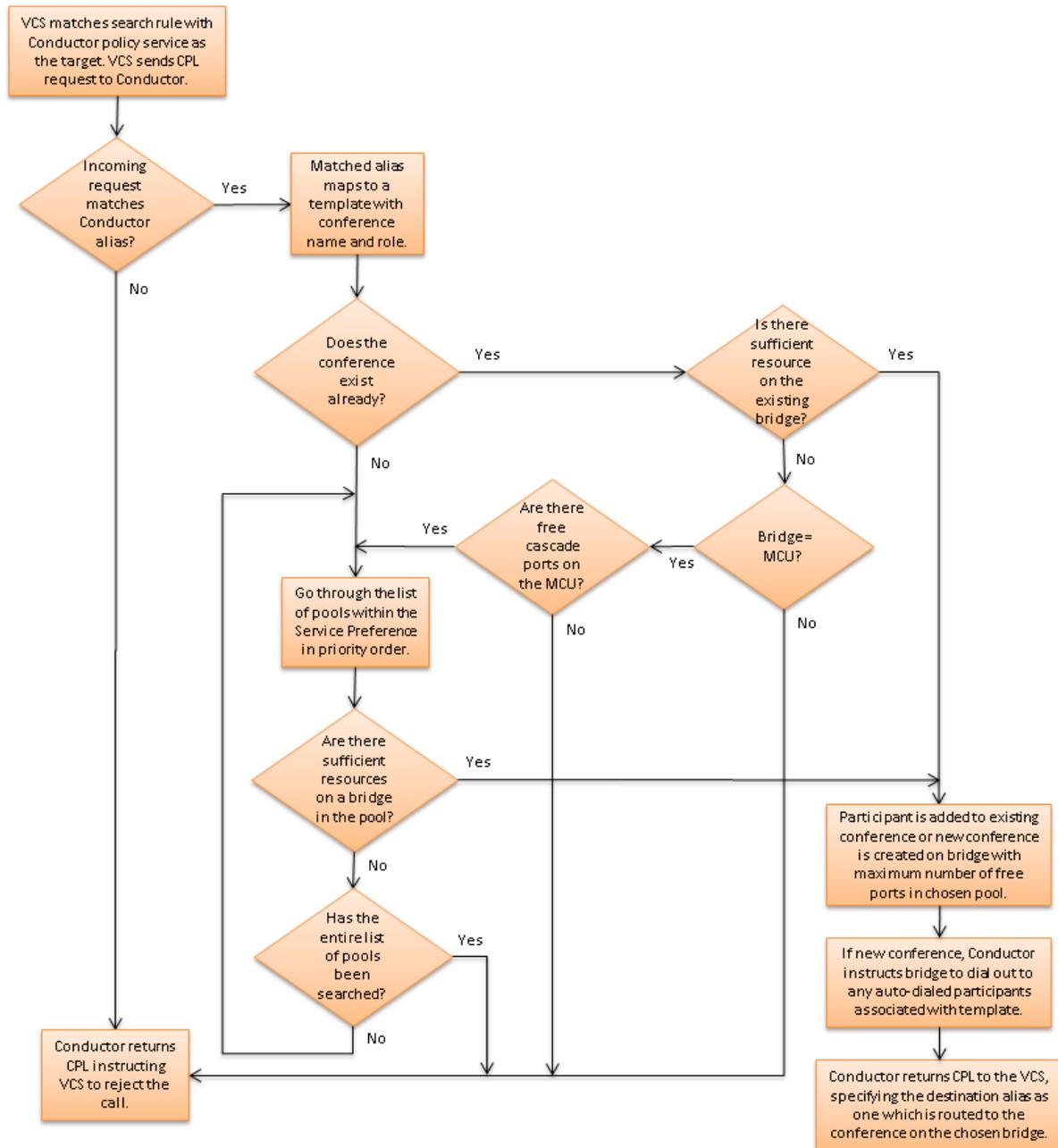
Call flow with the TelePresence Conductor

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:

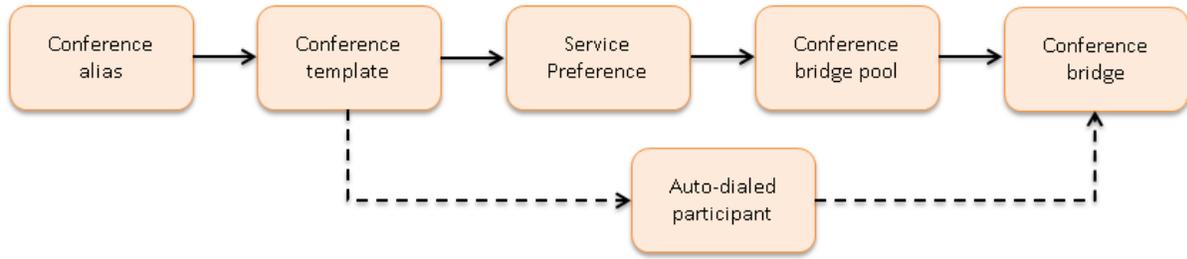


When these parts of the call flow are complete, the call is set up and media flows between the endpoint and the conference bridge.

TelePresence Conductor conference bridge selection process



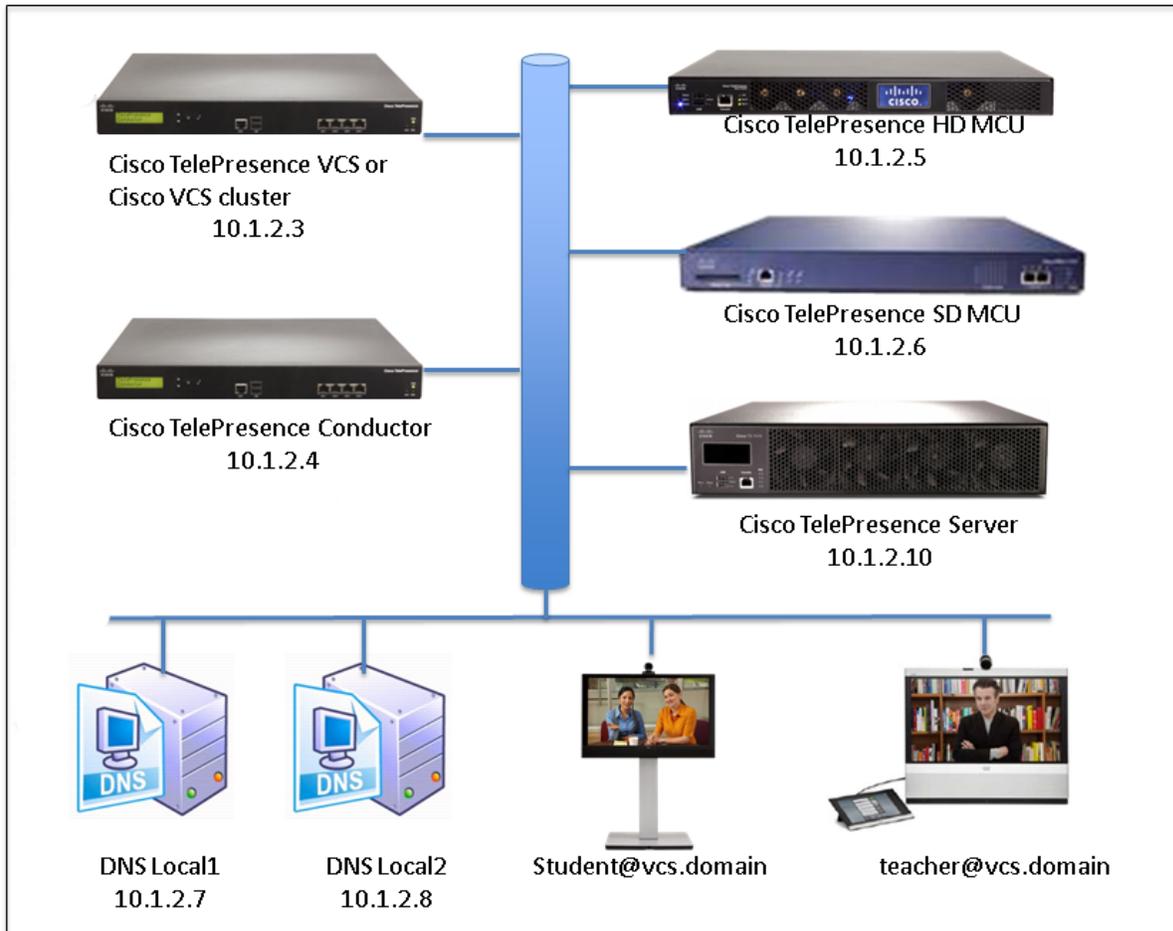
In a simplified format the set of steps for a conference to be created when the TelePresence Conductor receives an individual valid conference request is:



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

Example network deployment

The example network shown below is used as the basis for the deployment configuration described in this document.



Note that 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 `vcs.internal-domain.net` (which resolves to an IP address of 10.1.2.3 by the internal DNS servers).

Cisco TelePresence network elements

Cisco VCS

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

Conference bridges

Conference bridges are network devices that enable multiple video calls to come together in a multipoint video conference. TelePresence Conductor version XC2.2 supports the conference bridge types

TelePresence MCU and TelePresence Server.

Endpoints

These are devices that receive and make video calls. They can be software clients on PCs and Macs such as Jabber Video (Movi), desktop endpoints such as the EX90 and 9971, or room systems such as the MX300.

Deploying TelePresence Conductor with Cisco VCS

Prerequisites

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

- The Cisco VCS (or Cisco VCS cluster) must be running version X7.0 or later and must already be configured to act as an H.323 gatekeeper, a SIP registrar and proxy. Ensure that the system has been tested by registering at least three endpoints to it and that they are all capable of calling each other. For more information, see [VCS Administrator Guide](#).
- The TelePresence Conductor must be powered on, running version XC2.2 and accessible over the network. For assistance in reaching this stage, see [Cisco TelePresence Conductor Getting Started Guide](#).
- One or more conference bridges are powered on and accessible over the network. Basic configuration for the conference bridge should be completed as described in the relevant Getting Started Guide. These bridges must be dedicated for use by TelePresence Conductor – no other devices must try to route calls to them except via the TelePresence Conductor.
- The following Cisco TelePresence MCUs are supported by the TelePresence Conductor:
 - MCU 4200 series version 4.2 or later
 - MCU 4500 series version 4.2 or later
 - MCU 5300 series version 4.3(2.17) or later
 - MCU MSE 8420 version 4.2 or later
 - MCU MSE 8510 version 4.2 or later

Note: For all TelePresence MCUs we recommend software version 4.4 or later, otherwise some features will not be supported.

- The following Cisco TelePresence Servers are supported by the TelePresence Conductor:
 - TelePresence Server 7010 version 3.0(2.46) or later
 - TelePresence Server MSE 8710 version 3.0(2.46) or later
 - TelePresence Server version 3.1 or later on Virtual Machine
 - TelePresence Server version 3.1 or later on Multiparty Media 310/320
- Note:** For all TelePresence Servers we recommend software version 3.1 or later, otherwise some features will not be supported.

- This guide assumes the conference bridges are connected to the network on their port A.
- A web browser is available with access to the web interfaces of the Cisco VCS, TelePresence Conductor and conference bridges that are being configured.

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](#) 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 Cisco 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 in your deployment. This ensures that the dial plan is easy for an endpoint user to understand, and for administrators to manage.

TelePresence Conductor is compatible with the dial plan specified in the reference design, after 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 Cisco VCSs or Cisco VCS clusters to increase the resiliency of conference bridge dial out calls (to endpoints or during cascading) against Cisco VCS failure.

This deployment guide uses the following dial plan elements and configures the TelePresence Conductor and Cisco VCS accordingly:

Element	Format
Conference aliases for lecture chairpersons on TelePresence MCUs	teach.<name of lecture>@vcs.domain
Conference aliases for lecture guests on TelePresence MCUs	learn.<name of lecture>@vcs.domain
Conference aliases for high definition meeting participants on TelePresence MCUs	meet.<meeting name>.HD@vcs.domain
Conference aliases for standard definition meeting participants on TelePresence MCUs	meet.<meeting name>.SD@vcs.domain
Conference aliases for lecture chairpersons on TelePresence Servers	teachts.<name of lecture>@vcs.domain
Conference aliases for lecture guests on TelePresence Servers	learnts.<name of lecture>@vcs.domain
Conference aliases for high definition meeting participants on TelePresence Servers	meetts.<meeting name>.HD@vcs.domain
Conference aliases for standard definition meeting participants on TelePresence Servers	meetts.<meeting name>.SD@vcs.domain
Conference bridge prefixes for the TelePresence MCUs	HDMCU, SDMCU
Conference bridge prefixes for the TelePresence Servers	HDS, SDTS

Configuring the TelePresence MCUs

These tasks can be skipped, if only TelePresence Servers are used as conference bridges in your deployment.

Task 1: Creating a user

For the TelePresence Conductor to communicate with the TelePresence MCU it must use credentials for a user that has administrator rights. We recommend that you create a dedicated administrator level user for this task.

1. Go to the web interface of the TelePresence MCU you want to configure and log in as an administrator.
2. Go to **Users** and click **Add new user**.
3. Enter the following in the relevant fields:

User ID	Enter a username for the TelePresence Conductor to use.
Name	Enter a name for this user.
Password	Enter a password for the TelePresence Conductor to use.
Force user to change password on next login	Uncheck.
Privilege level	Select <i>administrator</i> .

User information

User ID: conductoradmin

Name: Conductor

Password: ●●●●●●●●

Re-enter password: ●●●●●●●●

Disable user account:

Lock password:

Force user to change password on next login:

Privilege level: administrator

E.164 phone number:

Associated video endpoint: <none>

Add user

4. Click **Add user**.
5. Repeat the steps for any other TelePresence MCUs.

Task 2: Installing an encryption key

The TelePresence MCU has the ability to use a secure connection for communications. These security features are enabled with the **Encryption** option key. You are recommended to install this option key for this deployment.

To verify that the key is installed or to install the key:

1. Go to **Settings > Upgrade**.
2. Go to the **Feature Management** section and verify that the **Encryption key** is installed. If the key is not

installed, enter the **Activation code** and click **Update features**.



To enable the use of encryption on the TelePresence MCU:

1. Go to **Settings > Encryption**.
2. Set **Encryption status** to *Enabled*.
3. Set **SRTP encryption** to *Secure transport (TLS) only*.
4. Click **Apply changes**.
5. Go to **Network > Services**.
6. Ensure that **Secure web (port 443)** is checked.
7. Ensure that **Encrypted SIP (TLS)** is checked.
SIP (TLS) must also be configured on the Cisco VCS in [Task 13: Adding each conference bridge as a neighbor zone \[p.24\]](#).
8. Ensure that **SIP (TCP)** is unchecked.
9. Ensure that **SIP (UDP)** is unchecked.
10. Click **Apply changes**.

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 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

UDP service		Port A
		IPv4
SNMP	<input type="checkbox"/>	161
SIP (UDP)	<input type="checkbox"/>	5060
H.323 gatekeeper	<input type="checkbox"/>	1719

Apply changes

- Repeat the steps for any other TelePresence MCUs

Task 3: Configuring SIP

- Go to **Settings > SIP**.
- Enter the following in the relevant fields, leave other fields as their default values:

SIP registrar usage	Select <i>Enabled</i> .
SIP registrar domain	Enter the Cisco VCS's SIP domain.
Username	Enter <i>sdmcu</i> for example.
Allow numeric ID registration for conferences	Uncheck.
SIP proxy address	Enter the Cisco VCS's IP address.
Maximum bit rate from Microsoft OCS/LCS clients	Select <i>limit disabled</i> .
Outgoing transport	Select <i>TCP</i> or <i>TLS</i> . We recommend that you use <i>TLS</i> .
Use local certificate for outgoing connections and registrations	Uncheck.

SIP	
SIP registrar usage	Enabled ▾
SIP registrar domain	vcs.domain
SIP registrar type	Standard SIP ▾
Username	sdmcu
Password	
Allow numeric ID registration for conferences	<input type="checkbox"/>
SIP call settings	
SIP proxy address	10.1.2.3
Maximum bit rate from Microsoft OCS/LCS clients	<limit disabled> ▾
Outgoing transport	<input type="radio"/> UDP <input type="radio"/> TCP <input checked="" type="radio"/> TLS
Use local certificate for outgoing connections and registrations	<input type="checkbox"/>

3. Click **Apply changes**.
4. Repeat the steps for any other TelePresence MCUs.

Task 4: Configuring H.323

1. Go to **Settings > H.323**.
2. Enter the following in the relevant fields, leave other fields as their default values:

H.323 gatekeeper usage	Select <i>Enabled</i> .
H.323 gatekeeper address	Enter the Cisco 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 <i>hdmcu<VCS SIP domain></i> .
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	<input checked="" type="checkbox"/> Port A IPv4 <input type="checkbox"/> Port A IPv6 <input type="checkbox"/> Port B IPv4 <input type="checkbox"/> Port B IPv6
(Mandatory) H.323 ID to register	hdmcu@vcs.domain
Use password	<input type="checkbox"/> Password: <input type="text"/>
Prefix for MCU registrations	<input type="text"/>
MCU service prefix	<input type="text"/> (optional)
Allow numeric ID registration for conferences	<input type="checkbox"/>
Send resource availability indications	<input type="checkbox"/> Thresholds: <input type="text"/> conferences <input type="text"/> video ports

3. Click **Apply changes**.
4. Repeat the steps for any other TelePresence MCUs.

Task 5: Changing miscellaneous settings

On all conference bridges:

1. Go to **Network > Services**.
2. Ensure that the protocol settings match those of the Cisco VCS, using either **SIP(TCP)**, **Encrypted SIP (TLS)** or **Incoming H.323** (if supported).
We recommend that you use encryption and follow the steps in [Task 2: Installing an encryption key \[p.13\]](#).
The protocol on the Cisco VCS is configured in [Task 13: Adding each conference bridge as a neighbor zone \[p.24\]](#).

1. Go to **Settings > Conferences**
2. Under **Conference Settings** 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 ▾

3. Click **Apply changes**.
4. Go to **Gatekeeper > Built in Gatekeeper**.
5. Under **Configuration** ensure **Status** is set to *Disabled*.

Note: The MCU 5300 series does not have a built-in Gatekeeper.



6. Click **Apply changes**.
7. Repeat the steps for any other TelePresence MCUs.

Configuring the TelePresence Server

These tasks can be skipped, if only TelePresence MCUs are used as conference bridges in your deployment.

Task 6: Creating a user

For the TelePresence Conductor to communicate with the TelePresence Server it must use credentials for a user that has administrator rights. We recommend that you create a dedicated administrator level user for this task.

1. Go to the web interface of the TelePresence Server you want to configure and log in as an administrator.
2. Go to **User > Add New User**.
3. Enter the following in the relevant fields:

User ID	Enter a username for the TelePresence Conductor to use.
Name	Enter a name for this user.
Password	Enter a password for the TelePresence Conductor to use.
Access rights	Select <i>Administrator</i> .

 A screenshot of a web form titled "Add new user". In the top right corner, there is a breadcrumb trail: "You are here: > Users > Add new user". The form has a light gray background and a title bar that says "User". It contains the following fields:

- User ID:** A text input field containing "conductoradmin".
- Name:** A text input field containing "Admin for Conductor".
- Password:** A password input field with masked characters (dots).
- Re-enter password:** A password input field with masked characters (dots).
- Access rights:** A dropdown menu with "Administrator" selected.

 At the bottom of the form is a button labeled "Add user".

4. Click **Add user**.
5. Repeat the steps for any other TelePresence Servers.

Task 7: Installing an encryption key

The TelePresence Server has the ability to use a secure connection for communications. These security features are enabled with the **Encryption** option key. You are recommended to install the option key for this deployment.

To verify that the key is installed or to install the key, perform the following tasks:

1. Go to **Configuration > Upgrade**.
2. Go to the **Feature management** section and verify that the **Encryption** key is installed. If the key is not installed, enter the **Activation code** and click **Update features**.

Feature management

The screenshot shows the 'Feature management' window. Under 'Activated features', 'MSE 8510 activation' is listed with a 'remove' link. Below it, 'Encryption' is highlighted with a red box and also has a 'remove' link. 'Third party interop' is also listed with a 'remove' link. Under 'License keys', 'Media port licenses x 80' and 'TS screen licenses x 16' are listed. There is an 'Activation code' input field and an 'Update features' button.

To verify that TLS is enabled on the TelePresence Server:

1. Go to **Network > Services**.
2. Ensure that **Encrypted SIP (TLS)** is checked.
3. Ensure that **SIP (TCP)** and **SIP (UDP)** are not checked.
4. Ensure that **HTTPS** is enabled on port 443.

The screenshot shows the 'Services' configuration page. It has a breadcrumb 'You are here: > Network > Services'. There are two sections: 'TCP service' and 'UDP service', both for 'Port A' and 'IPv4'.
 TCP service settings:
 - HTTP: 80
 - HTTPS: 443
 - Incoming H.323: 1720
 - SIP (TCP): 5060
 - Encrypted SIP (TLS): 5061
 - FTP: 21
 UDP service settings:
 - SIP (UDP): 5060
 At the bottom, there are 'Apply changes' and 'Reset to default' buttons.

5. Click **Apply changes**.

Task 8: Configuring SIP

Perform the following steps to enable SIP registration to a SIP registrar:

1. Go to **Configuration > SIP settings**.
2. Enter the following values into the relevant fields:

Outbound call configuration	Select <i>Use trunk</i> from the drop-down list.
Outbound address	Enter the IP address or FQDN of the Cisco VCS.
Outbound domain	Enter the appropriate domain name, for example vcs.domain .
Username	Enter for example TelePresence_Server .
Password	Leave blank.
Outbound transport	Select <i>TCP</i> or <i>TLS</i> . We recommend that you select <i>TLS</i> .
Negotiate SRTP using SDES	We recommend that you select <i>For secure transports (TLS) only</i> .
Use local certificate for outgoing connections and registrations	We recommend that you check the box.

SIP settings You are here: [Configuration](#) > [SIP settings](#)

SIP

Outbound call configuration	Use trunk
Outbound address	10.1.2.3
Outbound domain	vcs.domain
Username	TelePresence_Server
Password	
Outbound transport	TLS
Negotiate SRTP using SDES	For secure transports (TLS) only
Use local certificate for outgoing connections and registrations	<input checked="" type="checkbox"/>

3. Click **Apply changes**.
4. Repeat the steps for any other TelePresence Servers.

Task 9: Configuring H.323

Perform the following steps to enable use of an H.323 gatekeeper (for out-dial calls):

Note: TelePresence Servers that do not support H.323 do not require this configuration.

1. Go to **Configuration > H323 Settings**.
2. Enter the following values into the relevant fields:

Use gatekeeper	Check the box.
Address	Enter the IP address or FQDN of the Cisco VCS that is the gatekeeper.
H.323 ID to register	Enter the H323 registration ID for this TelePresence Server.
Password	Leave blank unless required by your Cisco VCS policy.

H.323 settings

H.323 gatekeeper

Use gatekeeper

Address

H.323 ID to register

Password

3. Click **Apply changes**.
4. Repeat the steps for any other TelePresence Servers.

Task 10: Configuring the operational mode

(This task is not relevant for Cisco TelePresence Server on Virtual Machine or Cisco TelePresence Server on Multiparty Media 310/320.)

1. Go to **Configuration > Operation mode**.
2. Select *Remotely managed* from the drop down list. This enables the TelePresence Conductor to manage the TelePresence Server.

Operation mode

Operation mode

3. Click **Apply changes**.
4. For the changes to take effect, the TelePresence Server must be restarted. Go to **Configuration > Shutdown**.
5. Click **Shutdown TelePresence Server**.
6. Click **Confirm TelePresence Server shutdown**.
7. Click **Restart TelePresence Server**.
8. After about 3 minutes, the TelePresence Server will be available to the TelePresence Conductor.
9. Repeat the steps for any other TelePresence Servers.

Configuring the Cisco VCS

Before configuring the Cisco VCS you must create a new user on the TelePresence Conductor, as described in the task below.

Task 11: Creating a new user on the TelePresence Conductor

We recommend that you set up an administrator account on the TelePresence Conductor that has API access only. This account will be used for all communications between the Cisco VCS and TelePresence Conductor.

1. Log into the TelePresence Conductor as a user with administrator rights.
2. Go to **Users > Administrator Accounts**.
3. Click **New**.

4. Enter the following in the relevant fields:

Name	Enter a name for this user.
Access level	Select <i>Read-write</i> .
Password	Enter a password for this account.
Web access	Set to <i>No</i> .
API access	Set to <i>Yes</i> .
State	Set to <i>Enabled</i> .

The screenshot shows a configuration window with the following fields and values:

- Name:** VCS_API_ACCT
- Access level:** Read-write
- Password:** [masked]
- Confirm password:** [masked]
- Web access:** No
- API access:** Yes
- State:** Enabled

Buttons: Save, Cancel

Task 12: Adding the TelePresence Conductor as a policy service

A policy service is in essence a location to which the Cisco VCS can send HTTP or HTTPS requests that contain various details about a call. 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 Cisco VCS with the TelePresence Conductor as a policy service:

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

Name	Enter a name for the policy service, for example Conductor Policy Service .
Protocol	Select <i>HTTPS</i> .

Certificate verification mode	<p>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 interface at Maintenance > Security certificates > Server certificate.</p> <p>Note: Setting Certificate verification mode to <i>Off</i> makes HTTPS communication highly insecure and is not recommended for production systems.</p>
HTTPS certificate revocation list CRL checking	Select <i>Off</i> .
Server 1 address	Enter the TelePresence Conductor's IP address.
Path	Enter <code>api/conference_controller/conference/conference_factory.cpl</code>
Username	Enter the username of the TelePresence Conductor administration user. This appears on the TelePresence Conductor's Administrator accounts page (Users > Administrator accounts).
Password	Enter the password of the TelePresence Conductor administration user.
Default CPL	Enter <code><reject status='504' reason='Conductor policy service unavailable' /></code>

Create policy service

Configuration

<p>Name</p> <p>Description</p> <p>Protocol</p> <p>Certificate verification mode</p> <p>HTTPS certificate revocation list (CRL) checking</p> <p>Server 1 address</p> <p>Server 2 address</p> <p>Server 3 address</p> <p>Path</p> <p>Status path</p> <p>Username</p> <p>Password</p> <p>Default CPL</p>	<p>★ Conductor Policy Service ⓘ</p> <p>ⓘ</p> <p>HTTPS ⓘ</p> <p>On ⓘ</p> <p>Off ⓘ</p> <p>★ 10.1.2.4 ⓘ</p> <p>ⓘ</p> <p>ⓘ</p> <p>★ api/conference_controller/conference/conference_fa ⓘ</p> <p>status ⓘ</p> <p>VCS_API_ACCT ⓘ</p> <p>ⓘ</p> <p><reject status='504' reason='Policy Service Unavailab ⓘ</p>
---	---

5. Click **Create policy service**.

Until the Cisco VCS updates its TelePresence Conductor status the status of the TelePresence Conductor policy service under **VCS configuration > Dial plan > Policy services** will list as 'Checking'. When the Cisco VCS queries the TelePresence Conductor for status this will change to 'Failed'. 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.

Task 13: Adding each conference bridge as a neighbor zone

To configure the Cisco VCS with neighbor zones for all conference bridges:

1. Go to **VCS configuration > Zones > Zones**.
2. Click **Create new zone**.
3. Enter the following in the relevant fields, leave other fields as their default values:

Name	Enter 'HD MCU' for example.
Type	Select <i>Neighbor</i> .
H.323 mode	Select <i>Off</i> if the conference bridge does not support H.323, otherwise, if H.323 is supported, leave as <i>On</i> .
SIP transport	We recommend that you select <i>TLS</i> and set the port to '5061'. Note: if you have selected a SIP transport of <i>TLS</i> , 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.
Peer 1 address	Enter the conference bridge's IP address.
Zone profile	If the Cisco VCS is running 7.0.x or later select <i>Infrastructure device</i> . These zone profiles perform no aliveness checking. Therefore, an 'Active' status given by this zone cannot be relied upon to indicate that Cisco VCS to conference bridge communication is possible.

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

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. Repeat the steps for all other conference bridges.

Task 14: Configuring a search rule with the TelePresence Conductor policy service as the target

Search rules define where the Cisco VCS routes calls. 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 **VCS configuration > Dial plans > Search rules**.
2. Click **New**.
3. Enter the following in the relevant fields, leave other fields as their default values:

Rule name	Enter 'To Conductor Policy Service' for example.
Priority	Enter '10' for example. The priority for this search rule must be higher than the priority for the local zone.
Mode	Select <i>Alias pattern match</i> .
Pattern type	Select <i>Regex</i> .
Pattern string	Enter <code>(meet meetts teach learn teachts learnts)\.*@<SIP domain></code> Note: Replace <code><SIP domain></code> 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> .

Create search rule

Configuration

Rule name	★ To Conductor Policy Service i
Description	<input type="text"/> i
Priority	★ 10 i
Protocol	Any i
Source	Any i
Request must be authenticated	No i
Mode	Alias pattern match i
Pattern type	Regex i
Pattern string	★ (meet meets teach learn teachts learnts)\. *@vcs\.domain i
Pattern behavior	Leave i
On successful match	Stop i
Target	Conductor Policy Service i
State	Enabled i

Create search rule
Cancel

4. Click **Create search rule**.

Task 15: Configuring a Cisco VCS search rule for each conference bridge

To configure the Search rule:

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

Rule name	Enter a name for the rule, for example To HD MCU .
Priority	Enter '15' for example. The priority for this search rule must be higher than the priority for the local zone.
Mode	Select <i>Alias pattern match</i> .
Pattern type	Select <i>Prefix</i> .
Pattern string	Enter HD MCU .
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. Repeat the steps for any other conference bridges.

Configuring the TelePresence Conductor

This section of the guide assumes that the TelePresence Conductor is reachable over the network. For assistance in reaching this stage, see [Cisco TelePresence Conductor Getting Started Guide](#).

The TelePresence Conductor policy service only lists 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 is configured with at least one conference bridge with a 'usable' status. This is to ensure that requests are not sent to members of a TelePresence Conductor cluster that have lost connectivity with the conference bridges.

Task 16: Changing the administrator password

1. Log into the TelePresence Conductor as the user 'admin' and with the default password 'TANDBERG'.
2. Go to **Users > Administrator accounts**.
3. Click **View/Edit** for the 'admin' user.

4. Enter a new password.
5. Click **Save**.

Note: the TelePresence Conductor will not handle conference requests if it has the administrator password set to its default value.

Task 17: Changing the root password

1. Log in to the TelePresence Conductor as root (default password = 'TANDBERG'). By default you can only do this using SSH or a serial connection.
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.

Note: the TelePresence Conductor will not handle conference requests if it has the root password set to its default value.

Task 18: Changing the system settings

1. Log into the TelePresence Conductor as a user with administrator rights.
2. Go to **System > DNS**.
3. Enter the following in the relevant fields:

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 i

Domain name i

DNS requests port * i

range start

DNS requests port * i

range end

Default DNS servers

Address 1 i

Address 2 i

Address 3 i

Per-domain DNS servers

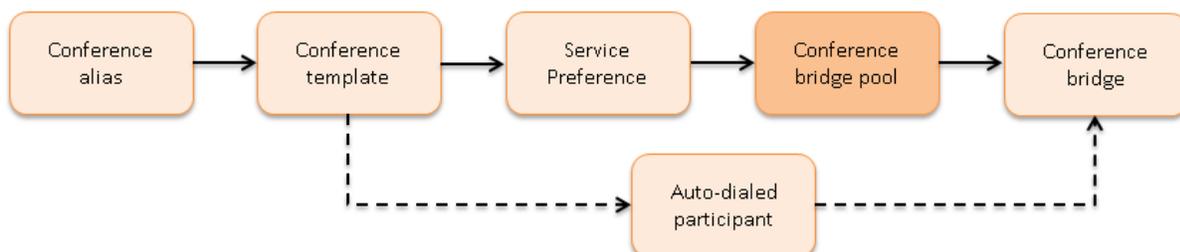
Address 1 i Dorr

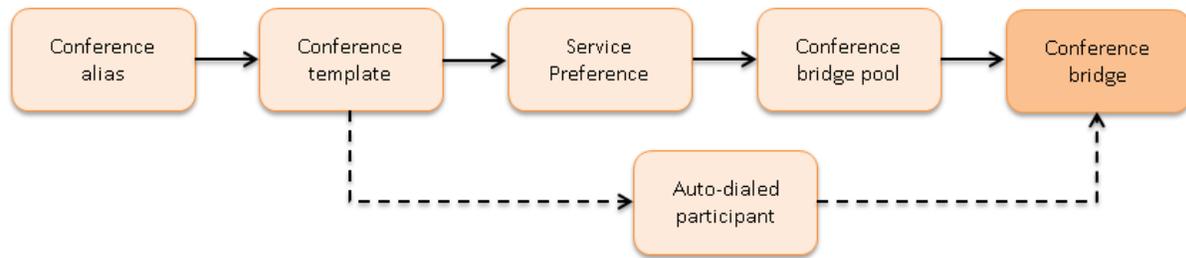
Address 2 i Dorr

Note: the FQDN of the TelePresence Conductor is <System host name>.<Domain name>.

4. Click **Save**.
5. Go to **System > Time** if the default servers are unreachable then it may be necessary to enter alternate NTP servers.
6. Ensure that under the **Status** section the State is *Synchronized*. This can take a couple of minutes.

Task 19: Setting up conference bridge pools





To set up a conference bridge pool, you need to create a conference bridge pool and then add one or more conference bridge(s) to it. The following examples show how to set up conference bridge pools for:

- TelePresence MCU hosted HD conferences
- TelePresence MCU hosted SD conferences
- TelePresence Server hosted HD conferences
- TelePresence Server hosted SD conferences

Creating a TelePresence MCU HD conference bridge pool

1. Go to **Conference configuration > Conference bridge pools**.
2. Click **New**.
3. In the **Pool name** field enter a name for the conference bridge pool, for example **HD MCU pool1**.
4. Choose the correct **Conference bridge type**, in this case *TelePresence MCU*.
5. Leave the **Location** set to *None*.

Conference bridge pools

Configuration

Pool name Description Conference bridge type Raise conference bridge resource alarm Location	<input style="width: 90%;" type="text" value="HD MCU pool"/> ⓘ <input style="width: 90%;" type="text"/> TelePresence MCU ⓘ <input checked="" type="checkbox"/> Threshold (%) 80 ⓘ None ⓘ
--	--

6. Click **Create pool**.

Adding a TelePresence MCU to the HD conference bridge pool

1. From the **Conference bridge pools** page click **Create conference bridge**.
2. Enter the following in the relevant fields:

Name	Enter a name for the conference bridge, for example HD MCU .
State	Select <i>Enabled</i> .
IP address or FQDN	Enter the IP address of the HD conference bridge.
Protocol	Select <i>HTTPS</i> .
Port	Enter '443'.

Conference bridge username	Enter the conference bridge admin username, for example <code>conductoradmin</code> . (This is created in Task 1: Creating a user [p.13] .)
Conference bridge Password	Enter the conference bridge password for this user.
Dial plan prefix	Enter <code>HDMCU</code> .
Dedicated content ports	Enter the appropriate value for your TelePresence MCU. To discover if a TelePresence MCU has any dedicated content ports follow the steps given in Appendix 1: Identifying dedicated content ports on a Cisco TelePresence MCU [p.64] .
SIP port	Enter the SIP port on which the conference bridge listens for SIP TLS traffic, typically this is '5061'.

Add conference bridge

Configuration

Name	* HD MCU <input style="width: 90%;" type="text" value="HD MCU"/>
Description	<input style="width: 90%;" type="text" value=""/>
State	Enabled <input style="font-size: 0.8em; vertical-align: middle;" type="button" value="v"/> <input style="font-size: 0.8em; vertical-align: middle;" type="button" value="i"/>
IP address or FQDN	* 10.1 2.5 <input style="width: 90%;" type="text" value="10.1 2.5"/>
Protocol	HTTPS <input style="font-size: 0.8em; vertical-align: middle;" type="button" value="v"/> <input style="font-size: 0.8em; vertical-align: middle;" type="button" value="i"/>
Port	* 443 <input style="width: 90%;" type="text" value="443"/>
Conference bridge username	* conductoradmin <input style="width: 90%;" type="text" value="conductoradmin"/>
Conference bridge password	***** <input style="width: 90%;" type="text" value="*****"/>
Dial plan prefix	HDMCU <input style="width: 90%;" type="text" value="HDMCU "/>
Conference bridge type	* TelePresence MCU <input style="font-size: 0.8em; vertical-align: middle;" type="button" value="v"/> <input style="font-size: 0.8em; vertical-align: middle;" type="button" value="i"/>
Conference bridge pool	* HD MCU pool <input style="font-size: 0.8em; vertical-align: middle;" type="button" value="v"/> <input style="font-size: 0.8em; vertical-align: middle;" type="button" value="i"/>
Dedicated content ports	* 0 <input style="width: 90%;" type="text" value="0"/>
SIP port	* 5061 <input style="width: 90%;" type="text" value="5061"/>
H.323 cascade call routing	Gatekeeper routed <input style="font-size: 0.8em; vertical-align: middle;" type="button" value="v"/> <input style="font-size: 0.8em; vertical-align: middle;" type="button" value="i"/>

3. Click **Create conference bridge**.
4. Ensure that under the **Conference bridges in this pool** section, under the **Status** header the conference bridge is listed as *Active*.
5. Repeat the steps to add any further TelePresence MCUs to the conference bridge pool.

Creating a TelePresence MCU SD conference bridge pool.

Repeat the steps under [Creating a TelePresence MCU HD conference bridge pool \[p.31\]](#) to create a TelePresence MCU SD conference bridge pool. Enter the same values for the fields, apart from the **Pool name**, which should be `SD MCU pool1`, for example.

Adding a TelePresence MCU to the SD conference bridge pool

Repeat the steps under [Adding a TelePresence MCU to the HD conference bridge pool \[p.31\]](#) to add a TelePresence MCU to the SD conference bridge pool. Enter the same values for the fields, apart from:

Name	Enter a name for the conference bridge, for example <code>SD MCU</code> .
IP address or FQDN	Enter the IP address of the SD conference bridge.
Dial plan prefix	Enter <code>SDMCU</code> .

Creating a TelePresence Server HD conference bridge pool

1. Go to **Conference configuration > Conference bridge pools**.
2. Click **New**.
3. In the **Pool name** field enter a name for the conference bridge pool, for example `HD TS pool1`.
4. Choose the correct **Conference bridge type**, in this case *TelePresence Server*.
5. Leave the **Location** set to *None*.

The screenshot shows the 'Conference bridge pools' configuration interface. The 'Configuration' tab is selected. The following fields are visible and populated:

- Pool name:** HD TS pool
- Description:** (empty)
- Conference bridge type:** TelePresence Server
- Raise conference bridge resource alarm:**
- Threshold (%):** 80
- Location:** None

6. Click **Create pool**.

Adding a TelePresence Server to the HD conference bridge pool

Before adding a TelePresence Server to the conference bridge pool, ensure that the **Operation mode** on the TelePresence Server is set to *Remotely managed* (see [Task 10: Configuring the operational mode \[p.21\]](#)).

1. Click **Create conference bridge**.
2. Enter the following in the relevant fields:

Name	Enter a name for the conference bridge, for example <code>HD TS</code> .
State	Select <i>Enabled</i> .
IP address or FQDN	Enter the IP address of the HD conference bridge.
Protocol	Select <i>HTTPS</i> .
Port	Enter '443'.
Conference bridge username	Enter the conference bridge admin username, for example <code>conductoradmin</code> . (This is created in Task 6: Creating a user [p.18] .)
Conference bridge Password	Enter the conference bridge password for this user.

Dial plan prefix	Enter HDTS .
SIP port	Enter the SIP port on which the conference bridge listens for SIP TLS traffic, typically this is '5061'.

Add conference bridge

Configuration

Name	* HD TS i
Description	<input type="text"/> i
State	Enabled v i
IP address or FQDN	* 10.1.2.10 i
Protocol	HTTPS v i
Port	* 443 i
Conference bridge username	* conductoradmin i
Conference bridge password	***** i
Dial plan prefix	HDTS i
Conference bridge type	* TelePresence Server v i
Conference bridge pool	* HD TS pool v i
SIP port	* 5061 i

3. Click **Create conference bridge**.
4. Ensure that under the **Conference bridges in this pool** section, under the **Status** header the conference bridge is listed as *Active*.
5. Repeat the steps to add any further TelePresence Servers to the conference bridge pool.

Creating a TelePresence Server SD conference bridge pool

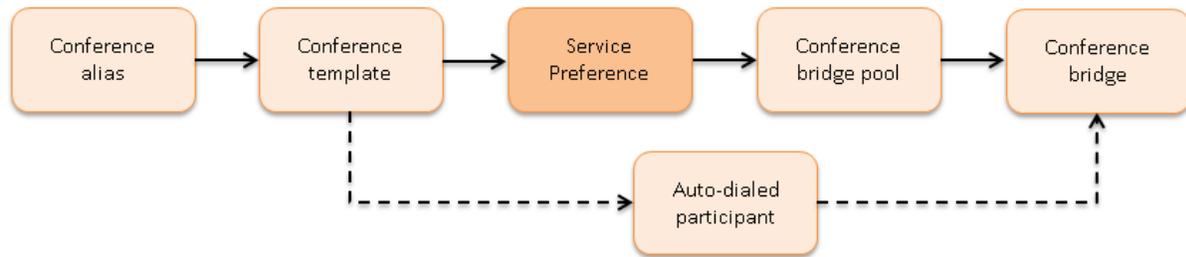
Repeat the steps under [Creating a TelePresence Server HD conference bridge pool \[p.33\]](#) to create a TelePresence Server SD conference bridge pool. Enter the same values for the fields, apart from the **Pool name**, which should be **SD TS pool1**, for example.

Adding a TelePresence Server to the SD conference bridge pool

Repeat the steps under [Adding a TelePresence Server to the HD conference bridge pool \[p.33\]](#) to add a TelePresence Server to the SD conference bridge pool. Enter the same values for the fields, apart from:

Name	Enter a name for the conference bridge, for example SD TS .
IP address or FQDN	Enter the IP address of the SD conference bridge.
Dial plan prefix	Enter SDTS .

Task 20: Creating Service Preferences



A Service Preference is a prioritized list of conference bridge pools that defines the order in which resources are used for conferences. During the configuration process, the conference bridge type is chosen as either *TelePresence MCU* or *TelePresence Server*. You cannot mix the two types of conference bridges. For TelePresence MCUs a conference can be cascaded from one TelePresence MCU to another, taking into account the prioritized list of conference bridge pools. Cascading between TelePresence Servers is not supported, because TelePresence Server version 3.0 does not have this feature.

The following examples show how to create Service Preferences for:

- TelePresence MCU hosted HD conferences
- TelePresence MCU hosted SD conferences
- TelePresence Server hosted HD conferences
- TelePresence Server hosted SD conferences

Creating a Service Preference for TelePresence MCU hosted HD conferences

1. Go to **Conference configuration > Service Preferences**.
2. Click **New**.
3. In the **Service Preference name** field enter **Prefer HD with SD fallback**.
4. In the **Conference bridge type** field, choose *TelePresence MCU*.
5. Click **Add Service Preference**.
6. In the **Pools** section of the page under **Pool name** select *HD MCU pool*.
7. Click **Add selected pool**.
8. In the **Pools** section of the page under **Pool name** select *SD MCU pool*.
9. Click **Add selected pool**.

Service Preferences

Service Preference

Service Preference name ⓘ

Description ⓘ

Conference bridge type ⓘ

Pools

Priority	Pool name
<input type="checkbox"/> 1	HD MCU pool
<input type="checkbox"/> 2	SD MCU pool
Please select ▼	

10. Click **Save**.

Creating a Service Preference for TelePresence MCU hosted SD conferences

1. Go to **Conference configuration > Service Preferences**.
2. Click **New**.
3. In the **Service Preference name** field enter **Prefer SD with HD fallback**.
4. In the **Conference bridge type** field, choose *TelePresence MCU*.
5. Click **Add Service Preference**.
6. In the **Pools** section of the page under **Pool name** select *SD MCU pool*.
7. Click **Add selected pool**.
8. In the **Pools** section of the page under **Pool name** select *HD MCU pool*.
9. Click **Add selected pool**.

Service Preferences

Service Preference

Service Preference name ⓘ

Description ⓘ

Conference bridge type ⓘ

Pools

Priority	Pool name
<input type="checkbox"/> 1	SD MCU pool
<input type="checkbox"/> 2	HD MCU pool
Please select ▼	

10. Click **Save**.

Creating a Service Preference for TelePresence Server hosted HD conferences

1. Go to **Conference configuration > Service Preferences**.
2. Click **New**.

3. In the **Service Preference name** field enter **Prefer HD TS**.
4. In the **Conference bridge type** field, choose *TelePresence Server*.
5. Click **Add Service Preference**.
6. In the **Pools** section of the page under **Pool name** select *HD TS pool*.
7. Click **Add selected pool**.

Service Preferences

Service Preference

Service Preference name * Prefer HD TS ⓘ

Description ⓘ

Conference bridge type * TelePresence Server ⓘ

Pools

Priority	Pool name
<input type="checkbox"/> 1	HD TS pool Please select ▼

Add selected pool **Delete pool** **Select all** **Unselect all**

Save **Delete** **Cancel**

8. Click **Save**.

Creating a Service Preference for TelePresence Server hosted SD conferences

1. Go to **Conference configuration > Service Preferences**.
2. Click **New**.
3. In the **Service Preference name** field enter **Prefer SD TS**.
4. In the **Conference bridge type** field, choose *TelePresence Server*.
5. Click **Add Service Preference**.
6. In the **Pools** section of the page under **Pool name** select *SD TS pool*.
7. Click **Add selected pool**.

Service Preferences

Service Preference

Service Preference name * Prefer SD TS ⓘ

Description ⓘ

Conference bridge type * TelePresence Server ⓘ

Pools

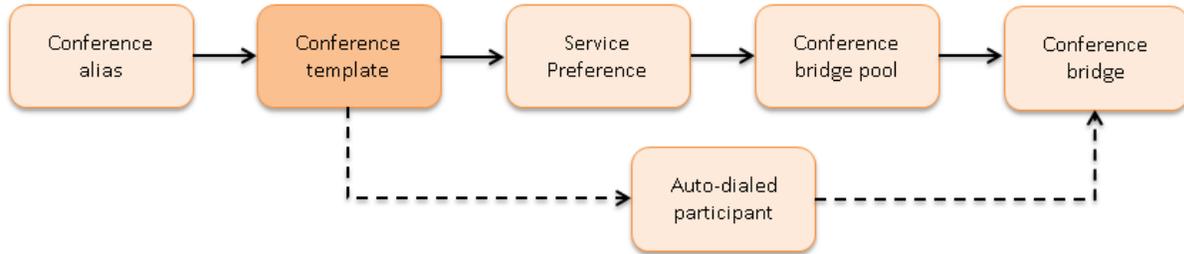
Priority	Pool name
<input type="checkbox"/> 1	SD TS pool Please select ▼

Add selected pool **Delete pool** **Select all** **Unselect all**

Save **Delete** **Cancel**

8. Click **Save**.

Task 21: Creating conference templates for Meeting-type conferences



A Meeting-type conference template provides all its participants with the same privileges and requires one or more conference aliases. The following examples show how to create conference templates for:

- 'HD Meetings' hosted on TelePresence MCUs
- 'SD Meetings' hosted on TelePresence MCUs
- 'HD Meetings' hosted on TelePresence Servers
- 'SD Meetings' hosted on TelePresence Servers

Creating a conference template for an 'HD Meeting' hosted on TelePresence MCUs

This template uses a Service Preference that prioritizes HD pools over SD pools for TelePresence MCU resources.

1. Go to **Conference configuration > Conference templates**.
2. Click **New**.
3. Enter the following in the relevant fields, leave other fields as their default values:

Name	Enter a name for the conference template, for example HD Meeting .
Conference type	Select <i>Meeting</i> .
Service Preference	Select <i>Prefer HD with SD fallback</i> .
Number of cascade ports to reserve	Enter '1' to enable cascading to one other TelePresence MCU. Enter '0' to disable cascading.

4. Click **Create conference template**.

Creating a conference template for an 'SD Meeting' hosted on TelePresence MCUs

This template uses a Service Preference that prioritizes SD pools over HD pools for TelePresence MCU resources.

Repeat the steps under [Creating a conference template for an 'HD Meeting' hosted on TelePresence MCUs \[p.38\]](#) to create a conference template for an 'SD Meeting' hosted on TelePresence MCUs. Enter the same values for the fields, apart from:

Name	Enter a name for the conference template, for example <i>SD meeting</i> .
Service Preference	Select <i>Prefer SD with HD fallback</i> .

Creating a conference template for an 'HD Meeting' hosted on TelePresence Servers

The following steps demonstrate how to create an HD meeting template for a TelePresence Server. Remember when configuring TelePresence Server pools and Service Preference that cascading between multiple TelePresence Servers is not supported.

1. Go to **Conference configuration > Conference templates**.
2. Click **New**.
3. Enter the following in the relevant fields, leave other fields as their default values:

Name	Enter a name for the conference template, for example <i>HD TS meeting</i> .
Conference type	Select <i>Meeting</i> .
Call Policy mode	<p>We recommend that you do not use this feature to define which participants can create a conference and that you use the field Allow conference to be created on the Conference aliases page instead.</p> <p>If you still want to use the feature, select <i>On</i> and see Appendix 3: Call policy mode [p.67] for more information.</p>

Service Preference	Select <i>Prefer HD TS</i> .
Participant quality	Choose one of the HD choices from the drop-down box. When using a CTS3000 you must select <i>Full HD (1080p 30fps / 720p 60fps video, multi-channel audio)</i> or a custom quality setting that has an audio quality level of multi-channel, otherwise insufficient resources will be allocated to display multiple screens.
Provision for multiscreen	Decide whether this conference will support multiscreen systems, or whether it will only display single screen systems and the center camera of a multiscreen system. The default is <i>No</i> . If <i>Yes</i> is selected, the endpoint does not support TIP (Telepresence Interoperability Protocol) and the expectation is for multiscreen systems to have all three screens active, then create a pre-configured endpoint to match each multiscreen system in the call. (To do this go to Conference configuration > Pre-configured endpoints). For more information on pre-configuring endpoints see Cisco TelePresence Conductor Administrator Guide .
Content quality	Select the maximum content quality allowed for this conference.

The screenshot shows the 'Conference templates' configuration page. The 'Modify conference template' section includes the following fields and values:

- Name: HD TS Meeting
- Description: (empty)
- Conference type: Meeting
- Call Policy mode: Off
- Service Preference: Prefer HD TS
- Limit number of participants: Maximum (with note: There are 0 auto-dialed participants associated with this template.)
- Limit the conference duration (minutes): Maximum
- Participant quality: Full HD
- Allow multiscreen: Yes
- Maximum screens: 3
- Optimize resources: Yes
- Content quality: Full HD
- Scheduled conference: No

At the bottom of the form, there are two buttons: 'Create conference template' and 'Cancel'.

4. Click **Create conference template**.

Creating a conference template for an 'SD Meeting' hosted on TelePresence Servers

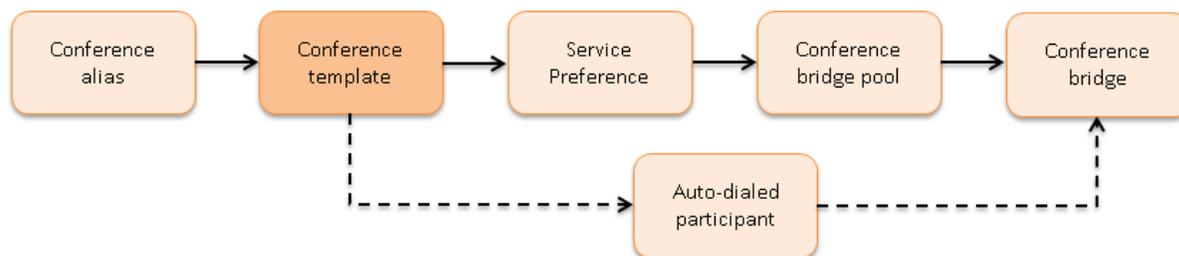
Repeat the steps under [Creating a conference template for an 'HD Meeting' hosted on TelePresence Servers \[p.39\]](#) to create a conference template for an 'SD Meeting' hosted on TelePresence Servers. Enter the same values for the fields, apart from:

Name	Enter a name for the conference template, for example SD TS meeting .
-------------	--

Service Preference Select *Prefer SD TS*.

Participant quality Choose one of the SD choices from the drop-down box.
 Note that CTS3000 endpoints require the audio level to be set to multi-channel to be allocated sufficient resources to display three screens. The pre-defined SD setting does not have an audio level of multi-channel, which will result in only the center screen of the CTS3000 to be displayed.

Task 22: Creating conference templates for Lecture-type conferences



A Lecture-type conference template defines two role types, Chairperson and Guest, with different privileges and requires at least one conference alias per role type. The following examples show how to create conference templates for:

- Lecture-type conferences hosted on TelePresence MCUs
- Lecture-type conferences hosted on TelePresence Servers

Creating a conference template for a Lecture-type conference hosted on HD TelePresence MCUs

The following steps set up a 'Lecture' template that uses an TelePresence MCU Service Preference:

1. Go to **Conference configuration > Conference templates**.
2. Click **New**.
3. Enter the following in the relevant fields, leave other fields as their default values:

Name	Enter a name for the conference template, for example Lecture .
Conference type	Select <i>Lecture</i> .
Number of chairperson participants to reserve	Enter '2' in this example.
Call Policy mode	We recommend that you do not use this feature to define which participants can create a conference and that you use the field Allow conference to be created on the Conference aliases page instead. If you still want to use the feature, select <i>On</i> and see Appendix 3: Call policy mode [p.67] for more information.
Service Preference	Select <i>Prefer HD with SD fallback</i> .
Number of cascade ports to reserve	Enter '1' to enable cascading to one other TelePresence MCU. Enter '0' to disable cascading.

4. Click **Create conference template**.
5. Click **View/Edit** for the Lecture template.
6. Click **Edit** under the **Advanced parameters** section.
7. Enter the following in the relevant fields, leave other fields as their default values:

Field	Input
PIN	Check the on box next to the field in the primary column then enter a PIN for the chair to use when entering the conference. Note: for TelePresence MCU software versions lower than 4.3 a Guest PIN must be specified if a Chair PIN is specified.

8. Click **Save** to exit the advance parameters.

Summary of configured parameters		
Parameter	Primary	Cascade
pin	55105	(Not set)

9. Click **Save** on the **Conference template** page.

Creating a conference template for a Lecture-type conference hosted on HD TelePresence Servers

1. Go to **Conference configuration > Conference templates**.
2. Click **New**.
3. Enter the following in the relevant fields, leave other fields as their default values:

Name	Enter a name for the conference template, for example Lecture - TS .
Conference type	Select Lecture .

Number of chairperson participants to reserve	Enter '2' in this example.
Call Policy mode	<p>We recommend that you do not use this feature to define which participants can create a conference and that you use the field Allow conference to be created on the Conference aliases page instead.</p> <p>If you still want to use the feature, select <i>On</i> and see Appendix 3: Call policy mode [p.67] for more information.</p>
Service Preference	Select <i>Prefer HD TS</i> .
Chairperson quality	<p>Enter the maximum quality setting to apply to chairpersons using this conference template.</p> <p>When using a CTS3000 you must select <i>Full HD (1080p 30fps / 720p 60fps video, multi-channel audio)</i> or a custom quality setting that has an audio quality level of multi-channel, otherwise insufficient resources will be allocated to display multiple screens.</p>
Guest quality	<p>Enter the maximum quality setting to apply to guests using this conference template.</p> <p>When using a CTS3000 you must select <i>Full HD (1080p 30fps / 720p 60fps video, multi-channel audio)</i> or a custom quality setting that has an audio quality level of multi-channel, otherwise insufficient resources will be allocated to display multiple screens.</p>
Provision for multiscreen	<p>Decide whether this conference will support multiscreen systems, or whether it will only display single screen systems and the center camera of a multiscreen system. The default is <i>No</i>.</p> <p>If <i>Yes</i> is selected, the endpoint does not support TIP (Telepresence Interoperability Protocol) and the expectation is for multiscreen systems to have all three screens active, then create a pre-configured endpoint to match each multiscreen system in the call. (To do this go to Conference configuration > Pre-configured endpoints).</p> <p>For more information on pre-configuring endpoints see Cisco TelePresence Conductor Administrator Guide.</p>
Content quality	Select the maximum quality allowed for this conference.

Conference templates

Modify conference template

Name	* Lecture - TS
Description	
Conference type	Lecture
Number of chairperson participants to reserve	* 2
Call Policy mode	Off
Service Preference	* Prefer HD TS
Limit number of participants	<input type="checkbox"/> Maximum <input type="text"/> There are 0 auto-dialec
Limit the conference duration (minutes)	<input type="checkbox"/> Maximum <input type="text"/>
Chairperson quality	HD (720p 30fps video, stereo audio)
Guest quality	HD (720p 30fps video, stereo audio)
Provision for multiscreen	No
Optimize resources	Yes
Content quality	HD (720p 30fps)
Scheduled conference	No

4. Click **Create conference template**.
5. Click **View/Edit** for the Lecture - TS template.
6. Click **Edit** under the **Advanced parameters** section.
7. Enter the following in the relevant field, leave other fields as their default values:

PIN Check the on box next to the **Pin** field and then enter a PIN for the chair to use when entering the conference.

Advanced template parameters

Modify advanced template parameters

Template name	Lecture - TS
Conference bridge type	TelePresence Server
Guest PIN	<input type="checkbox"/> <input type="text"/>
PIN	<input checked="" type="checkbox"/> 55105

8. Click **Save** to exit the advance parameters.

Advanced parameters

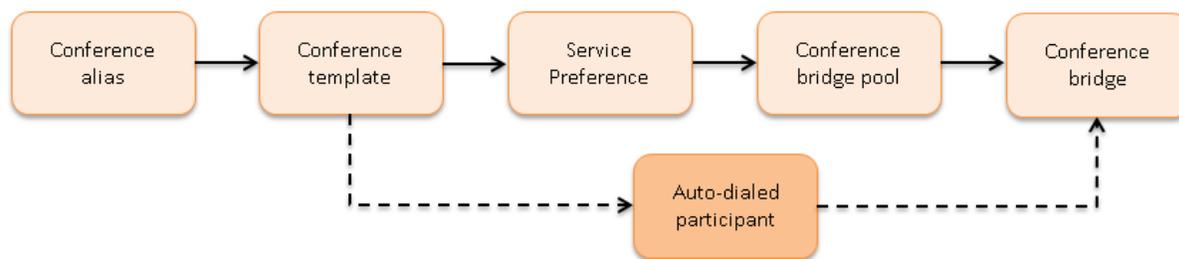
Summary of configured parameters

Parameter	Primary
pin	55105

Edit

9. Click **Save** on the **Conference template** page.

Task 23: Creating the auto-dialed participants



An auto-dialed participant is a participant that is automatically dialed from the conferencing resource at the start of the conference. The auto-dialed participant is associated with templates and is commonly used for dialing an endpoint, an external audio bridge, or a recording device.

Although the TelePresence Conductor and its conference bridges use SIP to call auto-dialed participants, H.323 endpoints can be called into a conference through the Cisco VCS's inter-working feature.

The following examples show how to create an auto-dialed participant for:

- an endpoint to join the 'HD Meeting'
- a recording device to join the 'Lecture' hosted on TelePresence MCUs
- a recording device to join the 'Lecture - TS' hosted on TelePresence Servers

Note: auto-dialed participants that are multiscreen endpoints are not supported. For multiscreen auto-dialed participants only the center screen is displayed in the conference.

Creating an auto-dialed participant for an endpoint

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

Name	Enter a name for the auto-dialed participant, for example <i>Invite boss to meeting</i> .
Conference template	Select <i>HD Meeting</i> .
Conference name match	Enter <i>meet\.boss\.(HD SD)</i> .
Address	Enter <i>boss@<SIP domain></i> .
Protocol	Select a protocol supported by the video network (<i>SIP</i> is recommended).
Role type	Select <i>Participant</i> .
Keep conference alive	Select <i>Yes</i> .

Auto-dialed participants

Modify participant

Name	★ Invite boss to meeting ⓘ
Description	ⓘ
Conference template	★ HD Meeting ⓘ Conference bridge type: TelePresence MCU
Conference name match (must use regex)	★ meet\.boss\.(HD SD) ⓘ
Participant address	★ boss@vcs.domain ⓘ
Protocol	SIP ⓘ
Role type	Participant ⓘ
DTMF sequence	ⓘ
Keep conference alive	Yes ⓘ
State	Enabled ⓘ

4. Click **Create participant**.
5. Click **View/Edit** for the 'Invite boss to meeting' auto-dialed participant.
6. At the bottom of the page, there is a chart with the templates that are associated with this auto-dialed participant. Verify this association is correct.

Template associated with this auto-dialed participant	Description	Type
HD Meeting		Meeting

7. Click **Save**.

Creating an auto-dialed participant for a recording device joining a TelePresence MCU hosted conference

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

Name	Enter a name for the auto-dialed participant, for example TCS - Recording device .
Conference template	Select <i>Lecture</i> to use with the 'Lecture' meeting on TelePresence MCUs.
Conference name match	Enter (.*) This will match on all conference names.
Address	Enter TCSrecording@<SIP domain> .
Protocol	Select a protocol supported by the video network (<i>SIP</i> is recommended).
Role type	Select <i>Guest</i> .
Keep conference alive	Select <i>No</i> .

Auto-dialed participants

Modify participant

Name	* TCS - Recording device
Description	
Conference template	* Lecture <small>Conference bridge type: TelePresence MCU</small>
Conference name match (must use regex)	* (.*)
Participant address	* TCSrecording@vcs.domain
Protocol	SIP
Role type	Guest
DTMF sequence	
Keep conference alive	No
State	Enabled

4. Click **Create participant**.
5. Click **View/Edit** for the 'TCS - Recording device' auto-dialed participant.
6. Click **Edit** under the **Advanced parameters** section.
7. Enter the following in the relevant fields, leave other fields as their default values:

Appear as a recording device Check the **on** box next to the field and then change the value to *True* from the drop-down list.

Advanced auto-dialed participant parameters

Modify advanced auto-dialed parameters

Participant name	TCS Recording device
Conference bridge type	TelePresence MCU
Appear as a recording device	<input checked="" type="checkbox"/> on value <input type="text" value="true"/>

8. Click **Save** to exit the advance parameters.

Advanced parameters

Summary of configured parameters

actAsRecorder	true
---------------	------

[Edit](#)

9. At the bottom of the page, there is a chart with the templates that are associated with this auto-dialed participant. Verify this association is correct.

Template associated with this auto-dialed participant	Description	Type
Lecture		Lecture

10. Click **Save** on the **Auto-dialed participants** page.

Creating an auto-dialed participant for a recording device joining a TelePresence Server hosted conference

1. Go to **Conference configuration > Auto-dialed participants**.
2. Click **New**.

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

Name	Enter a name for the auto-dialed participant, for example TCS - Recording device for TS .
Conference template	Select <i>Lecture</i> to use with the Lecture meeting on the TelePresence MCU
Conference name match	Enter (.*) This will match on all conference names.
Address	Enter TCSrecording@<SIP domain>
Protocol	Select a protocol supported by the video network (<i>SIP</i> is recommended).
Role type	Select <i>Guest</i> .
Keep conference alive	Select <i>No</i> .
Maximum quality	Enter the maximum quality setting to apply to this auto-dialed participant.

Auto-dialed participants

[Modify participant](#)

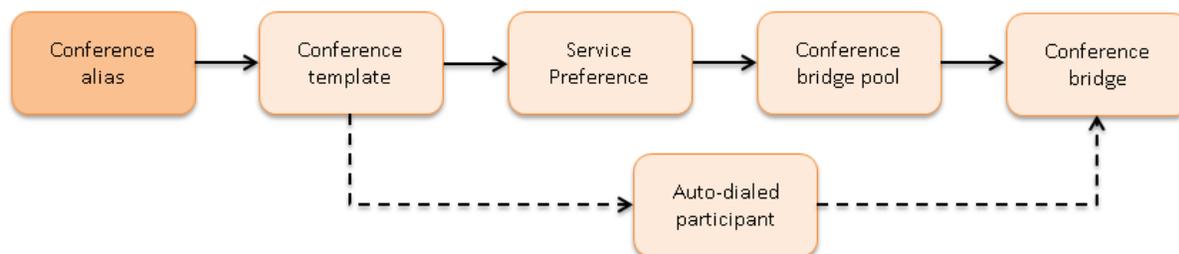
Name	* TCS - Recording device for TS <i>i</i>
Description	<input type="text"/> <i>i</i>
Conference template	* Lecture - TS <i>i</i> Conference bridge type: TelePresence Server
Conference name match (must use regex)	* (.*) <i>i</i>
Participant address	* TCSrecording@vcs.domain <i>i</i>
Protocol	SIP <i>i</i>
Role type	Guest <i>i</i>
DTMF sequence	<input type="text"/> <i>i</i>
Keep conference alive	No <i>i</i>
Maximum quality	HD (720p 30fps video, stereo audio) <i>i</i>
State	Enabled <i>i</i>

4. Click **Create participant**.
5. At the bottom of the page, there is a chart with the templates that are associated with this auto-dialed participant. Verify this association is correct.

Template associated with this auto-dialed participant	Description	Type
Lecture - TS		Lecture

6. Click **Save** on the **Auto-dialed participants** page.

Task 24: Creating conference aliases for the Meeting-type conferences



Meeting-type conferences require one or more conference aliases for the role-type of 'Participant'. The following examples show how to create a conference alias for:

- TelePresence MCU hosted 'HD Meeting' conference template
- TelePresence MCU hosted 'SD Meeting' conference template
- TelePresence Server hosted 'HD Meeting' conference template
- TelePresence Server hosted 'SD Meeting' conference template

Creating a conference alias for the TelePresence MCU hosted 'HD Meeting' template

1. Go to **Conference configuration > Conference aliases**.
2. Click **New**.
3. Enter the following in the relevant fields, leave other fields as their default values:

Name	Enter a name for the alias, for example HD Meeting .
Incoming alias	Enter (543. meet\. .*\.HD)@<SIP domain> . This pattern will either match a numerical alias of 543 and any single digit or meet.any_characters.HD@vcs.domain .
Conference name	Enter \1 .
Priority	Enter '25' for example.
Conference template	Select HD Meeting .
Role type	Select Participant .
Allow conference to be created	Select Yes if participants dialing this alias should be able to create the conference. If you select No , you need to define another alias resulting in the same conference with Allow conference to be created set to Yes or the conference must be created via the TelePresence Conductor's API. See Appendix 4: Allow conference to be created [p.73] for more information.

Conference aliases

Modify conference alias

Name	* HD Meeting
Description	
Incoming alias (must use regex)	* (543 meet\.*\HD)@vcs.domain
Conference name	* \1
Priority	* 25
Conference template	* HD Meeting
Role type	Participant
Allow conference to be created	Yes

Conference bridge type: TelePresence MCU

Create conference alias Cancel

- Click **Create conference alias**.

Creating a conference alias for the TelePresence MCU hosted 'SD Meeting' template

Repeat the steps under [Creating a conference alias for the TelePresence MCU hosted 'HD Meeting' template \[p.49\]](#) to create a conference alias for the 'SD Meeting' hosted on TelePresence MCUs. Enter the same values for the fields, apart from:

Name	Enter a name for the alias, for example <i>SD Meeting</i> .
Incoming alias	Enter <i>(544. meet\.*\SD)@<SIP domain></i> . This pattern will either match a numerical alias of 544 and any single digit or <i>meet.any_characters.SD@vcs.domain</i> .
Priority	Enter '40' for example.
Conference template	Select <i>SD Meeting</i> .

Creating a conference alias for the TelePresence Server hosted 'HD TS Meeting' template

The following steps create a conference alias that uses the 'HD TS Meeting' template and hosts the conference on a TelePresence Server:

- Go to **Conference configuration > Conference aliases**.
- Click **New**.
- Enter the following in the relevant fields, leave other fields as their default values:

Name	Enter a name for the alias, for example <i>HD TS Meeting</i> .
Incoming alias	Enter <i>(643. meetts\.*\HD)@<SIP domain></i> . This pattern will either match a numerical alias of 643 and any single digit or <i>meetts.any_characters.HD@vcs.domain</i> .
Conference name	Enter <i>\1</i> .
Priority	Enter '30' for example.
Conference template	Select <i>HD TS Meeting</i> .
Role type	Select <i>Participant</i> .

Allow conference to be created Select *Yes* if participants dialing this alias should be able to create the conference. If you select *No*, you need to define another alias resulting in the same conference with **Allow conference to be created** set to *Yes* or the conference must be created via the TelePresence Conductor's API.

See [Appendix 4: Allow conference to be created \[p.73\]](#) for more information.

Conference aliases

Modify conference alias

Name	HD TS Meeting
Description	
Incoming alias (must use regex)	(643 meetts\.*\HD)@vcs.domain
Conference name	11
Priority	30
Conference template	HD TS Meeting
Role type	Participant
Allow conference to be created	Yes

Conference bridge type: TelePresence MCU

Buttons: Create conference alias | Cancel

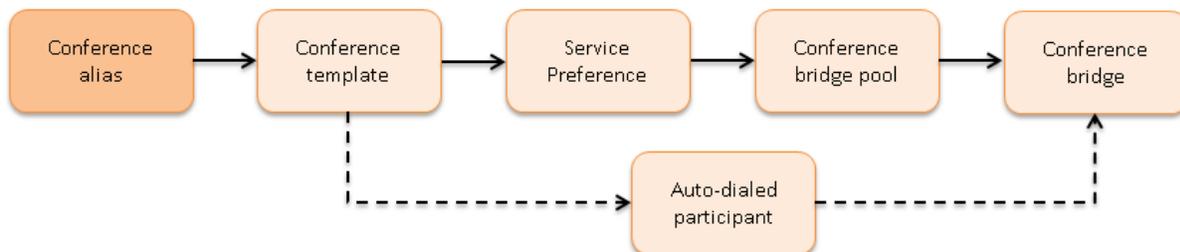
4. Click **Create conference alias**.

Creating a conference alias for the TelePresence Server hosted 'SD TS Meeting' template

Repeat the steps under [Creating a conference alias for the TelePresence Server hosted 'HD TS Meeting' template \[p.50\]](#) to create a conference alias for the 'SD Meeting' hosted on TelePresence Servers. Enter the same values for the fields, apart from:

Name	Enter a name for the alias, for example <i>SD TS Meeting</i> .
Incoming alias	Enter <i>(852. meetts\.*\SD)@<SIP domain></i> . This pattern will either match a numerical alias of 852 and any single digit or <i>meetts.any_characters.SD@vcs.domain</i> .
Priority	Enter '45' for example.
Conference template	Select <i>SD TS Meeting</i> .

Task 25: Creating conference aliases for the Lecture-type templates



Lecture-type conferences require one or more conference aliases per role-type. The role types are 'Chairperson' and 'Guest'. The following examples show how to create a conference alias for:

- TelePresence MCU hosted 'Lecture' template with a role of 'Chairperson'
- TelePresence MCU hosted 'Lecture' template with a role of 'Guest'
- TelePresence Server hosted 'Lecture-TS' template with a role of 'Chairperson'
- TelePresence Server hosted 'Lecture - TS' template with a role of 'Guest'

Creating a conference alias for the TelePresence MCU hosted 'Lecture' template with a role of 'Chairperson'

1. Go to **Conference configuration > Conference aliases**.
2. Click **New**.
3. Enter the following in the relevant fields, leave other fields as their default values:

Name	Enter a name for the alias, for example Training - Teacher .
Incoming alias	Enter (231. teach\..*)@<SIP domain> . This pattern will either match a numerical alias of 231 and any single digit or teach.any_characters@vcs.domain .
Conference name	Enter training .
Priority	Enter '10' for example.
Conference template	Select <i>Lecture</i> .
Role type	Select <i>Chairperson</i> .
Allow conference to be created	Select <i>Yes</i> if participants dialing this alias should be able to create the conference. If you select <i>No</i> , you need to define another alias resulting in the same conference with Allow conference to be created set to <i>Yes</i> or the conference must be created via the TelePresence Conductor's API. See Appendix 4: Allow conference to be created [p.73] for more information.

Conference aliases

[Modify conference alias](#)

Name	* Training - Teacher ?
Description	<input type="text"/> ?
Incoming alias (must use regex)	* (231. teach\..*)@vcs.domain ?
Conference name	* training ?
Priority	* 10 ?
Conference template	* Lecture ? Conference bridge type: TelePresence MCU
Role type	Chairperson ?
Allow conference to be created	Yes ?

4. Click **Create conference alias**.

Creating a conference alias for the TelePresence MCU hosted 'Lecture' template with a role of 'Guest'

1. Go to **Conference configuration > Conference aliases**.
2. Click **New**.
3. Enter the following in the relevant fields, leave other fields as their default values:

Name	Enter a name for the alias, for example Training - Students .
Incoming alias	Enter (388. learn\..)*@<SIP domain> . This pattern will either match a numerical alias of 388 and any single digit or learn.any_characters@vcs.domain .
Conference name	Enter Training for example.
Priority	Enter '20' for example.
Conference template	Select Lecture .
Role type	Select Guest .
Allow conference to be created	Select Yes if participants dialing this alias should be able to create the conference. If you select No , you need to define another alias resulting in the same conference with Allow conference to be created set to Yes or the conference must be created via the TelePresence Conductor's API. See Appendix 4: Allow conference to be created [p.73] for more information.

Conference aliases

Modify conference alias

Name	★ Training - Students i
Description	
Incoming alias (must use regex)	★ (388. learn\..)*@vcs.domain i
Conference name	★ Training i
Priority	★ 20 i
Conference template	★ Lecture i Conference bridge type: TelePresence MCU
Role type	Guest i
Allow conference to be created	Yes i

Create conference alias
Cancel

4. Click **Create conference alias**.

Creating a conference alias for the TelePresence Server hosted 'Lecture - TS' template with a role of 'Chairperson'

1. Go to **Conference configuration > Conference aliases**.
2. Click **New**.

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

Name	Enter a name for the alias, for example Training - Teacher - TS .
Incoming alias	Enter (232. teachts\..*)@<SIP domain> . This pattern will either match a numerical alias of 232 and any single digit or teachts.any_characters@vcs.domain .
Conference name	Enter TrainingTS for example.
Priority	Enter '15' for example.
Conference template	Select Lecture - TS .
Role type	Select Chairperson .
Allow conference to be created	Select Yes if participants dialing this alias should be able to create the conference. If you select No , you need to define another alias resulting in the same conference with Allow conference to be created set to Yes or the conference must be created via the TelePresence Conductor's API. See Appendix 4: Allow conference to be created [p.73] for more information.

Conference aliases

Modify conference alias

Name	* Training - Teacher - TS i
Description	<input type="text"/> i
Incoming alias (must use regex)	* (232. teachts\..*)@vcs.domain i
Conference name	* TrainingTS i
Priority	* 15 i
Conference template	* Lecture - TS i Conference bridge type: TelePresence Server
Role type	Chairperson i
Allow conference to be created	Yes i

Create conference alias
Cancel

4. Click **Create conference alias**.

Creating a conference alias for the TelePresence Server hosted 'Lecture - TS' template with a role of 'Guest'

1. Go to **Conference configuration > Conference aliases**.
2. Click **New**.
3. Enter the following in the relevant fields, leave other fields as their default values:

Name	Enter a name for the alias, for example Training - Students - TS .
-------------	---

Incoming alias	Enter <code>(389. learnts . .)@<SIP domain></code> . This pattern will either match a numerical alias of 389 and any single digit or <code>learnts.any_characters@vcs.domain</code> .
Conference name	Enter <code>TrainingTS</code> for example.
Priority	Enter '23' for example.
Conference template	Select <code>Lecture - TS</code> .
Role type	Select <code>Guest</code> .
Allow conference to be created	Select <code>Yes</code> if participants dialing this alias should be able to create the conference. If you select <code>No</code> , you need to define another alias resulting in the same conference with Allow conference to be created set to <code>Yes</code> or the conference must be created via the TelePresence Conductor's API. See Appendix 4: Allow conference to be created [p.73] for more information.

Conference aliases

Modify conference alias

Name	* Training - Students - TS i
Description	<input type="text"/> i
Incoming alias (must use regex)	* (389. learnts . .)@vcs.domain i
Conference name	* TrainingTS i
Priority	* 23 i
Conference template	* Lecture - TS i Conference bridge type: TelePresence Server
Role type	Guest i
Allow conference to be created	Yes i

Create conference alias
Cancel

4. Click **Create conference alias**.

Testing system configuration

When the configuration described in the previous sections is complete, you should test that the system is working correctly.

Creating a Meeting-type conference

To test that two or more endpoints can join an HD TelePresence MCU conference based on a template with a type of Meeting, dial 5432@<SIP domain> or meet.test.HD@<SIP domain> from each endpoint. Both endpoints should be taken to the same conference.

To test that two or more endpoints can join an HD TelePresence Server conference based on a template with a type of Meeting, dial 6432@<SIP domain> or meetts.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 an HD Meeting TelePresence MCU conference is created, dial 5432@<SIP domain> or meet.boss.HD@<SIP domain> from an endpoint. The auto-dialed participant boss@<SIP domain> and TCSrecording@<SIP domain> should receive a call from the TelePresence MCU conference bridge.

To test that auto-dialed participants are called when an HD Meeting TelePresence Server conference is created, dial 6432@<SIP domain> or meetts.boss.HD@<SIP domain> from an endpoint. The auto-dialed participant boss@<SIP domain> and TCSrecording@<SIP domain> should receive a call from the TelePresence Server conference bridge.

Creating a Lecture-type conference

To test that two or more endpoints can use different aliases to join the same TelePresence MCU conference based on a template with a type of Lecture, have one endpoint dial 2311@<SIP domain> or teach.test@vcs.domain to represent the teacher and have the other endpoint dial 3881@<SIP domain> or learn.test@vcs.domain. All endpoints should be taken to the same conference. The endpoints that dialed 3881@vcs.domain or learn.test@vcs.domain will see a blank screen until the endpoint that dialed 2311@vcs.domain or teach.test@vcs.domain enters the conference.

To test that two or more endpoints can use different aliases to join the same TelePresence Server conference based on a template with a type of Lecture, have one endpoint dial 2321@<SIP domain> or teachts.test@vcs.domain to represent the teacher and have the other endpoint dial 3891@<SIP domain> or learnmts.test@vcs.domain. All endpoints should be taken to the same conference. The endpoints that dialed 3891@vcs.domain or learnmts.test@vcs.domain will see a blank screen until the endpoint that dialed 2321@vcs.domain or teachts.test@vcs.domain enters the conference.

Testing 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 Cisco VCS you can test this by adding callers to the conference until it is cascaded. Alternatively, you can increase the number of chairperson participants 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.

Note that cascading is only supported on TelePresence MCUs; this capability does not exist on TelePresence Servers.

Creating a system backup

To create a backup of TelePresence Conductor system data:

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

Log files are not included in the system backup file.

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

For more information see [Cisco TelePresence Conductor Administrator Guide \(D14826\)](#) or the TelePresence Conductor's online help.

Troubleshooting

Tracking a call from Cisco VCS to TelePresence Conductor

Event log

To see the events associated with a particular call on both Cisco VCS and TelePresence Conductor look at the search history on the Cisco 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 [Cisco TelePresence Conductor Administrator Guide](#).

Note that the call tag is specific to a call across multiple Cisco VCSs.

Diagnostic log

Use diagnostic logging (**Maintenance > Diagnostics > Diagnostic logging**) to see the call signaling in the Cisco VCS.

Tracking a conference on the TelePresence Conductor

Event log

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.

Diagnostic log

Use diagnostic logging (**Maintenance > Diagnostics > Diagnostic logging**) to see the call signaling in the TelePresence Conductor.

Specific issues

Unable to enable more than one conference bridge

If only a single conference bridge can be enabled, the reason could be that there is no valid release key installed on the TelePresence Conductor.

Contact your Cisco account representative to obtain release key and option keys.

TelePresence Conductor does not communicate with any conference bridges

If the TelePresence Conductor is running without a release key, only a single un-clustered conference bridge is supported.

If the only conference bridge that is enabled on the TelePresence Conductor is clustered, the conference bridge shows as *Unusable* on the [Conference bridge status](#) page ([Status > Conference bridges](#)) and the TelePresence Conductor is unable to communicate with any conference bridges.

Contact your Cisco account representative to obtain release key and option keys.

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 this:
SearchRule (1)
Name: To Conference Policy Service
 - If the search rule is not used, go to [VCS configuration > Dial plan > Search rules](#) and 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 Cisco 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.

Conference does not get created

If a conference does not get created, check the list of alarms on the TelePresence Conductor.

If the alarm "Invalid JSON found" has been raised on the TelePresence Conductor and any JSON strings entered into the **Custom parameter** field on the [Advanced template parameters](#) or [Advanced auto-dialed participant parameters](#) pages contain double quotes, see [Alarm "Invalid JSON found" raised for valid JSON string \[p.62\]](#).

Auto-dialed participant not dialed

If the auto-dialed participant does not get called:

1. Go to **Status > Search history** on the Cisco VCS and see which 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** field 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 Cisco 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 Cisco VCS

If the conference bridges are not registering with the Cisco 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,
 - Or 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 TelePresence MCUs \[p.12\]](#).

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 Cisco VCS.
2. Ensure that the conference bridge is configured exactly as described in the section [Configuring the TelePresence MCUs \[p.12\]](#).

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 TelePresence MCUs \[p.12\]](#).

Pre-configured endpoint cannot join conference

When you pre-configure single-screen and multiscreen endpoints on the TelePresence Conductor, you specify the address of each codec used by the endpoint.

In certain scenarios the address of the endpoint may change depending on where it registers to (for example if the domain portion of the URI is the IP address of the peer the endpoint is registering to). If not all addresses that the endpoint can be known as are listed in the pre-configured endpoints configuration in TelePresence Conductor, the TelePresence Conductor may not recognize its address and the endpoint will use the template default settings rather than the known endpoint settings.

To resolve this, you must ensure that all possible addresses that could be used by the codec are listed.

To do this:

1. On the TelePresence Conductor, go to **Conference configuration > Pre-configured endpoints**.
2. From the list of pre-configured endpoints select the endpoint in question.
3. In the **Codecs** section at the bottom of the page, click on the first codec.
4. In the **Optional address** fields, ensure that all possible addresses from which calls for this codec could be received are listed.
5. Click **Save**.
6. Repeat steps 3-5 for each codec configured for that endpoint.

Auto-dialed participant joins the conference before the PIN is provided to the TelePresence MCU

A conference template on the TelePresence Conductor is configured to require a PIN on the TelePresence MCU and it has an auto-dialed participant associated with it.

An endpoint dials into a conference that is based on the conference template and hangs up before the user provides a PIN. In this case the auto-dialed participant is called and joins into the conference even though there are no participants in the conference. The call to the auto-dialed participant stays up until the max timer is reached.

To work around this issue (for a TelePresence MCU version 4.4 or later) and cause the conference bridge to delay calling the auto-dialed participant until at least one participant has entered the PIN and successfully joined the conference:

1. On the TelePresence Conductor, go to **Conference configuration > Conference templates** and select the relevant conference template.
2. In the **Advanced parameters** section click **Edit**.
3. Tick the first box next to **Custom parameters** and enter the following text into the adjacent text-box:
`{"preconfiguredParticipantsDefer": true}`
4. Click **Save** on the **Advanced template parameters** page.
5. Click **Save** on the **Conference templates** page.

Alarm "Invalid JSON found" raised for valid JSON string

It may be possible for the alarm "Invalid JSON found" to be raised even though the JSON string that was entered into the **Custom parameter** field on the **Advanced template parameters** or **Advanced auto-dialed participant parameters** pages appears to have been entered correctly. The alarm is raised if the JSON string contains double quotes (") with the Unicode value of 147 instead of the Unicode value 34. The Unicode value 147 is used in some external editors from which you may have copied the JSON string.

Sending the JSON string with the unsupported double quotes to the conference bridge will prevent the conference from being created.

To work around this issue, re-type the double quotes contained in the JSON string within the user interface field.

Error messages

Error communicating with mcu error="Method not supported" – this may be because a physical TelePresence Server has been added as a TelePresence MCU bridge.

Unsupported conference bridge software version - this may be because a physical TelePresence MCU has been added as a TelePresence Server bridge.

Regular expression match and replace

A regular expression replace of `\1\2` will replace with 1st bracket match and follow it with the 2nd bracket match.

If a match of the 1st bracket match, followed by the insertion of the literal digit 2 followed by the 2nd bracket match is required, then named matches need to be used. These work as follows:

`(?P<id>123) 456 (789)` will store

123 as `\1`

789 as `\2`

123 as named replace: `<id>` (the name used inside the "<" and ">" is user selectable)

to replace, use:

`\g<id>`

so to replace the 1st bracket match, followed by the insertion of the literal digit 2 followed by the 2nd bracket match use:

`\g<id>2\2`

Appendix 1: 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. Go to the TelePresence MCU in a browser.
2. Log in as administrator.
3. Go 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

Appendix 2: Example call flows

H.323 call flow

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

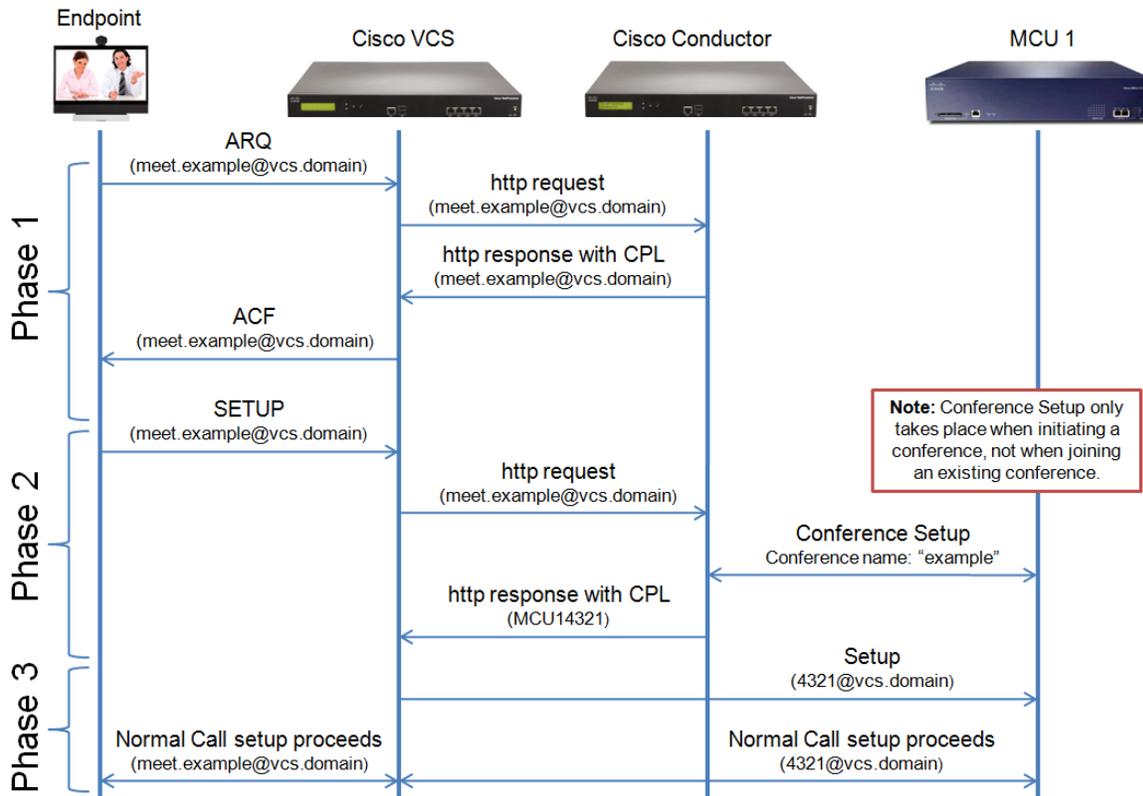


Figure 1: H.323 call flow

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 Cisco 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 Cisco 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, the TelePresence Conductor sets up a fresh conference on the conference bridge. TelePresence Conductor sends another piece of CPL to the Cisco VCS telling it to forward the call to MCU14321.

Phase 3

The Cisco 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 [p.65]) and proceeds from the point where the endpoint sends the SETUP message:

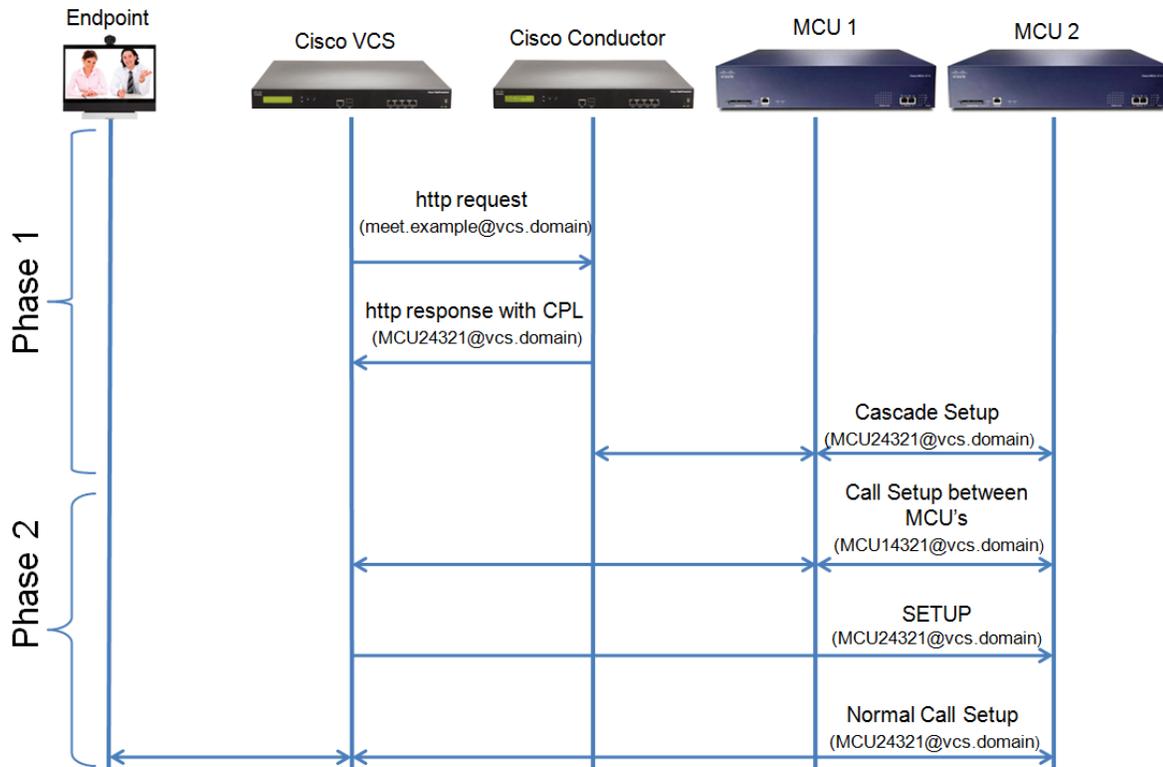


Figure 2: Cascade creation call flow

Phase 1

The Cisco 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 Cisco 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 Cisco 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.

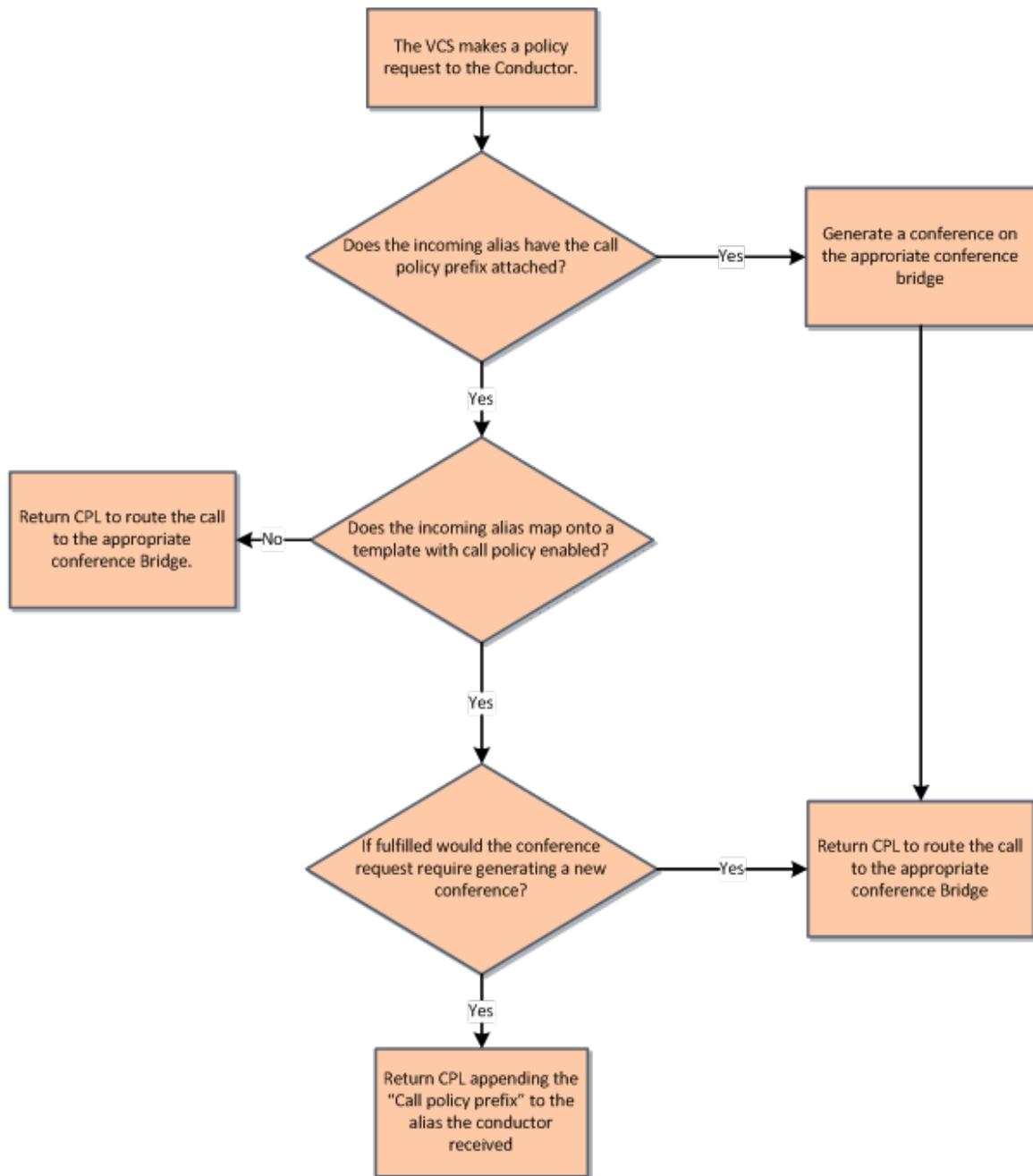
Appendix 3: Call policy mode

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

To make decisions on which participants are allowed to create a conference we recommend that you use the **Allow conference to be created** field on the TelePresence Conductor's conference alias instead of the Call Policy mode. For more information see [Appendix 4: Allow conference to be created \[p.73\]](#).

When a call policy mode enabled TelePresence Conductor receives a call that will generate a new conference, TelePresence Conductor returns call policy that attaches the call policy prefix to the dialed alias. This allows policy on the Cisco 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 Cisco VCS can control which users are allowed to create conferences.

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



There are three main ways a Cisco 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, see [External Policy on Cisco VCS Deployment Guide](#). 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

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**.
2. Click on the conference templates for which you want to enable Call Policy mode.
3. On the drop down menu for **Call Policy mode** select *On*.
4. Click **Save**.

Configuring the call policy prefix

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

1. Go to **Conference configuration > Call Policy**.
2. In the **Call Policy prefix** field 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 requires two search rules on the Cisco VCS pointing at the TelePresence Conductor policy service. The first has already been created in [Task 12: Adding the TelePresence Conductor as a policy service \[p.22\]](#). The second matches requests with the Call Policy prefix attached. To configure this (on the Cisco VCS):

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.)
4. Go to **VCS configuration > Dial plans > Search rules**.
5. Click **New**.
6. Enter the following in the relevant fields, leave other fields as their default values:

Rule name	Enter Authenticated users to Conductor Policy Service for example.
Description	Enter This search rule only matches authenticated users for example.
Priority	Enter '120' for example.
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 learn)\..*@<SIP domain></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	★ <input type="text" value="Authenticated users to Conductor Policy Service"/> ⓘ
Description	<input type="text" value="This search rule only matches authenticated users"/> ⓘ
Priority	★ <input type="text" value="120"/> ⓘ
Source	<input type="text" value="Any"/> ⓘ
Request must be authenticated	<input type="text" value="Yes"/> ⓘ
Mode	<input type="text" value="Alias pattern match"/> ⓘ
Pattern type	<input type="text" value="Regex"/> ⓘ
Pattern string	★ <input type="text" value="create\.(meet teach learn)\..*@vcs.domain"/> ⓘ
Pattern behavior	<input type="text" value="Leave"/> ⓘ
On successful match	<input type="text" value="Stop"/> ⓘ
Target	<input type="text" value="Conductor Policy Service"/> ⓘ
State	<input type="text" value="Enabled"/> ⓘ

7. Click **Create search rule**.

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

When using a call policy mode the Cisco 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** and select the search rule created in [Task 14: Configuring a search rule with the TelePresence Conductor policy service as the target \[p.26\]](#).
2. 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="10"/>	i
Protocol		<input type="text" value="Any"/>	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 learn)\. .*@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

3. 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:

1. Log into the Cisco VCS.
2. Go to **VCS configuration > Call Policy > Configuration**.
3. If it is not already selected, select *Local CPL* for **Call Policy mode**.
4. Click **Save**.
5. If the button is present, click **Delete uploaded file**.
6. Go to **VCS configuration > Call Policy > Rules**.
7. Click **New**.
8. Enter the following in the relevant fields, leave other fields as their default values:

Source pattern	Enter <code>. *@<SIP domain></code>
Destination pattern	Enter <code>create\.(meet teach learn)\. *@<SIP domain></code>
Action	Select <i>Allow</i>

9. Click **Add**.

10. **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.

Click **New**.

11. Enter the following in the relevant fields, leave other fields as their default values:

Source pattern	Enter <code>.*@.*</code>
Destination pattern	Enter <code>create\.(meet teach teach)\..*@<SIP domain></code>
Action	Select <i>Reject</i>

12. Click **Add**.

Appendix 4: Allow conference to be created

Similar to the Call Policy mode, which is only applicable to deployments using the Cisco VCS's external policy service interface, the field **Allow conference to be created** on the **Conference aliases** page, allows call control to define whether participants can create a conference or just join the conference after it has been created. We recommend that you use the **Allow conference to be created** field instead of the Call Policy feature.

For each alias associated with a conference template, define whether the participants dialing the alias are allowed to create the conference or not.

To define an alias for participants who are allowed to create a conference:

1. On the TelePresence Conductor go to **Conference configuration > Conference aliases**.
2. Fill in all required fields with the relevant values according to [Task 24: Creating conference aliases for the Meeting-type conferences \[p.49\]](#) or [Task 25: Creating conference aliases for the Lecture-type templates \[p.51\]](#).
3. For the field **Allow conference to be created** select **Yes**.

To define an alias for participants who are not allowed to create a conference and can only join the conference after it has been created:

1. On the TelePresence Conductor go to **Conference configuration > Conference aliases**.
2. Fill in all required fields with the relevant values according to [Task 24: Creating conference aliases for the Meeting-type conferences \[p.49\]](#) or [Task 25: Creating conference aliases for the Lecture-type templates \[p.51\]](#).
3. For the field **Allow conference to be created** select **No**.

Where a conference alias has been set up with **Allow conference to be created** set to **No**, a conference can be started using either one of the following methods:

- by making a call to a second conference alias that matches to the same conference and has **Allow conference to be created** set to **Yes**.
The two conference aliases must either have the same **Conference name** defined or a regex replace string (in the **Conference name** field) that results in the same conference name for both conference aliases.
- by creating the conference via the TelePresence Conductor's API, for example using Cisco TMS.

Note: participants dialing the conference alias that has **Allow conference to be created** set to **No** before the conference is created on the TelePresence Conductor will be rejected.

Document revision history

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

Revision	Date	Description
05	August 2013	Updated for XC2.2 and renamed to Cisco TelePresence Conductor with Cisco VCS (Policy Service) Deployment Guide.
04	May 2013	Updated for XC2.1
03	December 2012	Updated for XC2.0 and renamed as <i>Cisco TelePresence Conductor with Cisco TelePresence Video Communication Server (VCS) Deployment Guide</i> . Information regarding deployments with Cisco Unified Call Manager is now in <i>Cisco TelePresence Conductor with Cisco Unified Communications Manager Deployment Guide (D14998)</i> .
02	May 2012	Updated for XC1.2.
01	May 2012	Initial release.

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