Release Notes for the Cisco TelePresence Exchange System Release 1.1

Revised February 21, 2013

These release notes describe the new features and caveats of the Cisco TelePresence Exchange System release 1.1. For lists of open and resolved caveats that are pertinent to this release, see the "Caveats" section on page 20.

Contents

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Introduction

The Cisco TelePresence Exchange System is an integrated video service-creation platform that enables service providers and strategic partners to offer secure cloud-based managed and hosted Cisco TelePresence and business video services. The Cisco TelePresence Exchange System is a software environment that provides the following benefits:

- Simplifies end-to-end subscriber service provisioning
- Optimizes intelligent call routing for endpoints and network bandwidth
- Manages the call processing and allocation of media resources for conferencing
- Consolidates a centralized control point for management, billing, and administration
- Presents an open application programming interface (API) for application integration such as scheduling, management of active meetings, and billing

Based on proven technology and powered by a fully redundant and horizontally scalable architecture, it delivers an open, scalable, and robust multi-tenant solution that can grow in scale and functions based on service needs. As a result, it accelerates time to market by simplifying the process of new services production and promotes service innovation through APIs that support service customizing.

For more details on the Cisco TelePresence Exchange System and its supported features and functionality, see the "Product Overview" chapter of the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*, at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/1_1/install_admin/book/b_install_admin.html.

System Requirements

For information on the required hardware and minimum software releases that the Cisco TelePresence Exchange System solution requires, see the *System Requirements and Compatibility Matrix for the Cisco TelePresence Exchange System* at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/compatibility/matrix/ctxmatrix.html.

Related Documentation

For more information about the Cisco TelePresence Exchange System, see the following documentation:

- *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*, at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/1_1/install_admin/book/b_install_admin.html
- *API User Guide for the Cisco TelePresence Exchange System Release 1.1*, at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/1_1/api_guide/api_guide_11.html

To access the documentation suite for the Cisco TelePresence Exchange System, go to the following URL: http://www.cisco.com/go/ctx-docs.

For more information about the Cisco TelePresence Exchange System solution, see the following documentation:

- Cisco TelePresence Manager documentation, at http://www.cisco.com/en/US/products/ps7074/tsd_products_support_series_home.html

- Cisco TelePresence Multipoint Switch, at http://www.cisco.com/en/US/products/ps7315/tsd_products_support_series_home.html
- Cisco TelePresence MSE 8000 Series documentation, at http://www.cisco.com/en/US/products/ps11340/tsd_products_support_series_home.html
- Cisco Catalyst 4900 Series Switches documentation, at http://www.cisco.com/en/US/products/ps6021/tsd_products_support_series_home.html
- Cisco ASR 1000 Series Aggregation Services Router documentation, at http://www.cisco.com/en/US/products/ps9343/tsd_products_support_series_home.html
- *Cisco Application Control Engine (ACE) Quick Start Guide*, *Cisco ACE Server Load Balancing Configuration Guide,* and *Cisco ACE 4700 Series Appliance Administration Guide*, at http://www.cisco.com/en/US/products/ps7027/tsd_products_support_series_home.html
- *Cisco Unified Communications Manager Security Guide, Release 7.1(2)*, at http://www.cisco.com/en/US/partner/products/sw/voicesw/ps556/prod_maintenance_guides_list.html

Installation and Upgrade Notes

Installing Cisco TelePresence Exchange System Release 1.1 for the First Time

For information related to installing the 1.1 version of the Cisco TelePresence Exchange System, see the "Preparing for Installation" chapter of the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*, at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/1_1/install_admin/book/b_install_admin.html.

Upgrading to Cisco TelePresence Exchange System Release 1.1

The upgrade feature allows you to perform and monitor the progress of the actual upgrade for all six nodes in the server cluster by using the user interface of the Cisco TelePresence Exchange System Administrative Console.

For information related to upgrading to the 1.1 version of the Cisco TelePresence Exchange System, see the "Upgrading the Software" chapter of the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*, at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/1_1/install_admin/book/b_install_admin.html.

For information about how the Cisco TelePresence Exchange System migrates data from Release 1.0 to Release 1.1, and how the administration console graphical user interface has changed between the two releases, see the "Data Migration" appendix of the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*.

Important Notes

The following sections highlight items that might affect full operation of the Cisco TelePresence Exchange System:

Do Not Change the Region and Service Provider of a Resource that Has Active Meetings

When there are active meetings on a resource, do not change the region and service provider of the resource, either directly or indirectly (by modifying the resource group of the resource).

Enabling Cisco TelePresence Endpoints Running TC Release 5.x to Join Meetings Hosted on the Cisco TelePresence Multipoint Switch

Cisco TelePresence endpoints running TC release 5.x require a configuration change on Cisco TelePresence Multipoint Switch version 1.8 in order to join meetings hosted on the switch. For new installations, the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1* includes the procedure to make the configuration change when you configure the Cisco TelePresence Multipoint Switch to work with the Cisco TelePresence Exchange System.

If you have already configured a Cisco TelePresence Multipoint Switch to work with the Cisco TelePresence Exchange System, do the following additional procedure to enable Cisco TelePresence TC5.x endpoints to join meetings hosted on the Cisco TelePresence Multipoint Switch:

Procedure

Step 1 In CTMS Administration, from the left navigation pane, choose **Manage > Default Meeting Settings**.

The Default Settings window displays.

Step 2 For Supported Endpoint Types, select **Cisco TelePresence TC 5.0 (and later) and CTS 1.8 (and later) endpoints**.

Step 3 To save the modified setting, click **Apply**.

Licensing

The Cisco TelePresence Exchange System requires the installation of a license to enable Meet-Me and direct-dial services. The system checks the license before scheduling a meeting or initiating a Meet-Me or direct-dial call. The system blocks these operations if a valid license is not detected.

The Cisco TelePresence Exchange System comes preinstalled with a 30-day evaluation license. After 30 days, you must install a permanent license to continue to use the Meet-Me and direct-dial services. The permanent license is perpetual, meaning that it does not expire and does not need to be renewed.

The license is locked to the call engine servers. If you replace a call engine server, you need to request a new license file for the replacement server.

Meet-Me Meeting Scheduling Start-Time Considerations

For Meet-Me meetings, which have a pre-configured start and end time, the Cisco TelePresence Exchange System reserves resources based on fifteen-minute increments on the hour (for example, 9:00, 9:15, 9:30, or 9:45). When scheduling these types of meetings, choose a start and end time consistent with these time periods. For example, choose a start time of 9:30 am or 9:45 am rather than 9:40 am.

Special Considerations for Interoperability with the Cisco TelePresence Manager

To ensure proper interoperability between the Cisco TelePresence Manager and the Cisco TelePresence Exchange System, a Cisco support engineer must perform an additional configuration to enable the API on the Cisco TelePresence Manager during system installation. To arrange for this support, contact your local Cisco system engineer or file a support case at Cisco.com.

Be aware that if the necessary configuration is not done, the Cisco TelePresence Exchange System might fail to authenticate with the Cisco TelePresence Manager or might report the following API exception value and cause code: ERC_CTSMAN_COMMUNICATION_FAILURE (exception value), CTSMAN_INTERCOMPANY_NOT_CONFIGURED (cause code).

Limitations and Restrictions

The following sections detail limitations and restrictions related to the use of and interoperability with the Cisco TelePresence Exchange System:

Administration Console Access During Database Failover

In the event of a database failure, the administration console may take up to two minutes to respond after database failover.

IVR Prompt Length Restrictions and Recommendations

Added February 7, 2013

In response to requests from the Cisco TelePresence Exchange System, the IVR router plays most of the IVR prompts in their entirety, regardless of the length of the prompt recording. However, some prompts are subject to a maximum system-defined length, either alone or in combination with other prompts that are played after them. Any audio in the recording that exceeds the system-defined length is cut off.

To avoid having prompt audio cut off, we recommend that the recordings for these prompts not exceed certain lengths, as listed in Table 1.

Table 1 IVR Prompt Length Restrictions

Prompt	Recommended Length	Scenario
Valid Meeting Prompt	3 seconds	When a caller enters a valid meeting ID, the Valid Meeting Prompt plays for up to 3 seconds, then the system initiates a transfer to the meeting.
Helpdesk Prompt	3 seconds	When a caller enters * in the IVR menu, the Helpdesk prompt plays[1] and the system initiates a transfer to the service provider help desk route.
Invalid Meeting Prompt	4 seconds	When a caller enters three incorrect meeting IDs, the Invalid Meeting Prompt plays, followed by the Helpdesk prompt. After 7 seconds, the prompt that is currently playing is cut off, and the system initiates a transfer to the service provider help desk route.
Incorrect Host PIN Prompt	4 seconds	When a caller enters three incorrect host PINs, the Invalid Meeting Prompt plays, followed by the Helpdesk prompt. After 7 seconds, the prompt that is currently playing is cut off, and the system initiates a transfer to the service provider help desk route.

1. Although in this scenario the Helpdesk prompt plays for up to 7 seconds, we recommend no more than 3 seconds for this recording so that it is not cut off in other scenarios.

The Enter Host PIN prompt is played repeatedly until the host joins the meeting or the meeting times out. There is no length restriction on this prompt. However, because the system waits until the prompt plays completely before it checks whether the host has joined the meeting, an excessively long prompt may affect the customer experience. The recommended length is 20-30 seconds, which may include silence.

Meeting Scheduling Race Condition Limitation

If you attempt to schedule two meetings at the same time for the same time slot and they both require immediate allocation, you will receive an error message, and must try again. This will only happen for two Rendezvous guaranteed meetings or two Meet-Me guaranteed meetings scheduled within 5 minutes of the start time.

Presentation Viewing Limitation on Cisco Cius and Cisco Unified IP Phone 9971/9951

Participants who use a Cisco Cius, Cisco Unified IP Phone 9971, or Cisco Unified IP Phone 9951 to attend a Meet-Me meeting hosted on a Cisco TelePresence MCU MSE 8510 bridge are sometimes unable to view presentations shared by remote endpoints.

When scheduling a Meet-Me meeting, you can work around this issue by selecting a Custom Screen Layout option other than the default (single-screen) layout.

If you are seeing this issue when the meeting is active, you can use Active Meeting Management to modify the Custom Screen Layout to use any layout other than the single screen layout. See the "Active Meeting Management" section on page 9 for more information.

Resource Group Configuration Limitations

When configuring resource groups and regions, adhere to the following rules:

- Within a region, all media bridge resources that are reserved for large meetings should be associated with the same resource group.
- Within a region, all media bridge resources that are not reserved for large meetings should be associated with the same resource group.
- Media bridge resources that are reserved for large meetings and media bridge resources that are not reserved for large meetings should never be associated with the same resource group.
- A region should be configured with a maximum of one resource group that is associated with only media bridge resources that are reserved for large meetings.
- A region should be configured with a maximum of one resource group that is associated with only media bridge resources that are not reserved for large meetings.

In practice, the above rules mean that a region may be configured with up to two resource groups, where one of the resource groups is associated with only media bridge resources that are reserved for large meetings and the other resource group is associated with only media bridge resources that are not reserved for large meetings. If any of the rules are violated, meetings hosted in this region may fail at attend time. For more information, use Bug Toolkit to see CSCua53934.

Solution Caveats and Endpoint Limitations

Note the following caveats and endpoint limitations that may affect the operation or functionality of the Cisco TelePresence Exchange System solution:

Table 2 Solution Caveats

Product	Identifier	Headline	Fixed In Version
Cisco TelePresence Server MSE 8710 (TPS)	CSCtz01891	Calls dropped after eng failover as TPS sends Re-Invite to failed Engine	2.3.x[1]

1. At time of release, this TPS version was not supported for use with Release 1.1. For up-to-date compatibility information, see the *System Requirements and Compatibility Matrix for the Cisco TelePresence Exchange System* at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/compatibility/matrix/ctxmatrix.html.

Table 3 Endpoint Limitations

Product	Meeting Type	Problem
E20	SIP direct-dial	E20 cannot receive shared presentations from CTS.
E20	8710 dial-out	E20 drops presentation sharing from C60 or EX60 when E20 does hold/resume. Resharing presentation corrects problem.
E20	8710 dial-in	E20 drops presentation sharing from C60 or EX60 when E20 does hold/resume and re-share. Resharing presentation does not correct problem.
Cisco Jabber	8510 dial-out/ dial-in (both)	Cisco Jabber presentation sharing does not work with anything but other Cisco Jabber endpoints.
1700 MXP/ 3000 MXP	8510 dial-out/ dial-in (both)	1700 MXP/3000 MXP presentation sharing does not work with anything but Cisco Jabber and other MXP endpoints.

New Functionality in Cisco TelePresence Exchange System Release 1.1

This section includes information about new functionality in Cisco TelePresence Exchange System Release 1.1 only. For information about functionality that was introduced in other Cisco TelePresence Exchange System releases, see the applicable release notes at http://www.cisco.com/en/US/products/ps11276/prod_release_notes_list.html.

See the following sections:

Active Meeting Management

The active meeting management feature enables real-time management of Meet-Me and Rendezvous meetings that are currently in progress. The feature allows you to perform functions such as locking or unlocking the meeting to control whether additional participants can join, muting or unmuting participants, increasing the media bridge resource capacity of the meeting, dialing out to additional endpoints or dropping endpoints, and increasing the duration of the meeting.

The active meeting management options are enabled only after you add a valid active meeting management (ActiveMeetingMgmt) feature license.

A new service desk user role enables system administrators to create users who only manage active meetings. Access to the rest of the administration console is restricted for these users.

The active meeting management API provides programmatic access to all of the new functions for monitoring and controlling active meetings.

For instructions on configuring and using the active meeting management feature, see the "Managing Active Meetings" section in the "Configuring Collaboration Services" chapter of the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*, at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/1_1/install_admin/book/b_install_admin.html.

For additional information about the active meeting management API, see the "Active Meeting Management" chapter of the *API User Guide for the Cisco TelePresence Exchange System Release 1.1*, at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/1_1/api_guide/api_guide_11.html .

Administration Console Enhancements

This release contains the following general administration console enhancements:

ADMIN and SYSTEM Users Can Delete Meetings

Users with the ADMIN or SYSTEM role can delete a meeting as long as there are no participants currently attending the meeting.

When you delete a meeting, the Cisco TelePresence Exchange System cancels the meeting if it has not been previously cancelled, frees up any ports of organization bandwidth or segments of guaranteed media bridge capacity associated with the meeting, and removes the meeting details and diagnostics.

Default IVR Prompts for Lab Use

Release 1.1 comes preloaded with a default Cisco IVR prompt set called CTX Default Ivr Prompts. You can rename, replace or delete the default Cisco IVR prompt set. However, the system will replace the prompt set in its original form the next time the call engine servers restart. For this reason, we recommend that you do not rename the default set or create a different set with the same name.

Note The default Cisco prompts are provided for lab use only. In production, you must use one or more custom prompt sets rather than the default Cisco prompts.

For additional information on IVR prompts, see the "Configuring IVR Prompts" section in the "Configuring Collaboration Services" chapter of the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*, at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/1_1/install_admin/book/b_install_admin.html.

Global Configuration Window

The Global Configuration window allows you to configure settings which apply to the system as a whole, rather than to a specific entity such as a service provider, organization, or meeting.

The ISDN Dial Out Prefix setting that was previously available on the Advanced Configuration window are now on the Global Configuration window, along with additional new settings described in this section:

- MeetMe Default Screens, page 10
- Number of Rows to Display Per List Page, page 11
- SIP Load Balancer Address, page 12

Meetings Tab on Endpoint Window

When viewing the configuration for an endpoint on the Endpoint window in the administration console, you can select the Meetings tab to view a list of all meetings associated with the endpoint.

You can also use the Meetings tab to delete meetings associated with an endpoint. In order to delete an endpoint, you must first delete all associated meetings, including both past and future meetings.

MeetMe Default Screens

The MeetMe Default Screens setting on the System > Global Configuration window allows you control over the number of segments that the system reserves for unprovisioned endpoints that do not have a media profile associated with them (in other words, for dial-in calls or for dial-out situations where no media profile is specified by the meeting scheduler).

In these cases, if the meeting is hosted on a CTMS bridge, the system reserves MeetMe Default Screens + 1 segments. (The additional segment is to account for the possibility of 30 FPS presentation sharing.) If the meeting is hosted on a TPS bridge, the system reserves MeetMe Default Screens. The system always reserves one screen for these endpoints on an MCU MSE 8510 bridge.

Note Lowering the MeetMe Default Screens value may cause capacity problems for all meetings concurrently hosted on the bridge if unprovisioned endpoints with more screens than are reserved join the meeting. We recommend that you provision all endpoints that have more screens than the MeetMe Max Screens value.

The MeetMe Default Screens setting is also used to calculate maximum capacity for Rendezvous meetings. In this case, the system multiplies the value that you specify for Number of Endpoints by a value based on bridge type and MeetMe Default Screens (MeetMe Default Screens + 1 for CTMS, MeetMe Default Screens for TPS, or 1 for Cisco TelePresence MCU MSE 8510). This gives a "worst-case" estimation assuming that all endpoints that join will use the same amount of bandwidth. The system then adds the value of the Additional Capacity field on to the total to determine the maximum capacity.

At attend time, the system uses the value of MeetMe Default Screens to determine the number of segments to allocate when an unprovisioned or remote endpoint joins the meeting, using the same bridge type-based calculation (MeetMe Default Screens + 1 for CTMS, MeetMe Default Screens for TPS, or 1 for MSE 8510).

For more information on capacity reservation and allocation, see the "Organization Bandwidth, Endpoint Capacity, Protocols and Bridge Selection" appendix in the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*, at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/1_1/install_admin/book/b_install_admin.html.

Note The MeetMe Default Screens field description was not included in the online help at release time. For help using the field, see the "Configuring Meet-Me Default Screens" section in the "Configuring the System Settings" chapter of the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*.

Migrate Endpoints

The Migrate Endpoints window enables you to change the media profile configuration of one or more endpoints that belong to the same organization at the same time. You can use this window to quickly migrate the media profiles of CTS endpoints to the new (Native Interop) versions. For more information on the (Native Interop) media profiles, see the "Native Interop" section on page 14.

Number of Rows to Display Per List Page

You can control the number of rows that display on the page at one time in lists in the administration console (for example, on the Alarms, Time Zones, or Meetings list pages) by changing the value of the Number of Rows to Display Per List Page field on the System > Global Configuration window. The new value takes effect for all users of the administration console.

SIP Load Balancer Address

The SIP Load Balancer Address setting on the System > Global Configuration window allows you to configure an address for call engines that employ the Cisco Application Control Engine (ACE) to load balance all SIP traffic. Generally, the administrator defines the SIP load-balancer address on the system after installation and in situations in which the IP address of the call engines or the ACE changes. After changing the address, you must restart the call engine servers in order for the change to take effect.

HTTP load balancing (previously configured via the Meet-Me External HTTP Address in Cisco TelePresence Exchange System 1.0 releases) is not available in release 1.1. The SIP Load Balancer Address setting covers the SIP load-balancing portion of the functionality that was previously configured by using the Meet-Me External HTTP Address field.

Note The SIP Load Balancer Address field description was not included in the online help at release time. For help using the field, see the "Configuring a SIP Load Balancer Address" section in the "Configuring the System Settings" chapter of the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1.*

Advanced Resource Management

The Advanced Resource Management feature provides greater flexibility and control of how media bridge resources are allocated for Meet-Me and Rendezvous meetings. This feature introduces the concept of resource groups and reservation types.

Before configuring resource groups or adding Meet-Me or Rendezvous meetings, you must define reservation types. The reservation type determines whether the Cisco TelePresence Exchange System provides a guaranteed or best-effort level of service when reserving a media bridge resource for a Meet-Me or Rendezvous meeting. The reservation type levels of service are defined as follows:

- Guaranteed—When you create a guaranteed Meet-Me meeting, the system reserves media bridge resources for the specified meeting duration. For a guaranteed Rendezvous meeting, the system reserves resources for the meeting that can never be used for other meetings.
- Best-effort—When you create a best-effort Meet-Me or Rendezvous meeting, the system does not reserve any media bridge resources in advance for the meeting. Instead, the system allocates resources when the first participant joins the meeting and deallocates resources when the last participant leaves the meeting. For a best-effort meeting, the system may fail to allocate resources to the meeting because all the available resources may be in use by other best-effort meetings for the given time period.

When configuring a resource group, you choose a specific service provider and region and one or more reservation types to be associated with the group. You then configure the allowable amount of dedicated media resources and meeting booking capacity for each reservation type chosen. Assigning both a guaranteed and best-effort reservation type to a single resource group allows you to dedicate a specific percentage of the resources to guaranteed meetings and another percentage to best-effort meetings. For best-effort meetings, you have the capability to overbook the media bridge resources. Overbooking assumes that all Meet-Me and Rendezvous meetings associated with a specific reservation type will not be active at the same time. By having different levels of overbookings, you can provide different service levels (for example, Gold, Silver, and Bronze) whereby the higher service levels have lower overbooking and thus have a lower probability of booking failure.

After the resource group has been created, you configure specific media bridge resources to be associated with the group. Based on the set of requirements configured for a Meet-Me or Rendezvous meeting (such as service provider, region, reservation type, and endpoint requirements), the system selects the best-fit resource group and associated media bridge resources to use for the meeting.

For information on how to configure resource groups, see the "Configuring Resource Groups" section in the "Configuring Customers" chapter of the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*, at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/1_1/install_admin/book/b_install_admin.html.

For more information about reservation types, see the "Configuring Reservation Types" section in the "Configuring Collaboration Services" chapter of the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*.

Implicit Meeting Extension

When the Implicit Meeting Extension feature is enabled for a Meet-Me meeting, the Cisco TelePresence Exchange System checks for available resources shortly before the two minute end-of-meeting warning. If sufficient resources are available, the system displays a notification indicating to participants that the meeting has been extended, and the meeting continues for a specified length of time. (Participants in meetings hosted on the Cisco TelePresence Multipoint Switch will not see the meeting extension notification, however.)

You can configure the number and length of implicit extensions. The extension length must be in increments of 15 minutes. The maximum number of extensions times the extension length must not exceed 24 hours.

How Implicit Meeting Extension Works

Each time the meeting approaches an extension period, the system checks both currently active and upcoming scheduled meetings to ensure that sufficient resources are available. For example, you can schedule a meeting with a duration of 60 minutes and two 15-minute implicit extensions. The system checks for resources once the meeting has been underway for approximately 58 minutes. If resources are available, the meeting duration is extended to 75 minutes, and the system checks again after the meeting has been underway for approximately 73 minutes. If the extension fails, the system displays the two minute end-of-meeting warning to participants, and ends the meeting after two minutes. Otherwise, the meeting duration is extended to 90 minutes.

You can configure the meeting extension policy at the service provider level, organization level, or meeting level. The policy is hierarchical, so you can configure a meeting to inherit its settings from the organization. In addition, you can configure an organization to inherit its settings from the service provider. If you want a meeting to inherit organization settings, you must enable the inheritance option at the meeting level. Similarly, if you want an organization to inherit service provider settings, you must enable the inheritance option at the organization level.

Combining Explicit and Implicit Meeting Extension Features

You can also use the Active Meeting Management feature controls to explicitly increase the duration of a Meet-Me meeting that is in progress. You can use both implicit and explicit extensions for the same meeting. After each implicit extension, the system updates the duration of the meeting that is displayed via Active Meeting Management. After each explicit change to the meeting duration, the system resets the number of implicit extensions for the meeting. For example, on an active meeting that was originally scheduled for 60 minutes with two 15-minute implicit extensions enabled, if the system automatically

extends the meeting to 75 minutes, and then a service desk user manually changes the meeting duration to 90 minutes, the system could perform two more implicit extensions, if resources are available, for a total duration of up to 120 minutes.

Note You can use Active Meeting Management to modify the quantity and duration of implicit extensions if implicit extensions were enabled before the meeting began. Changes to the duration of implicit extensions will apply to the next extension period if the system has already checked for resources for an extension or has started the extension. You cannot enable implicit extensions while the meeting is active if they were disabled before the meeting began.

Host PINs

The Host PINs feature enables you to designate a host participant role for a Meet-Me or Rendezvous meeting in order to restrict which participant can start the meeting. More than one participant can be designated as a host. The system categorizes participants as either a host or a guest.

If a guest participant joins the meeting before a host, the system places the participant in a queue and prevents the participant from joining the meeting. Once a host joins the meeting, the meeting starts and all the guests in the queue join the meeting. You can configure whether or not to drop all participants when all the hosts leave the meeting.

You can configure the host settings at the service provider level, organization level, or meeting level. The settings are hierarchical, so you can configure a meeting to inherit its settings from the organization. In addition, you can configure an organization to inherit its settings from the service provider. If you want a meeting to inherit organization settings, you must enable the inheritance option at the meeting level. Similarly, if you want an organization to inherit service provider settings, you must enable the inheritance option at the organization level.

To use the Host PINs feature, you must enable the host options and create a host personal identification number (PIN) for the meeting. You can choose to have the system generate the host PIN or you can enter a customized host PIN. Host PINs can be reset as many times as needed.

The system designates a participant as a host if one of the following conditions apply:

- A participant manually enters the host PIN when joining the meeting.
- A participant joins the meeting from a provisioned endpoint that has been designated as a host. For this condition, the participant does not need to manually enter a host PIN to join the meeting.

Native Interop

Cisco TelePresence Exchange System Release 1.1 introduces new dial-out protocol support. In conjunction with the Cisco TelePresence System (CTS) Software Release 1.8, Release 1.1 also supports a broader set of media protocols for CTS endpoints. The Cisco TelePresence Exchange System bridge selection algorithm allows you to make optimum use of bridge resources based on the endpoint media profiles configured for a meeting.

The Cisco TelePresence Exchange System supports five signaling protocols:

- SIP—Session Initiation Protocol, a standards-based protocol.
- TIP—TelePresence Interoperability Protocol, a standards-based protocol developed by Cisco. Uses a limited version of SIP for initial signaling, followed by further signaling encoded in the media stream.

- MUX—A Cisco-proprietary protocol that was a precursor to TIP.
- H.323—A standards-based protocol.
- ISDN—A standards-based protocol.

In Release 1.1, the system can dial out to Cisco TelePresence System endpoints by using the MUX, TIP or SIP protocols, and can perform Guest Dial Out to SIP and TIP endpoints, in addition to H.323 and ISDN dial-outs which have been supported since Release 1.0.

Each endpoint may support one or more protocols, and each bridge type used by the system supports a particular set of protocols. Release 1.1 supports the following combinations of bridge types and protocols.

Cisco TelePresence Multipoint Switch (CTMS)	TIP or MUX
Cisco TelePresence Server MCU MSE 8510	SIP, H.323, ISDN
Cisco TelePresence Server MSE 8710	SIP, TIP, MUX, H.323, ISDN[1]

1. ISDN over H.323; requires Cisco TelePresence ISDN GW MSE 8321.

Note This functionality requires specific versions of solution components including CTMS, Cisco TelePresence Server, and MSE software. For

The system can use a mixture of protocols within a meeting on a bridge, except in the case of the Cisco TelePresence Multipoint Switch, which requires that all endpoints use the same protocol (either TIP or MUX).

The endpoint media profiles that you specify when scheduling a meeting determine which types of bridge are capable of hosting the meeting. If the first bridge type that the system tries does not have sufficient resources available, the system attempts to allocate resources from another bridge type that supports the same capabilities. The system considers CTMS resources to be the "cheapest" and MSE 8710 the most "expensive" and attempts to allocate resources from the cheapest capable bridge type first.

Release 1.1 includes pre-defined media profiles for various endpoints, and you can add additional media profiles. For the Cisco TelePresence System endpoints, the pre-defined media profiles that correspond to CTS Release 1.8 capabilities have "(Native Interop)" in the name and description fields.

For additional information about endpoint interoperability, including the bridge selection order and the mechanisms that the system uses to determine which protocol to use in dial-in and dial-out situations, see the "Organization Bandwidth, Endpoint Capacity, Protocols and Bridge Selection" appendix of the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*, at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/1_1/install_admin/book/b_install_admin.html.

Multilingual IVR Prompt Set Support

In Release 1.1, you can create one IVR prompt set per service number, and you can associate more than one service number to a service provider. This allows you to create prompt sets in different languages for different meeting scenarios.

When scheduling a meeting, you must select a service number for the meeting. The service number therefore determines the language that meeting participants hear when interacting with the IVR service. All meeting participants who interact with the IVR use the prompt set associated with the service number configured for the meeting, including those who dial in by the using One-Button-to-Push (OBTP) functionality on their endpoints. If a participant attempts to dial a service number that is not associated with the meeting, they hear the prompts associated with the service number that they dialed.

Rendezvous Meetings

Also called a *timeless* or *reservationless* meeting, a Rendezvous meeting is not limited to a single start time. Once created, a Rendezvous meeting starts whenever participants join the meeting. As soon as the first participant joins a Rendezvous meeting, it is considered *active*.

When you create a Rendezvous meeting, you can specify a maximum meeting instance duration or allow the system to enforce the default maximum of 1440 minutes (24 hours). At the end of the maximum meeting instance duration, all calls in the meeting are dropped, but users can immediately rejoin the meeting as part of a new meeting instance.

For a Rendezvous meeting, the system starts a new meeting instance and allocates media bridge resources when the first participant joins the meeting. Likewise, the system deallocates resources and ends the current meeting instance when the last participant leaves the meeting.

When you create a Rendezvous meeting, you specify a reservation type. The reservation type determines whether the system provides a guaranteed or best-effort level of service defined as follows:

- Guaranteed—For a guaranteed Rendezvous meeting, the system reserves resources to the meeting that can never be used for other meetings.
- Best-effort—When you create a best-effort Rendezvous meeting, the system does not reserve any media bridge resources in advance for the meeting. Instead, resource allocation occurs when the first participant joins the meeting. For any attempt to start a best-effort meeting, the system may fail to allocate resources to the meeting because all the available resources may be in use by other best-effort meetings for the given time period.

For more information about reservation types, see the "Configuring Reservation Types" section in the "Configuring Collaboration Services" chapter of the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*, at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/1_1/install_admin/book/b_install_admin.html. For information on how to schedule a Rendezvous meeting, see the "Scheduling Rendezvous Meetings" section in the same chapter.

Note Standing meetings are no longer supported on the Cisco TelePresence Exchange System as previously defined, but they have been replaced with an equivalent type of meeting. Instead of creating a standing meeting, create a guaranteed Rendezvous meeting.

Welcome Message for Cisco TelePresence Server MSE 8710

This release enhances the Cisco TelePresence Server MSE 8710 (TPS) Field Descriptions table by adding a new Conference Name field that displays a welcome message on the TPS screen. Additionally, the meeting subject will be displayed below the welcome message to the first caller to join the meeting (lobby screen). For information related to this field, see the "Configuring Media Resources" chapter of

the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*, at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/1_1/install_admin/book/b_install_admin.html.

Note You can only edit this field if you are provisioning a new resource or an existing resource. The system must be in maintenance mode.

Whitelists

The Whitelisting feature enables you to define policies that control incoming and outgoing calls between service providers (also known as inter-service provider calls) or between organizations associated with a single service provider (also known as intra-service provider calls). The policies apply to provisioned and unprovisioned endpoints. The Cisco TelePresence Exchange System call detail records (CDRs) identify calls that are rejected due to policy restriction.

For calls from one service provider to another, a flag at the organization level enables you to allow or deny outgoing calls to other service providers. (This includes Meet-Me, Rendezvous, and direct-dial calls, because the Cisco TelePresence Exchange System cannot determine the type of an outgoing call.) Two additional flags at the organization level enable you to allow or deny incoming direct-dial calls from another service provider and to allow or deny incoming Meet-Me and Rendezvous calls from another service provider.

For calls within a single service provider, you can define whitelist groups that determine which organizations can dial each other directly. (Meet-Me and Rendezvous meeting calls are always allowed between organizations that belong to the same service provider.) A whitelist group can contain more than one organization, and an organization can belong to more than one whitelist group.

Note If you provide hosted endpoint service and support unprovisioned endpoints, the Cisco Aggregation Services Router (ASR) that serves as the Session Border Controller must be configured such that each organization uses a single Carrier Information Code (CIC) in order for calls from unprovisioned endpoints to be mapped to the correct organization.

If you upgrade to Release 1.1 from an earlier version of the Cisco TelePresence Exchange System, no restrictions are placed on inter-service provider calls for existing organizations (the new inter-SP flags are all checked by default). For intra-service provider calls, the behavior is as follows:

- Organizations that were previously configured with the Direct Dial Enabled check box checked will have the new Enable Whitelists check box unchecked by default in Release 1.1 (the system places no restrictions on intra-service provider direct-dial calls).
- Organizations that were configured with the Direct Dial Enabled check box unchecked will have the new Enable Whitelists check box checked by default (intra-service provider direct dial calls be restricted by default until you add organizations to the same white list group).

For instructions on configuring flags for inter-service provider calls, see the "Configuring Organizations" section in the "Configuring Customers" chapter of the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*, at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/1_1/install_admin/book/b_install_admin.html. For instructions on configuring whitelist groups for intra-service provider calls, see the "Configuring Whitelist Groups" section in the same chapter.

Changed Functionality in Cisco TelePresence Exchange System Release 1.1

This section includes information about changed functionality in Cisco TelePresence Exchange System release 1.1 only. For information about functionality changes that were introduced in other Cisco TelePresence Exchange System releases, see the applicable release notes at http://www.cisco.com/en/US/products/ps11276/prod_release_notes_list.html.

See the following sections:

Command Reference Changes

Table 4 lists the new commands for Release 1.1. For detailed information about the commands, see the "Command Reference" appendix of the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*, at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/1_1/install_admin/book/b_install_admin.html.

Table 4 New Commands for Release 1.1

Command	Description
utils patch download	Downloads the patch to the local patch repository of the server.
utils patch history	Lists the sequence of the patches that have been installed and uninstalled.
utils patch install	Installs the patch on the administration server or call engine server.
utils patch show-patches	Displays the names of the patch files that are downloaded in to the repository.
utils patch uninstall	Uninstalls the patch that has already been installed on either the administration server or call engine server.
utils service corosync status	Checks the status of a corosync service on the current server.
utils service crm status	Monitors the Cluster Resource Manager (CRM) status on all the Cisco TelePresence Exchange System servers.
utils service database sync	Starts a Cisco Tomcat service that is down.
utils service nodemanager status	Checks the status of a Cisco Tomcat service after installation or during general operations.
utils service nodemanager stop	Gracefully stops a Cisco Tomcat service.

The following commands are modified to support the updates to split brain recovery, corosync service, and CRM:

- **utils service database drbd disable-ha**
- **utils service database drbd enable-ha**
- **utils service database status**

The following commands are deprecated:

- **utils service list**
- **utils service start**
- **utils service stop**

Enhancements to Call Routing Dial Patterns Capability

For direct dial and SIP dial out calls, you can now specify the rules that the system uses to match dial patterns to be based on either a destination number or destination domain (the characters that follow the @ symbol in the SIP URI). If the dial pattern rule is specified for a destination number, you can further configure the dial pattern rule to exactly match the dial pattern of the destination number or to match only the prefix, suffix, or regular expression of the destination number. If the dial pattern rule is specified for a destination domain, you can only configure the dial pattern rule to exactly match the characters that follow the @ symbol in the SIP URI.

Prior to Release 1.1, you could only specify the dial pattern rule to be based on exactly matching a destination number.

For additional information on dial patterns, see the "Configuring Dial Patterns" section in the "Configuring Call Routing" chapter of the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*, at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/1_1/install_admin/book/b_install_admin.html.

Enhancements to Meeting Diagnostics

Available only to super users, system administrators, and administrators, this release enhances the meeting diagnostics for the following functions:

- Log Collection Session—Displays all the new and previous archives for all the administration and call engine servers that are in the cluster.

Note Service desk users can also start and stop a log collection session.

For detailed instructions on how to start a log collection session for the appropriate servers, see the "Starting a Log Collection Session" section in the "Meeting Diagnostics" chapter of the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*, at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/1_1/install_admin/book/b_install_admin.html.

- Reservation Pool Usage—Displays the number of ports or segments that is reserved for a given resource pool and the time buckets. All Meet-Me pools (also called *scheduled*) display the start and end times in 15-minute interval time buckets. Because Rendezvous pools (also called *timeless*) are not limited to a start time, the reservation information is displayed in a single time bucket. For each resource type, this window also displays the maximum number of single-screen and three-screen endpoints that can be reserved for the specified time period.

If you specify the start time and duration of the meeting, the View Reservation Pool Usage window displays all of the scheduled pools within that time period and the Rendezvous pools. If you check the **Rendezvous Meeting** check box, this window displays all of the future scheduled pools and the Rendezvous pools.

For detailed instructions on how to view the reservation pool usage for a meeting, see the "Viewing Allocation Pool Usage" section in the "Meeting Diagnostics" chapter of the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*.

- Allocation Pool Usage—Displays the allocated ports or segments for a given resource pool and resource group.

  For detailed instructions on how to view the allocation pool usage, see the "Viewing Allocation Pool Usage" in the "Meeting Diagnostics" chapter of the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*.

Testing a Media Bridge Resource

In Cisco TelePresence Exchange System Release 1.0(3), the Request Specific Resource check box (and associated drop-down list of resources) allowed you to select a specific media bridge resource for a meeting. The intent of the field was to allow for testing a specific resource within a short period of time after scheduling a meeting. The Request Specific Resource check box and drop-down list have been removed from Release 1.1.

In Release 1.1, in order to allow for testing of a specific resource without interfering with the Cisco TelePresence Exchange System capacity reservation and allocation algorithm, we recommend that you create a specific test region on the system, place only the resource to be tested in the region, and use the test region when scheduling a test meeting. When you are done testing the resource, remove the resource from the test region, and delete the test region.

Before upgrading to Release 1.1, we strongly recommend that you cancel or delete any future meetings that are configured to use a specific resource. If you do not cancel such meetings, the meetings may cause capacity contention with other meetings on the system.

Caveats

This section addresses the open caveats in this release and provides information on how to use the Bug Toolkit to find further details on those caveats. Topics in this section include:

Open Caveats

Table 5 describes the open caveats in this release of the Cisco TelePresence Exchange System. Caveats are listed in order by severity, then by component, then by caveat number.

Table 5 Open Caveats in Cisco TelePresence Exchange System Release 1.1

Identifier	Component	Severity	Headline
CSCtz96865	admin_ui	3	User ejected from GUI while attempting AMM with higher CDR
CSCua53920	admin_ui	3	Need to press Save button twice to save Max Capacity change in any medi
CSCto88766	call_control	3	Endpoint didn't drop when meeting ends after back to back engine failure
CSCtz95262	call_control	3	Participant view shows 3 entries for T3 end point in a TPS meeting
CSCua56273	call_control	3	Mute status shows as "Unknown" for TPS and CTMS meetings
CSCua32284	other	3	Incorrect available capacity calculation
CSCua32355	other	3	Call drops while resuming CTMS dialout calls after engine failover
CSCua32398	other	3	Polycom dial-in endpoint drops in TPS meeting
CSCua32441	other	3	CDR data is different for API vs. Admin list page vs. Admin csv export
CSCua32472	other	3	Participant view shows misleading dialout endpoint status
CSCua32482	other	3	Participant view shows wrong participant no during double engine failove
CSCua32521	other	3	Some processes fail to start for Consumer TP system
CSCua34798	other	3	Meeting got opened late in DB failover scenario
CSCua34846	other	3	snmpd fails after querying object ifCounterDiscontinuityGroup
CSCua53872	other	3	Instance id got changed for TPS rendezvous meeting
CSCua53911	other	3	Test Connection to CTS-Man fails wth exception when it is onlne
CSCua53934	other	3	Reservation type associated to mutliple resource groups cause failure
CSCua32456	platform_os	3	Corosync coredumps when system is not in service
CSCua34782	platform_os	3	Location_0 was presented as “Null”, sometimes in SIP dialout scenarios
CSCtz94171	system_management	3	Failure to Delete CTMS Resource after Upgrade

Resolved Caveats

For the latest information on resolved caveats for this release, access Bug Toolkit as described in the “Accessing Bug Toolkit” section on page 21.

Accessing Bug Toolkit

You can use the Bug Toolkit to find information about caveats for this release, including a description of the problems and available workarounds. The Bug Toolkit lists both open and resolved caveats.

To access Bug Toolkit, you need the following items:

- Internet connection
- Web browser
- Cisco.com user ID and password

To use the Bug Toolkit, do the steps in the following procedure.

Procedure

Step 1 To access the Bug Toolkit, go to the following link:

http://tools.cisco.com/Support/BugToolKit/action.do?hdnAction=searchBugs

Step 2 Log in with your Cisco.com user ID and password.

Step 3 To look for information about a specific problem, enter the bug ID number in the **Search for Bug ID** field and click **Go**.

Step 4 To look for information when you do not know the bug ID number, do the following:

a. From the Select Product Category menu, choose **TelePresence**.

b. From the Select Products menu, choose the desired product.

c. From the Software Version menu, choose the version number.

d. Under Advanced Options, choose either **Use default settings** or **Use custom settings**.

- When you select **Use default settings**, the system searches for severity 1, 2, and 3 bugs, open and fixed bugs, and only those bugs containing bug details.
- When you select **Use custom settings**, you can specify the severity and status parameters or search for keywords within the bug headline and description.

Documentation Updates

This section describes omissions and changes to the published documentation for the Cisco TelePresence Exchange System Release 1.1

API User Guide for the Cisco TelePresence Exchange System Release 1.1: apiCallDetailRecord meetingInstanceID

Added January 18, 2013

The description for the meetingInstanceID parameter in the apiCallDetailRecord Element table in the "Call Detail Record API" chapter is incorrect. Use the following description instead:

This indicates the instance number of a Rendezvous meeting. The number starts at 0 for the first instance and increments by one for each successive instance of the Rendezvous meeting. For other meeting types, the value is always 0.

API User Guide for the Cisco TelePresence Exchange System Release 1.1: sendEndpointTextToParticipant

Added February 21, 2013

The "sendEndpointTextToParticipant" section in the "Active Meeting Management API" chapter incorrectly states that you can use the sendEndpointTextToParticipant method to display text on one or more specified endpoints, and that you can specify multiple participants in the input parameters.

When using the sendEndpointTextToParticipant method, specify only a single participant per request.

Changes to the Online Help for the Administration Console

The online help for the administration console captures an early version of a subset of the *Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1*. For the most recent content, see the guide, which is available at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/1_1/install_admin/book/b_install_admin.html.

Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1: Cisco TelePresence Multipoint Switch Dial Out

Added February 7, 2013

The "Dial Out and Dial In" section of the "Product Overview" chapter omits the following information about the Cisco TelePresence Multipoint Switch dial out behavior.

If your meeting is hosted on a Cisco TelePresence Multipoint Switch (CTMS), the Cisco TelePresence Exchange System dials out to each endpoint only one time and attempts to connect for up to 30 seconds. If the connection fails for any reason, the system does not make any further attempts to redial.

Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1: Incorrect Header Editors

Added January 18, 2013

The "Configuring Editors" and "Creating Adjacencies" sections of the "Configuring Cisco Session Border Controllers" chapter include incorrect outbound header editors in the configuration examples. You do not need to configure an outbound SIP header editor or include an outbound header editor in the adjacency configuration. However, you must either configure the default header editor to be an empty blacklist, or configure an outbound header editor that is an empty blacklist and apply it to the adjacencies for the Cisco Application Control Engine and for both Cisco TelePresence Exchange System call engine servers.

Use the following header editor example configuration instead of the example in the "Configuring Editors" section.

```
Router(config)# sbc mmsbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# sip header-editor in1
Router(config-sbc-sbe-mep-hdr)# blacklist
Router(config-sbc-sbe-mep-hdr)# store-rule entry1
Router(config-sbc-sbe-mep-hdr-ele-act)# condition header-name session-expires header-value
regex-match ";\(.*\)" store-as refreshparam
Router(config-sbc-sbe-mep-hdr-ele-act)# exit
Router(config-sbc-sbe-mep-hdr)# header session-expires entry 1
Router(config-sbc-sbe-mep-hdr-ele)# action replace-value value 1800
Router(config-sbc-sbe-sip-hdr-ele-act)# condition variable refreshparam is-defined eq
false
Router(config-sbc-sbe-mep-hdr-ele-act)# exit
Router(config-sbc-sbe-mep-hdr)# header session-expires entry 2
Router(config-sbc-sbe-mep-hdr-ele)# action replace-value value “1800;${refreshparam}”
Router(config-sbc-sbe-sip-hdr-ele-act)# condition variable refreshparam is-defined eq true
```

Installation and Administration Guide for the Cisco TelePresence Exchange System Release 1.1: IVR Prompt Flow

Added February 7, 2013

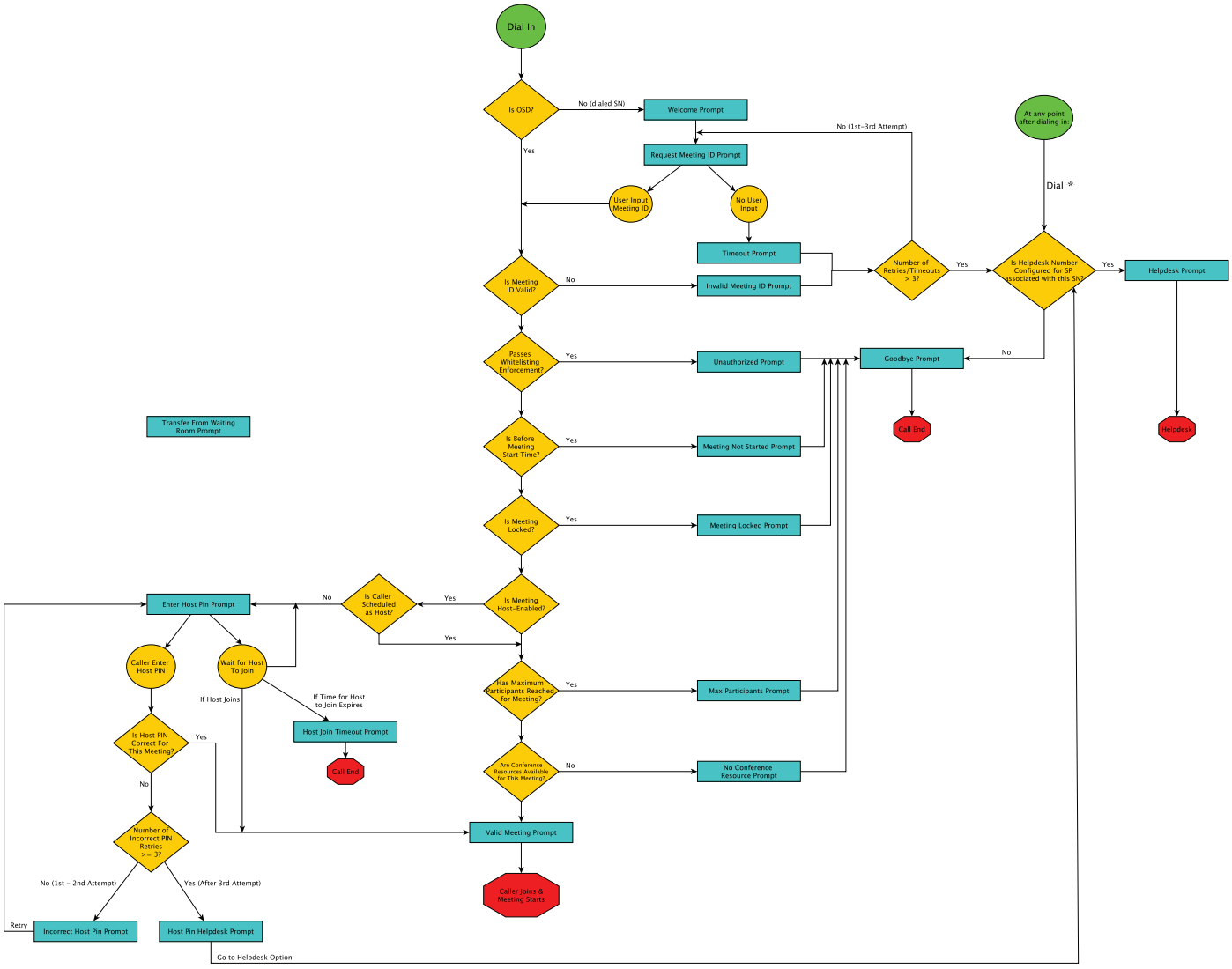
The “Configuring IVR Prompts” section of the “Configuring Collaboration Services” chapter omits the following information about how the IVR prompts are used by the system.

Figure 1 illustrates the possible paths through the IVR prompt set for a caller dialing in to the system, starting with whether the caller used one-string dial (OSD) to append the conference ID to the service number (for example, 18005551212**12345678) or dialed the service number (SN) alone. Alternatively, the caller can dial * at any time to reach the Helpdesk prompt and transfer, if a help desk route has been configured for the service provider associated with the service number.

Note that the Transfer from Waiting Room Prompt, although available in the default prompt set, is not used in this release.

Tip If you are viewing the html version of these Release Notes and have difficulty seeing the figure, see the PDF version at http://www.cisco.com/en/US/docs/telepresence/tx/exchange_system/1_1/rn_ctx11.pdf.

Figure 1 IVR Prompt Flow in Cisco TelePresence Exchange System Release 1.1



Obtaining Documentation and Submitting a Service Request

For information on obtaining documentation, submitting a service request, and gathering additional information, see the monthly *What's New in Cisco Product Documentation*, which also lists all new and revised Cisco technical documentation, at:

http://www.cisco.com/en/US/docs/general/whatsnew/whatsnew.html

Subscribe to the *What's New in Cisco Product Documentation* as an RSS feed and set content to be delivered directly to your desktop using a reader application. The RSS feeds are a free service. Cisco currently supports RSS Version 2.0.

This document is to be used in conjunction with the documents listed in the "Related Documentation" section on page 2.

Release Notes for the Cisco TelePresence Exchange System Release 1.1