



# **Cisco TelePresence Conductor with Cisco VCS (B2BUA)**

## **Deployment Guide**

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**XC2.2  
X7.0 and later**

**D15014.03**

**Revised May 2014**

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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 call routing to an appropriate conference bridge. 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 TelePresence Conductor's B2BUA. For more information on the deployment using the Cisco VCS's external policy server interface, see [Cisco TelePresence Conductor with Cisco VCS \(Policy Service\) 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).

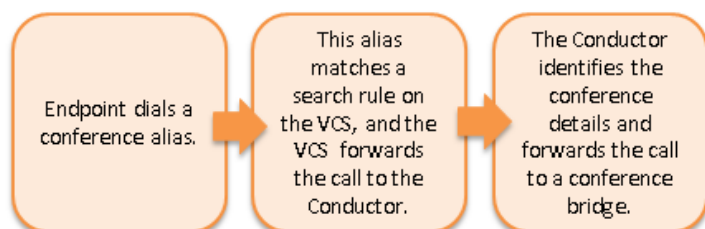
## Further reading

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 \(B2BUA\) Deployment Guide](#) (D15034).

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

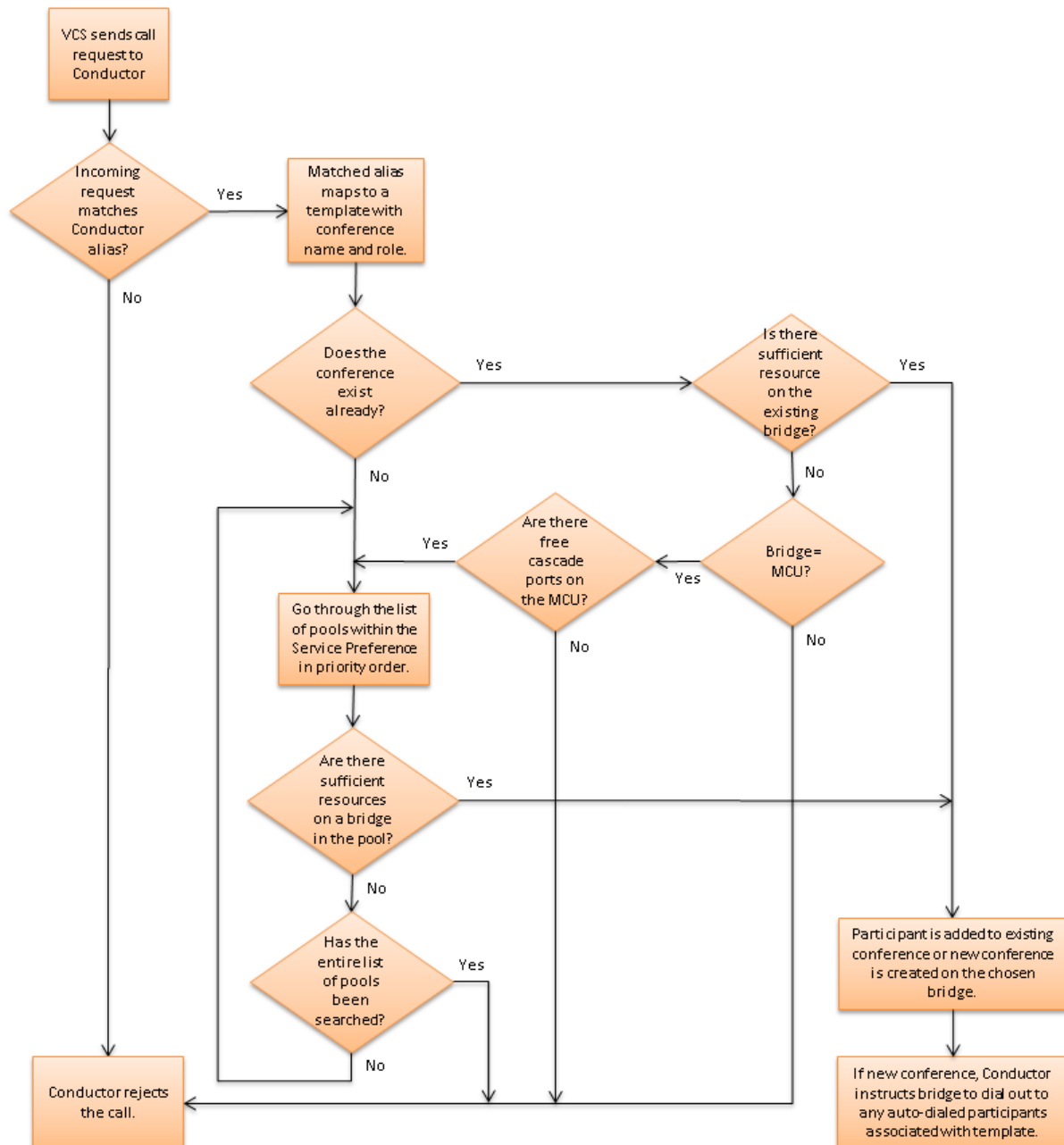
## Call flow with the TelePresence Conductor

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:

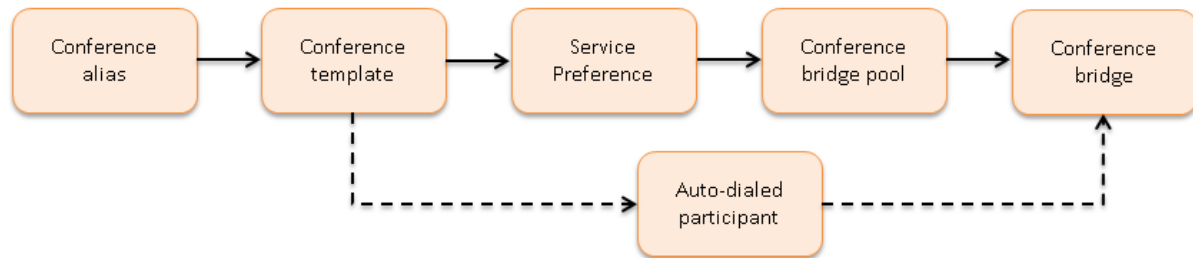


When these parts of the call flow are complete, the call is set up and media flows between the endpoint and the conference bridge.

## TelePresence Conductor conference bridge selection process



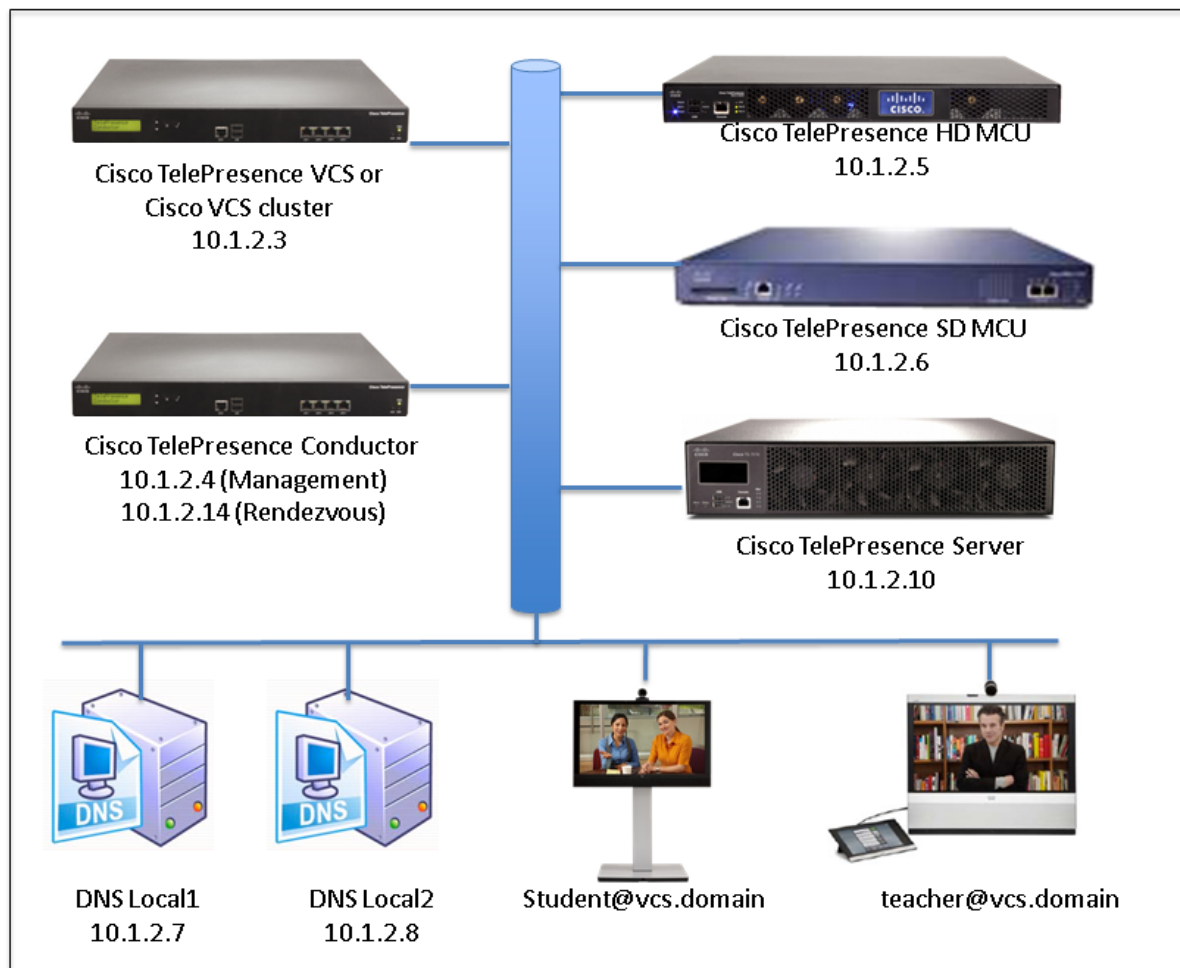
In a simplified format the set of steps for a conference to be created when the TelePresence Conductor receives an individual valid conference request is:



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 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](#).
- The TelePresence Conductor must have enough unique IP addresses configured to fulfill the requirements for creating a Location supporting rendezvous type calls for use by Cisco VCS. The TelePresence Conductor must have an IP address for management plus an IP address for all VCS rendezvous conferences.
- One or more conference bridges are powered on and accessible over HTTP/HTTPS and SIP TLS. 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.

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 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

## 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**.

- Enter the following in the relevant fields:

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

- Click **Add user**.
- 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 must install this option key in order for this deployment to work.

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

- Go to **Settings > Upgrade**.
- 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:

- Go to **Settings > Encryption**.
- 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 [Configuring the Cisco VCS \[p.19\]](#).
8. Ensure that **SIP (TCP)** is unchecked.
9. Ensure that **SIP (UDP)** is unchecked.
10. Ensure that **Incoming H.323** is checked. . H.323 is required for TelePresence MCU cascading to work.
11. 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 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 checked="" type="checkbox"/>	1719

Apply changes

12. Repeat the steps for any other TelePresence MCUs

### Task 3: Configuring SIP

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

<b>SIP registrar usage</b>	Select <i>Disabled</i> .
<b>SIP proxy address</b>	Leave blank.
<b>Outgoing transport</b>	Select <i>TLS</i> .
<b>Use local certificate for outgoing connections and registrations</b>	Check this box.

SIP	
SIP registrar usage	Disabled ▼
SIP registrar domain	<input type="text"/>
SIP registrar type	Standard SIP ▼
Username	<input type="text"/>
Password	<input type="password"/>
Allow numeric ID registration for conferences	<input type="checkbox"/>
SIP call settings	
SIP proxy address	<input type="text"/>
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 checked="" type="checkbox"/>

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

## Task 4: Disabling H.323 registration

- Go to **Settings > H.323**.
- Set **H.323 gatekeeper usage** to *Disabled*.
- Leave all other fields as their default values.

H.323	
H.323 gatekeeper usage	Disabled ▼
H.323 gatekeeper address	<input type="text"/>
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	<input type="text"/>
Use password	<input type="checkbox"/> Password: <input type="password"/>
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

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

## Task 5: Changing miscellaneous settings

On all conference bridges:

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 ▼
<b>Media port reservation</b>	<b>Disabled ▼</b>

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.

Configuration	
Status	Disabled ▼

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:

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

**Add new user** You are here: [Users](#) > [Add new user](#)

User

User ID

conductoradmin

Name

Admin for Conductor

Password

••••••••

Re-enter password

••••••••

Access rights

Administrator ▼

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 must install the option key in order for this deployment to work.

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

Feature management

Activated features

MSE 8510 activation (00000-00000-00000-00000)

**Encryption** (00000-00000-00000-00000) remove

Third party interop (00000-00000-00000-00000) remove

License keys

Media port licenses x 80 (00000-00000-00000-00000)

TS screen licenses x 16 (00000-00000-00000-00000)

Activation code

Update features

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 **Incoming H.323**, **SIP (TCP)** and **SIP (UDP)** are not checked.



- Ensure that **HTTPS** is enabled on port 443.

**Services**
You are here: [Network](#) > [Services](#)

Port A	
TCP service	IPv4
HTTP	<input checked="" type="checkbox"/> 80
HTTPS	<input checked="" type="checkbox"/> 443
Incoming H.323	<input type="checkbox"/> 1720
SIP (TCP)	<input type="checkbox"/> 5060
Encrypted SIP (TLS)	<input checked="" type="checkbox"/> 5061
FTP	<input checked="" type="checkbox"/> 21

Port A	
UDP service	IPv4
SIP (UDP)	<input type="checkbox"/> 5060

- Click **Apply changes**.

## Task 8: Configuring SIP

For the TelePresence Server to support auto-dialed participants, the TelePresence Server needs to know where to direct signaling requests.

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

<b>Outbound call configuration</b>	Select <i>Call direct</i> from the drop-down list.
<b>Outbound address</b>	Leave blank.
<b>Outbound domain</b>	Leave blank.
<b>Username</b>	Leave blank.
<b>Password</b>	Leave blank.
<b>Outbound transport</b>	Select <i>TLS</i> .
<b>Negotiate SRTP using SDES</b>	Select <i>For secure transports (TLS) only</i> .
<b>Use local certificate for outgoing connections and registrations</b>	Check the box.

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

**SIP**

Outbound call configuration	Call direct
Outbound address	<input type="text"/>
Outbound domain	<input type="text"/>
Username	<input type="text"/>
Password	<input type="password"/>
Outbound transport	TLS
Negotiate SRTP using SDP	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: Disabling H.323 registration

Perform the following steps to disable H323 registration to a gatekeeper:

1. Go to **Configuration > H323 Settings**.
2. Uncheck the box for **Use gatekeeper**.
3. Leave all other fields as their default values.
4. Click **Apply changes**.
5. 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

Remotely managed

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

### Task 11: Adding the TelePresence Conductor as a neighbor zone

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:

<b>Name</b>	Enter 'Conductor' for example.
<b>Type</b>	Select <i>Neighbor</i> .
<b>H.323 mode</b>	Select <i>Off</i> .
<b>SIP transport</b>	Select <i>TLS</i> . Set the port to 5061.
<b>Peer 1 address</b>	Enter the TelePresence Conductor's rendezvous IP address for Cisco VCS (not the TelePresence Conductor's primary LAN IP address used to manage the TelePresence Conductor). This will be added on the TelePresence Conductor in <a href="#">Task 16: Adding an IP address for Cisco VCS rendezvous conferences on TelePresence Conductor [p.25]</a> .
<b>Zone profile</b>	Select <i>Custom</i> .
<b>Automatically respond to SIP searches</b>	Select <i>On</i> .

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

**Configuration**

Name

★ Conductor

Type

★ Neighbor

Hop count

★ 15

**H.323**

Mode

Off

Port

★ 1719

**SIP**

Mode

On

Port

★ 5061

Transport

TLS

TLS verify mode

Off

Accept proxied registrations

Allow

Media encryption mode

Auto

**Authentication**

Authentication policy

Do not check credentials

SIP authentication trust mode

Off

**Location**

Peer 1 address: 10.1.2.14

Peer 2 address:

Peer 3 address:

Peer 4 address:

Peer 5 address:

Peer 6 address:

**Advanced**

Zone profile: Custom

Monitor peer status: Yes

Call signaling routed mode: Auto

Automatically respond to H.323 searches: Off

Automatically respond to SIP searches: On

Empty INVITE allowed: On

SIP poison mode: Off

SIP encryption mode: Auto

SIP SDP attribute line limit mode: Off

SIP SDP attribute line limit length: 130

SIP multipart MIME strip mode: Off

SIP UPDATE strip mode: Off

Interworking SIP search strategy: Options

SIP UDP/BFCP filter mode: Off

SIP Duo Video filter mode: Off

SIP record route address type: IP

SIP Proxy-Require header strip list:

4. Click **Create zone**.

## Task 12: Configuring a search rule with the TelePresence Conductor neighbor zone 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:

<b>Rule name</b>	Enter 'To Conductor' for example.
------------------	-----------------------------------

<b>Priority</b>	Enter '10' for example.
-----------------	-------------------------

<b>Mode</b>	Select <i>Alias pattern match</i> .
<b>Pattern type</b>	Select <i>Regex</i> .
<b>Pattern string</b>	Enter <code>(meet meetts teach learn teachts learnts)\..*@&lt;SIP domain&gt;</code> <b>Note:</b> Replace <SIP domain> with the appropriate SIP domain for your network.
<b>Pattern behavior</b>	Select <i>Leave</i> .
<b>On successful match</b>	Select <i>Stop</i> .
<b>Target</b>	Select <i>Conductor</i> .

### Create search rule

Configuration

Rule name

★ To Conductor ⓘ

Description

ⓘ

Priority

★ 10 ⓘ

Protocol

Any ⓘ

Source

Any ⓘ

Request must be authenticated

No ⓘ

Mode

Alias pattern match ⓘ

Pattern type

Regex ⓘ

Pattern string

★ (meet|meetts|teach|learn|teachts|learnts)\..\*@vcs\domain ⓘ

Pattern behavior

Leave ⓘ

On successful match

Stop ⓘ

Target

Conductor ⓘ

State

Enabled ⓘ

Create search rule

Cancel

- Click **Create search rule**.

## 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 will only accept calls 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 13: 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 14: 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 15: 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:

<b>System host name</b>	Enter the hostname of your TelePresence Conductor.
<b>Domain name</b>	Enter the domain for your TelePresence Conductor.
<b>Address 1</b>	Enter the IP address of the DNS server.
<b>Address 2</b>	Enter the IP address of your backup DNS server.

## DNS

**DNS settings**

System host name	<input type="text" value="conductor"/>	
Domain name	<input type="text" value="vcs.domain"/>	
DNS requests port	<input type="text" value="1024"/>	
range start		
DNS requests port	<input type="text" value="65535"/>	
range end		

**Default DNS servers**

Address 1	<input type="text" value="10.1.2.7"/>	
Address 2	<input type="text" value="10.1.2.8"/>	
Address 3	<input type="text"/>	

**Per-domain DNS servers**

Address 1	<input type="text"/>		Domr
Address 2	<input type="text"/>		Domr

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

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



## Task 16: Adding an IP address for Cisco VCS rendezvous conferences on TelePresence Conductor

1. Go to **System > IP**.
2. In the **Additional addresses for LAN 1** section click **New**.

**IP**

**Network configuration**

IPv4 gateway: 10.1.2.1

**Primary LAN 1 IP address**

IPv4 address: 10.1.2.4

IPv4 subnet mask: 255.255.255.128

IPv4 address range: 10.1.2.0 - 10.1.2.127

**Additional addresses for LAN 1**

IP address

**New** Delete address Select all Unselect all

Save

3. Enter the new IP address to be used.  
**Note:** the IP address must be on the same subnet as the primary TelePresence Conductor IP interface, and must be reserved for use by this TelePresence Conductor alone. It will be used as the **Peer 1 address** in the Cisco VCS neighbor zone created later on in the process.
4. Click **Add address**.

**Additional IP addresses**

**Additional address for LAN 1**


Address: 10.1.2.14

**Add address** Cancel


IP address needs to be on the same subnet as Conductor

5. In the **Additional addresses for LAN 1** list, verify that the IP address was added correctly.


**IP**


 **Saved:** Network interface settings saved, however a **restart** is required for them to take effect.

**Network configuration**

IPv4 gateway  

**Primary LAN 1 IP address**

IPv4 address  

IPv4 subnet mask  

IPv4 address range

**Additional addresses for LAN 1**

**IP address** ▼

☐ **10.1.2.14**

6. Go to **Maintenance > Restart options**.
7. Click **Restart** so that network interface changes are applied.
8. Wait for the TelePresence Conductor to restart.
9. To verify the new TelePresence Conductor IP address is active on the network, ping the IP address from another device.

## Task 17: Configuring a Location for Cisco VCS

A Location is used to allow the Cisco VCS to forward conference call requests directly to the TelePresence Conductor back-to-back-user-agent (B2BUA). A single Location can be set up for all traffic between any Cisco VCS (or Cisco VCS cluster) and the TelePresence Conductor.

Because the Cisco VCS only supports rendezvous conferences (not ad hoc), a conference alias is required. Endpoints registered to a Cisco VCS are able to initiate a Multiway conference, which is similar to an ad hoc conference initiated by endpoints registered to a Unified CM, but still uses a conference alias to create a spontaneous rendezvous conference.

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

<b>Location name</b>	Enter a name for the Location, for example <b>vcs_location</b> .
<b>Conference type</b>	Select <i>Rendezvous</i> .
<b>Rendezvous IP address</b>	From the drop down list, select the TelePresence Conductor IP address to be used for Cisco VCS calls. This must match the <b>Destination address</b> of the neighbor zone on the Cisco VCS.

**Trunk IP address** This is only required if TelePresence Conductor needs to forward auto-dialed participants or any other out-dialed calls such as those initiated by Cisco TMS to the Cisco VCS.  
Enter the IP address of the Cisco VCS.

**Trunk port** Enter '5061'.

**Trunk protocol** Select *TLS*.

### Locations

#### Modify Location

Location name  ⓘ  
 Description  ⓘ  
 Conference type  ⓘ

#### Rendezvous conference settings

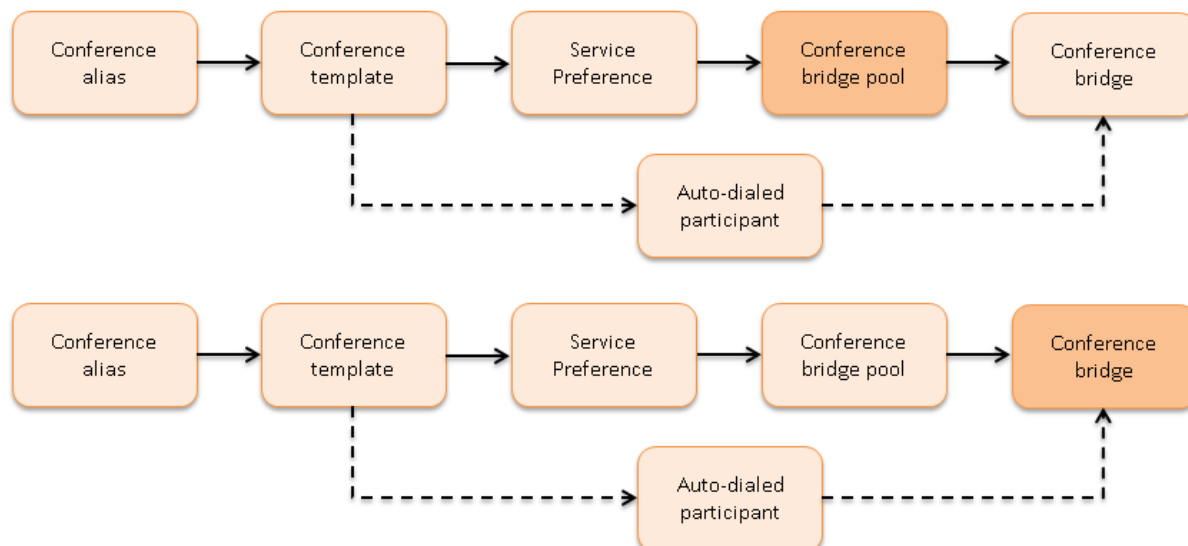
Rendezvous IP address (local)  ⓘ

#### Unified CM trunk settings for outdial

Outdial local IP address   
 Trunk IP address  ⓘ  
 Trunk port  ⓘ  
 Trunk transport protocol  ⓘ

4. Click **Add location**.

## Task 18: 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. Select from the drop-down list the **Location** configured for the Cisco VCS.

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:

<b>Name</b>	Enter a name for the conference bridge, for example <b>HD MCU</b> .
<b>State</b>	Select <i>Enabled</i> .
<b>IP address or FQDN</b>	Enter the IP address of the HD conference bridge.
<b>Protocol</b>	Select <i>HTTPS</i> .
<b>Port</b>	Enter '443'.
<b>Conference bridge username</b>	Enter the conference bridge admin username, for example <b>conductoradmin</b> . (This is created in <a href="#">Task 1: Creating a user [p.11]</a> .)
<b>Conference bridge Password</b>	Enter the conference bridge password for this user.

<b>Dedicated content ports</b>	Enter the appropriate value for your TelePresence MCU. To discover if a TelePresence MCU has any dedicated content ports follow the steps given in <a href="#">Appendix 1: Identifying dedicated content ports on a Cisco TelePresence MCU [p.60]</a> .
<b>SIP port</b>	Enter the SIP port on which the conference bridge listens for SIP TLS traffic, typically this is '5061'.
<b>H.323 cascade call routing</b>	Select <i>Direct</i> .

**Add conference bridge**

**Configuration**

Name	★ HD MCU ⓘ
Description	ⓘ
State	Enabled ⓘ
IP address or FQDN	★ 10.1.2.5 ⓘ
Protocol	HTTPS ⓘ
Port	★ 443 ⓘ
Conference bridge username	★ conductoradmin ⓘ
Conference bridge password	..... ⓘ
Dial plan prefix	ⓘ
Conference bridge type	★ TelePresence MCU ⓘ
Conference bridge pool	★ HD MCU pool ⓘ
Dedicated content ports	★ 0 ⓘ
SIP port	★ 5061 ⓘ
H.323 cascade call routing	Direct ⓘ

Create conference bridge Cancel

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 \[p.28\]](#) 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 pool, 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 \[p.28\]](#) to add a TelePresence MCU to the SD conference bridge pool. Enter the same values for the fields, apart from:

<b>Name</b>	Enter a name for the conference bridge, for example SD MCU.
<b>IP address or FQDN</b>	Enter the IP address of the SD conference bridge.

## 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. Select from the drop-down list the **Location** configured for the Cisco VCS.

**Conference bridge pools**

**Configuration**

Pool name: HD TS pool

Description:

Conference bridge type: TelePresence Server

Raise conference bridge resource alarm: ☒

Location: VCS location

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 \[p.18\]](#)).

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

<b>Name</b>	Enter a name for the conference bridge, for example <b>HD TS</b> .
<b>State</b>	Select <i>Enabled</i> .
<b>IP address or FQDN</b>	Enter the IP address of the HD conference bridge.
<b>Protocol</b>	Select <i>HTTPS</i> .
<b>Port</b>	Enter '443'.
<b>Conference bridge username</b>	Enter the conference bridge admin username, for example <b>conductoradmin</b> . (This is created in <a href="#">Task 6: Creating a user [p.15]</a> .)
<b>Conference bridge Password</b>	Enter the conference bridge password for this user.
<b>SIP port</b>	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
Description	
State	Enabled
IP address or FQDN	★ 10.1.2.10
Protocol	HTTPS
Port	★ 443
Conference bridge username	★ conductoradmin
Conference bridge password	•••••
Dial plan prefix	
Conference bridge type	★ TelePresence Server
Conference bridge pool	★ HD TS pool
SIP port	★ 5061

Create conference bridge Cancel

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

### Creating a TelePresence Server SD conference bridge pool

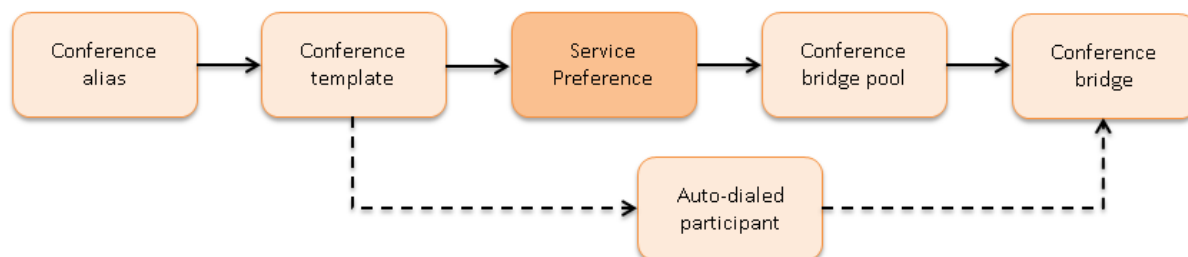
Repeat the steps under [Creating a TelePresence Server HD conference bridge pool \[p.30\]](#) to create a TelePresence Server SD conference bridge pool. Enter the same values for the fields, apart from the **Pool name**, which should be **SD TS pool1**, for example.

### Adding a TelePresence Server to the SD conference bridge pool

Repeat the steps under [Adding a TelePresence Server to the HD conference bridge pool \[p.30\]](#) to add a TelePresence Server to the SD conference bridge pool. Enter the same values for the fields, apart from:

<b>Name</b>	Enter a name for the conference bridge, for example <b>SD TS</b> .
<b>IP address or FQDN</b>	Enter the IP address of the SD conference bridge.

## Task 19: 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 Presences 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	<a href="#">HD MCU pool</a>
<input type="checkbox"/> 2	<a href="#">SD MCU pool</a>
<input type="text" value="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	<a href="#">SD MCU pool</a>
<input type="checkbox"/> 2	<a href="#">HD MCU pool</a>
	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:  ⓘ

Description:  ⓘ

Conference bridge type:  ⓘ

**Pools**

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

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**

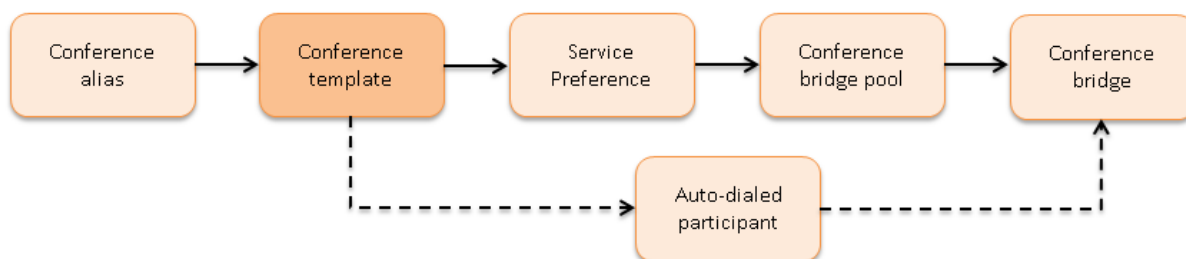
Priority	Pool name
<input type="checkbox"/> 1	<a href="#">SD TS pool</a>
	Please select ▼

Add selected pool Delete pool Select all Unselect all

Save Delete Cancel

8. Click **Save**.

## Task 20: 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:

<b>Name</b>	Enter a name for the conference template, for example <b>HD Meeting</b> .
-------------	---

<b>Conference type</b>	Select <i>Meeting</i> .
<b>Service Preference</b>	Select <i>Prefer HD with SD fallback</i> .
<b>Number of cascade ports to reserve</b>	Enter '1' to enable cascading to one other TelePresence MCU. Enter '0' to disable cascading.

**Conference templates**

**Modify conference template**

Name	★ HD Meeting ⓘ
Description	ⓘ
Conference type	Meeting ⓘ
Call Policy mode	Off ⓘ
Service Preference	★ Prefer HD with SD fallback ⓘ
Number of cascade ports to reserve	★ 1 ⓘ
Limit number of participants	<input type="checkbox"/> Maximum ⓘ There are 0 aut
Limit the conference duration (minutes)	<input type="checkbox"/> Maximum ⓘ
Allow content	Yes ⓘ
Scheduled conference	No ⓘ

- 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 \[p.34\]](#) to create a conference template for an 'SD Meeting' hosted on TelePresence MCUs. Enter the same values for the fields, apart from:

<b>Name</b>	Enter a name for the conference template, for example <i>SD meeting</i> .
<b>Service Preference</b>	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.

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

<b>Name</b>	Enter a name for the conference template, for example <i>HD TS meeting</i> .
<b>Conference type</b>	Select <i>Meeting</i> .

<b>Call Policy mode</b>	This feature is not supported for this deployment. To define which participants can create a conference use the field <b>Allow conference to be created</b> on the <a href="#">Conference aliases</a> page instead.
<b>Service Preference</b>	Select <i>Prefer HD TS</i> .
<b>Participant quality</b>	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.
<b>Provision for multiscreen</b>	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 Yes 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 <a href="#">Conference configuration &gt; Pre-configured endpoints</a> ).  For more information on pre-configuring endpoints see <a href="#">Cisco TelePresence Conductor Administrator Guide</a> .
<b>Content quality</b>	Select the maximum content quality allowed for this conference.

**Conference templates** You are here: [Conference configuration](#) > [Conference templates](#) > New

**Modify conference template**

Name	<input type="text" value="HD TS Meeting"/>	
Description	<input type="text"/>	
Conference type	Meeting	
Call Policy mode	Off	
Service Preference	<input checked="" type="radio"/> Prefer HD TS  Conference bridge type: TelePresence Server	
Limit number of participants	<input type="checkbox"/> Maximum <input type="text" value=""/>	There are 0 auto-dialed participants associated with this template.
Limit the conference duration (minutes)	<input type="checkbox"/> Maximum <input type="text" value=""/>	
Participant quality	Full HD	
Allow multiscreen	Yes	
Maximum screens	<input checked="" type="radio"/> 3	
Optimize resources	Yes	
Content quality	Full HD	
Scheduled conference	No	

**Advanced parameters**

Advanced parameters can be edited after the template has been created.

#### 4. Click **Create conference template**.

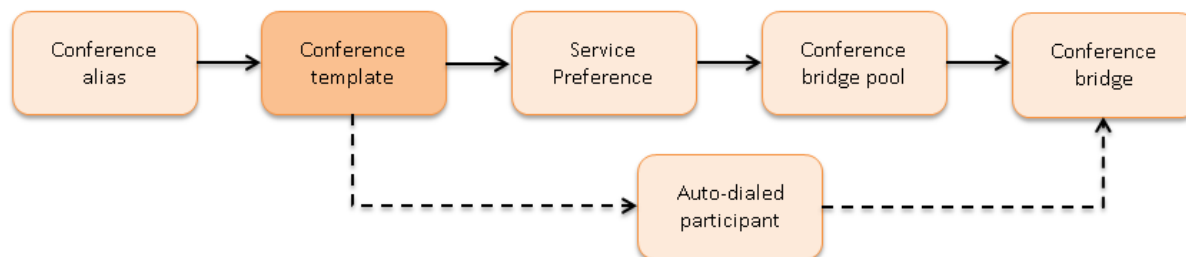
### Creating a conference template for an 'SD Meeting' hosted on TelePresence Servers

Repeat the steps under [Creating a conference template for an 'HD Meeting' hosted on TelePresence Servers \[p.35\]](#) to create a conference template for an 'SD Meeting' hosted on TelePresence Servers. Enter the same

values for the fields, apart from:

<b>Name</b>	Enter a name for the conference template, for example <b>SD TS meeting</b> .
<b>Service Preference</b>	Select <i>Prefer SD TS</i> .
<b>Participant quality</b>	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 21: 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:

<b>Name</b>	Enter a name for the conference template, for example <b>Lecture</b> .
<b>Conference type</b>	Select <i>Lecture</i> .
<b>Number of chairperson participants to reserve</b>	Enter '2' in this example.
<b>Call Policy mode</b>	This feature is not supported for this deployment. To define which participants can create a conference use the field <b>Allow conference to be created</b> on the <b>Conference aliases</b> page instead.
<b>Service Preference</b>	Select <i>Prefer HD with SD fallback</i> .
<b>Number of cascade ports to reserve</b>	Enter '1' to enable cascading to one other TelePresence MCU. Enter '0' to disable cascading.

**Conference templates**

**Modify conference template**

Name	★ Lecture ⓘ
Description	ⓘ
Conference type	Lecture ⓘ
Number of chairperson participants to reserve	★ 2 ⓘ
Call Policy mode	Off ⓘ
Service Preference	★ Prefer HD with SD fallback ⓘ
Number of cascade ports to reserve	★ 1 ⓘ
Limit number of participants	<input type="checkbox"/> Maximum ⓘ There are 0
Limit the conference duration (minutes)	<input type="checkbox"/> Maximum ⓘ
Allow content	Yes ⓘ
Scheduled conference	No ⓘ

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.

New participants will see custom layout	<input type="checkbox"/> false ⓘ	<input type="checkbox"/> false ⓘ
PIN	<input checked="" type="checkbox"/> 55105 ⓘ	<input type="checkbox"/> ⓘ

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

**Advanced 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:

<b>Name</b>	Enter a name for the conference template, for example <b>Lecture - TS</b> .
<b>Conference type</b>	Select <i>Lecture</i> .
<b>Number of chairperson participants to reserve</b>	Enter '2' in this example.
<b>Call Policy mode</b>	This feature is not supported for this deployment. To define which participants can create a conference use the field <b>Allow conference to be created</b> on the <a href="#">Conference aliases</a> page instead.
<b>Service Preference</b>	Select <i>Prefer HD TS</i> .
<b>Chairperson quality</b>	Enter the maximum quality setting to apply to chairpersons using this conference template. 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.
<b>Guest quality</b>	Enter the maximum quality setting to apply to guests using this conference template. 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.
<b>Provision for multiscreen</b>	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 Yes 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 <a href="#">Conference configuration &gt; Pre-configured endpoints</a> ).  For more information on pre-configuring endpoints see <a href="#">Cisco TelePresence Conductor Administrator Guide</a> .
<b>Content quality</b>	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	
Limit the conference duration (minutes)	<input type="checkbox"/> Maximum	
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	

Conference bridge type: TelePresence Server

- Click **Create conference template**.
- Click **View/Edit** for the Lecture - TS template.
- Click **Edit** under the **Advanced parameters** section.
- 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

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

Advanced parameters

Summary of configured parameters

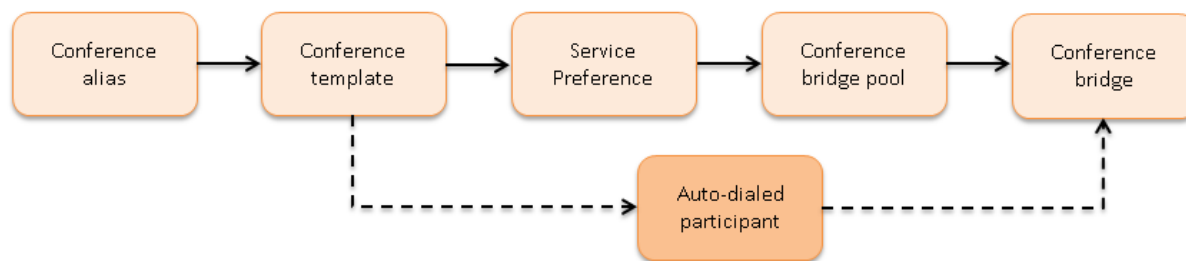
Parameter	Primary
pin	55105

Edit

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



## Task 22: 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:

<b>Name</b>	Enter a name for the auto-dialed participant, for example <b>Invite boss to meeting</b> .
<b>Conference template</b>	Select <i>HD Meeting</i> .
<b>Conference name match</b>	Enter <code>meet\.boss\.(HD SD)</code> .
<b>Address</b>	Enter <code>boss@&lt;SIP domain&gt;</code> .
<b>Protocol</b>	Select <i>SIP</i> .
<b>Role type</b>	Select <i>Participant</i> .
<b>Keep conference alive</b>	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 ⓘ

- Click **Create participant**.
- Click **View/Edit** for the 'Invite boss to meeting' auto-dialed participant.
- 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
<a href="#">HD Meeting</a>		Meeting

- Click **Save**.

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

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

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

**Auto-dialed participants**

**Modify participant**

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

- Click **Create participant**.
- Click **View/Edit** for the 'TCS - Recording device' auto-dialed participant.
- Click **Edit** under the **Advanced parameters** section.
- 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	<div>on</div> <div>value</div> <div> <input checked="" type="checkbox"/> true ⓘ </div>

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

**Advanced parameters**

**Summary of configured parameters**

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

[Edit](#)

- 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
<a href="#">Lecture</a>		Lecture

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

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

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

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

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

**Auto-dialed participants**

[Modify participant](#)

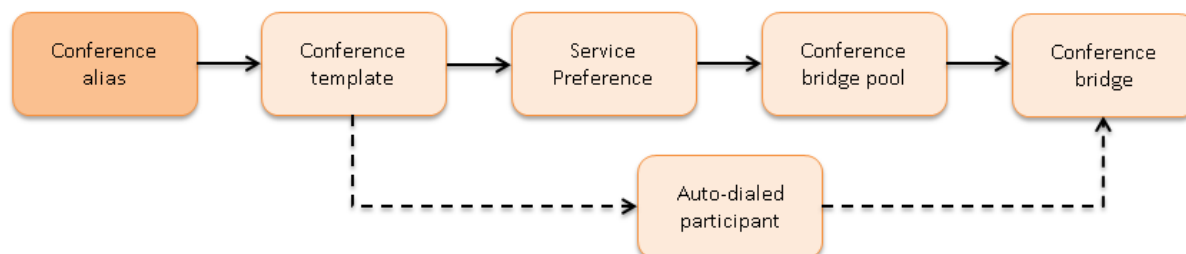
Name	* TCS - Recording device for TS
Description	
Conference template	* Lecture - TS  Conference bridge type: TelePresence Server
Conference name match (must use regex)	* (.*?)
Participant address	* TCSrecording@vcs.domain
Protocol	SIP
Role type	Guest
DTMF sequence	
Keep conference alive	No
Maximum quality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

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
<a href="#">Lecture - TS</a>		Lecture

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

## Task 23: 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:

<b>Name</b>	Enter a name for the alias, for example <b>HD Meeting</b> .
<b>Incoming alias</b>	Enter <b>(543. meet\..*\.\HD)@&lt;SIP domain&gt;</b> . This pattern will either match a numerical alias of 543 and any single digit or <b>meet.any_characters.HD@vcs.domain</b> .
<b>Conference name</b>	Enter <b>\1</b> .
<b>Priority</b>	Enter '25' for example.
<b>Conference template</b>	Select <b>HD Meeting</b> .
<b>Role type</b>	Select <b>Participant</b> .
<b>Allow conference to be created</b>	<p>Select <b>Yes</b> if participants dialing this alias should be able to create the conference. If you select <b>No</b>, you need to define another alias resulting in the same conference with <b>Allow conference to be created</b> set to <b>Yes</b> or the conference must be created via the TelePresence Conductor's API.</p> <p>See <a href="#">Appendix 2: Allow conference to be created [p.61]</a> for more information.</p>

**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 ⓘ Conference bridge type: TelePresence MCU
Role type	Participant ⓘ
Allow conference to be created	Yes ⓘ

- 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 \[p.45\]](#) to create a conference alias for the 'SD Meeting' hosted on TelePresence MCUs. Enter the same values for the fields, apart from:

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

<b>Name</b>	Enter a name for the alias, for example <i>HD TS Meeting</i> .
<b>Incoming alias</b>	Enter <i>(643. meetts\..*\.\HD)@&lt;SIP domain&gt;</i> . This pattern will either match a numerical alias of 643 and any single digit or <i>meetts.any_characters.HD @vcs.domain</i> .
<b>Conference name</b>	Enter <i>\1</i> .
<b>Priority</b>	Enter '30' for example.
<b>Conference template</b>	Select <i>HD TS Meeting</i> .
<b>Role type</b>	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 2: Allow conference to be created \[p.61\]](#) 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	★ \1 ⓘ
Priority	★ 30 ⓘ
Conference template	★ HD TS Meeting ⓘ Conference bridge type: TelePresence MCU
Role type	Participant ⓘ
Allow conference to be created	Yes ⓘ

Create conference alias Cancel

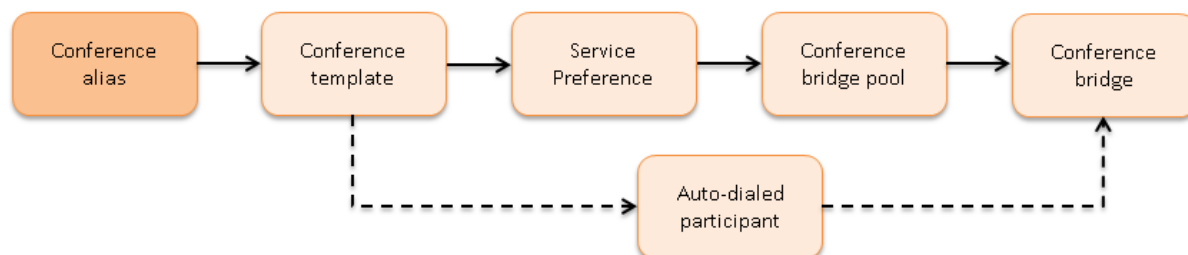
4. Click **Create conference alias**.

### 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 \[p.46\]](#) to create a conference alias for the 'SD Meeting' hosted on TelePresence Servers. Enter the same values for the fields, apart from:

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

## Task 24: 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:

<b>Name</b>	Enter a name for the alias, for example <b>Training - Teacher</b> .
<b>Incoming alias</b>	Enter <b>(231. teach\..*)@&lt;SIP domain&gt;</b> . This pattern will either match a numerical alias of 231 and any single digit or <b>teach.any_characters@vcs.domain</b> .
<b>Conference name</b>	Enter <b>training</b> .
<b>Priority</b>	Enter '10' for example.
<b>Conference template</b>	Select <i>Lecture</i> .
<b>Role type</b>	Select <i>Chairperson</i> .
<b>Allow conference to be created</b>	Select <b>Yes</b> if participants dialing this alias should be able to create the conference. If you select <b>No</b> , you need to define another alias resulting in the same conference with <b>Allow conference to be created</b> set to <b>Yes</b> or the conference must be created via the TelePresence Conductor's API.  See <a href="#">Appendix 2: Allow conference to be created [p.61]</a> for more information.

#### Conference aliases

Modify conference alias

Name

Description

Incoming alias (must use regex)

Conference name

Priority

Conference template

Role type

Allow conference to be created

★ Training - Teacher

★ (231.|teach\..\*)@vcs.domain

★ training

★ 10

★ Lecture

★ Chairperson

★ Yes

Conference bridge type: TelePresence MCU

Create conference alias

Cancel

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:

<b>Name</b>	Enter a name for the alias, for example <b>Training - Students</b> .
<b>Incoming alias</b>	Enter <b>(388. learn\..)@&lt;SIP domain&gt;</b> . This pattern will either match a numerical alias of 388 and any single digit or <b>learn.any_characters@vcs.domain</b> .
<b>Conference name</b>	Enter <b>Training</b> for example.
<b>Priority</b>	Enter '20' for example.
<b>Conference template</b>	Select <i>Lecture</i> .
<b>Role type</b>	Select <i>Guest</i> .
<b>Allow conference to be created</b>	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 <b>Allow conference to be created</b> set to Yes or the conference must be created via the TelePresence Conductor's API.  See <a href="#">Appendix 2: Allow conference to be created [p.61]</a> for more information.

**Conference aliases**

**Modify conference alias**

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

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:

<b>Name</b>	Enter a name for the alias, for example <b>Training - Teacher - TS</b> .
<b>Incoming alias</b>	Enter <b>(232. teachts\..*)@&lt;SIP domain&gt;</b> . This pattern will either match a numerical alias of 232 and any single digit or <b>teachts.any_characters@vcs.domain</b> .
<b>Conference name</b>	Enter <b>TrainingTS</b> for example.
<b>Priority</b>	Enter '15' for example.
<b>Conference template</b>	Select <b>Lecture - TS</b> .
<b>Role type</b>	Select <b>Chairperson</b> .
<b>Allow conference to be created</b>	Select <b>Yes</b> if participants dialing this alias should be able to create the conference. If you select <b>No</b> , you need to define another alias resulting in the same conference with <b>Allow conference to be created</b> set to <b>Yes</b> or the conference must be created via the TelePresence Conductor's API.  See <a href="#">Appendix 2: Allow conference to be created [p.61]</a> for more information.

### Conference aliases

**Modify conference alias**

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

4. Click **Create conference alias**.

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

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

<b>Name</b>	Enter a name for the alias, for example <b>Training - Students - TS</b> .
-------------	---

<b>Incoming alias</b>	Enter <code>(389. learnts\..)@&lt;SIP domain&gt;</code> . This pattern will either match a numerical alias of 389 and any single digit or <code>learnts.any_characters@vcs.domain</code> .
<b>Conference name</b>	Enter <code>TrainingTS</code> for example.
<b>Priority</b>	Enter '23' for example.
<b>Conference template</b>	Select <i>Lecture - TS</i> .
<b>Role type</b>	Select <i>Guest</i> .
<b>Allow conference to be created</b>	Select Yes 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 <b>Allow conference to be created</b> set to <i>Yes</i> or the conference must be created via the TelePresence Conductor's API.  See <a href="#">Appendix 2: Allow conference to be created [p.61]</a> for more information.

### Conference aliases

Modify conference alias

Name

★ Training - Students - TS ⓘ

Description

ⓘ

Incoming alias (must use regex)

★ (389.|learnts\..)@vcs.domain ⓘ

Conference name

★ TrainingTS ⓘ

Priority

★ 23 ⓘ

Conference template

★ Lecture - TS ⓘ Conference bridge type: TelePresence Server

Role type

Guest ⓘ

Allow conference to be created

Yes ⓘ

Create conference alias

Cancel

- 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 Back-to-back-user-agent (B2BUA).

## 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.

## 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 \[p.58\]](#).

## Auto-dialed participant not connected

If the auto-dialed participant does not get called:

1. On the TelePresence Conductor go to **Conference configuration > Auto-dialed participants** and verify that the settings for the auto-dialed participant are correct, specifically check that:
  - **Participant address** is correct.
  - **Conference name match** will match a valid conference.
  - **State** of the participant is *Enabled*.
2. On the TelePresence Conductor go to **Status > Logs > Event Log > All events** to check whether the TelePresence Conductor tried to call the auto-dialed participant.
3. On the TelePresence MCU, verify how the conference bridge will dial the auto-dialed participant and perform the relevant steps:

Method of dialing auto-dialed participant	Configuration to verify
SIP via Cisco VCS trunk	<p>On the TelePresence Conductor go to <b>Conference configuration &gt; Locations</b> and verify that</p> <ul style="list-style-type: none"> <li>• the <b>Conference type</b> is <i>Rendezvous</i> or <i>Both</i></li> <li>• the <b>SIP trunk settings for out-dial calls</b> are set correctly to route the auto-dialed participant back to Unified CM.</li> </ul> <p>On the TelePresence MCU go to <b>Settings &gt; SIP</b> and ensure the conference bridge is not registered to a SIP Proxy by having the <b>SIP registrar usage</b> field set to <i>Disabled</i>.</p>
SIP via a proxy	<p>On the TelePresence MCU</p> <ul style="list-style-type: none"> <li>• go to <b>Network &gt; Services</b> and verify that <b>SIP (TLS)</b> is ticked</li> <li>• go to <b>Settings &gt; SIP</b> and verify that the TelePresence MCU has the correct <b>SIP proxy address</b> defined and <b>Outgoing transport</b> set to <i>TLS</i></li> <li>• check that the TelePresence MCU is registered to the SIP proxy</li> <li>• check that the TelePresence MCU can make outbound calls via that proxy</li> </ul>



Method of dialing auto-dialed participant	Configuration to verify
H323 via a gatekeeper	<p>On the TelePresence MCU</p> <ul style="list-style-type: none"> <li>go to <b>Network &gt; Services</b> and verify that <b>Incoming H.323</b> is ticked</li> <li>go to <b>Settings &gt; H323</b> and verify that <ul style="list-style-type: none"> <li><b>H.323 gatekeeper usage</b> is <i>Enabled</i></li> <li><b>Gatekeeper address</b> contains the correct address</li> <li><b>H.323 ID to register</b> is correct</li> </ul> </li> <li>check that the TelePresence MCU is registered to the H323 gatekeeper</li> <li>check that the TelePresence MCU can make outbound calls via that gatekeeper</li> </ul>

- On the TelePresence Server go to **Configuration > SIP Settings** and verify that the **Outbound call configuration** is set to *Call direct*.

## 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:

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

## Encrypted calls drop on Cisco TelePresence System (CTS) Series endpoints

When a CTS is registered to a Unified CM, which has a SIP trunk to a Cisco VCS and this Cisco VCS is connected to a TelePresence Conductor via the TelePresence Conductor's B2BUA, calls will drop when the conference that the CTS dials into is hosted on a TelePresence Server and when the SIP session is refreshed. In this scenario we recommend that you create a SIP trunk directly from the Unified CM to the TelePresence Conductor as detailed in the [Optimized Conferencing for Cisco Unified Communications Manager Solution Guide](#).

## 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 `\12\2` will replace with 12th 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: 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.

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 23: Creating conference aliases for the Meeting-type conferences \[p.45\]](#) or [Task 24: Creating conference aliases for the Lecture-type templates \[p.47\]](#).
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 23: Creating conference aliases for the Meeting-type conferences \[p.45\]](#) or [Task 24: Creating conference aliases for the Lecture-type templates \[p.47\]](#).
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
03	August 2013	Updated for XC2.2.
02	August 2013	Corrected the zone profile setting on the Cisco VCS and added the description of an issue to the troubleshooting section.
01	May 2013	Initial release.

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