Cisco TelePresence Server 2.2(1.48)

Software release notes

D14817.01

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Document revision history

Revision	Date	Description
D14817.01	August 2011	Updated resolved caveats and interoperability section for first maintenance release of Cisco TelePresence Server 2.2.
D14817	August 2011	Republished to include interoperability information.
D14817	June 2011	These notes accompany the release of version 2.2 of the Cisco TelePresence Server software.

Introduction

The Cisco TelePresence Server software applies to the following Cisco TelePresence Server hardware:

- Cisco TelePresence Server MSE 8710
- Cisco TelePresence Server 7010

This document accompanies version 2.2 of the software. In particular, the document refers to build number (1.48) of the software which is the first maintenance release.

Version 2.2 is a feature release, that is, it incorporates a number of new features as well as resolving a number of caveats discovered during the maintenance of version 2.1.

This document lists and describes the new features, resolved caveats, and any known limitations or open caveats. The document also contains instructions for upgrading to this version of the software and for reversing the upgrade if necessary.

New features and functionality in version 2.2

- HD and Full HD modes
- Increased cluster size
- Content improvements
- Lobby screen
- Conference ending notification
- Improvements to telepresence experience
  - Faster audio switching
  - Centered PIPs
- Improved support for Cisco CTS systems
  - Support for TIP endpoint identification and ad hoc TIP calls
  - Encrypted calls
  - Roster list
  - Conference locking
  - Multichannel audio
- Conference size limits
- Improved text rendering
- IPv6 support
- Send DTMF to endpoints on connect
- TelePresence Server to CUCM trunking
- New help system

HD and Full HD modes

In earlier releases the maximum supported video definition of the TelePresence Server was 720p30 (720 horizontal lines, progressive scanning, 30 frames per second) which is now called ‘HD mode’.

This release introduces a new mode called ‘Full HD’ mode which allows for the following maximum video definitions:

- 1080p30 (1080 horizontal lines, progressive scanning, 30 frames per second)
- 720p60 (720 horizontal lines, progressive scanning, 60 frames per second)

Lower resolutions than 720 are also supported at 60 fps in ‘HD mode’.

There are no additional licensing requirements to use Full HD mode, with each screen license enabling a screen in either mode.

A Cisco TelePresence Server can support up to 16 screens in HD mode or up to 12 screens in Full HD mode.

Increased cluster size

The clustering feature is improved in this release. This version allows for a cluster of four Cisco TelePresence Server MSE 8710 blades to act as a single TelePresence Server, where previously the maximum cluster size was three blades.

This means that a cluster can now support up to 64 video participants in HD mode or up to 48 participants in Full HD mode.

The number of additional audio and content channels also increases when four blades are in the cluster.

Content improvements

The TelePresence Server now dynamically reassigns content ports to ensure that as many participants as possible can see the presentation.

In HD mode, a single Cisco TelePresence Server hardware unit supports 10 content ports and 16 video ports. This specification means that, in previous releases, some participants in full conferences would not have been able to see the presentation.

Version 2.2 of the Cisco TelePresence Server software mitigates this limitation by mixing the content stream into the main video stream for single screen endpoints, thereby freeing up content ports to be reassigned to other endpoints.

Multiple screen endpoints are given preference over the content ports and are guaranteed to receive the content as an extended video stream.

Single screen endpoints that receive the mixed stream now see this stream in a new layout. The layout shows the content in the largest area of the screen and reduces the video streams of other participants to smaller pictures along the bottom of the screen.

This capability of the TelePresence Server is not constrained to individual conferences; if necessary, the TelePresence Server can free up content ports for one conference by mixing the streams to endpoints in other conferences.

Content port priority is granted to endpoints in the following order:

1. Multiple screen systems
2. Single screen / immersive systems
3. All other endpoint types.

To ensure stability, the TelePresence Server does not reassign content ports between endpoints of equal priority.

Lobby screen

The lobby screen is a new feature in this release. It is a static image with conference details overlaid that will be seen by participants before entering the conference.

The lobby screen shows the conference title and an optional lobby message. It also shows the conference start and end times if the conference is scheduled and not permanent.

You can enable or disable the lobby screen for each conference, or allow individual conferences to use the default lobby screen behavior that you set for the TelePresence Server.

You can set the lobby screen date format on a server-wide basis, and the lobby message for each conference.

Conference ending notification

The TelePresence Server can now be configured to notify all participants that the conference is coming to an end. You can enable or disable this setting for each conference, or allow individual conferences to use the default settings for the TelePresence Server.

You can use a custom conference ending notification on a server-wide basis. Cisco CTS endpoints render the conference ending message locally rather than using the TelePresence Server’s setting.

Improvements to telepresence experience

Faster audio switching

Improvements to the audio mixing capabilities enable the TelePresence Server to prepare the outgoing stream more quickly than in previous versions. The delay when the active speaker changes is reduced and participants experience a more natural flow of conversation.

Centered PIPs

The automatic layout of the composed video stream is improved. When participants' video streams are displayed in PIPs across the bottom of the layout, those PIPs are centered where possible.

Improvements to support for Cisco CTS systems

Support for TIP endpoint identification and ad-hoc TIP calls

Cisco TelePresence Server 2.2 supports TIP endpoint identification and ad-hoc TIP calls.

This means that CTS systems, using the newly released CTS 1.7.4 software that includes TIP identification, are better supported in this release. This feature requires Cisco Unified Communications Manager 8.5 or later.

This feature enhances the interoperability and scalability of the TelePresence Server, CUCM, and CTS solution.

In TelePresence Server releases prior to 2.2, Cisco CTS systems needed to be preconfigured using **Add Cisco endpoint** on the TelePresence Server.

The TelePresence Server can now accept ad-hoc calls from newer CTS systems, or you can preconfigure them using **Add new endpoint** (rather than **Add Cisco endpoint**).

The older way of preconfiguring CTS systems has been retained, via the **Add legacy Cisco CTS endpoint** feature, to ensure compatibility with CTS versions prior to 1.7.4 and CUCM versions prior to 8.5.

Encrypted calls to CTS endpoints

The TelePresence Server can now set up encrypted calls to Cisco CTS systems, provided that the server has the Encryption feature key installed.

Roster list display

The TelePresence Server now sends a roster list of the conference participants to the CTS system. The list displays on the phone attached to the CTS system.

This feature requires the HTTP service to be enabled on the TelePresence Server's IPv4 interface.

Conference locking

You can use the phone attached to the CTS system to lock the conference, which means that nobody else can join the conference. This feature requires the HTTP service to be enabled on the TelePresence Server's IPv4 interface.

You can also lock the conference via the TelePresence Server’s web interface or the XML-RPC API.

Multichannel audio for CTS rooms

The TelePresence Server now supports multichannel audio to and from CTS 3000 series systems, which provides a better meeting experience.

Conference size limits

This release of the TelePresence Server enables an administrator to define the maximum number of participants allowed in a conference. There is an audio only participant limit and a video participant limit which, when combined, define the conference's participant limit. No limits are defined for content channels.

Note that the TelePresence Server does not reserve resources and cannot guarantee that the configured limits may always be met; it's possible that the number of participants may be limited by the available resources before the configured limit is achieved.

The TelePresence Server considers multiple screen endpoints as multiple participants.

Improved text rendering

In prior releases the TelePresence Server used a bitmap font to display text on endpoint screens. The bitmaps were scaled as appropriate for the screens which did not always provide a visually pleasing result.

Version 2.2 introduces support for a TrueType font. To improve text rendering you'll need to upload the supplied font file to the upgraded TelePresence Server.

The TelePresence Server defaults to the former method of rendering text if you choose not to upload the font, or to remove it.

IPv6 support

The TelePresence Server now supports IPv6. You can manually or automatically configure IPv6 or IPv4 addresses on the TelePresence Server's Ethernet interface. You can also set an IPv6 address for the name server(s) and the default gateway.

The web interface supports the CIDR format of IPv6 address. For IPv6 addresses it's likely that you'll want to automatically generate the address, for which you can select DHCPv6 or SLAAC, depending on which automatic address allocation system the network is using.

Send DTMF to endpoints on connect

The TelePresence Server now has the ability to send a string of DTMF tones when calling out to an endpoint, which facilitates dialing into systems where additional numbers are required to complete the connection (such as a PIN).

You can enter the DTMF string on the call out parameters of a pre-configured endpoint, or you can specify it via the relevant API call.

Trunking to CUCM (Cisco Unified Communications Manager)

Version 2.2 of the Cisco TelePresence Server software introduces a new method of making calls via Cisco Unified Communications Manager.

You can now trunk the TelePresence Server to CUCM, where previously you were limited to either registering the TelePresence Server directly with CUCM or registering it with a VCS that was trunked to CUCM.

Newly supported in this release	Cisco TelePresence Server	Trunk	Cisco Unified Communications Manager		
Previously supported (preferred option)	Cisco TelePresence Server	Registered	Cisco TelePresence Video Communications Server (VCS)	Trunk	Cisco Unified Communications Manager

Previously supported (not recommended)	

This feature is accessed in the TelePresence Server's configuration for outbound SIP calls. Instead of simply enabling or disabling a SIP registrar, as before, you now have three options for **Outbound call configuration**:

Use registrar Corresponds to previous release's Use SIP registrar (checked).

Use trunk New option to trunk to the address/domain combination provided.

Call direct Corresponds to previous release's Use SIP registrar (not checked).

This new feature also affects endpoint configuration. When you are pre-configuring endpoints on the TelePresence Server you now have an unchecked option to **Call direct** instead of a checked option to **Use gatekeeper**. The endpoints will use either the trunk or the registrar by default, depending on your server-wide setting for **Outbound call configuration**.

The new trunking feature implies support for the following:

1. Either registering the TelePresence Server to CUCM for SIP, or setting a proxy for all outgoing messages.
2. Not registering individual conferences with SIP registrar.
3. Making all outgoing calls via CUCM.
4. Creating a dial plan on CUCM to route calls to the correct TelePresence Server.
5. 32 bit authentication tags.

New help system on TelePresence Server

The TelePresence Server has a new help system that includes better navigation and findability features than the previous system.

Resolved caveats

The following issues were found in previous releases of Cisco TelePresence Server and are resolved in version 2.2 (1.48).

Resolved since version 2.2 (1.43), June 2011

Internal reference	Identifier	Summary
113635	CSCtq83179	On a small proportion of hardware units, the TelePresence Server would occasionally restart unexpectedly. This issue has been resolved in this release.
116011	CSCtr11850	The upgrade process would sometimes freeze. This issue has been resolved in this release.
115974, 115611	CSCtr11853, CSCtr11860	Some issues were being seen when using Internet Explorer 9 to access the web interface and online help of the TelePresence Server. These issues have been resolved in this release.
116887	CSCts13246	The TelePresence Server was slightly too sensitive to a change in audio level that registered as a change in speaker, particularly on a CTS 3000 system. This issue has been resolved in this release.
115691, 113473	CSCts15761, CSCts13241	There were some video issues due to instability on the Backplane and Internal links. These have been resolved in this release.

Resolved since version 2.1 (1.37), January 2011

Internal reference	Identifier	Summary
110961	CSCtr08930	Content handover to a Movi user would fail. This issue with the handling of a particular content protocol between Movi and the TelePresence Server is now fixed.
110962	CSCtr08917	TLS encrypted SIP calls to Movi would occasionally disconnect after a few minutes with the error “call decryption failure”. This issue has been fixed.
112559	CSCtr09001	Incoming H.323 calls that inadvertently matched pre-configured CTS endpoints would cause the TelePresence Server to fail. This issue has been resolved by not attempting to match H.323 calls to CTS endpoints, which always use SIP.
113193	CSCtr09895	Small amounts of packet loss in video streams from various endpoints would cause horizontal lines of corruption in the decoded video. This issue has been resolved and the TelePresence Server is now much more tolerant of this level of packet loss.
113322	CSCtr08913	Audio from a Cisco Unified IP Phone 9971 was not being heard by other participants after hold/resume. The TelePresence Server was not mixing the received audio stream into the output stream after the resume. The issue has been resolved.
113431	CSCtq51023	Content video stream would freeze on CTS endpoints during content handover between other endpoints. This has been resolved.
113723	CSCtq42930	Occasional call failures from a TelePresence Server to an external EX90 endpoint, via a VCS, were caused by a race condition that prevented the TelePresence Server from acknowledging the HTTP

Internal reference	Identifier	Summary
		200 OK response from the VCS.
115656	CSCtq82470	The API call system.info returned the incorrect gateKeeperOK status. This call has been fixed.
115717	CSCtr08923	The API call conference.status returns a participant address field that often isn't an address. The address supplied by the endpoint is now used instead of the matched call-in parameter..

Known limitations

HD quality indicators on CTS endpoints

The lobby screen is a static image that is designed for HD mode, that is, 720 pixels high. When a CTS endpoint displays the lobby screen, it may go on to incorrectly report the quality of the received video stream. The quality indicator may show four bars – for 720p video – even though the endpoint is actually receiving 1080p video and should display five bars.

Encryption required causes issues with some endpoints

Some endpoints such as the Sony XG-80 and HG-90, and the TANDBERG Classic 6000s are unable to join conferences in which encryption is required, even when encryption is enabled on the endpoint. (TANDBERG is now part of Cisco).

Setting these conferences to have optional encryption allows these endpoints to join using encryption.

Using a Room to start a conference that is registered to a SIP registrar may cause issues

If you are using a room to start a conference and the following are true then the configured endpoint may not be called successfully.

You enable SIP registrar in the Room configuration with a numeric ID.

You configure a SIP endpoint (in the top section of the Rooms configuration page).

This is because the endpoint may be called before the conference has completed registration with the SIP registrar. This is less likely to happen if you go to **Configuration > System settings**, and then enter a username and password in the SIP registrar section, because this speeds up conference registration with the SIP registrar.

Occasionally calls to additional pre-configured SIP endpoints may also fail, if registration is particularly slow.

Clustering limitations

Currently slot 10 does not support clustering.

If a slave fails while there is high utilization on the cluster, some participants in the cluster may receive audio only.

Open caveats

The following issues currently apply to this version of the TelePresence Server software.

Internal reference	Identifier	Summary
118688	CSCts23629	The participant media summary CDR event incorrectly reports the `position` attribute of a stream to/from a single screen endpoint. This issue does not impact operation.
112982	CSCtr11845	Blades occasionally report changes to backplane link status when another blade in the same chassis reboots. This issue should not impact normal operation.
113932	CSCtr11856	SIP calls out from the TelePresence Server to a LifeSize endpoint may exhibit garbled audio, in both directions, if the calling parties use the G.722.1.C codec. This is unlikely to occur unless the call cannot fall back on another codec. Try using a different codec to work around this issue.

Interoperability

We endeavor to make the TelePresence Server interoperable with all relevant standards-based equipment. However, it is not possible to test all scenarios.

The following list describes the equipment and software revisions that were tested for interoperability with this release. The absence of a device or revision does not imply a lack of interoperability.

Endpoints

Equipment	Software revision	Comments
Aethra X3	11.2.2	Tested H.323 interoperability; no issues found.
Cisco C20, Cisco EX90	TC4.2.0	Tested H.323 and SIP interoperability; no issues found.
Cisco C90, Cisco EX60, Cisco C60	TC4.1.0	Tested H.323 and SIP interoperability; known limitations: • Video artefacts may be seen on the endpoint if the TelePresence Server sends XGA resolution video using the H.263+ video codec (not the default codec). Use 16:9 ratio video, or the H.264 video codec, to avoid this issue.
Cisco CTS 500	1.7.2, 1.7.4	Tested TIP interoperability; no issues found.
Cisco CTS 1300	1.7.4	Tested TIP interoperability; no issues found.
Cisco CTS 3000	1.7.2, 1.7.4	Tested TIP interoperability; no issues found.
Cisco E20	TE4.0.0	Tested SIP interoperability; no issues found.
Cisco TelePresence Movi	4.2.0	Tested SIP interoperability; no issues found. Tested H.323-SIP interworking; known limitation: • Movi may continue to show content (such as a presentation) even when the content has stopped. Using a SIP call without interworking works around this issue.
Cisco TelePresence MXP 1700	F9.0	Tested H.323 and SIP interoperability; no issues found.
Cisco TelePresence MXP 150	L6.0.2	Tested H.323 and SIP interoperability; no issues found.
Cisco TelePresence System T3	TCU 4.1 and TC4.1.1	Tested H.323 interoperability; no issues found.
Cisco Unified IP Phone 9971	9.1(2)	Tested SIP interoperability; no issues found.
Cisco Unified Personal Communicator	8.5(1)	Tested SIP interoperability; no issues found.
Cisco Unified Video Advantage	2.2(1.7)	Tested SIP interoperability; no issues found.

Equipment	Software revision	Comments
LifeSize Room	LS_RM1_4.7.13(22)	Tested H.323 and SIP interoperability; known limitations: • TLS encrypted SIP calls do not work. • The endpoint does not support BFCP hence SIP content does not work. • G.722.1 Annex C audio does not work.
LifeSize Room 200	4.7.17(1)	Tested H.323 and SIP interoperability; known limitations: • TLS encrypted SIP calls do not work. • The endpoint does not support BFCP, hence SIP content does not work. • G.722.1 Annex C audio does not work.
Polycom HDX 8000	2.6.1.3-5205, 3.0.0.2-13047	Tested H.323 and SIP interoperability; known limitations: • SIP content does not work on calls placed out from the TelePresence Server to this endpoint. • SIP content is supported on incoming calls from this endpoint; however, automatic content handover does not work when the endpoint tries to take over content sharing. Stopping and restarting the content works around this issue.
Polycom RPX 200	2.6	Tested H.323 interoperability; no issues found.
Polycom QDX 6000	3.0.2181	Tested H.323 interoperability; no issues found. The endpoint does not support SIP. It also does not support interworking via VCS.
Polycom ViewStation SP	7.5.4	Tested H.323 interoperability; no issues found.
Polycom VSX 7000e	9.0.6.1	Tested H.323 interoperability; no issues found.
Polycom VVX 1500	3.2.2.0481	Tested H.323 and SIP interoperability; no issues found.
Radvision Scopia XT1000	1.00.0014	Tested H.323 and SIP interoperability; no issues found.
Sony PCS-G50	2.64	Tested H.323 interoperability; no issues found. Tested interworking to SIP via VCS; known limitation: • Content is not supported when interworking to SIP via VCS.
Sony HG-90	2.22.00	Tested H.323 interoperability; no issues found. Tested interworking to SIP via VCS; known limitations: • Only supports Voice Only calls unless the bandwidth to the endpoint is greater than 410 kbps (however, the HG-90 does transmit video below this bitrate) • If the HG-90 is dialled in to a conference, it will become a Voice Only call if the TelePresence Server is set to transmit video in 4:3 ratio only.

Equipment	Software revision	Comments
Sony PCS-1	3.42	Tested H.323 and SIP interoperability; no issues found.

Infrastructure

Equipment	Software revision	Comments
Cisco TelePresence Content Server	S5.0	Tested H.323 and SIP interoperability; no issues found.

Gatekeepers

Equipment	Software revision	Comments
Cisco TelePresence Video Communication Server (VCS)	X6.1	No issues found.
Tandberg Gatekeeper	N5.2	No issues found.
Polycom PathNav	7.00.03	No issues found.
GNU Gatekeeper	2.3.1	No issues found.

Cisco Unified Communications Manager

Equipment	Software revision	Comments
Cisco Unified Communications Manager	7.1.5.10000-12	The following TelePresence Server 2.2 features are not supported when interoperating with this version of CUCM: • Autodetection of CTS endpoints • Trunking to CUCM
Cisco Unified Communications Manager	8.5.1.10000-26	No issues found.
Cisco Unified Communications Manager	8.6.1.10000-19	No issues found. Pre-release at the time of writing.

Updating to 2.2(1.48)

Note: You **must** back up your configuration **before** upgrading to 2.2(1.48).

You must also remember the administrator user name and password for the backup configuration. You will need these if you ever need to make use of this backup file.

If you are using Call Detail Records (CDR), or any other logs, for billing, auditing or other purposes, you must download and save your logged data. When you reboot the TelePresence Server, as part of the upgrade, you will delete all existing CDRs.

Prerequisites and software dependencies

- You need the software package (zipped image file) for this version.
- You should have the true type font file if you want to improve the text rendering. You can get this file, called **ts-font**, from the same place where you download your software, i.e. http://www.cisco.com/cisco/software/release.html?mdfid=283645287&flowid=21873&softwareid=280886992&release=2.2(1.43)&relind=AVAILABLE&rellifecycle=&reltype=latest.
- You should also have any licenses and feature keys you need for the upgrade.
- Take a backup of your current configuration and software package, including logs and keys.
- Make sure you have administrative access to the Cisco TelePresence Server(s) and the Cisco TelePresence MSE 8050 Supervisor that manages their chassis.
- If you are upgrading a cluster of TelePresence Servers, you must have access to all of them and you must upgrade them all to the identical build.
- Keep a list of the model numbers and serial numbers of your devices in case you need to contact support.
- Arrange a downtime window and notify users of when the service will be unavailable. The approximate duration of an upgrade is 10 to 20 minutes.

Upgrade via the web interface

1. Unzip the image file locally.
2. Log in to your TelePresence Server's web interface with administrative credentials.
3. The username is **admin** and there is no password on a new unit.
4. Go to **Configuration > Upgrade**.
5. In the **Main software image** section, locate the **New image file** field. Browse to and select the unzipped new image file.
6. Click **Upload software image**.

   The web browser uploads the file to the TelePresence Server, which may take a few minutes.

   Note: Do not browse away from the **Upgrade** page, or refresh the page, during the upload process. The upload may fail if you do.

   The web browser refreshes automatically and displays the message *Main image upload completed*. Close the message.
7. Go to **Configuration > Shutdown**. Click **Shut down TelePresence Server**. This option will now change to **Confirm TelePresence Server shutdown**. Click to confirm.

8. Click the **Restart TelePresence Server and upgrade** button. This button only appears in the **Upgrade** page during this process.

   The unit will reboot and upgrade itself which takes a few minutes.

   Note: You may be logged out due to inactivity. If this happens, log in again, go to **Configuration > Upgrade** and click **Restart Telepresence Server and upgrade**.

9. Go to the **Status** page to verify that your TelePresence Server is using the new version.

10. Restore your configuration if necessary; refer to the online help for details.

Upgrade via FTP

1. Unzip the image file locally.
2. Connect to the TelePresence Server via ftp.
   For example, enter **ftp *IP Address*** at the command prompt, or use an FTP client with a graphical user interface.
3. Supply the administrator username and password.
   Username is **admin** without a password (on a new unit).
4. Upload the image file.
   For example, enter **put *ImageFilename*** at the ftp prompt.
5. Reboot the hardware after the upload.
   You can reboot via the upgrade page on the web interface or use the power button on the hardware.
   The unit upgrades itself when it restarts.
6. Log in to the web interface and go to the **Status** page to verify that your TelePresence Server is using the new version.
7. Restore your configuration if necessary; refer to the online help for details.

Notes

- FTP is generally more reliable for upgrades than the web interface.
- You can monitor the upgrade progress via the serial port.

Downgrade instructions

If you need to reverse your upgrade, you can re-install the former version of the software.

The downgrade procedure is the same as the upgrade procedure except you will use the earlier software image.

You need the correct version of the software and your saved configuration before you proceed.

1. Follow the upgrade procedure using the earlier software image.
2. Restart the hardware and check the status via the web interface.

   The status report indicates the software version.
3. Restore your configuration from the saved XML file.

Upgrade the font

Your TelePresence Server may be shipped with the TrueType font pre-installed. You can check this on the **Status** or **Configuration > Upgrade** page.

If the font is not present, and you want to use TrueType text rendering on your TelePresence Server instead of the default text rendering method, you must upload the font file.

Note: You should do this when the TelePresence Server is not heavily loaded. Also, you must use the supplied font; do not attempt to load a different font file.

Uploading via the web interface:

1. Browse to the TelePresence Server and log in.
2. Click the button to locate and select **ts-font** (e.g. **Browse** or **Choose file**, depending on browser used).
3. Click **Upload font**.

   After a short while, the **Font file status** changes to *Present*.

Uploading via ftp:

1. Open an ftp prompt in the local folder where you downloaded the file **ts-font**.
2. Open the TelePresence Server and log in.
3. Enter the command `put ts-font font` at the ftp prompt.

   This command copies and renames the file because the TelePresence Server expects a file called **font**.

   After a short while, the **Font file status**, on the web interface's upgrade page, changes to *Present*.

Removing the font:

1. If you want to revert to the default text rendering, click **Delete font**.
2. Confirm that you want to remove the font file.

   The Font file status changes to *Not present*.

Checking for updates and getting help

If you experience any problems when configuring or using the product, consult the online help available from the user interface. The online help explains how the individual features and settings work.

If you cannot find the answer you need, check the web site at http://www.cisco.com/cisco/web/support/index.html where you will be able to:

- make sure that you are running the most up-to-date software,
- find further relevant documentation, for example product user guides, printable versions of the online help, reference guides, and articles that cover many frequently asked questions,
- get help from the Cisco Technical Support team. Make sure you have the following information ready before raising a case:
  - The serial number and product model number of the unit (if applicable)
  - The software build number which can be found on the product user interface (if applicable)
  - Your contact email address or telephone number
  - A full description of the problem

References and related documents

Published documentation for the Cisco TelePresence Server can be found at http://www.cisco.com/en/US/products/ps11339/tsd_products_support_series_home.html.