Release Notes for Cisco Unified MeetingPlace Audio Server Release 5.4(1.4)

Published May 22, 2009

These release notes contain information on new and changed support, new and changed functionality, limitations and restrictions, and open and resolved caveats for Cisco Unified MeetingPlace Audio Server Release 5.4(1.4), and for Cisco Unified MeetingPlace MeetingTime Release 5.4(1.1).

You can access the latest software upgrades for all versions of Audio Server and MeetingTime on the Cisco Software Center website at http://www.cisco.com/kobayashi/sw-center/sw-voice.shtml.

Note To access the software download page, you must be logged on to Cisco.com as a registered user.

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Introduction

Cisco Unified MeetingPlace Audio Server is the software that runs the Cisco Unified MeetingPlace system. The software is installed on either a Cisco Unified MeetingPlace 8106 or a Cisco Unified MeetingPlace 8112 hardware server. Additional software components—such as Cisco Unified MeetingPlace Web Conferencing and Cisco Unified MeetingPlace for Microsoft Outlook—are installed on a Cisco Media Convergence Server (MCS).

MeetingTime is the desktop software that allows system administrators to access and use the Audio Server system functions from customer-provided Windows computers.

System Requirements

- Requirements for Cisco Unified MeetingPlace Audio Server Release 5.4, page 2
- Compatibility Information, page 2
- Determining the Software Version, page 2

Requirements for Cisco Unified MeetingPlace Audio Server Release 5.4

System Requirements for Cisco Unified MeetingPlace Release 5.4 contains the most current information on Audio Server requirements. The document is available at http://www.cisco.com/en/US/products/sw/ps5664/ps5669/prod_installation_guides_list.html.

Compatibility Information

For information about the compatibility of Audio Server Release 5.4 with other Cisco Unified MeetingPlace components, refer to *Compatibility Matrix: Cisco Unified MeetingPlace Components* at http://www.cisco.com/en/US/products/sw/ps5664/ps5669/products_device_support_tables_list.html.

Determining the Software Version

- Cisco Unified MeetingPlace Audio Server, page 2
- Cisco Unified MeetingPlace Gateways and Services, page 3

Cisco Unified MeetingPlace Audio Server

To Determine the Cisco Unified MeetingPlace Audio Server Version in Use

Step 1 Open the command line interface (CLI) window for the Cisco Unified MeetingPlace Audio Server.

Step 2 Enter **release**.

The Audio Server version is displayed.

Cisco Unified MeetingPlace Gateways and Services

Caution If Cisco Unified MeetingPlace components have been upgraded recently, you must restart the Cisco Unified MeetingPlace Gateway System Integrity Manager (GWSIM). Otherwise, the version numbers may not be accurate.

To Determine the Cisco Unified MeetingPlace Gateway Versions and Services That Are in Use

Step 1 Open the command line interface (CLI) window for the Cisco Unified MeetingPlace Audio Server.

Step 2 Enter **gwstatus**.

The Cisco Unified MeetingPlace gateway versions and services that are in use are displayed.

Related Documentation

For descriptions and locations of Cisco Unified MeetingPlace documentation on Cisco.com, see the *Documentation Guide for Cisco Unified MeetingPlace*. The document is shipped with Cisco Unified MeetingPlace and is available at http://www.cisco.com/en/US/products/sw/ps5664/ps5669/products_documentation_roadmaps_list.html.

New and Changed Requirements and Support—Release 5.4(1.4)

This section contains information about new and changed requirements and support in the Cisco Unified MeetingPlace Audio Server Release 5.4(1.4) time frame only. Refer to the release notes of the applicable version for information on new and changed support with earlier versions of Audio Server. Release notes for all versions of Audio Server are available at http://www.cisco.com/en/US/products/sw/ps5664/ps5669/prod_release_notes_list.html.

Audio Server Documentation

The section lists new product documentation available with this release.

Compatibility Matrix for Cisco Unified MeetingPlace Components

For the most current list of supported version combinations of Cisco Unified MeetingPlace Audio Server and the Cisco Unified MeetingPlace components, refer to *Compatibility Matrix: Cisco Unified MeetingPlace Components* at http://www.cisco.com/en/US/products/sw/ps5664/ps5669/products_device_support_tables_list.html.

The document replaces information in the "Cisco Unified MeetingPlace Component Compatibility Matrix" section in the "Introducing Cisco Unified MeetingPlace" chapter of the *Installation and Upgrade Guide for Cisco Unified MeetingPlace*.

Configuration of Extended RSNA Prefixes Supported

Cisco Unified MeetingPlace Audio Server supports the configuration of extended RSNA prefixes indicating specific Audio Server/codec support combinations. With extended prefixes, you can control the codec that is used in redirecting calls from one server to another.

RSNA allows multiple Cisco Unified MeetingPlace Audio Servers to appear as one server to the user community. All users who host (as a profile user) or attend (as a profile user or as a guest) a reservationless meeting can access the meeting by dialing the phone number of the server that is local to that user, regardless of which server is hosting the meeting. Users are then redirected to the server that is hosting the meeting.

For information on using extended prefixes, see the "Administration Guide for Cisco Unified MeetingPlace Audio Server: About RSNA" section on page 13 under "Documentation Updates: Changes."

Video Integration Upgrade Required

If the system has Cisco Unified MeetingPlace Video Integration installed, you must upgrade to Video Integration Release 5.4(107).

Web Conferencing Upgrade Required

If the system has Cisco Unified MeetingPlace Web Conferencing installed, you must upgrade to Web Conferencing Release 5.4(154).

New Functionality—Release 5.4(1.4)

There is no new functionality for Cisco Unified MeetingPlace Audio Server Release 5.4(1.4). See the "Resolved Caveats—Audio Server Release 5.4(1.4) and MeetingTime Release 5.4(1.1)" section on page 8.

Refer to the release notes of the applicable version for information on new functionality in earlier versions of Audio Server. Release notes for all versions of Audio Server are available at http://www.cisco.com/en/US/products/sw/ps5664/ps5669/prod_release_notes_list.html.

Changed Functionality—Release 5.4(1.4)

There is no changed functionality for Cisco Unified MeetingPlace Audio Server Release 5.4(1.4). See the "Resolved Caveats—Audio Server Release 5.4(1.4) and MeetingTime Release 5.4(1.1)" section on page 8.

Refer to the release notes of the applicable version for information on changed functionality in earlier versions of Audio Server. Release notes for all versions of Audio Server are available at http://www.cisco.com/en/US/products/sw/ps5664/ps5669/prod_release_notes_list.html.

Installation and Upgrade Information

- Installing Cisco Unified MeetingPlace Audio Server Release 5.4 for the First Time, page 5
- Installing or Upgrading to MeetingTime Release 5.4, page 5
- Upgrading to Cisco Unified MeetingPlace Audio Server Release 5.4, page 6

Installing Cisco Unified MeetingPlace Audio Server Release 5.4 for the First Time

Cisco Systems ships new Cisco Unified MeetingPlace Audio Server systems with Audio Server software Release 5.4 already installed.

For instructions on installing the hardware, refer to the *Installation and Upgrade Guide for Cisco Unified MeetingPlace Audio Server Release 5.4* at http://www.cisco.com/en/US/products/sw/ps5664/ps5669/prod_installation_guides_list.html.

Installing or Upgrading to MeetingTime Release 5.4

You must first download the MeetingTime software and decompress it.

To Download and Decompress MeetingTime Release 5.4 Software

Step 1 Go to the Cisco Software Center web site at http://www.cisco.com/kobayashi/sw-center/sw-voice.shtml.

Step 2 In the Cisco Unified MeetingPlace section, click the **Cisco Unified MeetingPlace 5.4** link.

Step 3 Download the file for the current version of MeetingTime.

Step 4 Using any commercially available ISO file unpackaging program, expand the ISO file to the directory of your choice on your local hard disk.

If you cannot find a program to expand ISO files, you can use any commercially available CD-ROM burner program, and burn the file onto a CD.

Step 5 Confirm that the file **SetupMeetingTime<version>.exe** was created (on the local disk or optional CD-ROM).

If you do not see the file, contact Cisco Network Consulting Engineering (NCE) for assistance.

For instructions on installing and upgrading MeetingTime, refer to the "Installing MeetingTime" appendix of the *Administration Guide for Cisco Unified MeetingPlace Audio Server Release 5.4* at http://www.cisco.com/en/US/products/sw/ps5664/ps5669/prod_maintenance_guides_list.html.

Upgrading to Cisco Unified MeetingPlace Audio Server Release 5.4

Upgrading to Cisco Unified MeetingPlace Audio Server Release 5.4 is supported from Releases 5.3 and 5.2.

You must first download the Audio Server software and make a CD from which the Cisco Unified MeetingPlace 8100 series server can upgrade the software.

To Download Audio Server Release 5.4 Software and Make a CD

Step 1 Go to the Cisco Software Center website at http://www.cisco.com/kobayashi/sw-center/sw-voice.shtml.

Step 2 In the Cisco Unified MeetingPlace section, click the **Cisco Unified MeetingPlace 5.4** link.

Step 3 Download the file for the current version of Audio Server.

Step 4 Using any commercially available CD-ROM burner program, burn the ISO image onto a CD. Do not exceed a speed of 8x when burning the image onto the CD.

Step 5 After burning the CD, confirm that the directory structure **\UPDATE\<version>** was created.

If you do not see the structure, contact Cisco Network Consulting Engineering (NCE) for assistance.

For instructions on upgrading a Cisco Unified MeetingPlace 8100 series server, see the "Upgrading the Cisco Unified MeetingPlace Audio Server Software" chapter of the *Installation and Upgrade Guide for Cisco Unified MeetingPlace Audio Server Release 5.4* at http://www.cisco.com/en/US/products/sw/ps5664/ps5669/prod_installation_guides_list.html.

Limitations and Restrictions

- QSIG Support, page 6
- Meeting Attachments and User Profiles, page 6

QSIG Support

In Cisco Unified MeetingPlace Release 5.3 and later, QSIG is available only in the unshifted timeslot mode; that is, the B-channels must be numbered 1 to 15 and 17 to 31. The B-channel 16 is not available. If your version of QSIG does not allow this, you can configure your PBX to use ETSI ISDN (non-QSIG) instead.

Meeting Attachments and User Profiles

When a meeting is scheduled with attachments, users will receive the attachments in their e-mail notifications for the meeting even though the Include Attachments attribute in their user profiles is set to No (not to receive attachments with meeting notifications).

The Include Attachments attribute in user profiles is available on the MeetingTime Configure tab under Receiving Notifications.

Caveats

This section lists Severity 1, 2, and 3 caveats.

You can find the latest caveat information for Cisco Unified MeetingPlace Audio Server version 5.4(1.4)—in addition to caveats of any severity for any release—by using Bug Toolkit, an online tool available for customers to query defects according to their own needs. Bug Toolkit is available at http://www.cisco.com/pcgi-bin/Support/Bugtool/launch_bugtool.pl. For information on using Bug Toolkit, see the "Using Bug Toolkit" section on page 10.

Note To access Bug Toolkit, you must be logged on to Cisco.com as a registered user.

This section contains caveat information for Audio Server Release 5.4(1.4) and MeetingTime Release 5.4(1.4) only. Refer to the release notes of the applicable version for caveat information for earlier versions of Audio Server and MeetingTime. Release notes for all versions of Audio Server are available at http://www.cisco.com/en/US/products/sw/ps5664/ps5669/prod_release_notes_list.html.

Open Caveats—Audio Server Release 5.4(1.4) and MeetingTime Release 5.4(1.1)

(Caveats are listed in order by component, then by caveat number.)

Table 1 Cisco Unified MeetingPlace Audio Server 5.4(1.4) and MeetingTime 5.4(1.1) Open Caveats

Caveat Number	Component	Description
CSCsa36524	meetingtime	Spkr detection display errs by MeetingTime (poss chan 0-11) w/ T1 PRI
CSCsa69318	meetingtime	Can't use GUI with keyboard only with Meeting Time 5.2, 5.3
CSCse41754	meetingtime	MeetingTime Receptionist Whiteboard Update Failure
CSCsg77916	meetingtime	MeetingTime loses connection upon moving callers to waiting room
CSClt22404	mp-server	C3 - H323 RAS meeting message recording problem
CSClt23223	mp-server	Participant does not get notification for canceled mtgs he was invited
CSCsa32358	mp-server	Meeting Recording Missing after an unsuccessful prompt
CSCsa35611	mp-server	IVR: Receive the list of comments and attachment feature not working
CSCsa35637	mp-server	IVR: Can not change purge date to a future date
CSCsa35643	mp-server	delete all recording, comments and attachments does not function
CSCsa35651	mp-server	IVR= unable to change alternate notification method
CSCsa37810	mp-server	Disable allow 3rd party initiation does not take affect-2key
CSCsa51569	mp-server	Back-end / No-Show SMA/ Cancellation notification is not sent
CSCsa99055	mp-server	"System will call" and "Schedule immediate meeting from VUI"
CSCsb73833	mp-server	0-out works from breakout session but shouldn't

Table 1 *Cisco Unified MeetingPlace Audio Server 5.4(1.4) and MeetingTime 5.4(1.1) Open Caveats (continued)*

Caveat Number	Component	Description
CSCsc01057	mp-server	Non-profiled Users Able to Login to Profile-only Meeting
CSCsc82347	mp-server	Users cannot record their name when entering meeting
CSCse30011	mp-server	Need to always send video extension for meetings scheduled with video
CSCse42486	mp-server	Spanish prompt asking users to enter their profile password gets cut off
CSCse45848	mp-server	Mtgs with auto-start recording show false recording state with 1st party
CSCse64501	mp-server	Admin password reset back to "cisco" after upgrade to MP 5.4
CSCsf30159	mp-server	Lecture style meeting. MP audio link is muted into MCU
CSCsg50560	mp-server	password may be exposed when upgrading via Remote File ftp
CSCsg51884	mp-server	MT user parameter, Attend preference for video prof. Outdial not working
CSCsh30808	mp-server	Long schedule delay in load test when meeting are dropping and joining
CSCsh37393	mp-server	CUPC/MP video: Unable to mute video participant
CSCsh57350	mp-server	Root permission changed to zero (Protection violation)
CSCsh88771	mp-server	"Find me" feature does not work in Immediate or Reservationless meeting
CSCsi00573	mp-server	MP Server restart/segmentation fault in a runtime library
CSCsi09760	mp-server	Hardware watchdog timeout due to OS issues

Resolved Caveats—Audio Server Release 5.4(1.4) and MeetingTime Release 5.4(1.1)

(Caveats are listed in order by severity, then by component, then by caveat number.)

Table 2 *Cisco Unified MeetingPlace Audio Server 5.4(1.4) and MeetingTime 5.4(1.1) Resolved Caveats*

Caveat Number	Severity	Component	Description
CSCsf17886	1	mp-server	MP Audio Server restarts when moving lecture meeting attendees from WR
CSCsg36585	1	mp-server	System restarts when lecture mtg guest moved to Q&A queue with MtgTime
CSCsg62708	1	mp-server	ConfSchd deadbeef during Continuous Meeting outdial.
CSCsg98082	1	mp-server	Conference Scheduler crashes under RSNA load
CSCsh14484	1	mp-server	Sched fails with alloc error on optimized reservationless system
CSCsh83181	1	mp-server	VUI crashes during setup of a reservationless meeting
CSCsd45453	2	mp-server	Users getting silence when dialing in on certain MP ports for IP calls
CSCse65319	2	mp-server	Users cannot create conferences, MAJ 0x3008d
CSCsg29506	2	mp-server	Flex Menus - get profile state bypasses expired passwords.
CSCsg61200	2	mp-server	Video Portion of MeetingPlace Meeting Fails to auto-extend
CSCsg67593	2	mp-server	Incorrect prompt played in reservationless meeting
CSCsh15930	2	mp-server	Remote Server part entry fails with memory allocation error

Table 2 Cisco Unified MeetingPlace Audio Server 5.4(1.4) and MeetingTime 5.4(1.1) Resolved Caveats (continued)

Caveat Number	Severity	Component	Description
CSCsh21941	2	mp-server	Non-VUI reservationless sched fails with "not found" (157) error
CSCsh36649	2	mp-server	Audio server upgraded to 5.4 version crashes with ENOMEM errors
CSCsh38164	2	mp-server	RSNA intermittently failing
CSCsh44073	2	mp-server	Stop using RSH
CSCsh51737	2	mp-server	System startup script executes world-writable scripts
CSCsh52670	2	mp-server	smart blade low in available timeslots
CSCse40832	3	meetingtime	Contacts are not switched correctly between user and group defaults
CSCsg21243	3	meetingtime	VUI doesn't use Allow Internet Access profile setting
CSCsg97838	3	meetingtime	Allow Internet Access not correct in sched details or review tab
CSCsg21932	3	mp-docs-audio	UseRsvnless field needed for raw profile report
CSClt23121	3	mp-server	formtype=Query returns error template always after error condition
CSCse16335	3	mp-server	MP outdialing to original phone number when selecting disconnect option
CSCse78152	3	mp-server	Disconnecting an outdial to an autoanswer device caused issues ...
CSCse87389	3	mp-server	Cannot modify invitee notification settings when resched recurr mtg
CSCse90996	3	mp-server	A problem with an active DSP is reported as an Unused DSP
CSCse93676	3	mp-server	Can't add two site at once in MSM setup after deleting same sites.
CSCse95549	3	mp-server	Resched recurr mtgs logs "Trouble reading conf. Conf = 6342, Stat = 2,..
CSCsf31553	3	mp-server	Cptrace -C Delete by xxxx does not reflect real user who deleted mtg
CSCsg00929	3	mp-server	displayname update does not take effect consistently
CSCsg15536	3	mp-server	display name does not show correctly on MPWeb after multiple ANI events
CSCsg31923	3	mp-server	large unique ID causes incorrect cptrace -D
CSCsg36617	3	mp-server	Participant list in web meeting room is missing video participants
CSCsg39956	3	mp-server	ANI is not displayed for guests dialing into a meeting
CSCsg54338	3	mp-server	ConfPartPool needs to account for video
CSCsg57054	3	mp-server	Slow prompts in large reservationless configuration
CSCsg57143	3	mp-server	Shadow server fails DB login after network connection lost
CSCsg66104	3	mp-server	System restarted when a report was run
CSCsg72391	3	mp-server	ConfSchd sometimes calculates video extension time incorrectly
CSCsg80801	3	mp-server	Expired profile password change on a Reservationless system did not work
CSCsg93357	3	mp-server	ports on span unresponsive, incoming calls rejected
CSCsg93832	3	mp-server	Video parties dropped if no audio join before disc empty port time
CSCsh23836	3	mp-server	Cannot start Immediate meeting with reservationless=NO and outdial=NO
CSCsh23883	3	mp-server	Cannot add password in Reservationless meeting with password=Yes

Table 2 Cisco Unified MeetingPlace Audio Server 5.4(1.4) and MeetingTime 5.4(1.1) Resolved Caveats (continued)

Caveat Number	Severity	Component	Description
CSCsh37617	3	mp-server	Web Server enabled on MA Blade
CSCsh42039	3	mp-server	RSNA transfer failed, erc=3173
CSCsh47085	3	mp-server	Remove extra entries from vpunit .rhosts file
CSCsh48201	3	mp-server	Remove /test_scripts from root path
CSCsh50834	3	mp-server	server restarts when Web part leaves conf during reserved sched request
CSCsh51786	3	mp-server	Meeting events not sent to breakout rooms
CSCsh63010	3	mp-server	Error When Attempting to Change Announcement Options
CSCsh71034	3	mp-server	Return meeting ID for try-before-buy request
CSCsh77914	3	mp-server	Create new language pack CD image
CSCsh87022	3	mp-server	User is incorrectly displayed in Waiting room

Using Bug Toolkit

To access Bug Toolkit, you need an Internet connection, web browser, and Cisco.com user ID and password. For more detailed information on using Bug Toolkit, click Help in any Bug Toolkit window.

To Use Bug Toolkit

Step 1 Open your web browser and go to http://www.cisco.com/cgi-bin/Support/Bugtool/launch_bugtool.pl.

Step 2 Click the **Launch Bug Toolkit** link.

Step 3 To look for information about a specific caveat, enter the ID number in the Enter Known Bug ID field.

To view all caveats for a Cisco Unified MeetingPlace component, go to the "Search for Bugs in Other Cisco Software and Hardware Products" section, and enter **meetingplace** in the Product Name field.

Step 4 In the list, select **Cisco Unified MeetingPlace**, then click **Next**.

Step 5 On the Cisco Unified MeetingPlace search page, set options to limit your search results. You can choose any or all of the available options:

a. Choose the applicable Cisco Unified MeetingPlace version:
- Choose the major version for the major releases (for example, 5.3). A major release contains significant new features, enhancements, architectural changes, and/or defect fixes.
- Choose the revision for more specific information. A revision (maintenance) release primarily contains defect fixes to address specific problems, but it may also include new features or enhancements.

b. Choose the applicable features or components. Make your selection from the Available list and click **Add** to place your selection in the Limit Search To list.

c. Enter keywords for which to search caveat titles and descriptions.

Note To make queries less specific, use the All wildcard for the major version/revision, features/components, and keyword options.

d. Choose the applicable advanced options:
   - Bug Severity—The default specifies severity levels 1 through 3.
   - Bug Status Group—Check the **Fixed** check box for resolved caveats.
   - Release Note Enclosure—The default specifies Valid Release Note Enclosure.

Step 6 Click **Next**. Bug Toolkit returns a list of caveats.

Note You can modify your results by submitting another query and using different criteria. You can also save your query for future use.

Documentation Updates: Errors

This section lists errors in the current Cisco Unified MeetingPlace Audio Server documentation and gives corrected information. The correct information will be incorporated in a future documentation release, or as otherwise noted.

Administration Guide for Cisco Unified MeetingPlace Audio Server: About Planning for Outages

In the "About Planning for Outages" section in the "Managing and Maintaining Cisco Unified MeetingPlace" chapter, disregard the caution.

Administration Guide for Cisco Unified MeetingPlace Audio Server: Raw Participant Join Leave Report

In the "Raw Participant Join Leave Report" section in the "Raw Data Export Specifications" appendix, added to the entry for the "Type of Value, and Size" column for the Device field name as follows (the new information is in bold):

Field Name	Description	Type of Value, and Size
Device	Device and port number used to join the meeting	0-1151 (voice port) 4080 (MeetingTime) 4081 (Cisco Unified MeetingPlace Web) 4082 (Cisco Unified MeetingPlace Web Conferencing option—to enable data conferencing) 4084 (Cisco Unified MeetingPlace Video) **4086 (Cisco Unified MeetingPlace Web meeting room)**

Administration Guide for Cisco Unified MeetingPlace Audio Server: Configuring T1 PRI Port Groups

In the "Configuring a T1 PRI Cisco Unified MeetingPlace System" section on page 1-14 in the "Configuring the Cisco Unified MeetingPlace Audio Server" chapter, added the following note to step 2:

Note Using the **blade -p** command automatically sets the port group default to port group 3 and sets the protocol table default to 2 (ATT). However, if you want to select another protocol, such as Bell PRI, you will need to use the **port** command to configure port group 3 to select a different protocol. This is described in the "Configuring Port Groups" section on page 1-17.

In the "Configuring Port Groups" section on page 1-17 in the "Configuring the Cisco Unified MeetingPlace Audio Server" chapter, in step 2, changed the second line in the example to this:

```
Enter port group record number [0...31, <cr> for all] : 3
```

Added the following note after step 2:

Note Be sure to enter the correct port group number. The port group numbers are as follows: 0 (T1 CAS), 1 (IP), 2 (E1), and 3 (T1 PRI).

Administration Guide for Cisco Unified MeetingPlace Audio Server: Configuring E1 Port Groups

In the "Configuring an E1 Cisco Unified MeetingPlace System" section on page 1-25 in the "Configuring the Cisco Unified MeetingPlace Audio Server" chapter, added the following note to step 2:

Note Using the **blade -e** command automatically sets the port group default to port group 2 and sets the protocol table default to zero, which is correct for E1 (Euro ISDN). However, if you want to select another protocol, such as QSIG, you will need to use the **port** command to configure port group 2 to select a different protocol. This is described in the "Configuring Port Groups" section on page 1-29.

In the "Configuring Port Groups" section on page 1-29 in the "Configuring the Cisco Unified MeetingPlace Audio Server" chapter, in step 2, changed the second line in the example to this:

```
Enter port group record number [0...31, <cr> for all] : 2
```

Added the following note after step 2:

Note Be sure to enter the correct port group number. The port group numbers are as follows: 0 (T1 CAS), 1 (IP), 2 (E1), and 3 (T1 PRI).

Configuration Guide for Cisco Unified MeetingPlace Audio Server: Configuring a Pure IP Cisco Unified MeetingPlace System

In the "To Configure a Pure IP Cisco Unified MeetingPlace System" procedure in the "Configuring a Pure IP Cisco Unified MeetingPlace System" section in the "Configuring the Cisco Unified MeetingPlace Audio Server System" chapter, the note in Step 6 gives the incorrect port number.

The correct version of the second sentence of the note is:

> For a Cisco Unified MeetingPlace 8106, the blade slot being configured is slot 6, the number of ports is 120, and the first port is 0.

The note in Step 10 also gives the incorrect port number. The correct version of the second sentence of the note is:

> For a Cisco Unified MeetingPlace 8106, the blade slot being configured is slot 6 and the first port is 0.

Documentation Updates: Changes

This section lists changes to the current Cisco Unified MeetingPlace Audio Server documentation. The changed information will be incorporated in a future documentation release, or as otherwise noted.

Administration Guide for Cisco Unified MeetingPlace Audio Server: About RSNA

In the "Setting Up Cisco Unified MeetingPlace Audio Server" chapter of the *Administration Guide for Cisco Unified MeetingPlace Audio Server Release 5.4*, disregard the "About RSNA" section and use the following information instead, which also includes information about using extended prefixes.

Introduction

RSNA allows multiple Cisco Unified MeetingPlace Audio Servers to appear as one server to the user community. All users who host (as a profile user) or attend (as a profile user or as a guest) a reservationless meeting can access the meeting by dialing the phone number of the server that is local to that user, regardless of which server is hosting the meeting. Users are then redirected to the server that is hosting the meeting.

You can also configure extended RSNA prefixes indicating specific Audio Server/codec support combinations. With extended prefixes, you can control the codec that is used in redirecting calls from one server to another.

If you are setting up RSNA for the first time, see the following sections:

- RSNA Requirements, page 14
- Extended RSNA Prefix Configuration Example, page 15 (if you are using extended prefixes)
- Setting Up RSNA for the First Time, page 17
- Troubleshooting RSNA, page 23

If you are adding extended prefixes to an existing RSNA configuration, see the following sections:

- Extended RSNA Prefix Requirements, page 14

- Extended RSNA Prefix Configuration Example, page 15
- Adding Extended Prefixes to an Existing RSNA Configuration, page 21
- Troubleshooting RSNA, page 23

RSNA Requirements

- RSNA requires that all voice communication with the Cisco Unified MeetingPlace server uses VoIP and SIP. Any PSTN or H.323 end points must go through a SIP integration to communicate with the system. This implementation depends on the use of the SIP "REFER method" call transfer mechanism, as specified in RFC 3515. All participating SIP end points, including a Cisco Unified MeetingPlace H.323/SIP IP Gateway server, any SIP integrations that communicate with Cisco Unified MeetingPlace, and any phones or other terminals that communicate directly with the system by using SIP must support the SIP REFER method.

  A SIP phone that does not support the REFER method does not transfer correctly and cannot attend meetings that require a transfer. In this situation, users are asked to dial the phone number of the hosting server, as if they entered an unrecognized meeting ID.
- System administrators must ensure the following:
  - All participating Cisco Unified MeetingPlace servers have synchronized user databases
  - All user profiles for active users are present on all systems
- Use Cisco Unified MeetingPlace Directory Services to synchronize user databases across multiple Cisco Unified MeetingPlace systems. The following fields in each user profile on all servers must be set identically:
  - User ID
  - Scheduling Home Server
  - This configuration assumes that profile numbers are assigned across the system (for example, profile ID 1234 on servers A and B both refer to the same person). This configuration also assumes that either server can authenticate the user for both servers.
  - Passwords must be synchronized. If the password is not the same across the system, the user is provisionally authenticated (and must reenter the password for access as a profile user after exiting the meeting). Also, if users want to rerecord their name on a remote server, they need to know their password on that server.
- If you are using extended RSNA prefixes, all requirements for extended prefixes. See the following section, "Extended RSNA Prefix Requirements."

Extended RSNA Prefix Requirements

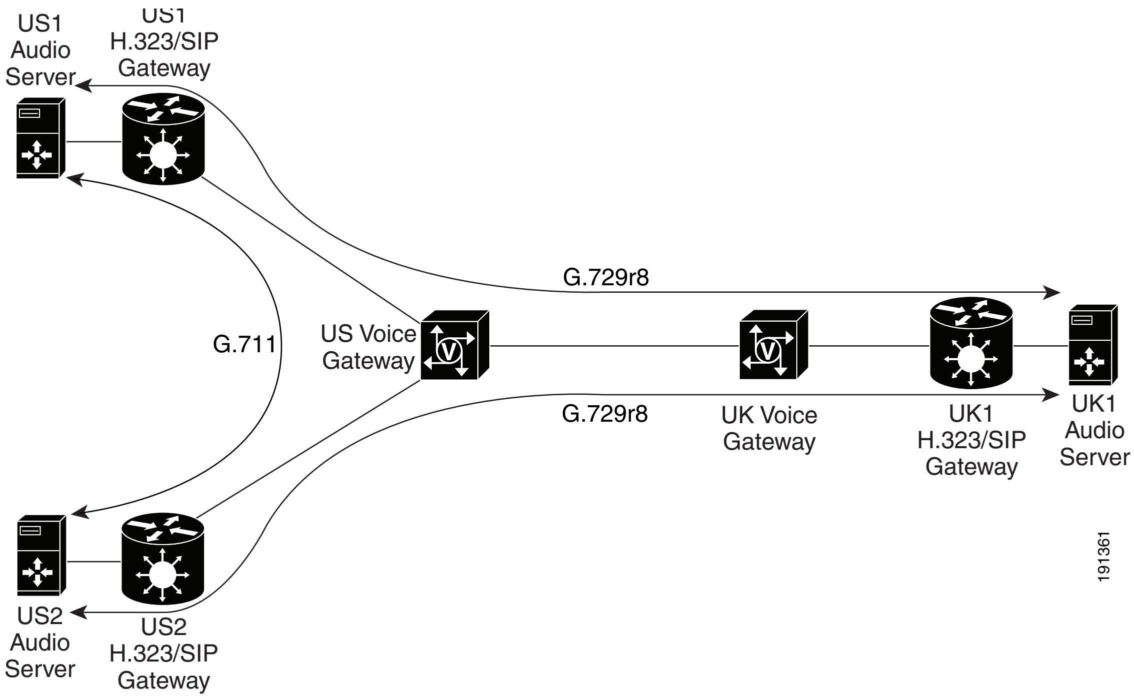
- The following Cisco Unified MeetingPlace component versions:
  - MeetingTime 5.4(1.1)
  - H.323/SIP IP Gateway 5.3(1.4)
- Voice gateways involved in the RSNA transfer must support transcoding and run IOS 12.4.9T or a later release.
- All requirements for the RSNA feature. See the preceding section, "RSNA Requirements."

Extended RSNA Prefix Configuration Example

Extended RSNA prefixes allow you to specify the codec that is used during the call transfer by using a codec configuration prefix value in the dial peers on each voice gateway. In MeetingTime, you configure a server record on each Cisco Unified MeetingPlace Audio Server for every other server that will participate in RSNA transfers. In the server record, you specify a value for the Extended RSNA Codec, and MeetingTime returns an extended prefix value. You use the extended prefix value as the destination pattern when configuring the dial peers for transferring calls to that server. In each dial peer, you also specify which codec to use on the transferred calls.

The example in Figure 1 shows a topology where there are two Cisco Unified MeetingPlace Audio Servers in the United States and one in the United Kingdom. In this scenario, calls that are transferred between the U.S. servers (US1 and US2) should use the G.711 codec, and calls that are transferred between the U.S. servers and the UK server (or vice versa) should use the G.729r8 codec.

Figure 1 RSNA Transfer Paths Utilizing Different Codecs



On the Audio Servers in this example, configuring an Extended RSNA Codec in MeetingTime for every other server results in the extended prefix values shown in Table 3.

Table 3 Extended Prefix Values for Other Servers on US1

Server	Extended Prefix
US1	0101
US2	0201
UK1	0301

Note that the actual codec used when transferring calls depends on the dial-peer configuration on the gateway, not on the value of the Extended RSNA Codec field. Any codec (other than None) can be chosen for the Extended RSNA Codec field in order to generate the extended prefix; however, the codec you choose for a given server should be consistent across all of the Audio Servers in order to generate the same extended prefix for that server (for example, use G.711uLaw when configuring the Other Servers record for US1 on both US2 and UK1).

Given these prefix values, the U.S. voice gateway would be configured with dial peers such as the following to match the extended prefixes:

```
dial-peer voice 210 voip
 description rsna_refer_to_us1
 destination-pattern 0101
 session protocol sipv2
 session target ipv4:10.10.10.1
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
dial-peer voice 220 voip
 description rsna_refer_to_us2
 destination-pattern 0201
 session protocol sipv2
 session target ipv4:10.10.10.2
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
dial-peer voice 230 voip
 description rsna_refer_to_UK1
 destination-pattern 0301
 session protocol sipv2
 session target ipv4:10.10.10.3
 dtmf-relay rtp-nte
 codec g729r8
 no vad
!
```

The outgoing dial peers on the UK voice gateway will look something like this:

```
dial-peer voice 210 voip
 description rsna_refer_to_us1
 destination-pattern 0101
 session protocol sipv2
 session target ipv4:10.10.10.1
 dtmf-relay rtp-nte
 codec g729r8
 no vad
!
dial-peer voice 220 voip
 description rsna_refer_to_us2
 destination-pattern 0201
 session protocol sipv2
 session target ipv4:10.10.10.2
 dtmf-relay rtp-nte
 codec g729r8
 no vad
!
```

Setting Up RSNA for the First Time

Note If the system is already configured to use RSNA and you are adding extended prefixes, see the "Adding Extended Prefixes to an Existing RSNA Configuration" section on page 21 instead.

Setting up RSNA involves configuring settings on the Cisco Unified MeetingPlace Audio Servers to enable RSNA and adding dial peers to the Cisco voice gateways in your network to handle RSNA transfers.

This section contains the following information:

- Enabling RSNA, page 17
- Configuring a Cisco Voice Gateway for RSNA Transfers, page 20

Enabling RSNA

Do the three procedures in this section in the order listed to enable single number access on the Cisco Unified MeetingPlace Audio Servers.

Note *For Cisco Unified MeetingPlace 8112 only:* Setting the RSNA Enabled attribute to No on a Cisco Unified MeetingPlace 8112 applies only to the server that is transferring a call (not receiving a call).

In the first procedure, you create an Other MeetingPlace Servers record for each remote server that will participate in RSNA.

Caution Do not create an Other MeetingPlace Servers record with an ID Number that matches the Scheduling Home Server value for users homed on the local server.

To Create Other Cisco Unified MeetingPlace Servers Records

Step 1 In the MeetingTime Configure tab, select the **Other MeetingPlace Servers** view.

Step 2 Create a new record with the following values:

Attribute	Value
Name	Enter the name of the remote server.
ID Number	Enter a number between 101 and 32767. This number must match the value of the Scheduling Home Server attribute set in the profile for users homed on the remote server. **Note** Setting this field to 0 (zero) prevents this record from being used. This field must contain a value greater than 100. For example, 102 and 121 are valid values; however, 007 is invalid.
Phone Number	Access phone number for the remote server (used to play a prompt if a transfer fails).
Ethernet Address	Enter the Ethernet address of the remote server (the same address used to generate license keys on that server).
VoIP Gateway IP Address 1	Enter the IP address of the remote server VoIP gateway. **Note** Must be a valid and functioning address.

Attribute	Value
(Optional) VoIP Gateway IP Address 2	Enter the IP address of a second remote server VoIP gateway, if there is one. **Note** Must be a valid and functioning address. **Note** When configuring one VoIP gateway, enter the IP address of the gateway for VoIP Gateway IP Address 1, and leave VoIP Gateway IP Address 2 blank. When configuring two VoIP gateways, enter different IP addresses for VoIP Gateway IP Address 1 and VoIP Gateway IP Address 2. Do not enter a value for VoIP Gateway IP Address 2 if VoIP Gateway IP Address 1 is blank or set to 0.0.0.0.
Will Accept SNA Transfers?	Choose **Yes**.
Extended RSNA Codec	Choose a codec in order to generate an Extended RSNA Prefix value for the remote server. If you do not want to use the Extended RSNA Prefix feature, choose **None**. After you choose the codec and save your changes, MeetingTime displays a value in the Extended Prefix field. You will use this value when configuring dial peers for the RSNA transfer on the voice gateway. **Note** The actual codec used when transferring calls depends on the dial-peer configuration on the gateway, not on the value of the Extended RSNA Codec field. Any codec (other than None) can be chosen for the Extended RSNA Codec field in order to generate the extended prefix; however, the codec you choose for this server record should be consistent with the value you choose for the remote server in MeetingTime on other Audio Servers in order to generate the same extended prefix on each Audio Server.

Step 3 Repeat Step 1 and Step 2 for each participating server to create Other MeetingPlace Servers records for all remote servers.

Step 4 Click **Save Changes**.

Make sure that all ID Number values and Scheduling Home Server values are identical across participating servers.

In the second procedure, you turn on the RSNA transfer mechanism by enabling the system for RSNA.

To Enable the System for RSNA

Step 1 In the MeetingTime Configure tab, select the **Usage Parameters** view.

Step 2 Scroll to the **SNA Settings** attributes. For the RSNA Enabled attribute, choose **Yes**.

Step 3 Click **Save Changes**.

In the third procedure, you enable remote users to use the RSNA feature by setting attributes in the MeetingTime Configure tab. Each user is assigned to a specific server identified by the Scheduling Home Server value in the user profile. This value is used to look up an Other MeetingPlace Servers record (which you created in the "To Create Other Cisco Unified MeetingPlace Servers Records" procedure on page 17).

To Configure Users and Groups for RSNA

Step 1 In the MeetingTime Configure tab, select the **User Profile** or **User Group** view.

Step 2 For the Scheduling Home Server attribute (in the Identification attributes category), enter a number between 101 and 32767 that matches the ID Number value set in Step 2 in the "To Create Other Cisco Unified MeetingPlace Servers Records" procedure on page 17.

This value must be the same as the ID Number value and must not be 0 (zero). The range should start at 101, and not 1. (For example, 1 to 32767 should instead be 101 to 32767.)

Caution Confirm that the records on the home server are configured correctly. If they are incorrect, the call will not be transferred and the reservationless meeting will be held on the local server of the user.

Step 3 Confirm that the Use Reservationless attribute (in the Restrictions attributes category) is set to either **Yes** or **Group Dflt(Yes)**.

Step 4 Click **Save Changes**.

Step 5 Confirm that the following values are set correctly:

Attribute	Value
RSNA Enabled (in the Configure tab, Usage Parameters view)	Yes For more information, see the "To Enable the System for RSNA" procedure on page 18.
Will Accept SNA Transfers (in the Configure tab, Other MeetingPlace Servers view)	Yes For more information, see the "To Create Other Cisco Unified MeetingPlace Servers Records" procedure on page 17.

You can also configure a complementary "reservationless WebConnect" service for Cisco Unified MeetingPlace Web Conferencing. For more information, refer to the *Configuration Guide for Cisco Unified MeetingPlace Web Conferencing* at http://www.cisco.com/en/US/products/sw/ps5664/ps5669/products_installation_and_configuration_guides_list.html. However, because WebConnect and RSNA work independently, you must ensure that each is configured correctly to be synchronized.

Configuring a Cisco Voice Gateway for RSNA Transfers

In this section, you add dial peers to the Cisco voice gateways in your network to handle RSNA transfers.

To Configure a Cisco Voice Gateway for RSNA Transfers

Step 1 Telnet into the router, and enter the requested login information.

Step 2 Enter **show running-config**.

The current configuration of the Cisco voice gateway displays.

Step 3 To ensure that the router sends calls from any endpoint to a particular Cisco Unified MeetingPlace H.323/SIP IP Gateway, enter the following configuration information:

.

Enter	Notes
dial-peer voice [*number*] voip	Required. [*number*] is a numeric tag that differs depending on how you have set up your voice network.
description [*text*]	Optional. [*text*] is a description of this configuration; for example, *Send call to US1 IPGW.*
destination-pattern [*0000*]	[*0000*] is the destination pattern (dialed number) in your dial plan.
session protocol sipv2	Required.
session target ipv4:[*000.00.0000*]	[*000.00.0000*] is the IP address of the Cisco Unified MeetingPlace H.323/SIP IP Gateway server to which calls will be sent.
dtmf-relay rtp-nte	Required. Allows DTMF relay using NTE RTP packets. DTMF tones are encoded in the NTE format and transported in the same RTP channel as the voice.
req-qos [*type*]	Optional. [*type*] is the type of requested Quality of Service (QOS); for example, *controlled-load.*
codec [codec]	[*codec*] is the voice-compression (codec) type; for example, *g711ulaw.*
no vad	Optional but recommended. Disables voice-activity detection.

Step 4 To set up a dial peer to handle the SIP REFER request, enter the following configuration information:

Enter	Notes
dial-peer voice [*number*] voip	Required. [*number*] is a numeric tag that differs depending on how you have set up your voice network.
description [*text*]	Optional. [*text*] is a description of this configuration; for example, *SIP REFER to US1*.
application session	Required for IOS releases prior to 12.3.

Enter	Notes
destination-pattern [*0000*]	If you are using the Extended RSNA Prefix feature, [*0000*] must match the Extended Prefix string displayed in MeetingTime on the originating Cisco Unified MeetingPlace Audio Server, as shown in the “Other MeetingPlace Servers” configuration for the server being transferred to. If you are not using the Extended RSNA Prefix feature, [*0000*] must match the RSNA Prefix configured on the Cisco Unified MeetingPlace H.323/SIP IP Gateway server.
session protocol sipv2	Required.
session target ipv4:[*000.00.0000*]	*000.00.0000* is the IP address of the Cisco Unified MeetingPlace H.323/SIP IP Gateway server to transfer calls to.
dtmf-relay [rtp-nte]	Required. Allows DTMF relay using NTE RTP packets. DTMF tones are encoded in the NTE format and transported in the same RTP channel as the voice.
req-qos [*type*]	Optional. [*type*] is the type of requested Quality of Service (QOS); for example, *controlled-load*.
codec [*codec*]	[*codec*] is the voice-compression (codec) type; for example, *g711ulaw*.
no vad	Optional but recommended. Disables voice-activity detection.

Step 5 Enter **exit**, then enter **exit** again.

Step 6 Enter **write memory** to apply the configuration settings.

Adding Extended Prefixes to an Existing RSNA Configuration

Note The instructions in this section assume the system is already set up to use RSNA. If it is not, see the “Setting Up RSNA for the First Time” section on page 17 instead. The instructions there include options for using extended prefixes.

Do the two procedures in this section in the order listed to use extended prefixes.

In the first procedure, on each Audio Server that participates in RSNA transfers, you configure the Extended RSNA Codec in MeetingTime for all other participating servers.

To Configure the Extended RSNA Codec

Step 1 In the MeetingTime Configure tab, select **Other MeetingPlace Servers**.

Step 2 Click **Query**.

Step 3 Click the forward arrow to select a server that will participate in RSNA transfers.

Step 4 For the Extended RSNA Codec field, select a codec to be used when calls are transferred to this server.

Step 5 Click **Save Changes**.

Step 6 Note the value returned in the Extended Prefix field. You will use this value when configuring a dial peer on the voice gateway.

Step 7 Repeat Step 3 through Step 6 for all remaining Audio Servers that participate in RSNA transfers.

Step 8 Repeat Step 1 through Step 7 on each remaining Audio Server that participates in RSNA transfers.

In the second procedure, for each extended prefix generated in the preceding procedure, you configure the Cisco voice gateways with dial peers to transfer calls using the proper codec.

To Configure the Voice Gateway with a Dial Peer for the Extended RSNA Prefix

---

Step 1 Telnet into the router, and enter the requested login information.

Step 2 Enter **show running-config**.

The current configuration of the Cisco voice gateway displays.

Step 3 To set up a dial peer to handle the SIP REFER request, enter the following configuration information:

Enter	Notes
dial-peer voice [*number*] voip	Required. [*number*] is a numeric tag that differs depending on how you have set up your voice network.
description [*text*]	Optional. [*text*] is a description of this configuration; for example, *SIP REFER to US1*.
destination-pattern [*0000*]	[*0000*] must match the extended prefix string displayed in MeetingTime in Step 6 of the "To Configure the Extended RSNA Codec" procedure on page 21.
session protocol sipv2	Required.
session target ipv4:[*000.00.0000*]	*000.00.0000* is the IP address of the Cisco Unified MeetingPlace H.323/SIP IP Gateway server to transfer calls to.
dtmf-relay [rtp-nte]	Required. Allows DTMF relay using NTE RTP packets. DTMF tones are encoded in the NTE format and transported in the same RTP channel as the voice.
req-qos [*type*]	Optional. [*type*] is the type of requested Quality of Service (QOS); for example, *controlled-load*.
codec [*codec*]	[*codec*] is the voice-compression (codec) type; for example, *g711ulaw*.
no vad	Optional but recommended. Disables voice activity detection.

Step 4 Repeat Step 3 for each unique extended prefix string.

Step 5 Enter **exit**, then enter **exit** again.

Step 6 Enter **write memory** to apply the configuration settings.

Troubleshooting RSNA

This section contains the following information:

- Resolving Recorded User Name Problems, page 23
- Viewing Alarm Codes for RSNA, page 23

Resolving Recorded User Name Problems

The first time that logged-in users attend a meeting hosted on each remote server, they are asked to record their name. As a result, users may have different recorded names on each server.

If users do not like the recorded name on a particular server, have them note the meeting with the undesirable recording, and tell their system administrator. The system administrator should then find that meeting, determine which server is the problem, and give users the phone number to call to access that server. Users can then enter their profile on that server and rerecord their user name.

Viewing Alarm Codes for RSNA

Table 4 describes alarms that may be generated by RSNA. These descriptions may help system administrators to troubleshoot problems.

Table 4 RSNA Alarm Codes

Alarm	Description
0x30130 (196912) "RSNA transfer loop? UserID=%d MtgOwner=%d"	(Minor alarm) RSNA transfer loop. Indicates an inter-server configuration conflict or user database synchronization problem.
0x30134 (196916) "RSNA userunknown, user=%08x%08x%08x%08x"	(Minor alarm) User identified in an RSNA transfer is unknown. Indicates a user database inter-server synchronization problem.
0x30137 (196919) "RSNA: Password mismatch, user=%08x%08x%08x%08x"	(Minor alarm) Password provided in the transfer information block does not match the local user's password. Indicates an inter-server database synchronization problem.
0x30138 (196920) "RSNA: Info block verification failed, ex=%#x"	(Minor alarm) Corrupted information block received from the transferring server. Indicates the information was corrupted in transmission, a software version mismatch, clock skew, or an attempt at forgery. Look up the secondary exception code (ex=) for more specifics.
0x901bd 16/0, "stdafx.cpp", 69 (0x10, 0, 0, 0) IPGW software exception, unit 16	(Major alarm) Incompatible software versions are in use. This can happen if an older version of the Audio Server software that does not support the Extended RSNA Prefix feature is used with a version of the Cisco Unified MeetingPlace H.323/SIP IP Gateway.

Configuration Guide for Cisco Unified MeetingPlace Audio Server: Configuring the Shadow Server

In the section called "Checking the Licenses on the Shadow Server" in the "Configuring a Cisco Unified MeetingPlace Shadow Server" chapter, deleted the note after step 9. Step 9 now says:

Step 9 Compare the output from the shadow server with the output from the primary server. The license information for the shadow server and the primary server must be identical. If the shadow server does not have the same licenses as the primary server, do not proceed. Contact Cisco TAC.

Configuration Guide for Cisco Unified MeetingPlace Audio Server: Restoring the Database Backup

In the section called "Restoring the Database Backup" in the "Configuring a Cisco Unified MeetingPlace Shadow Server" chapter, changed the note after step 5. The note now says:

Note Because the backup is taken from a live system, the **restore** command must perform various consistency checks on the database keys, voice file system, and conference reservations database. On a very large system, this can take more than 24 hours.

Documentation Updates: Omissions

This section lists new and additional information that is not included in the current Cisco Unified MeetingPlace Audio Server documentation. The new and additional information will be incorporated in a future documentation release.

Administration Guide for Cisco Unified MeetingPlace Audio Server: Raw Profile Information Data

In Table C-1 in the "Raw Profile Information Data" section in the "Raw Data Export Specifications" appendix, the following information should be included:

Field Name	Description	Type of Value, and Size
UseReservationless	Whether this user can schedule and attend reservationless meetings	Yes/No/gd

Troubleshooting Information

Troubleshooting information for Cisco Unified MeetingPlace Audio Server can be found in the *Administration Guide for Cisco Unified MeetingPlace Audio Server Release 5.4* at http://www.cisco.com/en/US/products/sw/ps5664/ps5669/prod_maintenance_guides_list.html.

Obtaining Documentation, Obtaining Support, and Security Guidelines

For information on obtaining documentation, obtaining support, providing documentation feedback, security guidelines, and also recommended aliases and general Cisco documents, 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