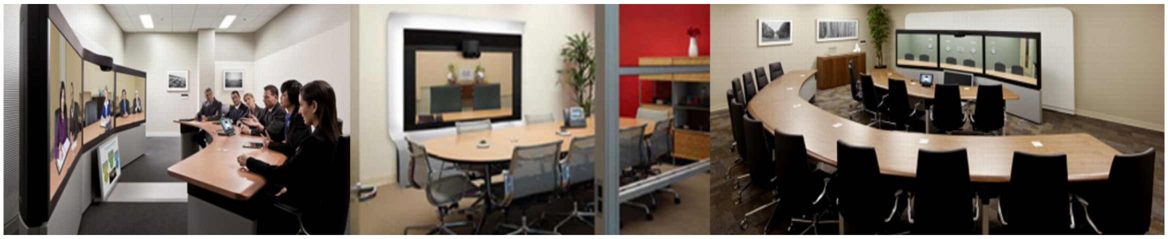


CHAPTER 9

Configuring Audio

Revised: November 2013

Contents

This chapter describes how to configure audio for Cisco WebEx Enabled TelePresence.

The following sections describe the audio deployment scenarios:

Prerequisites

To configure SIP or PSTN Audio, the following are required:

- VCS Control/Expressway must be configured.

  For details, refer to: Chapter 4, "Configuring Cisco TelePresence Video Communication Server Control and Expressway".
- When using Unified CM, make sure:
  - SIP trunk is configured between Unified CM and VCS Control.

  For details, see Configuring a SIP Trunk Between Unified CM and VCS Control, page 4-5
  - Your regions are configured for g.711.
- If configuring PSTN audio, Gateway must be registered to VCS or Unified CM.
- MCUs/TelePresence Servers must be registered to VCS.
  - No support for MCUs/TelePresence Servers trunked to Unified CM.
- Endpoints registered to VCS and/or Unified CM and able to call into MCUs/TelePresence Servers
- Familiarity with all of required products

- If configuring TSP audio and the TSP provider offers a waiting room feature, the TSP provider must configure it to allow multiple hosts to log in to the audio conference, or the human host must be trained to not log in as a host. If multiple hosts are not enabled, each host that dials in disconnects the host that dialed in before it. For example, if the MCU dials in first, when the human host dials in later, they will disconnect the MCU.

  The human host still maintains host privileges on the WebEx client and can mute/unmute participants through that user interface if needed.

Note Cisco Conductor is not supported at this time.

Configuring SIP Audio for Cisco WebEx Enabled TelePresence

The following section describes the steps required for configuring SIP audio for Cisco WebEx Enabled TelePresence.

This section describes the following:

- Configuring the WebEx Site in Cisco TMS to Use SIP Audio
- Enabling Hybrid Audio on the WebEx Site, page 9-3

Note SIP audio only supports WebEx audio (TSP audio is not supported).

Configuring the WebEx Site in Cisco TMS to Use SIP Audio

To configure Cisco TMS to use SIP for the WebEx site, do the following:

Step 1 Log into Cisco TMS.

Step 2 Go to **Administrative Tools** > **Configuration** > **WebEx Settings**.

The WebEx Settings page appears.

Step 3 Click the name of the WebEx site you want to configure.

The WebEx Site Configuration page appears.

Step 4 If a new site, enter the Site Name, Host Name and other required fields.

Step 5 For TSP Audio, select **No**.

Step 6 Click **Save**.

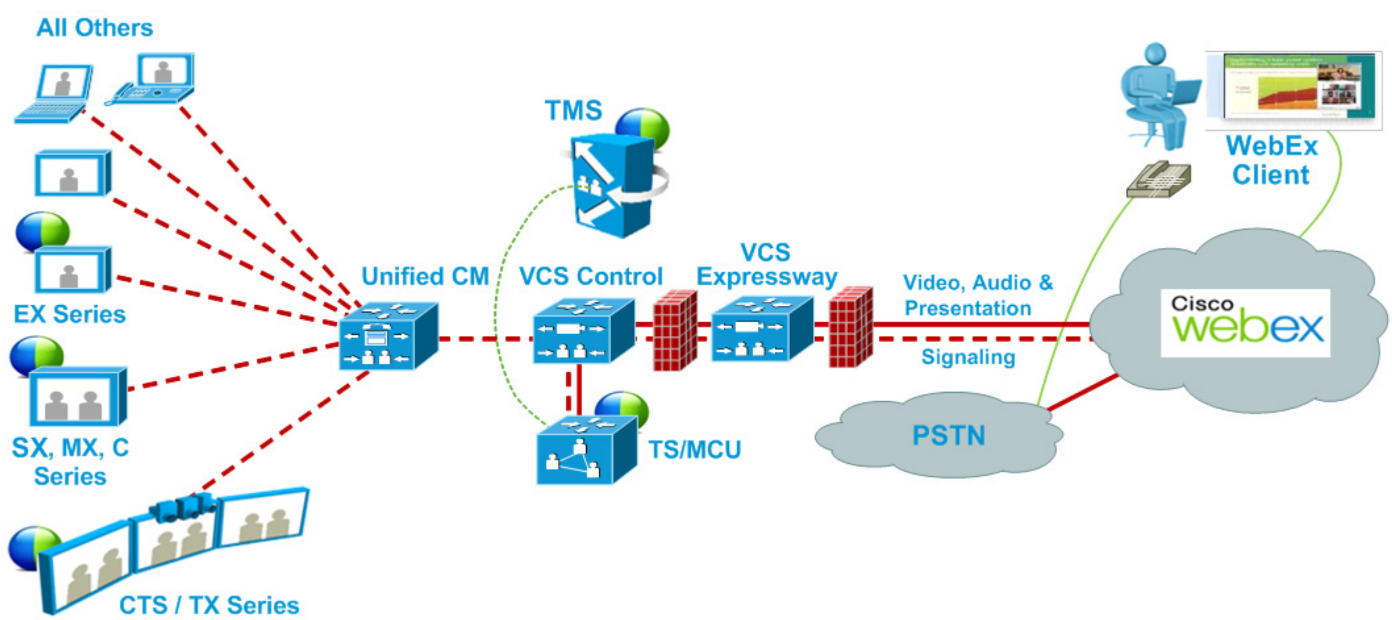
Enabling Hybrid Audio on the WebEx Site

To use SIP audio, your WebEx site must be enabled for **Hybrid Audio**. Hybrid Audio is also required to provide your WebEx participants the option of using their computer to connect to the audio portion of a meeting.

This configuration must be done by the WebEx team. Contact the WebEx team for assistance, or submit an online ticket at:

https://support.webex.com/MyAccountWeb/GPLWebForm.do

Figure 9-1 SIP Audio Deployment with Endpoints Registered to Unified CM



Configuring PSTN Audio for Cisco WebEx Enabled TelePresence

The following section describes the steps required for configuring PSTN audio for Cisco WebEx Enabled TelePresence.

This section describes the following:

- Configuring the WebEx Site in Cisco TMS to Use PSTN Audio
- Enabling Hybrid Mode on the WebEx Site, page 9-4
- Configuring PSTN Calls to Pass Through a PSTN Gateway to WebEx, page 9-4

Note Cisco WebEx Enabled TelePresence always dials a fully qualified E.164 number beginning with the international escape character (+). For example: +14085551212. Make sure that VCS and/or Unified CM call routing is set up accordingly.

Configuring the WebEx Site in Cisco TMS to Use PSTN Audio

To configure Cisco TMS to use PSTN for the WebEx site, do the following:

Step 1 Log into Cisco TMS.

Step 2 Go to **Administrative Tools** > **Configuration** > **WebEx Settings**.

The WebEx Settings page appears.

Step 3 Click the name of the WebEx site you want to configure.

The WebEx Site Configuration page appears.

Step 4 If a new site, enter the Site Name, Host Name and other required fields.

Step 5 For TSP Audio, select **Yes**.

Step 6 Click **Save**.

Caution If the meeting organizer chooses a TelePresence Server when scheduling the meeting, Cisco TMS will automatically attempt to schedule the meeting using MCU. If an MCU is not available, the meeting will not be scheduled successfully.

Enabling Hybrid Mode on the WebEx Site

If you want WebEx participants to have the option of using their computer to join the audio portion of a meeting, your WebEx site must be set to **Hybrid** mode. This configuration must be done by the WebEx team. Contact the WebEx team for assistance.

Configuring PSTN Calls to Pass Through a PSTN Gateway to WebEx

WebEx always provides a fully qualified E.164 number beginning with the international escape character (+). For example: +14085551212. VCS and/or Unified CM call routing must be properly configured to ensure PSTN calls are routed correctly.

Two deployments models are supported for routing PSTN calls to pass through a PSTN gateway to WebEx:

- Configuring PSTN Calls to Pass through a PSTN Gateway Registered to VCS, page 9-4
- Configuring PSTN Calls to Pass through a PSTN Gateway Registered to Unified CM, page 9-6

Configuring PSTN Calls to Pass through a PSTN Gateway Registered to VCS

To configure PSTN calls to pass through a PSTN Gateway registered to VCS, do the following:

Step 1 On VCS, create a transform or search rule that transforms the globally routable number provided by WebEx (example: +14085551212) to a number with the tech-prefix of the gateway registered to VCS (example: 9#14085551212).

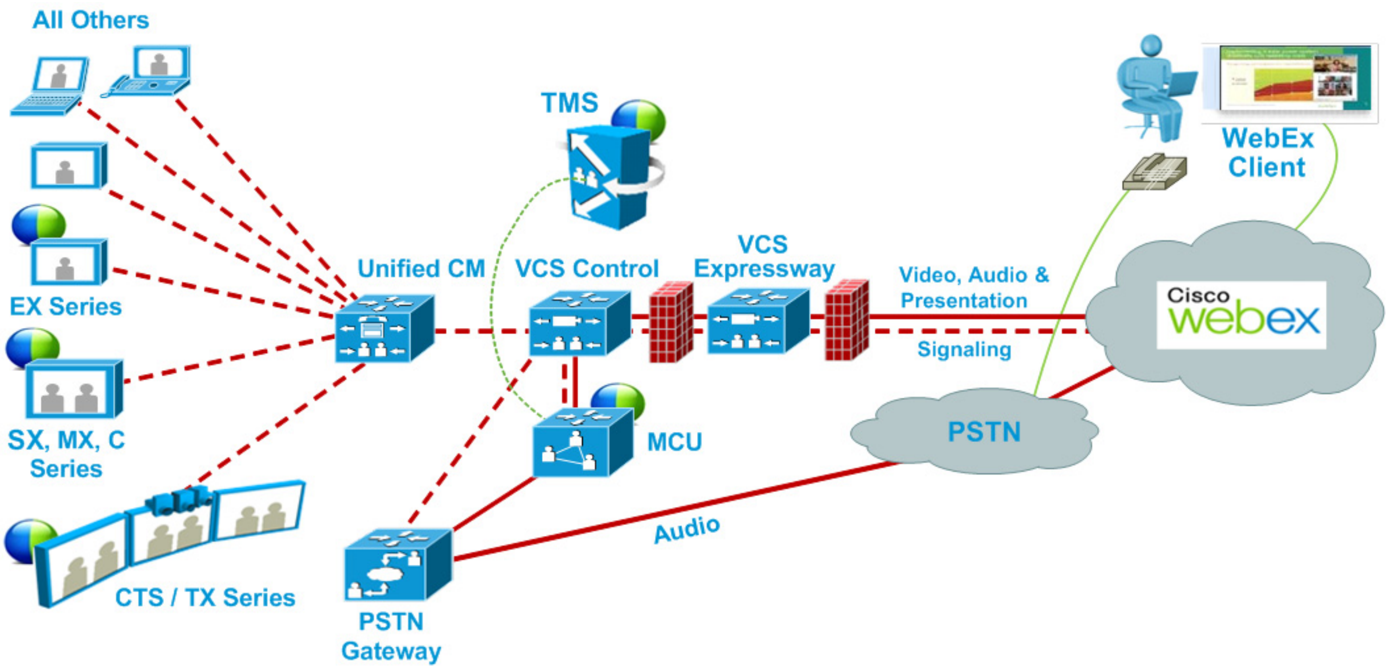
This example transforms **+14085551212@example.webex.com** to **9#14085551212@example.webex.com** using the Regex pattern type:

- Pattern string:**\+(\d+@.*)**
- Replace string: **9#\1**

For more information about configuring traversal zones, search rules and transforms in VCS, refer to the "Cisco TelePresence Video Communication Server Basic Configuration (Control with Expressway) Deployment Guide" at:

https://www.cisco.com/en/US/docs/telepresence/infrastructure/vcs/config_guide/Cisco_VCS_Basic_Configuration_Cisco_VCS_Control_with_Cisco_VCS_Expressway_Deployment_Guide_X7-2.pdf

Figure 9-2 PSTN Audio Deployment with Gateway Registered to VCS and Endpoints Registered to Unified CM



Configuring VCS Control for ISDN Gateways

If you are going to use an ISDN gateway to pass PSTN calls through to WebEx, you must configure the Interworking setting in VCS Control.

Note This step is required only for ISDN gateways.

To configure VCS Control for ISDN Gateways, do the following:

Step 1 Log in to VCS Control.

Step 2 Go to **VCS Configuration** > **Protocols** > **Interworking**.

Step 3 For H.323 <-> SIP interworking mode select **On** and click **Save**.

Note An option key is required in order to save this configuration.

Configuring PSTN Calls to Pass through a PSTN Gateway Registered to Unified CM

To configure PSTN calls to pass through a PSTN Gateway registered to Unified CM, do the following:

Step 1 On VCS, create a search rule that takes the globally routable number with the international escape character (+) provided by WebEx (example: +14085551212) and routes it to Unified CM.

Step 2 On Unified CM, create a route pattern according to your dial plan to route these types of calls to the appropriate PSTN gateway registered to Unified CM.

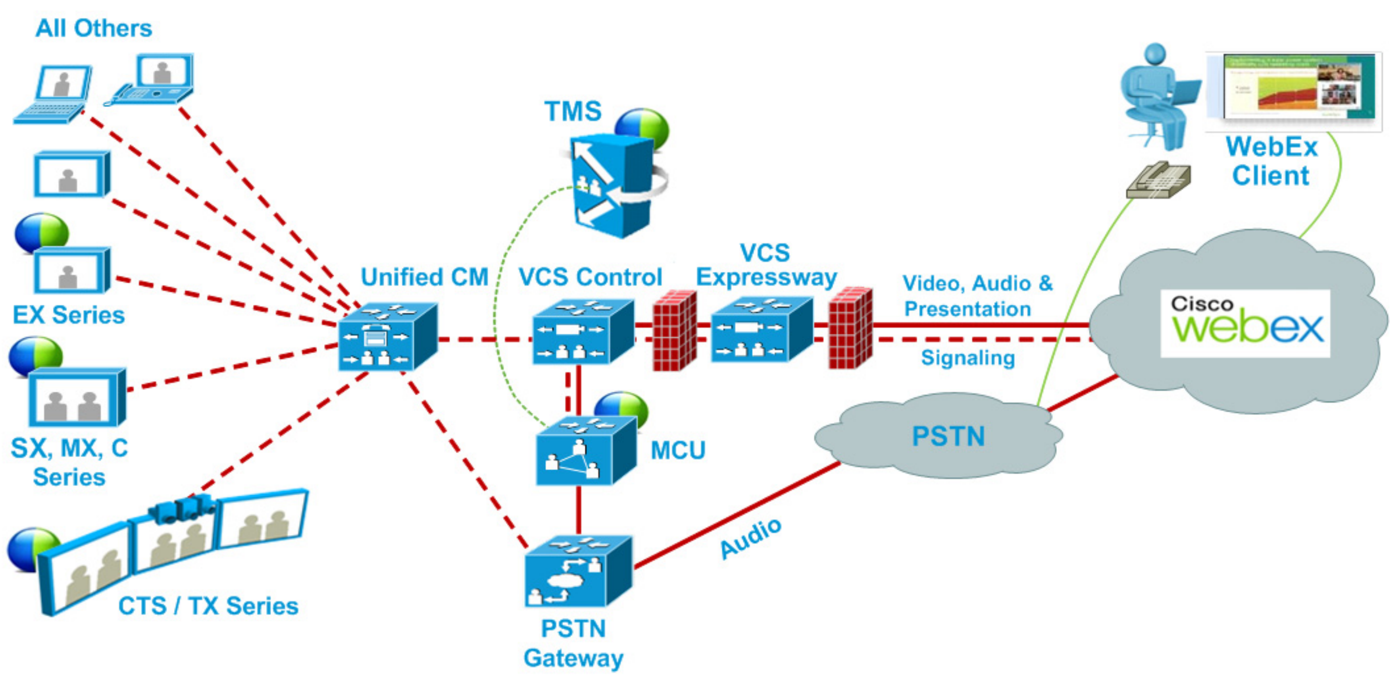
For more information about configuring search rules on VCS, refer to the "Cisco TelePresence Video Communication Server Basic Configuration (Control with Expressway) Deployment Guide" at:

https://www.cisco.com/en/US/docs/telepresence/infrastructure/vcs/config_guide/Cisco_VCS_Basic_Configuration_Cisco_VCS_Control_with_Cisco_VCS_Expressway_Deployment_Guide_X7-2.pdf

For more information about configuring route patterns in Unified CM, refer to the documentation for your Unified CM version:

https://www.cisco.com/en/US/products/sw/voicesw/ps556/prod_maintenance_guides_list.html

Figure 9-3 PSTN Audio Deployment with Gateway and Endpoints Registered to Unified CM



Configuring VCS Control for ISDN Gateways

If you are going to use an ISDN gateway to pass PSTN calls through to WebEx, you must configure the Interworking setting in VCS Control.

Note This step is required only for ISDN gateways.

To configure VCS Control for ISDN Gateways, do the following:

Step 1 Log in to VCS Control.

Step 2 Go to **VCS Configuration** > **Protocols** > **Interworking**.

Step 3 For H.323 <-> SIP interworking mode select **On** and click **Save**.

Note An option key is required in order to save this configuration.

Verifying the Outbound Dialing Configuration for VCS and MCU/TelePresence Server

To verify that outbound dialing is properly configured, do the following:

Step 1 Immediately after call is placed, on the VCS Control navigate to Status -> Search history

Step 2 Determine if the call appears in the search history:

a. If the call does not appear here, then the MCU never dialed out. Enable SIP/H323 logs on the MCU. Replace the call, stop SIP/H323 logging, and download the logs.

b. If the call does appear here, click on view for the call under the actions header. This will display the detailed search history.

Step 3 Within the detailed search history, in the first subsearch it should show transforms. The value listed below here is the exact URI that you are calling after the transforms take effect. Later on, you should do a subsearch that points to the Zone for the external dial-out (usually Unified CM). The alias listed here is exactly as you are presenting the call to the other side. Make sure that the other side is expecting the call in this formation (ie: no "+" characters or anything the other side might not support).

Step 4 If the search on the other side shows "Found: False" look at the Reason. If the Reason is Not Found, then the other side is returning a 404. In this case, make sure that the following are taking place:

a. The VCS is passing the URI EXACTLY as the other side is expecting

b. The other side is configured to allow the call.

Configuring TSP Audio for Cisco WebEx Enabled TelePresence

To deploy Telephony Service Provider (TSP) audio, PSTN audio is required. Follow the steps in Configuring PSTN Audio for Cisco WebEx Enabled TelePresence and then contact WebEx cloud services to assist you with the TSP configuration.

Note The TSP provider must support the Call-in User Merge. Call-in User Merge allows TSP partners to pass the attendee ID via DTMF code, rather than prompting the user via the audio. The WebEx Meeting Manager prompts the user to enter the DTMF code, followed by the attendee ID.

There are four required parts to TSP audio configuration:

- Configuring MACC Domain Index and Open TSP Meeting Room Webex Settings
- Configuring the TSP Dial String
- Configuring How the Conference is Opened
- Configuring TSP Audio for the Meeting Organizer

For more information, refer to:

- Overview of TSP Audio Configuration and Meetings

Note TSP audio requires that the MCU/TS is able to make an outbound call to establish an audio cascade between TelePresence and the TSP partner audio bridge. To ensure that the MCU/TS can make the call, please review the section: Configuring PSTN Calls to Pass through a PSTN Gateway Registered to VCS, page 9-4.

Configuring MACC Domain Index and Open TSP Meeting Room Webex Settings

WebEx cloud services must configure these settings for you. Contact WebEx cloud services for more information.

Configuring the TSP Dial String

During a meeting that uses TSP audio, TelePresence equipment dials into the TSP partner's bridge and navigates the menu hierarchy to connect to the conference. The audio (IVR) prompts for each TSP provider are different. As a result, a DTMF dial string must be created.

DTMF Dial String

A static DTMF dial string must be created and tested by your TSP audio provider and then provided to Cisco WebEx cloud services. WebEx cloud services then configures the dial string parameters in the WebEx cloud for your WebEx site. The following is an example sequence of what needs to be provided:

1. MCU/TelePresence Server dials the phone number
2. Pause 2 seconds
3. Enter [participant code] DTMF values (Example: 12345678)
4. Enter #
5. Pause 6 seconds
6. Enter #
7. Pause 25 seconds
8. Enter #1

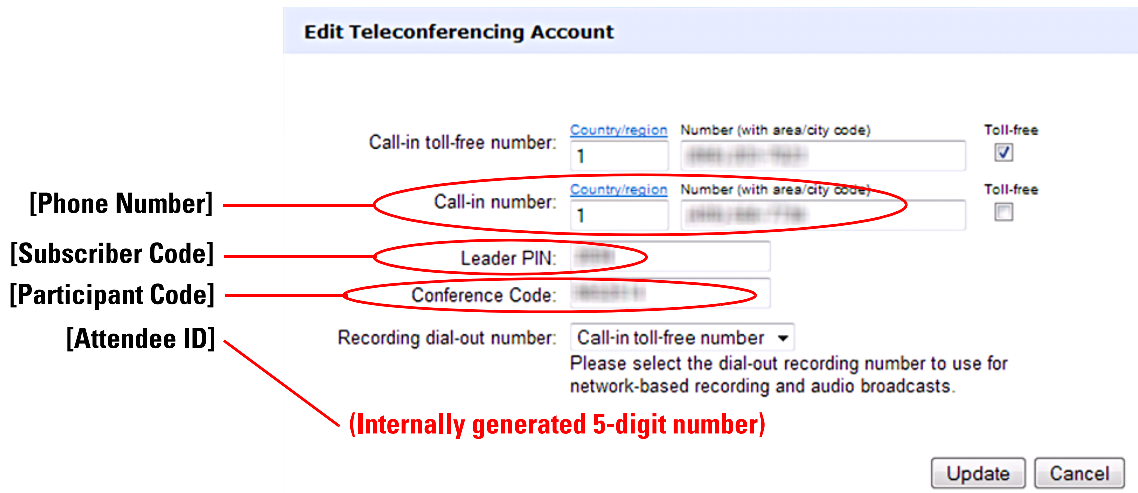
9. Pause 1 second
10. Enter [attendee ID] DTMF values (Example: 44356)

For more information, contact Cisco WebEx cloud services.

Variables Available to the Dial String

The following variables are available for use with the DTMF dial string that is created by your TSP audio provider and configured by WebEx cloud services.

Figure 9-4 WebEx Host Account / TSP Audio Account



Configuring How the Conference is Opened

TSP providers typically wait for the WebEx host to call in before opening up the conference.

Until the host dials in (by entering the host key) participants are in a waiting room. If the host is late or never dials in and unlocks via WebEx, the meeting will never get unlocked.

Contact your TSP provider to determine if they have a waiting room. If they do have a waiting room, there are two methods for ensuring the conference is opened for a meeting:

- **Method 1**: Configure the DTMF dial string for the MCU/TelePresence Server to enter the meeting as the host and unlock the meeting.
  - WebEx cloud services works with the TSP partner to create the proper DTMF dial string.
  - If the WebEx host has already entered meeting, the DTMF dial string of the MCU/TelePresence Server will be heard by meeting participants.

Note A DTMF dial string is required, whether or not you are configuring the dial string for the MCU/TelePresence Server to enter the meeting as host. Contact WebEx cloud services for more information.

- **Method 2**: The WebEx TSP server sends the **W2A_UpdateConference=2** API command to the TSP partner's bridge to unlock the meeting.

– The TSP partner may have to recode their TSP adapter in order to recognize and properly execute the unlock conference command. Contact your TSP provider to determine if they support this API command.

How TSP Integration Methods Affect Call Scenarios

The following table describes common scenarios and the results depending on which method is used to open the conference.

Table 9-1 Scenarios and Results for TSP Methods

Scenario	Expected Result	If method 1 is used	If method 2 is used
MCU/TelePresence Server is the first caller into the audio conference	Successful join	The MCU/TelePresence Server will have host role in the TSP audio conference	The MCU/TelePresence Server will not have the host role in audio.
One or more attendees have already joined the audio conference (waiting room) before the MCU/TelePresence Server dials in.	Successful join	The MCU/TelePresence Server will have host role in the TSP audio conference	The MCU/TelePresence Server will not have the host role in audio.
The host has already joined the audio conference before the MCU/TelePresence Server dials in.	Successful join	Users who have already joined the audio conference may hear the “extra” DTMF tones broadcast into the audio conference, which is the MCU/TelePresence Server following the DTMF sequence as though it were the host.	No such extra DTMF tones will be heard.
The host (who had already joined the audio conference before the MCU/TelePresence Server dials in), hangs up while the conference is still underway.	Varies	Audio conference may terminate. Depends on TSP implementation - some may not terminate. Depends on host's selection in WebEx GUI upon leaving conference (option to keep conference running)	Since method 2 is being used, the partner should keep the conference running until either: **a.** all attendees have left the conference or **b.** TSP API sends W2A_CloseConference

Scenario	Expected Result	If method 1 is used	If method 2 is used
DTMF failure	Fail to join		
The host joins via WebEx before the MCU/TelePresence Server dials in, and the host uses the WebEx GUI to lock the conference. (WebEx has decided to respect the hosts' locking of the conference in this case.)	Fail to join	MCU should fail to join	MCU/TelePresence Server should fail to join

Configuring TSP Audio for the Meeting Organizer

Each meeting organizer who needs to schedule WebEx Enabled TelePresence meetings that use TSP audio, must log in to the WebEx site and configure their account to use TSP audio. This is a one-time configuration.

Prerequisites

The meeting organizer must have the following information, provided by the TSP audio service provider:

- Call-in toll-free number
- Call-in number
- Host access code
- Attendee access code

Configuring TSP Audio

To configure TSP audio, do the following:

Step 1 Open a browser and go to your WebEx site. (Example: *http://example.webex.com*)

Step 2 In the upper part of the page, click **My WebEx**. 

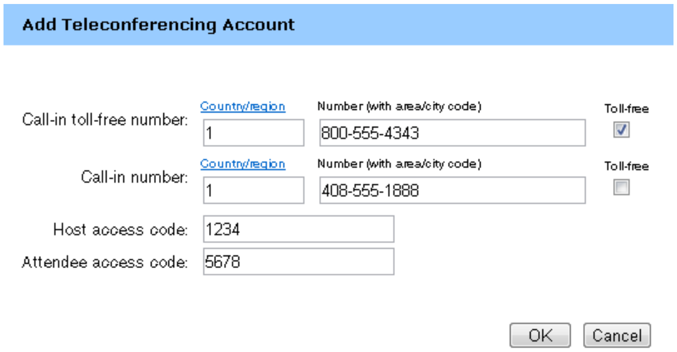
Step 3 Enter the **Username** and **Password** for your WebEx account and click **Log In**.

Step 4 In the left-hand side of the page, click **My Audio**. 

Step 5 In the Teleconferencing Service Accounts section, click **Add account**.

Step 6 In the Add Teleconferencing Account window, enter the appropriate phone numbers and access codes for the host and attendees, as provided by the TSP audio service provider.

Figure 9-5 Add Teleconferencing Account window

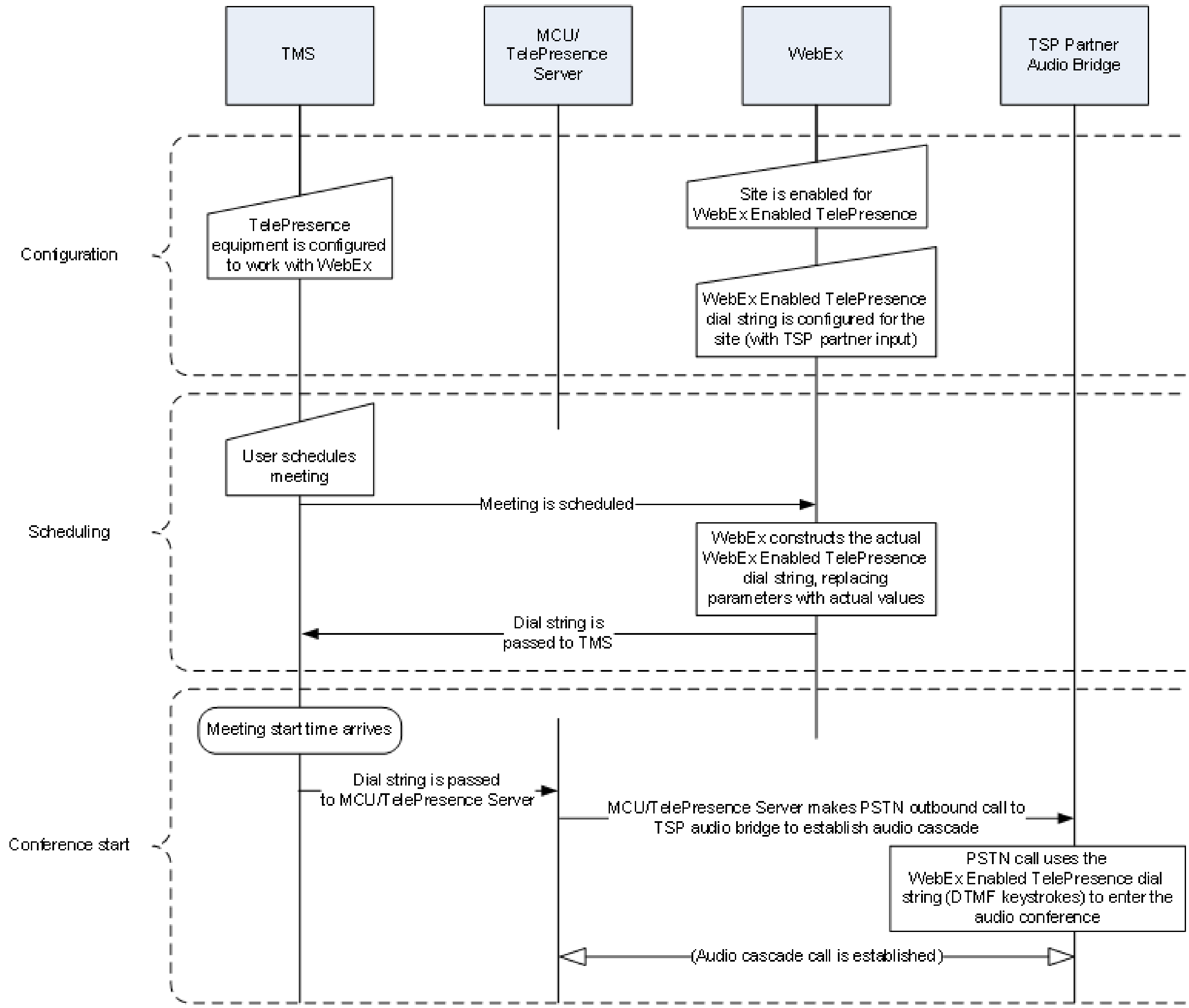


Step 7 Click **OK**.

Overview of TSP Audio Configuration and Meetings

The following diagram provides an overview of which components are configured for TSP audio, as well as what takes place when a meeting is scheduled and starts.

Figure 9-6 TSP Audio Configuration, Scheduling and Meeting Start Flow



How a TSP Meeting Works

A meeting that uses TSP Audio takes place the following way:

1. The meeting is scheduled.
2. A dial string is passed back to the MCU/TelePresence Server.
3. At the scheduled start time, the MCU/TelePresence Server starts the meeting.
4. TelePresence connects into WebEx via SIP.
5. The TSP partner starts the audio conference on their bridge and they open up the conference.
6. At the same time as TelePresence connects to WebEx via SIP, it also dials via PSTN into the TSP partner bridge using the DTMF dial string.

Behavior of TSP Audio Meetings When the MCU or TelePresence Server Dials in as Host

The MCU/TelePresence Server will attempt to redial the connection for any reason up to a maximum number of retries. In the case where the MCU/TelePresence Server joins as host, it is important to note that if the MCU/TelePresence Server is the host and this call is disconnected for any reason, the TSP partner may tear down the audio conference (all participants may be disconnected). The MCU/TelePresence Server will immediately dial back in and re-establish the audio conference, but the participants may need to call back in again. The word “may” is used here because we understand this to be configurable on the TSP and/or the behavior may differ from one TSP provider to another.