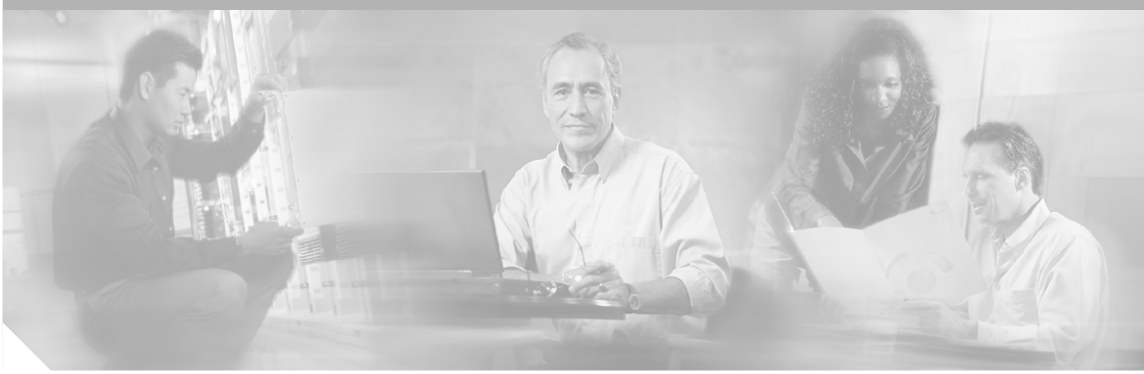

Systems Test Architecture Reference Manual for EMEA IPT

IP Communications Systems Test Release 3.0

Corporate Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
http://www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

Text Part Number: OL-6867-03

CONTENTS

Preface

Overview

Systems Test Architecture Reference Manual for EMEA IPT describes the components and configurations that have been tested and verified as part of IP Communications Systems Test Release 3.0 for EMEA Internet Protocol Telephony (IPT). This manual also includes related information for call flows, troubleshooting, failover behavior, and call load testing.

Audience

This manual is intended for system administrators who are familiar with the various hardware and software components that are included in IP Communications Systems Test Release 3.0 and that are discussed in this manual. It assumes that readers have the technical and product knowledge to install, configure, manage, and troubleshoot the systems described.

Organization

This manual is organized as follows:

Chapter 1, "Tested Scenarios and Site Models"	Describes the tested IPT site scenarios and the site models that make up these scenarios; includes topology diagrams and lists of the hardware and software components in each site model
Chapter 2, "Cisco CallManager Configuration"	Provides an overview of how Cisco CallManager was set up for the Multi-Site Single-Cluster Distributed scenario
Chapter 3, "Cisco Unity Configuration"	Provides an overview of how Cisco Unity was set up
Chapter 4, "Cisco CallManager Express Configuration"	Provides an overview of how Cisco CallManager Express was set up, and includes configuration files for using Cisco CallManager Express with PRI, BRI, and FXO
Chapter 5, "Cisco IP Manager Assistant Configuration"	Provides an overview of how Cisco IPMA was set up for the Multi-Site Single-Cluster Distributed scenario
Chapter 6, "Wireless Configuration"	Provides an overview of how the Cisco Aironet Access Point (AP) 1231, the Cisco IP Phone 7920, and the Cisco Secure Access Control Server (ACS) 3.2 were configured for wireless operation
Chapter 7, "IP Video Telephony Configuration"	Provides an overview of how IP Video Telephony was configured
Chapter 8, "Cisco Enterprise Gateway Configuration"	Provides an overview of how Cisco EGW 2200 Enterprise Gateways and related components were configured
Chapter 9, "Using Microsoft Active Directory 2003 with an IPT Solution"	Provides guidelines and references for using Microsoft Active Directory 2003 with an IPT solution

Chapter 10, "Troubleshooting and Technical Tips"	Provides guidance and resources for diagnosing and correcting errors
Chapter 11, "Cisco CallManager Failure, Failover, and Recovery"	Provides an overview of failover testing
Chapter 12, "Call Load Testing"	Describes results of load testing
Appendix A, "Release Versions of Components"	Shows the release versions of the hardware and software components used in IP Communications Systems Test Release 3.0 for North America IPT and EMEA IPT

Related Documentation

The following documents are available at this URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/ip_tele/gblink/system/gbtst3x/index.htm

- *Systems Test Architecture Reference Manual for North America IPT: IP Communications Systems Test Release 3.0*—Describes the components and configurations that have been tested and verified as part of IP Communications Systems Test Release 3.0 for North America IPT.
- *Systems Release Notes for North America and EMEA IPT: IP Communications Systems Test Release 3.0*—Provides late-breaking information, including resolved and known caveats, and important notes.

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Access to all tools on the Cisco Technical Support Website requires a Cisco.com user ID and password. If you have a valid service contract but do not have a user ID or password, you can register at this URL:

http://tools.cisco.com/RPF/register/register.do

Submitting a Service Request

Using the online TAC Service Request Tool is the fastest way to open S3 and S4 service requests. (S3 and S4 service requests are those in which your network is minimally impaired or for which you require product information.) After you describe your situation, the TAC Service Request Tool automatically provides recommended solutions. If your issue is not resolved using the recommended resources, your service request will be assigned to a Cisco TAC engineer. The TAC Service Request Tool is located at this URL:

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For S1 or S2 service requests or if you do not have Internet access, contact the Cisco TAC by telephone. (S1 or S2 service requests are those in which your production network is down or severely degraded.) Cisco TAC engineers are assigned immediately to S1 and S2 service requests to help keep your business operations running smoothly.

To open a service request by telephone, use one of the following numbers:

Asia-Pacific: +61 2 8446 7411 (Australia: 1 800 805 227)
EMEA: +32 2 704 55 55
USA: 1 800 553 2447

For a complete list of Cisco TAC contacts, go to this URL:

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Definitions of Service Request Severity

To ensure that all service requests are reported in a standard format, Cisco has established severity definitions.

Severity 1 (S1)—Your network is "down," or there is a critical impact to your business operations. You and Cisco will commit all necessary resources around the clock to resolve the situation.

Severity 2 (S2)—Operation of an existing network is severely degraded, or significant aspects of your business operation are negatively affected by inadequate performance of Cisco products. You and Cisco will commit full-time resources during normal business hours to resolve the situation.

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- Cisco Marketplace provides a variety of Cisco books, reference guides, and logo merchandise. Visit Cisco Marketplace, the company store, at this URL:

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- The Cisco *Product Catalog* describes the networking products offered by Cisco Systems, as well as ordering and customer support services. Access the Cisco Product Catalog at this URL:

  http://cisco.com/univercd/cc/td/doc/pcat/

- *Cisco Press* publishes a wide range of general networking, training and certification titles. Both new and experienced users will benefit from these publications. For current Cisco Press titles and other information, go to Cisco Press at this URL:

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- *Packet* magazine is the Cisco Systems technical user magazine for maximizing Internet and networking investments. Each quarter, Packet delivers coverage of the latest industry trends, technology breakthroughs, and Cisco products and solutions, as well as network deployment and troubleshooting tips, configuration examples, customer case studies, certification and training information, and links to scores of in-depth online resources. You can access Packet magazine at this URL:

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- *iQ Magazine* is the quarterly publication from Cisco Systems designed to help growing companies learn how they can use technology to increase revenue, streamline their business, and expand services. The publication identifies the challenges facing these companies and the technologies to help solve them, using real-world case studies and business strategies to help readers make sound technology investment decisions. You can access iQ Magazine at this URL:

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- *Internet Protocol Journal* is a quarterly journal published by Cisco Systems for engineering professionals involved in designing, developing, and operating public and private internets and intranets. You can access the Internet Protocol Journal at this URL:

  http://www.cisco.com/ipj

- World-class networking training is available from Cisco. You can view current offerings at this URL:

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Tested Scenarios and Site Models

This chapter describes the following Internet Protocol Telephony (IPT) site scenarios that were tested and verified as part of IP Communications Systems Test Release 3.0 for EMEA IPT:

- Single Site scenario
- Multi-Site Centralized scenario
- Multi-Site Single-Cluster Distributed

Each scenario is composed of one or more site models, which this chapter also describes.

For additional guidelines, recommendations, and best practices for implementing enterprise networking solutions, refer to the Cisco Solution Reference Network Design (SRND) guides and related documents, which are available at this URL:

www.cisco.com/go/srnd

For a list of the release versions of the components used in the site models, refer to Appendix A, “Release Versions of Components.”

For information about using Cisco IPMA and Lotus Domino, refer to the following chapters in *Systems Test Architecture Reference Manual For EMEA IPT, IP Communications Systems Test Release 3.0*. These components were tested for EMEA IPT but also are functional in North America IPT.

- Cisco Unity Configuration
- Cisco IP Manager Assistant Configuration

For additional configuration information, refer to the following chapters in *Systems Test Architecture Reference Manual For North America IPT: IP Communications Systems Test Release 3.0*. These chapters describe components that were tested for North America IPT but that also are functional in EMEA IPT.

- Cisco Unity Express Configuration
- Cisco Personal Assistant Configuration
- Cisco Customer Response Applications Configuration
- Cisco MeetingPlace Configuration
- Fax, Modem, and TTY/TDD Configurations
- Quality of Service Configuration

This chapter includes the following topics:

Purpose of Solution Tests

An efficient, effective, and reliable IPT solution requires many interrelated hardware and software components. The Single Site, Multi-Site Centralized, and Multi-Site Single-Cluster Distributed scenarios described in this manual provide you with models and guidance as you implement an IPT system for your organization. For each scenario, Cisco has selected, installed, configured, and tested hardware and software designed to work together seamlessly and to provide a complete and optimized IPT solution.

Each scenario and test addresses the following issues:

- End-to-end functionality
- Operability in a real-world environment
- Scalability
- Stability
- Stress
- Load

- Redundancy
- Reliability
- Usability
- Availability
- Installability
- Upgradeability
- Serviceability
- Regression

Overview of Test Scenarios

The following sections describe the IPT site scenarios that were tested and verified as part of IP Communications Systems Test Release 3.0 for EMEA IPT:

- Single Site Scenario, page 1-3
- Multi-Site Centralized Scenario, page 1-4
- Multi-Site Single-Cluster Distributed Scenario, page 1-6

Single Site Scenario

A Single Site scenario consists of a Cisco CallManager located at a single site or campus, with no telephony services provided over an IP WAN. A LAN or a metropolitan area network (MAN) carries voice traffic throughout the site. If an IP WAN is used, it is for data traffic only. Calls beyond the LAN or MAN use the public switched telephone network (PSTN).

A Single Site scenario can consist of either of these site models:

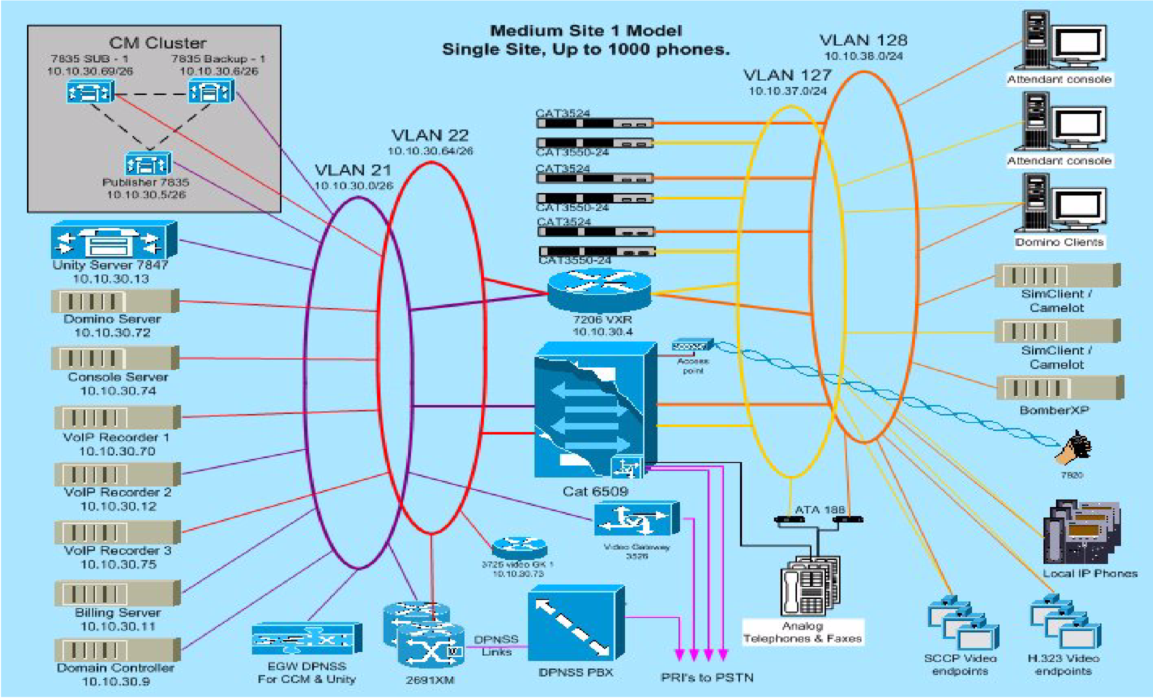
- Small Site model. For more information, see the "Small Site Model" section on page 1-9.
- Medium Site model. For more information, see the "Medium Site 1 Model" section on page 1-12.

The tested Single Site scenarios have the following design characteristics:

- Support for up to 1,000 phones at a medium site and 500 phones at a small site
- Cisco CallManager cluster for redundancy and system scaling

Figure 1-1 provides an overview of the Single Site scenario (Medium Site model).

Figure 1-1 Single Site Scenario (Medium Site Model)



Multi-Site Centralized Scenario

A Multi-Site Centralized scenario consists of a multi-site IP over ATM/Frame Relay WAN with centralized call processing. In this scenario, a single Cisco CallManager cluster provides call processing services for multiple remote sites and uses the WAN to carry IP telephony traffic between the sites. The WAN also carries call control signaling between the central site and the remote sites.

If the central site or the WAN goes down, remote sites can continue to have service through Survivable Remote Site Telephony (SRST), which runs on Cisco IOS gateways. Remote sites can also place calls over the PSTN if the WAN becomes temporarily oversubscribed.

The tested Multi-Site Centralized scenario is composed of one Central Site model and 50 Remote Site models.

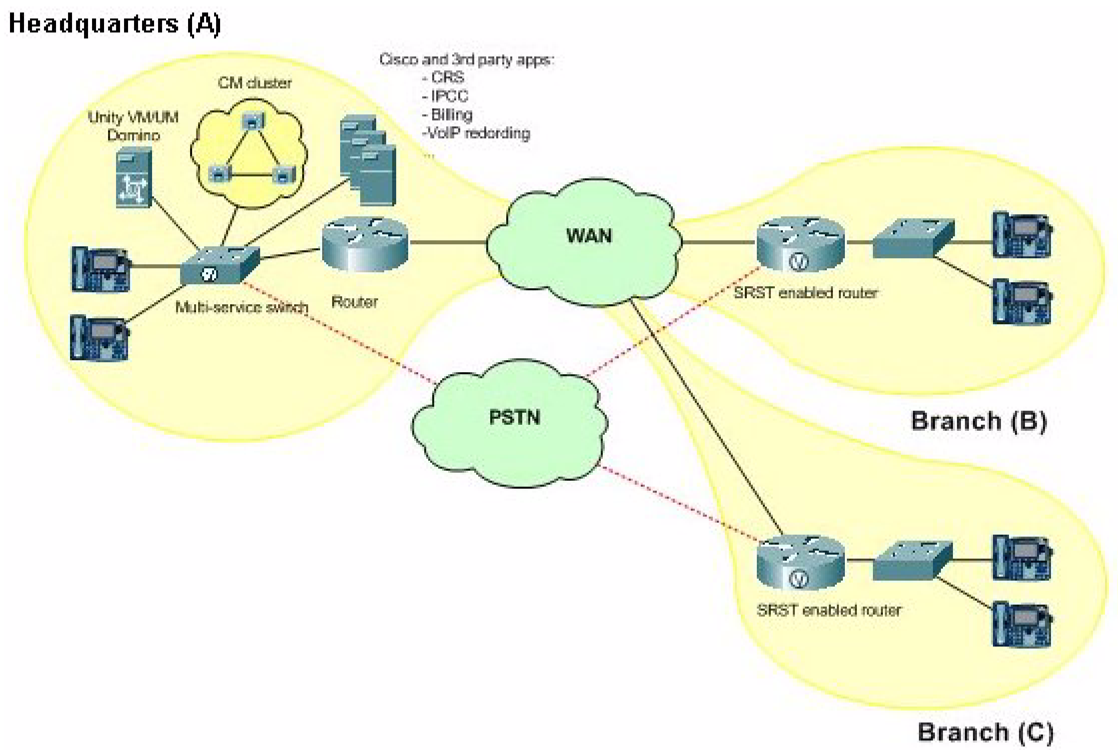
For more information about the Multi-Site Centralized scenario site models, see the "Central Site Model" section on page 1-18 and the "Remote Site Models" section on page 1-21.

The tested Multi-Site Centralized has the following design characteristics:

- Cisco CallManager cluster resides at the central site
- Support for up to 1,000 phones in each site
- Centralized operator services
- Billing and accounting services
- Voice recording available for any PSTN call
- Cisco Unity Unified Messaging resides at the central site
- Cisco Unity, in a deployment where Lotus Domino / Lotus Notes is used for unified messaging services
- Centralized dial plan and administration
- Call admission control based on locations (to protect voice quality of WAN calls)
- SRST for remote sites

Figure 1-2 provides an overview of the Multi-Site Centralized scenario.

Figure 1-2 Multi-Site Centralized Scenario



Multi-Site Single-Cluster Distributed Scenario

A Multi-Site Single-Cluster Distributed scenario consists of two or more sites. Each site includes one primary and one standby subscriber server. The subscriber servers are connected to a publisher server that resides in addition to the two subscriber servers at one. In this scenario, an IP over ATM WAN caries voice traffic and intra-cluster call control signaling between the sites.

This scenario provides local and remote failover operations for the two primary subscriber servers. Under normal operation, the primary subscriber server in each site manages the local phones in that site. With local failover, the local primary subscriber server fails over to the local standby server. If the primary subscriber server and its standby server fail, remote failover allows the subscriber server to failover to a subscriber server at the remote site. There is full cluster-wide functionality between the two sites for all other applications.

A redundant pair of Cisco EGW 2200 Enterprise Gateways is located in one of the sites to provide signal processing between Cisco CallManager and TDM PBX systems that use DPNSS. The Cisco EGW 2200s also use MGCP to control multiple Cisco gateways in each site, which are connected to multiple PBXs. The Cisco EGW 2200s interact with Cisco CallManager through a gatekeeper using H.323.

The Multi-Site Single-Cluster Distributed scenario is composed of these site models:

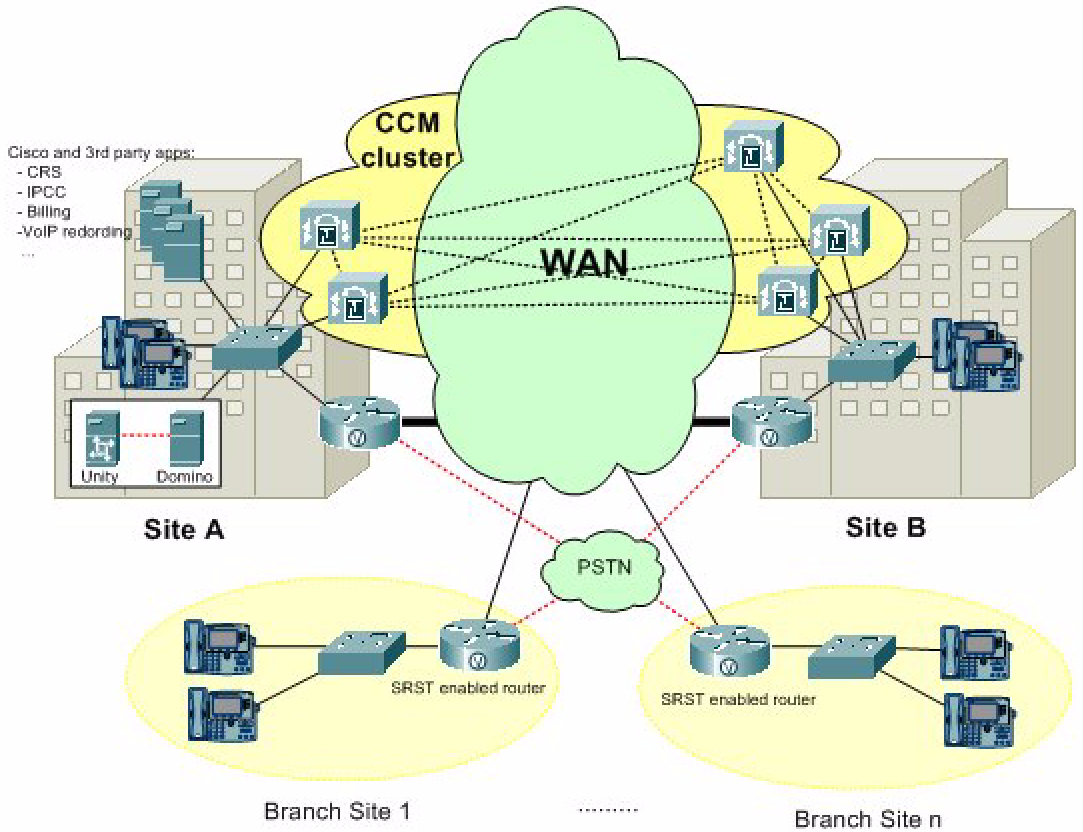
- Medium Site 1 model—Location of the publisher server. For more information, see the "Medium Site 1 Model" section on page 1-12
- Medium Site 2 model—Location of the subscriber servers. For more information, see the Medium Site 2 Model, page 1-15.

The tested Multi-Site Single-Cluster Distributed scenario has the following design characteristics:

- Separate Cisco CallManager clusters reside at each site
- Support for up to 1,000 phones at each site
- Centralized operator services
- Centralized billing and accounting services
- Voice recording available for any PSTN call
- Most services, including Cisco Unity Unified Messaging, Cisco IP Manager Assistant, Cisco Attendant Consoles, and voice over IP recording, run at each site and provide functionality to their respective sites.
- Cisco IP Manager Assistant and Cisco Attendant Consoles will failover to their respective servers at the other site. In the event of a failover, existing functionality will continue.
- Centralized dial plan and administration
- Call admission control based on locations (to protect voice quality of WAN calls)

Figure 1-3 provides an overview of the Multi-Site Single-Cluster Distributed scenario.

Figure 1-3 Multi-Site Single-Cluster Distributed Scenario



Site Models for the Test Scenarios

The following sections describe the site models that were used to create the various test scenarios.

Each section includes a table that lists the hardware and software components used in the model. The tables contain the following information for each component:

- Component—Hardware or software component
- Description—Information such model number, release number, protocol, and hardware platform
- Qty.—Quantity of the component used in the model

Table 1-1 provides an overview of the site models.

Table 1-1 Site Models

Name	Reference	Description
Small Site	See the "Small Site Model" section on page 1-9	Can stand alone as a Single Site scenario
Medium Site 1	See the "Medium Site 1 Model" section on page 1-12	Can stand alone as a Single Site scenario or be the location of the publisher server in a Multi-Site Single-Cluster Distributed scenario
Medium Site 2	See the "Medium Site 2 Model" section on page 1-15	Location of subscriber servers in a Multi-Site Single-Cluster Distributed scenario
Central Site	See the "Central Site Model" section on page 1-18	Central site in a Multi-Site Centralized scenario
Remote Site	See the "Remote Site Models" section on page 1-21	Remote sites in a Multi-Site Centralized scenario

Small Site Model

The Small Site model can stand alone as a Single Site scenario. This model contains approximately 500 phones.

Figure 1-4 shows the topology of the Small Site model.

Figure 1-4 Small Site Model Topology

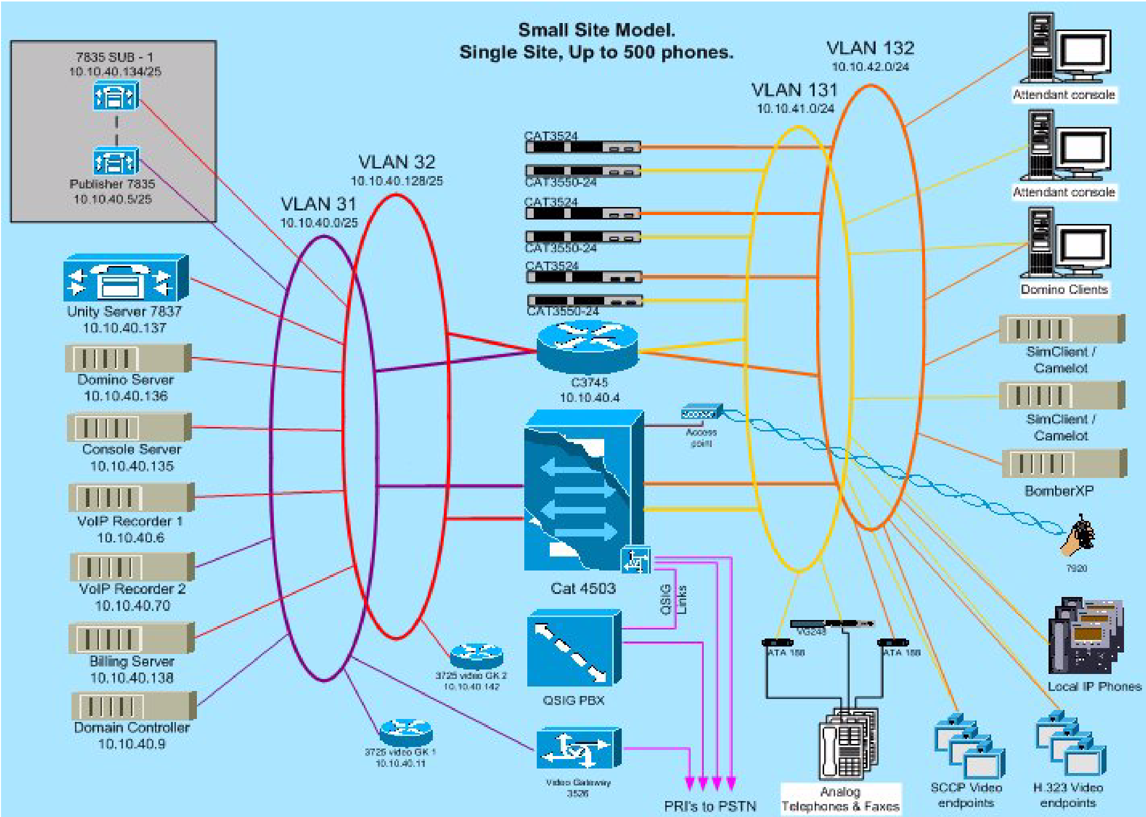


Table 1-2 lists the hardware and software components used in the Small Site model.

Table 1-2 Small Site Model Components

Component	Description	Qty.
Access Switch	Cisco Catalyst 3524	3
Access Switch	Cisco Catalyst 3550-24	3
Attendant Console	Third-party product	1
Billing Server	Third-party product	1

Table 1-2 Small Site Model Components (continued)

Component	Description	Qty.
Cisco Analog Telephone Adaptor (ATA)	ATA 188	2
Cisco CallManager	Cisco CallManager installed on an MCS-7835-1266	1
Cisco CallManager Cluster	Cisco CallManager installed on an MCS-7825-1133	1
Cisco IP Communicator	Cisco IP Communicator installed on a Pentium IV PC	1
Cisco IP Phone	Cisco IP Phone CP-7902G Cisco IP Phone CP-7905G Cisco IP Phone CP-7910 Cisco IP Phone CP-7912G Cisco IP Phone CP-7920G Cisco IP Phone CP-7935 Cisco IP Phone CP-7936G Cisco IP Phone CP-7940G Cisco IP Phone CP-7960G Cisco IP Phone CP-7970G	18
Cisco IP Softphone	Cisco Softphone installed on a Pentium IV PC	1
Cisco Unity Unified Messaging	Cisco Unity installed on a MCS-7837	1
Core Switch	Cisco Catalyst 4506 with Supervisor IV	1
Domain Controller	Windows 2000 Server	1
FXS Gateway	Cisco VG248	1
Gateway	Cisco 3725 (MGCP) (NM-HDV with VWIC-2MFT-T1)	1
Locale Installer	Cisco Locale Installer	1

Table 1-2 Small Site Model Components (continued)

Component	Description	Qty.
Lotus Domino	Lotus Domino	1
Music on Hold (MOH)	Installed on an MCS-7835-1266	1
Router	Cisco 3725	1
Video Endpoint (H.323)	Tandberg 880, 2500, or 6000	2
Video Endpoint (SCCP)	Cisco VT Advantage	1
	Tandberg 550 Tandberg 1000	3
Video Gateway	Cisco 3526	2
Video MCU	Cisco IPVC-3511	1
Voice over IP Recorder	Third-party product	1

Medium Site 1 Model

The Medium Site 1 model can stand alone as a Single Site scenario or, with the addition of IP over ATM WAN routers and gatekeepers, be the location of the publisher server in the Multi-Site Single-Cluster Distributed scenario. This model contains approximately 1,000 phones.

Figure 1-5 shows the topology of the Medium Site 1 model.

Figure 1-5 Medium Site 1 Model Topology

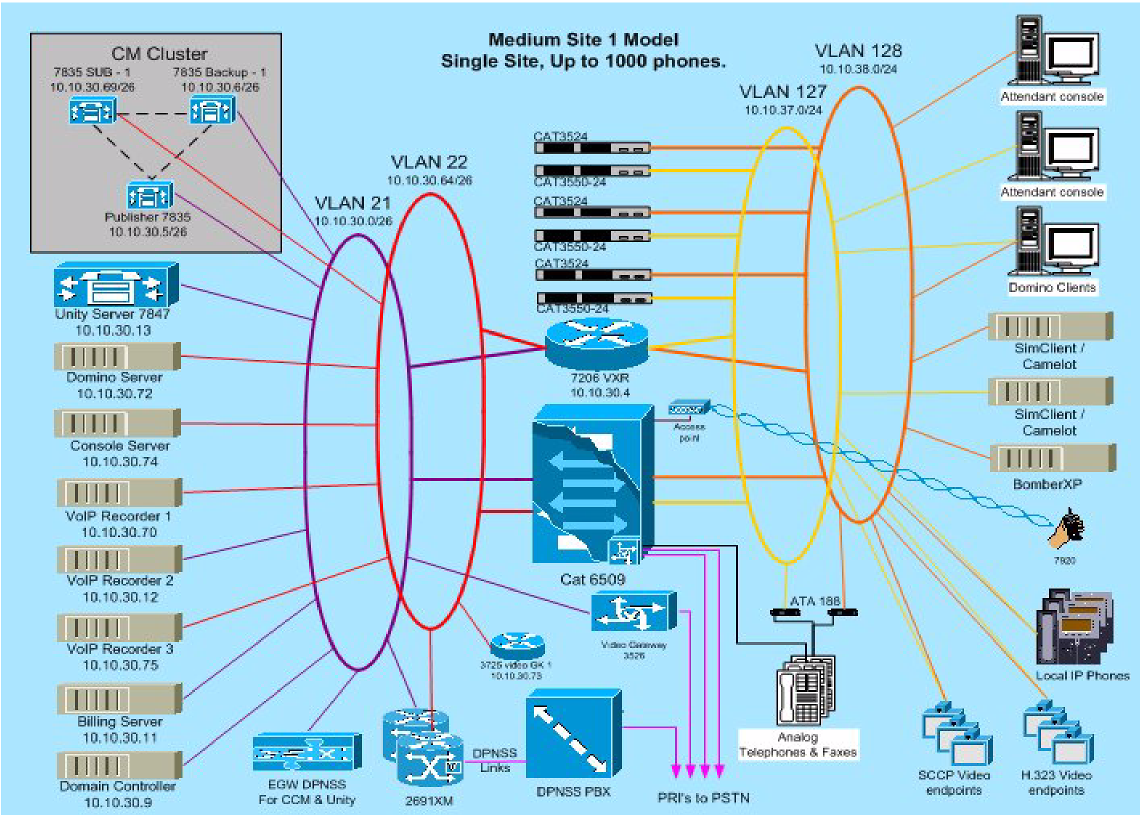


Table 1-3 lists the hardware and software components used in the Medium Site 1 model.

Table 1-3 Medium Site 1 Model Components

Component	Description	Qty.
Access Point	Cisco AP1121G	1
Access Switch	Cisco Catalyst 3524	3
	Cisco Catalyst 3500-24	3
Attendant Console	Third-party product	1

Table 1-3 Medium Site 1 Model Components (continued)

Component	Description	Qty.
Billing Server	Third-party product	1
Cisco ATA	ATA	2
Cisco CallManager	Cisco CallManager installed on an MCS-7835-1266	3
Cisco IP Communicator	Cisco IP Communicator installed on a Pentium IV PC	1
Cisco IP Phone	Cisco IP Phone CP-7902G Cisco IP Phone CP-7905G Cisco IP Phone CP-7910 Cisco IP Phone CP-7912G Cisco IP Phone CP-7920G Cisco IP Phone CP-7935 Cisco IP Phone CP-7936G Cisco IP Phone CP-7940G Cisco IP Phone CP-7960G Cisco IP Phone CP-7970G	18
Cisco IP Softphone	Cisco Softphone installed on a Pentium IV PC	1
Cisco Locale Installer	Cisco Locale Installer	1
Cisco Unity Unified Messaging	Cisco Unity installed on a MCS-7845H-2.4-ECS1	1
Core Switch	Cisco Catalyst 6509 running CATOS	1
Domain Controller	Windows 2000	1
Lotus Domino	Lotus Domino	1
Enterprise Gateway	Cisco EGW 2200 installed on an MCS-7835H-3.0-CC1	1
FXS Gateway	Cisco 6624 bundled	1
Gateway	Cisco 6608 bundled	1

Table 1-3 Medium Site 1 Model Components (continued)

Component	Description	Qty.
Gateway for EGW	Cisco 2691	1
Router	Cisco 7206	1
Video Endpoint (H.323)	Tandberg 880, 2500, or 6000	2
Video Endpoint (SCCP)	Cisco VT Advantage	1
	Tandberg 550 Tandberg 1000	2
Video Gateway	Cisco 3526	1
Video MCU	Cisco IPVC-3511	1
Voice over IP Recorder	Third-party product	1

Medium Site 2 Model

The Medium Site 2 model is the location of subscriber servers in the Multi-Site Single-Cluster Distributed scenario. This model contains approximately 1,000 phones.

Figure 1-6 shows the topology of the Medium Site 2 model. This figure also shows the Medium Site 1 model and how the two models interact.

Figure 1-6 Medium Site 2 Model Topology

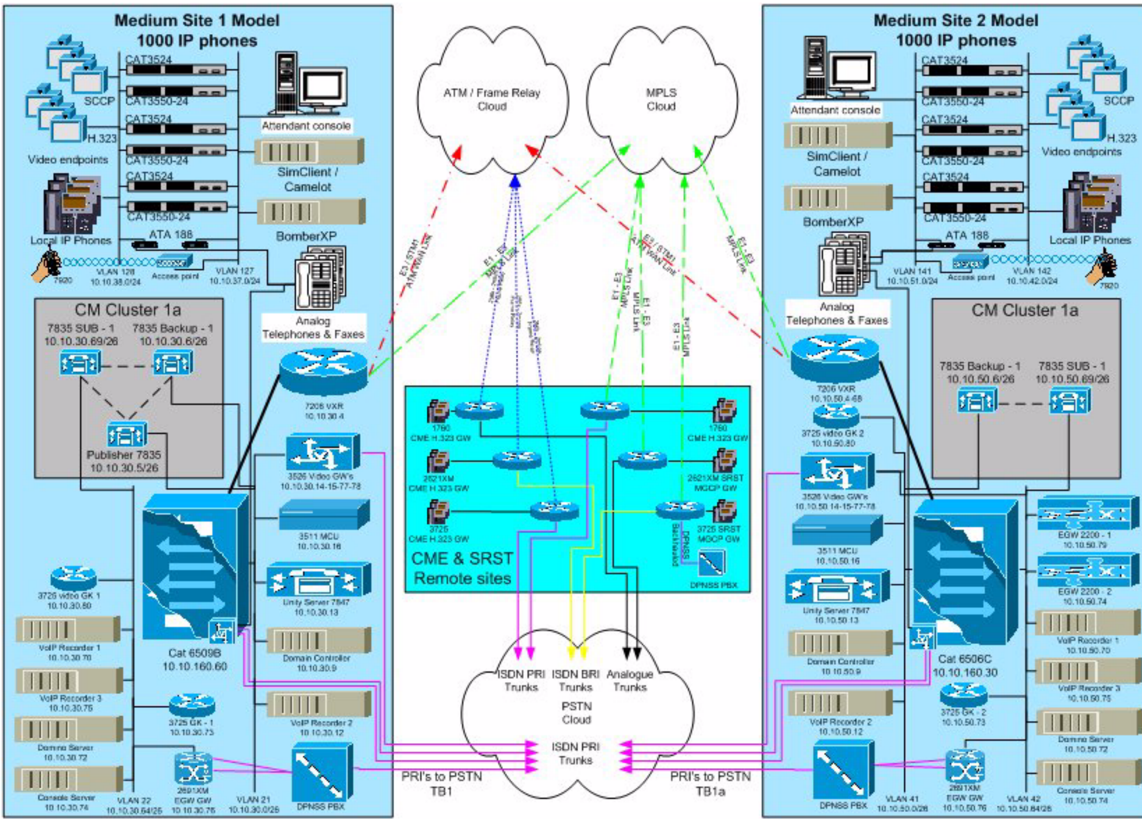


Table 1-4 lists the hardware and software components used in the Medium Site 2 model.

Table 1-4 Medium Site 2 Model Components

Component	Description	Qty.
Access Switch	Cisco Catalyst 4506 running IOS	1
	Cisco Catalyst 3524 WC5	3
	Cisco Catalyst 3500-24	3
Attendant Console	Third-Party Product	1

Table 1-4 Medium Site 2 Model Components (continued)

Component	Description	Qty.
Billing Server	Third-Party Product	1
Cisco ATA	ATA 188	2
Cisco CallManager	Cisco CallManager installed on an MCS-7835-1266	2
Cisco IP Communicator	Cisco IP Communicator installed on a Pentium IV PC	1
Cisco IP Phone	Cisco IP Phone CP-7902G Cisco IP Phone CP-7905G Cisco IP Phone CP-7910 Cisco IP Phone CP7912G Cisco IP Phone CP-7920G Cisco IP Phone CP-7935 Cisco IP Phone CP-7936G Cisco IP Phone CP-7940G Cisco IP Phone CP-7960G Cisco IP Phone CP-7970G	18
Cisco IP Softphone	Cisco Softphone installed on a Pentium IV PC	1
Cisco Locale Installer	Cisco Locale Installer	1
Cisco Unity Unified Messaging	Cisco Unity installed on an MCS-7847 ECSH	1
Core Switch	Cisco Catalyst 6509 running CATOS	1
Domain Controller	Windows 2000	1
Lotus Domino	Lotus Domino	1
Enterprise Gateway	Cisco EGW 2200 installed on an MCS-7835H-3.0-CC1	1

Table 1-4 Medium Site 2 Model Components (continued)

Component	Description	Qty.
FXS Gateway	Cisco Catalyst 6500 with WS-SVC-CMM-12AGW	1
	VG248	1
Gateway	Cisco Catalyst 6500 with WS-SVC-CMM-6E1	1
Router	7206 NPE400	1
Video Endpoint (H.323)	Tandberg 880, 2500, or 6000	2
Video Endpoint (SCCP)	Cisco VT Advantage	1
	Tandberg 550 Tandberg 1000	2
Voice over IP Recorder	Third-Party Product	1

Central Site Model

In the Multi-Site Centralized scenario, the Central Site model is the site where the Cisco CallManager or the Cisco CallManager cluster is located. The Central Site model provides the call processing services for the remote sites.

Figure 1-7 shows the topology of the Central Site model. This figure also includes topologies of typical Remote Site models.

Figure 1-7 Central Site and Remote Sites Topologies

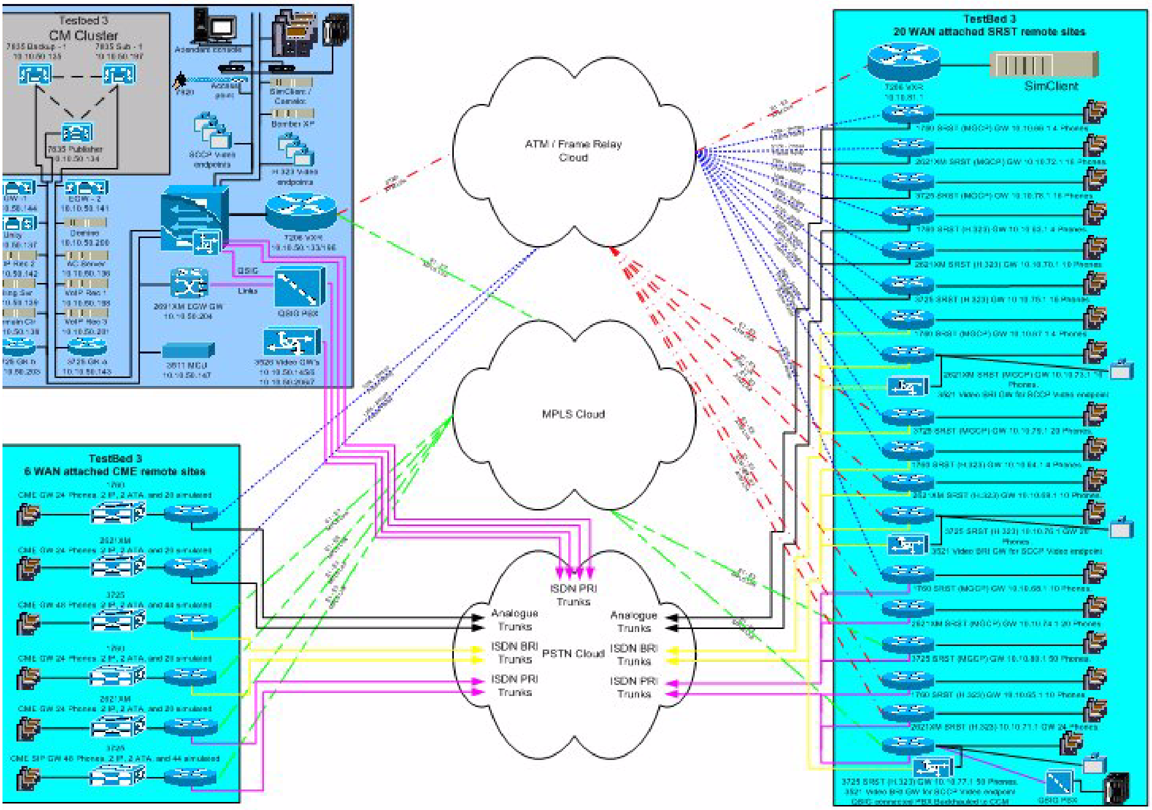


Table 1-5 lists the hardware and software components used in the Central Site model.

Table 1-5 Central Site Model Components

Component	Description	Qty.
Access Switch	Cisco Catalyst 6509	
	Cisco Catalyst 3524	3
	Cisco Catalyst 3500-24	3
Attendant Console	Third-party product	1

Table 1-5 Central Site Model Components (continued)

Component	Description	Qty.
Billing Server	Third-party product	1
Cisco ATA	ATA 188	2
Cisco CallManager	Cisco CallManager installed on an MCS-7835H-3.0-CC1	3
Cisco IP Communicator	Cisco IP Communicator installed on a Pentium IV PC	1
Cisco IP Phone	Cisco IP Phone CP-7902G Cisco IP Phone CP-7905G Cisco IP Phone CP-7910 Cisco IP Phone CP-7912G Cisco IP Phone CP-7920G Cisco IP Phone CP-7935 Cisco IP Phone CP-7936G Cisco IP Phone CP-7940G Cisco IP Phone CP-7960G Cisco IP Phone CP-7970G	18
Cisco IP Softphone	Cisco Softphone installed on a Pentium IV PC	1
Cisco Locale Installer	Cisco Locale Installer	1
Cisco Unity Unified Messaging	Cisco Unity installed on a MCS-7845H-2.4-ECS1	1
Core Switch	Cisco Catalyst 6509 running CATOS	1
Domain Controller	Windows 2000	1
Lotus Domino	Lotus Domino	1
Enterprise Gateway	Cisco EGW 2200 installed on an MCS-7835H-3.0-CC1	2

Table 1-5 Central Site Model Components (continued)

Component	Description	Qty.
FXS Gateway	Cisco Catalyst 6500 with WS-SVC-CMM-12AGW	1
	VG248	1
Gateway	Cisco Catalyst 6500 with WS-SVC-CMM-6E1	1
Router	7206 NPE400	1
Video Endpoint (H.323)	Tandberg 880, 2500, or 6000	2
Video Endpoint (SCCP)	Cisco VT Advantage	1
	Tandberg 550 Tandberg 1000	2
Video Gateway	Cisco 3526 IPVC Gateway	2
Video MCU	Cisco IPVC-3511	1
Voice over IP Recorder	Third-party product	1

Remote Site Models

In the Multi-Site Centralized scenario, the Remote Site model represents the sites other than the central site. (The Cisco CallManager cluster is located at the central site.)

Figure 1-7 shows topology of the Remote Site model. This figure also includes the topology of the Central Site model.

Table 1-6 lists the hardware and software components used in the Remote Site model.

Table 1-6 Remote Site Model Components

Component	Description	Qty.
Cisco ATA	ATA 188	2
Cisco IP Communicator	Cisco IP Communicator installed on a Pentium IV PC	1

Table 1-6 Remote Site Model Components (continued)

Component	Description	Qty.
Cisco IP Phone	Cisco IP Phone 7902G Cisco IP Phone 7905G Cisco IP Phone 7910 Cisco IP Phone 7912G Cisco IP Phone 7920G Cisco IP Phone 7935 Cisco IP Phone 7936G Cisco IP Phone 7940G Cisco IP Phone 7960G Cisco IP Phone 7970G	58
Cisco IP Softphone	Cisco Softphone installed on a Pentium IV PC	1
Router/Gateway	Cisco 1761 (H.323)	3
	Cisco 1761 (MGCP)	3
	Cisco 1761 (CME)	2
Router/Gateway (continued)	Cisco 2621XM (H.323)	3
	Cisco 2621XM (MGCP)	3
	Cisco 2621XM (CME)	2
	Cisco 3660 (H.323)	1
	Cisco 3660 (MGCP)	1
	Cisco 3725 (H.323)	3
	Cisco 3725 (MGCP)	3
	Cisco 3725 (CME)	2
Video Endpoint (H.323)	Tandberg 880, 2500, or 6000	10
Video Endpoint (SCCP)	Cisco VT Advantage	1
	Tandberg 550 Tandberg 1000	20

Table 1-6 Remote Site Model Components (continued)

Component	Description	Qty.
Video Gateway	Cisco IP/VC 3521	3
Voice over IP Recorder	Third-party product	4

Cisco CallManager Configuration

This chapter provides an overview of how Cisco CallManager was set up for the Multi-Site Single-Cluster Distributed scenario in IP Communications Systems Test Release 3.0 for EMEA IPT. This chapter does not include detailed installation and configuration instructions. Rather, it is intended to provide you with guidance as you set up the Cisco CallManager component of your IPT solution.

This chapter also does not include information about the Application menu in Cisco CallManager Administration. The web pages available from that menu use the default settings.

Cisco CallManager was installed on Cisco MCS-7835-1266 or Cisco MCS-7825-1133 servers and configured according to the instructions in the Cisco CallManager documentation.

For detailed information about installing, configuring, and administering Cisco CallManager, refer to the Cisco CallManager documentation at this URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_callmg/4_0/index.htm

For additional information about configuring Cisco CallManager for Cisco MeetingPlace, see the "Cisco CallManager Configuration for Cisco MeetingPlace" section on page 9-4.

For additional information about configuring Cisco CallManager for IP Video Telephony, see the "Cisco Call Manager Configuration for IP Video Telephony" section on page 7-9.

For additional information about configuring Cisco CallManager for Cisco Enterprise Gateway, see the "Configuring Cisco CallManager for Cisco EGWs" section on page 8-24.

This chapter includes the following topics:

Cisco CallManager System Configuration

The following sections provide an overview of how Cisco CallManager was configured on many of the System menu web pages that you access from Cisco CallManager Administration. These sections do not describe all of the System menu web pages or web page fields. Instead, they point out selected configuration information that will help you understand how Cisco CallManager was set up to perform most effectively.

The following sections do not discuss these System menu options:

- Date/Time Group—Use default settings
- Device Defaults—Use default settings
- AAR Group—Use default settings

System > Server Configuration

To access the Cisco CallManager Administration web pages for adding and configuring servers, choose **System > Server** from the Cisco CallManager Administration application.

The following Cisco CallManager servers were configured for the Multi-Site Single-Cluster Distributed scenario:

- PUBLISHER/TFTP
- SUBSCRIBER_SITE_1
- SUBSCRIBER_SITE_2
- BACKUP1_SITE_1
- BACKUP2_SITE_2

Table 2-1 describes the settings in the Server Configuration page.

Table 2-1 Cisco CallManager Server Configuration

Field	Setting
Host Name/IP Address	Name of the server For example, SUBSCRIBER_SITE_1
MAC Address	Blank
Description	Description of server For example, Cisco CallManager Server 1

System > Cisco CallManager Configuration

To access the Cisco CallManager Administration web pages for adding and configuring Cisco CallManagers, choose **System > Cisco CallManager** from the Cisco CallManager Administration application.

The following Cisco CallManagers were configured for the Multi-Site Single-Cluster Distributed scenario:

- PUBLISHER/TFTP
- SUBSCRIBER_SITE_1
- SUBSCRIBER_SITE_2
- BACKUP1_SITE_1
- BACKUP2_SITE_2

Table 2-2 describes the settings in the Cisco CallManager Configuration page.

Table 2-2 Cisco CallManager Configuration

Field	Setting
Cisco CallManager Name	Name of the Cisco CallManager For example, SUBSCRIBER_SITE_1
Description	Description of the Cisco CallManager For example, Cisco CallManager 1 Server
Starting Directory Number	2000
Ending Directory Number	2000
Partition	<None>
External Phone Number Mask	Blank
Auto-registration Disabled on this Cisco CallManager	Checked
Ethernet Phone Port	2000
Digital Port	2001
Analog Port	2002
MGCP Listen Port	2427
MGCP Keep-alive Port	2428

System > Cisco CallManager Group

To access the Cisco CallManager Administration web pages for adding and configuring Cisco CallManager Groups, choose **System > Cisco CallManager Group** from the Cisco CallManager Administration application.

The following Cisco CallManager groups were configured for the Multi-Site Single-Cluster Distributed scenario:

- Central_1
- Central _1a
- IPMATB1Group
- Default

Table 2-3 describes the settings in the Cisco CallManager Group Configuration page.

Table 2-3 Cisco CallManager Group Configuration

Field	Setting
Cisco CallManager Group	Name of the Cisco CallManager group
Auto-registration Cisco CallManager Group	Checked for Default Unchecked for other groups
Selected Cisco CallManagers (ordered by highest priority)	For group Central_1: SUBSCRIBER_SITE_1, BACKUP_SITE_1, BACKUP_SITE_2 For group Central_1a: SUBSCRIBER_SITE_2, BACKUP_SITE_2, BACKUP_SITE_1 For group IPMATB1Group: SUBSCRIBER_SITE_1, BACKUP_SITE_1 For group Default: PUBLISHER

System > Region

To access the Cisco CallManager Administration web pages for adding and configuring regions, choose **System > Region** from the Cisco CallManager Administration application.

Note If you will be using video devices, Cisco recommends that you set up specific regions for the video devices. The codec between these regions must be G.711.

The following regions were configured for the Multi-Site Single-Cluster Distributed scenario:

- Default
- Central_1a
- Rem_01
- Rem_02
- VCentral
- VCentral_1a

Table 2-4 describes the settings in the Region Configuration page for the Default region.

Table 2-4 Region Configuration for Default Region

Region	Audio Codec Setting	Video Call Bandwidth Setting
Default (Within this Region)	G.711	384
Central_1a Region	G.729	384
Rem_01 Region	G.729	None
Rem_02 Region	G.729	None
VCentral	G.711	384
VCentral_1a	G.711	384

System > Device Pool

To access the Cisco CallManager Administration web pages for adding and configuring device pools, choose **System > Device Pool** from the Cisco CallManager Administration application.

Note If you will be using video regions, you must create separate device pools within these regions.

The following device pools were configured for the Multi-Site Single-Cluster Distributed scenario:

- Central_1a
- CME
- Default
- ipmatb1pool
- Rem01-2621M
- Rem01-3725M

Table 2-5 describes the settings in the Device Pool Configuration page.

Table 2-5 Device Pool Configuration

Field	Settings for Central_1a	Settings for CME	Settings for Default	Settings for ipmatb1pool	Settings for Rem01-2621M	Settings for Rem01-3725M
Device Pool Name	Central_1a	CME	Default	ipmatb1pool	Rem01-2621M	Rem01-3725M
Cisco CallManager Group	Central_1a	Central_1	Central_1	ipmatb1group	Central_1	Central_1a
Date/Time Group	CMLocal	CMLocal	CMLocal	CMLocal	CMLocal	CMLocal
Region	Central_1a	CME	Default	Default	Rem-01	Rem-02

Table 2-5 Device Pool Configuration (continued)

Field	Settings for Central_1a	Settings for CME	Settings for Default	Settings for ipmatb1pool	Settings for Rem01-2621M	Settings for Rem01-3725M
Softkey Template	Standard User with CallBack	Standard User with CallBack	Standard User with CallBack	Standard User with CallBack	Standard User with CallBack	Standard User with CallBack
SRST Reference	Disable	Disable	Disable	Disable	TB1-1-2621XMMGCP	TB1a-1-3725MGCP
Calling Search Space for Auto-Registration	Central_1aCSS	<None>	Central_1CSS	Central_1CSS	Rem01-ExtnCSS	Rem02-ExtnCSS
Media Resource Group List	MRGL_Central_1a	MRGL_Central	MRGL_Central	MRGL_Central	MRGL_Remote	MRGL_Remote
Network Hold MOH Audio Source	Sample Audio Source	Sample Audio Source	Sample Audio Source	Sample Audio Source	Sample Audio Source	Sample Audio Source
User Hold MOH Audio Source	Sample Audio Source	Sample Audio Source	Sample Audio Source	Sample Audio Source	Sample Audio Source	Sample Audio Source
Network Locale	United Kingdom	United Kingdom	United Kingdom	United Kingdom	United Kingdom	United Kingdom
User Locale	English United States	English United States	English United States	English United States	English United States	English United States
MLPP Indication	Default	Default	Default	Default	Default	Default
MLPP Preemption	Default	Default	Default	Default	Default	Default
MLPP Domain	blank	blank	blank	blank	blank	blank

System > Enterprise Parameters

To access the Cisco CallManager Administration web pages for configuring enterprise parameters, choose **System > Enterprise Parameters** from the Cisco CallManager Administration application.

Table 2-6 describes how the enterprise parameters that were configured for the Multi-Site Single-Cluster Distributed scenario. Parameters that are not shown used their default values. In this example, the Cisco CallManager cluster does not include a DNS server, so the values in the URL fields are Cisco CallManager IP addresses.

Table 2-6 Enterprise Parameters

Field	Value
URL Help	http://10.10.30.5/help
Enable Dependency Records	True
Default Network Locale	United Kingdom
URL Authentication	http://10.10.30.5/CCMCIP/authenticate.asp
URL Directories	http://10.10.30.5/CCMCIP/xmldirectory.asp
URL Information	http://10.10.30.5/CCMCIP/GetTelecasterHelpText.asp
URL Services	http://10.10.30.5/CCMCIP/getservicesmenu.asp
Enable All User Search	False

System > Location

To access the Cisco CallManager Administration web pages for configuring locations, choose **System > Locations** from the Cisco CallManager Administration application.

Table 2-7 describes the two locations that were configured for the Multi-Site Single-Cluster Distributed scenario.

Table 2-7 Locations

Location	Voice Bandwidth	Video Bandwidth
Rem01-2621H	512 Kbps	None
Rem02-3725M	1024 Kbps	None

System > SRST

To access the Cisco CallManager Administration web pages for configuring SRST sites, choose **System > SRST** from the Cisco CallManager Administration application.

Table 2-7 describes the two SRST sites that were configured for the Multi-Site Single-Cluster Distributed scenario.

Table 2-8 SRST

SRST Reference Name	IP Address	Port
TB1-1-2621XMMGCP	10.10.92.1	2000
TB1a-1-3725MGCP	10.10.93.1	2000

Cisco CallManager Route Plan Configuration

The following sections provide an overview of how Cisco CallManager was configured on many of the Route Plan menu web pages that you access from Cisco CallManager Administration. These sections do not describe all of the Route Plan menu web pages or web page fields. Instead, they point out selected configuration information that will help you understand how Cisco CallManager was set up to perform most effectively.

- Route Plan > Partition, page 2-11
- Route Plan > Calling Search Space, page 2-12
- Route Plan > Route/Hunt > Route Group, page 2-14
- Route Plan > Route Hunt > Route/Hunt List, page 2-14

- Route Plan > Route Pattern/Hunt Pilot, page 2-15
- Route Plan > Translation Pattern, page 2-18

The following sections do not discuss these Route Plan menu options:

- Application Dial Rules—Use default settings
- Route Filter—Not configured

Route Plan > Partition

To access the Cisco CallManager Administration web pages for adding and configuring partitions, choose **Route Plan > Partition** from the Cisco CallManager Administration application.

Table 2-9 describes the partitions that were configured for the Multi-Site Single-Cluster Distributed scenario.

Table 2-9 Partitions

Partition Name	Description
Central1a	All local DDI extensions (except managers) in Site B
CForward	Call forward partition
CForward1a	Call forward 1a partition
Console	Operator console 1
console1a	Operator console 1A
EGW	Calls to the Enterprise Gateway
IPMA	All Cisco IPMA route points and ports for Site A
IPMA1a	All Cisco IPMA route points and ports for Site B
Manager1	IPMA Managers in Site A
Manager1a	IPMA Managers in Site B
PBXPSTN1	Alternate PSTN route via PBX 1
PBXPSTN1a	Alternate PSTN route via PBX 1a

Table 2-9 Partitions (continued)

Partition Name	Description
PSTN-Rem01	Trunks for Remote 01
PSTN-Rem02	Trunks for Remote 02
PSTN1	All Local PRI trunks in Site A
PSTN1a	All Local PRI trunks in Site B
Rem-Extns	All remote extensions
Central1	All local DDI extensions (except managers) in Site A

Route Plan > Calling Search Space

To access the Cisco CallManager Administration web pages for adding and configuring calling search spaces, choose **Route Plan > Calling Search Space** from the Cisco CallManager Administration application.

Table 2-10 describes the calling search spaces that were configured for the Multi-Site Single-Cluster Distributed scenario.

Table 2-10 Calling Search Spaces

Calling Search Space Name	Description	Selected Partitions
consearch	Console search space Site A	IPMA, console1a, Central1, Central1a, EGW, Rem-Extns, console, PSTN1, PSTN1a
consearch1a	Console search space Site B	IPMA1a, Central1a, console1a, Central1, EGW, Rem-Extns, PSTN1a, console
Central1aCSS	Site B extn PSTN Console & IPMA	IPMA1a, IPMA, Central1a, Central1, EGW, console1a, Rem-Extns, PBXPSTN1a, console
Central1CSS	Site A extn PSTN Console & IPMA	IPMA, IPMA1a, Central1, Central1a, EGW, Rem-Extns, console, PBXPSTN1, console1a

Table 2-10 Calling Search Spaces (continued)

Calling Search Space Name	Description	Selected Partitions
CforwardCCS	Site A Call forward CSS	CForward, Central1, Central1a, Rem-Extns, EGW, IPMA, IPMA1a, console
CforwardCCS1a	Site B Call forward CSS	CForward1a, Central1, Central1a, EGW, Rem-Extns, IPMA1a, IPMA, console1a
IPMA1ACSS	Site B IPMA CSS	Manager1a, Manager1, Central1a, Central1, EGW, PBXPSTN1a
IPMACSS	Site A IPMA CSS	Manager1, Manager1a, Central1, Central1a, EGW, Rem-Extns, PBXPSTN1
PSTN1aCSS	Site B incoming PSTN calls	IPMA1a, IPMA, Central1a, console1a, EGW, Central1, console
PSTN1CSS	Site A incoming PSTN calls	IPMA, IPMA1a, Central1, console, EGW, Central1a, console1a
Rem01-Extn-CSS	Remote 1 Extn CCS	Rem-Extns, PSTN-Rem01, console, EGW, console1a, Central1, Central1a, IPMA, IPMA1a, PSTN1
Rem01-PSTN-CSS	Remote 1 PSTN CCS	Rem-Extns, console, console1a
Rem02-Extn-CSS	Remote 2 Extn CCS	Rem-Extns, PSTN-Rem02, console, EGW, console1a, Central1, Central1a, IPMA, IPMA1a PSTN1
Rem02-PSTN-CSS	Remote 2 PSTN CCS	Rem-Extns, console, console1a

Route Plan > Route/Hunt > Route Group

To access the Cisco CallManager Administration web pages for adding and configuring route groups, choose **Route Plan > Route/Hunt > Route Group** from the Cisco CallManager Administration application.

Table 2-11 shows the route groups that were configured for the Multi-Site Single-Cluster Distributed scenario.

Table 2-11 Route Groups

Route Group Name	Route Group Members
EGW_GRP	ggb1-level2 (H225 Trunk)
EGW1aGRP	gb1-level8 (H225 Trunk)
PSTN_Access	S0/DS1-0@SDA00027E3913FC S0/DS1-0@SDA00027E3913FB S0/DS1-0@SDA00027E3913FA S0/DS1-0@SDA00027E3913F9 S0/DS1-0@SDA00027E3913F8
PSTN1a_access	S2/DS1-0@GB1a_CAT_6K_CMM S1/DS1-2@GB1a_CAT_6K_CMM S1/DS1-1@GB1a_CAT_6K_CMM S1/DS1-0@GB1a_CAT_6K_CMM
Rem01-PSTN	AALN/S1/SU0/1@TB1SRST1-2621XMMGCP AALN/S1/SU0/0@TB1SRST1-2621XMMGCP

Route Plan > Route Hunt > Route/Hunt List

To access the Cisco CallManager Administration web pages for adding and configuring route lists, choose **Route Plan > Route/Hunt List** from the Cisco CallManager Administration application.

Table 2-12 describes the route/hunt lists that were configured for the Multi-Site Single-Cluster Distributed scenario.

Table 2-12 Route/Hunt Lists

Route/Hunt List Name	Description	Cisco CallManager Group	Selected Route Groups	Called Number Prefix Digits
EGW1-rte	EGW 1 route group list	Central_1	EGW_GRP	blank
EGW1a-rte	EGW 1a route group list	Central_1a	EGW1aGRP	blank
EGWPSTN1-rte	EGW / PSTN 1 route group list	Central_1	PSTN_Access EGW_GRP	For PSTN_ACCESS: blank For EGW_GRP: 702
EGWPSTN1a-rte	EGW / PSTN 1a route group list	Central_1a	PSTN1a_access EGW1a_GRP	For PSTN1a_ access: blank For EGW1a_GRP: 703
PSTN1_rte	PSTN1_route_list	Central_1	PSTN_Access	blank
PSTN1a_rte	PSTN1a route list	Central_1a	PSTN1a_access	blank

Route Plan > Route Pattern/Hunt Pilot

To access the Cisco CallManager Administration web pages for adding and configuring route patterns, choose **Route Plan > Route Pattern/Hunt Pilot** from the Cisco CallManager Administration application.

Fifty-three route pattern/hunt pilots were configured. Table 2-13 describes the settings in the Route Pattern Configuration page for six route patterns, which are used for:

- Calls to the PSTN using overlap sending
- Call forward pattern to the PSTN using overlap sending
- EnBlock PSTN local calls with backup through the Cisco EGW 2200 legacy PBXs on trunk failure or congestion

Table 2-13 Route Pattern/Hunt Pilot Configuration

Field	Site A PSTN Route Pattern/Hunt Pilot Setting	Site B PSTN Route Pattern/Hunt Pilot Setting	Site A Callforward Route Pattern/Hunt Pilot Setting	Site B Callforward Route Pattern/Hunt Pilot Setting	Site A PSTN with PBX Backup Route Pattern/Hunt Pilot Setting	Site B PSTN with PBX Backup Route Pattern/Hunt Pilot Setting
Route Pattern	9.	9.	9.!	9.!	9.[2-8]XXXXX	9.[2-8]XXXXX
Partition	PSTN1	PSTN1a	CForward	CForward1a	PBXPSTN1	PBXPSTN1a
Description	PSTN Access	PSTN1a Access	PSTN access for Call Forward All	PSTN access for Call Forward All 1a	Access Alt PBX route 1	Access Alt PBX route 1a
Numbering Plan	North American Numbering Plan	North American Numbering Plan	North American Numbering Plan	North American Numbering Plan	North American Numbering Plan	North American Numbering Plan
Route Filter	<None>	<None>	<None>	<None>	<None>	<None>
MLPP Precedence	Default	Default	Default	Default	Default	Default
Gateway or Route/Hunt List	PSTN1_rte	PSTN1a_rte	PSTN1_rte	PSTN1a_rte	EGWPSTN1-rte	EGWPSTN1a-rte
Route Option	Route This Pattern	Route This Pattern	Route This Pattern	Route This Pattern	Route This Pattern	Route This Pattern
Provide Outside Dial Tone	Checked	Checked	Checked	Checked	Checked	Checked
Allow Overlap Sending	Checked	Checked	Checked	Checked	Unchecked	Unchecked
Urgent Priority	Unchecked	Unchecked	Unchecked	Unchecked	Unchecked	Unchecked

Table 2-13 Route Pattern/Hunt Pilot Configuration (continued)

Field	Site A PSTN Route Pattern/Hunt Pilot Setting	Site B PSTN Route Pattern/Hunt Pilot Setting	Site A Callforward Route Pattern/Hunt Pilot Setting	Site B Callforward Route Pattern/Hunt Pilot Setting	Site A PSTN with PBX Backup Route Pattern/Hunt Pilot Setting	Site B PSTN with PBX Backup Route Pattern/Hunt Pilot Setting
Use Calling Party's External Phone Number Mask	Checked	Checked	Checked	Checked	Checked	Checked
Calling Party Transform Mask	blank	blank	blank	blank	blank	blank
Prefix Digits (Outgoing Calls)	blank	blank	blank	blank	blank	blank
Calling Line ID Presentation	Default	Default	Default	Default	Default	Default
Calling Name Presentation	Default	Default	Default	Default	Default	Default
Connected Line ID Presentation	Default	Default	Default	Default	Default	Default
Connected Name Presentation	Default	Default	Default	Default	Default	Default
Discard Digits	PreDot	PreDot	PreDot	PreDot	PreDot	PreDot

Table 2-13 Route Pattern/Hunt Pilot Configuration (continued)

Field	Site A PSTN Route Pattern/Hunt Pilot Setting	Site B PSTN Route Pattern/Hunt Pilot Setting	Site A Callforward Route Pattern/Hunt Pilot Setting	Site B Callforward Route Pattern/Hunt Pilot Setting	Site A PSTN with PBX Backup Route Pattern/Hunt Pilot Setting	Site B PSTN with PBX Backup Route Pattern/Hunt Pilot Setting
Called Party Transform Mask	blank	blank	blank	blank	blank	blank
Prefix Digits (Outgoing Calls)	blank	blank	blank	blank	blank	blank
ISDN Network-Specific Facilities Information Element fields	not configured	not configured	not configured	not configured	not configured	not configured

Route Plan > Translation Pattern

To access the Cisco CallManager Administration web pages for adding and configuring translation patterns, choose **Route Plan > Translation Pattern** from the Cisco CallManager Administration application.

Table 2-14 describes the settings in the Translation Pattern Configuration page for three of the translation patterns that were configured for the Multi-Site Single-Cluster Distributed scenario. Nine other translation patterns were configured but are not shown in this table.

Table 2-14 Translation Pattern Configuration

Field	*67.! Translation Translation Pattern Settings	0 Translation Pattern Settings	701.!Translation Pattern Settings
Translation Pattern	*67.!	0	701.!
Partition	PSTN1	Central1a	Central1
Description	CLI Restrict for Site A	Operator console 1a	Incoming call with EGW Node number
Numbering Plan	North American Numbering Plan	North American Numbering Plan	North American Numbering Plan
Route Filter	<None>	<None>	<None>
MLPP Precedence	Default	Default	Default
Calling Search Space	Central1CSS	Central1aCSS	Central1CSS
Route Option	Route This Pattern	Route This Pattern	Route This Pattern
Provide Outside Dial Tone	Unchecked	Unchecked	Unchecked
Use Calling Party's External Phone Number Mask	Checked	Unchecked	Unchecked
Calling Party Transform Mask	blank	blank	blank
Prefix Digits (Outgoing Calls)	blank	blank	blank
Calling Party Presentation	Restricted	Default	Default
Discard Digits	PreDot	<None>	PreDot
Called Party Transform Mask	blank	1691	blank
Prefix Digits (Outgoing Calls)	blank	blank	blank

Cisco CallManager Service Configuration

The following sections provide an overview of how Cisco CallManager was configured on many of the Service menu web pages that you access from Cisco CallManager Administration. These sections do not describe all of the Service menu web pages or web page fields. Instead, they point out selected configuration information that will help you understand how Cisco CallManager was set up to perform most effectively.

- Service > Media Resource > Conference Bridge, page 2-20
- Service > Media Resource > Media Termination Point, page 2-21
- Service > Media Resource > Music On Hold Audio Source, page 2-22
- Service > Media Resource > Music On Hold Server, page 2-22
- Service > Media Resource > Transcoder, page 2-23
- Service > Media Resource > Media Resource Group, page 2-24
- Service > Media Resource > Media Resource Group List, page 2-25
- Service > Service Parameters, page 2-25

The following sections do not discuss these Service menu options:

- Cisco IPMA Configuration Wizard—Not configured
- Cisco CM Attendant Console—Not configured

Service > Media Resource > Conference Bridge

To access the Cisco CallManager Administration web pages for adding and configuring conference bridges, choose **Service > Media Resource > Conference Bridge** from the Cisco CallManager Administration application.

Table 2-15 describes the settings in the Conference Bridge Configuration page for one software conference bridge and one hardware conference bridge that were configured for the Multi-Site Single-Cluster Distributed scenario. Four other software conference bridges and 6 other hardware conference bridges were configured but are not shown in this table.

Table 2-15 Conference Bridge Configuration

Field	Software Conference Bridge Settings	Hardware Conference Bridge Settings
Conference Bridge Type	Cisco Conference Bridge Software	Cisco Conference Bridge Hardware
Host Server	10.10.50.69	—
MAC Address	—	0003FEAAEDC4
Conference Bridge Name	CFB_10.10.50.69	CFB0003FEAAEDC4
Description	SW Conference Bridge 1 Site A	Catalyst CMM DSP Farm
Device Pool	Central_1a	Central_1a
Location	Central_1a	Central_1a

Service > Media Resource > Media Termination Point

To access the Cisco CallManager Administration web pages for adding and configuring media termination points, choose **Service > Media Resource > Media Termination Point** from the Cisco CallManager Administration application.

Table 2-16 describes the settings in the Media Termination Point Configuration page for one media termination point that was configured for the Multi-Site Single-Cluster Distributed scenario. Five other media termination points were configured but are not shown in this table.

Table 2-16 Media Termination Point Configuration

Field	Setting
Host Server	10.10.30.6
Media Termination Point Name	MTP_GB_1_BACK1
Description	MTP on Site A Backup server
Device Pool	Default

Service > Media Resource > Music On Hold Audio Source

To access the Cisco CallManager Administration web pages for adding and configuring music on hold (MOH) audio sources, choose **Service > Media Resource > Music On Hold Audio Source** from the Cisco CallManager Administration application.

Table 2-17 describes how the MOH audio source was configured in the Music On Hold (MOH) Audio Source Configuration page.

Table 2-17 Music On Hold Audio Source Configuration

Field	MOH Audio Source 1 Settings
MOH Audio Source File	SampleAudioSource
MOH Audio Source Name	SampleAudioSource
Play Continuously	Checked
Allow Multicasting	Checked

Service > Media Resource > Music On Hold Server

To access the Cisco CallManager web pages for adding and configuring music on hold (MOH) servers, choose **Service > Media Resource > Music On Hold Server** from the Cisco CallManager Administration application.

Table 2-18 describes the settings in the Music On Hold (MOH) Server Configuration page.

Table 2-18 Music On Hold Server Configuration

Field	Setting
Host Server	10.10.30.6
Music on Hold Server Name	MOH_GB_1_BACK1
Description	HOH on Site A Backup Server
Device Pool	Default
Location	Central_1
Maximum Half Duplex Streams	250

Table 2-18 Music On Hold Server Configuration (continued)

Field	Setting
Maximum Multicast Connections	30
Fixed Audio Source Device	blank
Run Flag	Yes
Enable Multicast Audio Sources on this MOH Server	Unchecked
Base Multicast IP Address	0.0.0.0
Base Multicast Port Number	0
Increment Multicast on	Port Number

Service > Media Resource > Transcoder

To access the Cisco CallManager Administration web pages for adding and configuring transcoders, choose **Service > Media Resource > Transcoder** from the Cisco CallManager Administration application.

Two transcoder audio sources were configured for the Multi-Site Single-Cluster Distributed scenario. Table 2-19 describes how the transcoders were configured in the Transcoder Configuration page.

Table 2-19 Transcoder Configuration

Field	Transcoder 1 Settings	Transcoder 2 Settings
Transcoder Type	Cisco Media Termination Point Hardware	Cisco Media Termination Point Hardware
Description	MTP00027E3913FF	MTP0003FEAAEDC4
MAC Address	00027E3913FF	0003FEAAEDC4
Device Pool	Default	Central1a
Special Load Information	blank	blank

Service > Media Resource > Media Resource Group

To access the Cisco CallManager Administration web pages for adding and configuring media resource groups, choose **Service > Media Resource > Media Resource Group** from the Cisco CallManager Administration application.

Table 2-20 describes the settings in the Media Resource Group Configuration page for the media resource groups that were configured for the Multi-Site Single-Cluster Distributed scenario.

Table 2-20 Media Resource Group Configuration

Field	MGL_Central_1a Settings	MRG_2 Settings	MRG_Central Settings
Media Resource Group Name	MGL_Central_1a	MRG_2	MRG_Central
Description	MGL_Central_1a	Media Resource group 2	Media Resource group 1
Selected Media Resource	CFB0003FEAAEDC4 (CFB) MOH_10.10.30.5 (MOH) MTP0003FEAAEDC4 (XCODE)	CFB00027E3913FD (CFB) CFB00027E3913FE (CFB CFB00027E39145A (CFB) CFB00027E39145B (CFB) MOH_10.10.30.5 (MOH) MOH_GB_1_BACK1 (MOH) MTP00027E3913FF (XCODE)	CFB00027E3913FD (CFB) CFB00027E3913FE (CFB) CFB00027E39145A (CFB) CFB00027E39145B (CFB) MOH_10.10.30.5 (MOH) MOH_GB_1_BACK1 (MOH) MTP00027E3913FF (XCODE)
Use Multicast for MOH Audio	Not checked	Not checked	Not checked

Service > Media Resource > Media Resource Group List

To access the Cisco CallManager Administration web pages for adding and configuring media resource group lists, choose **Service > Media Resource > Media Resource Group List** from the Cisco CallManager Administration application.

Table 2-21 describes the settings in the Media Resource Group List Configuration page for the media resource group lists that were configured for the Multi-Site Single-Cluster Distributed scenario.

Table 2-21 Media Resource Group List Configuration

Field	MRGL_Central Setting	MRGL_Central_1a Setting	MRGL_Remote Setting
Media Resource Group Name	MRGL_Central	MRGL_Central_1a	MRGL_Remote
Selected Media Resource Groups	MRG_Central	MGL_Central_1a	MRG_2

Service > Service Parameters

To access the Cisco CallManager Administration web pages for adding and configuring services on selected servers, choose **Service > Service Parameters** from the Cisco CallManager Administration application.

Table 2-22 describes the settings in the Music On Hold (MOH) Server Configuration page for the servers used in the Multi-Site Single-Cluster Distributed scenario.

Table 2-22 Server Parameter Configuration Settings for Each Server

Service for all Servers	Settings
Cisco CallManager	All default settings except the following: • CDR Enabled Flag—True • CDR Log Calls with Zero Duration Flag—True • Call Diagnostics Enabled—True • Disable Nonregistered SCCP KeepAlives—True • Device Status Poll Interval (msec)—60000 • MGCP Database Query Delay Timer (msec)—300 • H225 DTMF Duration (msec)—300 • H225 TCP Timer (sec)—15 • H245 TCS Timeout—25 • Send H225 User Info Message—H225 Info for Call Progress Tone • Forward Maximum Hop Count—255 • Forward No Answer Timer (sec)—24 • Maximum MeetMe Conference Unicast—10 • Media Exchange Interface Capability Timer (sec)—25 • Media Exchange Timer (sec)—25 • Strip G.729 Annex B (Silence Suppression) from Capabilities—True • Route Plan Initialization Timer (sec)—120 • Suppress Debug Info for Router Death1 • Preferred G729 Millisecond Packet Size—30 • Code Yellow Entry Latency (msec)—15 • Code Yellow Duration (min)—30 • Max Db Updates Allowed—100 • Automated Alternate Routing Enable—True

Table 2-22 Server Parameter Configuration Settings for Each Server (continued)

Service for all Servers	Settings
Cisco CTIManager	Suppress Debug Info for Router Death—1
Cisco IP Manager Assistant	Cisco IPMA RNA Forwarding Flag—True

Cisco CallManager Feature Configuration

The following sections provide an overview of how Cisco CallManager was configured on many of the Feature menu web pages that you access from Cisco CallManager Administration. These sections do not describe all of the Feature menu web pages or web page fields. Instead, they point out selected configuration information that will help you understand how Cisco CallManager was set up to perform most effectively.

- Feature > Cisco IP Phone Services, page 2-27
- Feature > Voice Mail > Cisco Voice Mail Port, page 2-28
- Feature > Voice Mail > Message Waiting, page 2-29
- Feature > Voice Mail > Voice Mail Pilot, page 2-30
- Feature > Voice Mail > Voice Mail Profile, page 2-30

The following sections do not discuss these Feature menu options:

- Call Park—Not configured
- Find and List Call Pickup Number—Not configured
- Find and List Meet-Me Numbers—Not configured

Feature > Cisco IP Phone Services

To access the Cisco CallManager Administration web pages for adding and configuring phone services, choose **Feature > Cisco IP Phone Services** from the Cisco CallManager Administration application.

Eight IP phone services were configured for Cisco and third-party applications. Table 2-23 describes the settings in the Cisco IP Phone Services Configuration page for a sample IP phone service.

Table 2-23 Cisco IP Phone Services Configuration

Field	Setting
Service Name	IPMA service
Service Description	Service for IPMA
Service URL	http://10.10.30.6/ma/servlet/MAService?cmd=doPhoneService&Name=#DEVICENAME#&locale=english_united_states
Parameters	blank
Character Set	Western European (Latin1)

Feature > Voice Mail > Cisco Voice Mail Port

To access the Cisco CallManager Administration web pages for adding and configuring Cisco voice mail ports, choose **Feature > Voice Mail > Cisco Voice Mail Port** from the Cisco CallManager Administration application.

Table 2-24 describes the settings in the Cisco Voice Mail Port Configuration page for one Cisco voice mail port that was configured for the Multi-Site Single-Cluster Distributed scenario. One hundred and twenty-three other Cisco voice mail ports were configured but are not shown in this table.

Table 2-24 Cisco Voice Mail Port Configuration

Field	Setting
Port name	Unity-VI1
Description	Unity port 1
Device Pool	Default
Calling Search Space	Central1CSS
AAR Calling Search Space	<None>
Location	Central_1

Table 2-24 Cisco Voice Mail Port Configuration (continued)

Field	Setting
Directory Number	1100
Partition	VMPilotPartition
Calling Search Space	Central1CSS
AAR Group	<None>
Display (Internal Caller ID)	Voicemail
External Number mask	blank

Feature > Voice Mail > Message Waiting

To access the Cisco CallManager Administration web pages for adding and configuring message waiting number, choose **Feature > Voice Mail > Message Waiting** from the Cisco CallManager Administration application.

Table 2-25 describes the settings in the Message Waiting Configuration page for two of the message waiting numbers that were configured for the Multi-Site Single-Cluster Distributed scenario. Two other message waiting numbers were configured but are not shown in this table.

Table 2-25 Message Waiting Configuration

Field	Message Waiting Number 1 Settings	Message Waiting Number 2 Settings
Message Waiting Number	1096	1097
Description	tb1aunity1onMWI	tb1aunity1OFFMWI
Message Waiting Indicator	On	Off
Partition	Central1a	Central1a
Calling Search Space	Central1aCSS	Central1aCSS

Feature > Voice Mail > Voice Mail Pilot

To access the Cisco CallManager Administration web pages for adding and configuring voice mail pilots, choose **Feature > Voice Mail > Voice Mail Pilot** from the Cisco CallManager Administration application.

Table 2-26 describes the settings in the Voice Mail Pilot Configuration page for one of the voice mail pilots that was configured for the Multi-Site Single-Cluster Distributed scenario. Two other Cisco voice mail pilots were configured but are not shown in this table.

Table 2-26 Voice Mail Pilot Configuration

Field	Setting
Voice Mail Pilot Number	1700
Description	Site B unity1 pilot
Calling Search Space	Central1aCSS
Make this the default Voice Mail Pilot for the system	Unchecked

Feature > Voice Mail > Voice Mail Profile

To access the Cisco CallManager Administration web pages for adding and configuring voice mail profiles, choose **Feature > Voice Mail > Cisco Voice Mail Profile** from the Cisco CallManager Administration application.

Table 2-27 describes the settings in the Voice Mail Profile Configuration page for one of the voice mail profiles that was configured for the Multi-Site Single-Cluster Distributed scenario. Two other Cisco voice mail profiles were configured but are not shown in this table.

Table 2-27 Voice Mail Profile Configuration

Field	Setting
Voice Mail Profile Name	VMnamesiteA
Description	VM name Site A
Voice mail Pilot	1100/Central1CSS

Table 2-27 Voice Mail Profile Configuration (continued)

Field	Setting
Voice Mail Box Mask	blank
Make this the default voice mail profile for the system	Unchecked

Cisco CallManager Device Configuration

The following sections provide an overview of how Cisco CallManager was configured on many of the Device menu web pages that you access from Cisco CallManager Administration. These sections do not describe all of the Device menu web pages or web page fields. Instead, they point out selected configuration information that will help you understand how Cisco CallManager was set up to perform most effectively.

- Device > CTI Route Point, page 2-31
- Device > Gatekeeper, page 2-34
- Device > Gateway, page 2-35
- Device > Phone, page 2-42
- Device > Trunk, page 2-47
- Device > Device Settings > Device Profile, page 2-50

The following sections do not discuss these Device menu options:

- Device Settings > Firmware Load Information—Use default settings
- Device Settings > Phone Button Template—Use default settings
- Device Settings > Softkey Template—Use default settings

Device > CTI Route Point

To access the Cisco CallManager Administration web pages for adding and configuring CTI route points, choose **Device > CTI Route Point** from the Cisco CallManager Administration application.

Table 2-28 describes the settings on the CTI Route Point Configuration page for four CTI route points that were configured for third-party attendant consoles in the Multi-Site Single-Cluster Distributed scenario. Table 2-29 shows how one of the directory numbers was configured for each example route point.

Fifteen other CTI route points were configured, including route points for Cisco CRA, Cisco Personal Assistant, and Cisco Emergency Responder. These route points are not shown in the following tables.

Table 2-28 CTI Route Point Configuration

Field	CTI Route Point 1 Setting	CTI Route Point 2 Setting	CTI Route Point 3 Setting	CTI Route Point 4 Setting
Device Name	Con_1_Ext_CallQ	Con_1_Int_CallQ	Con_1_PreCT_Ext	Con_1_PreCT_Int
Description	Console 1 Ext Call Q	Console 1 Int Call Q	Console 1 PreCT Ext	Console 1 PreCT Int
Device Pool	Default	Default	Default	Default
Calling Search Space	consearch	consearch	consearch	consearch
Location	<None>	<None>	<None>	<None>
Media Resource Group List	<None>	<None>	<None>	<None>
User Hold Audio Source	<None>	<None>	<None>	<None>
Network Hold Audio Source	<None>	<None>	<None>	<None>

Table 2-29 Directory Number Configuration for CTI Route Points

Field	CTI Route Point 1 Setting	CTI Route Point 2 Setting	CTI Route Point 3 Setting	CTI Route Point 4 Setting
Directory Number	1644	1643	1642	1641
Partition	console	console	console	console
Voice Mail Profile	<None>	<None>	<None>	<None>

Table 2-29 Directory Number Configuration for CTI Route Points (continued)

Field	CTI Route Point 1 Setting	CTI Route Point 2 Setting	CTI Route Point 3 Setting	CTI Route Point 4 Setting
Calling Search Space	National_css	PA_CSS	consearch	consearch
AAR Group	<None>	<None>	<None>	<None>
User Hold Audio Source	<None>	<None>	<None>	<None>
Network Hold Audio Source	<None>	<None>	<None>	<None>
Auto Answer	Not available on this device	Not available on this device	Not available on this device	Not available on this device
Forward All	default settings	default settings	default settings	default settings
Forward Busy	default settings	default settings	default settings	default settings
Forward No Answer	default settings	default settings	Checked with 1696 destination and consearch calling search space	Checked with 1695 destination and consearch calling search space
Forward On Failure	default settings	default settings	default settings	default settings
No Answer Ring Duration	blank	blank	blank	blank
Call Pickup Group	<None>	<None>	<None>	<None>
Target (Destination)	blank	blank	blank	blank
Calling Search Space	<None>	<None>	<None>	<None>
No Answer Ring Duration	blank	blank	blank	blank
Display (Internal Caller ID)	blank	blank	blank	blank

Table 2-29 Directory Number Configuration for CTI Route Points (continued)

Field	CTI Route Point 1 Setting	CTI Route Point 2 Setting	CTI Route Point 3 Setting	CTI Route Point 4 Setting
Line Text Label	Not available on this device	Not available on this device	Not available on this device	Not available on this device
External Phone Number Mask	blank	blank	blank	blank
Line Text Label	Not available on this device	Not available on this device	Not available on this device	Not available on this device
Message Waiting Lamp Policy	Not available on this device	Not available on this device	Not available on this device	Not available on this device
Ring Setting (Phone Idle)	Not available on this device	Not available on this device	Not available on this device	Not available on this device
Ring Setting (Phone Active)	Not available on this device	Not available on this device	Not available on this device	Not available on this device
Maximum Number of Calls	5000	5000	5000	5000
Busy Trigger	4500	4500	4500	4500
Caller Name	Checked	Checked	Checked	Checked
Redirected Number	Unchecked	Unchecked	Unchecked	Unchecked
Caller Number	Unchecked	Unchecked	Unchecked	Unchecked
Dialed Number	Unchecked	Unchecked	Unchecked	Unchecked
Character Set	Western European (Latin 1)	Western European (Latin 1)	Western European (Latin 1)	Western European (Latin 1)

Device > Gatekeeper

To access the Cisco CallManager Administration web pages for adding and configuring gatekeepers, choose **Device > Gatekeeper** from the Cisco CallManager Administration application.

A gatekeeper was configured at each site in the Multi-Site Single-Cluster Distributed scenario. Table 2-30 describes the settings in the Gatekeeper Configuration page for these gatekeepers.

Table 2-30 Gatekeeper Configuration

Field	Site A Gatekeeper Settings	Site B Gatekeeper Settings
Host Name/IP Address	10.10.30.73	10.10.50.73
Description	Site A 3745 gatekeeper	Site B 3745 gatekeeper
Registration Request Time To Live	60	60
Registration Retry Timeout	300	300
Enable Device	checked	checked

Device > Gateway

To access the Cisco CallManager Administration web pages for adding and configuring gateways, choose **Device > Gateway** from the Cisco CallManager Administration application.

This section shows how one analog gateway (Analog-6624) and one digital gateway (Digital-CMM) were configured for the Multi-Site Single-Cluster Distributed scenario. Ten other gateways were configured but are not described in this section.

Table 2-31 describes how the 6624 analog gateway was configured in the Gateway Configuration page. Table 2-32 describes how one of the directory numbers for this analog gateway was configured in the Port Configuration page. Table 2-33 describes how one of the directory numbers for this analog gateway was configured in the Directory Number Configuration page.

Note The NSE Type setting on the POTS Port Configuration page should be `IOS Gateways` for a 6624 analog gateway that is used for FAX.

Table 2-34 describes how the CMM digital gateway was configured in the Gateway Configuration page. Table 2-35 describes how one of the end points for this digital gateway was configured in the Gateway Configuration page.

Note The NSE Type setting on the POTS Port Configuration page should be **`IOS Gateways`** for a CMM digital gateway with a 24-port FXS port adapter that is used for FAX.

Table 2-31 Analog Gateway Configuration

Field	Setting
MAC Address	00E0140D1EEC
Description	SAA00E0140D1EEC
Device Pool	Default
Load Information	Blank
Network Locale	United Kingdom
Location	Central_
AAR Group	<None>
Calling Search Space	Central1CSS
AAR Calling Search Space	<None>
Media Resource Group List	MRGL_Central
Network Hold Audio Source	1 - SampleAudioSource
Port Selection Order	Top Down
MLPP Domain	blank
MLPP Indication	Not available on this device
MLPP Preemption	Not available on this device
SNMP Community String	Public
Disable SNMP Set operations	Unchecked

Table 2-32 Port Configuration for 6624 Analog Gateway

Field	Setting
Port Direction	boundBothways
Prefix DN	blank
Num Digits	0
Expected Digits	0
SMDI Port Number	0
Audio Signal Adjustment into IP Network	Minus1db
Audio Signal Adjustment from IP Network	Minus6db
Digit On Duration	100
Interdigit Duration	100
Hold Tone Silence Duration	0
Port Used for Voice Calls	Checked
Port Used for Modem Calls	Checked
Port Used for Fax Calls	Checked
Port Codec Parameter Selection	0
Fax Relay Enable	Checked
Fax Error Correction Mode Override	Checked
Maximum Fax Rate	14400bps
Fax Payload Size	20
Non Standard Facilities Country Code	65525
Non Standard Facilities Vendor Code	65525
Fax/Modem Packet Redundancy	Unchecked
Initial Playout Delay	40
Minimum Playout Delay	20
Maximum Playout Delay	150
Call Restart Timer	5000

Table 2-32 Port Configuration for 6624 Analog Gateway (continued)

Field	Setting
Offhook Validation Timer	100
Onhook Validation Timer	250
Hookflash Timer	1000
Echo TailLength	64 ms
Minimum ERL	6 db

Table 2-33 Directory Number Configuration for 6624 Analog Gateway

Field	Setting
Directory Number	2101
Partition	Central1
Voice Mail Profile	Default
Calling Search Space	Central1CSS
AAR Group	<None>
Auto Answer	Not available on this device
Network Hold Audio Source	1 - SampleAdioSource
Forward All	Calling search space set to CforwardCSSs
Forward Busy	default settings
Forward No Answer	default settings
No Answer Ring Duration	blank
Call Pickup Group	default settings
Target (Destination)	blank
Calling Search Space	<None>
No Answer Ring Duration	blank
Display (Internal Caller ID)	6624-1
Line Text Label	Not available on this device

Table 2-33 Directory Number Configuration for 6624 Analog Gateway (continued)

Field	Setting
External Phone Number Mask	0162845XXXX
Message Waiting Lamp Policy	Not available on this device
Ring Setting (Phone Idle)	Not available on this device
Ring Setting (Phone Active)	Not available on this device
Maximum Number of Calls	2
Busy Trigger	1
Caller Name	Checked
Redirected Number	Unchecked
Caller Number	Unchecked
Dialed Number	Checked
Character Set	Western European (Latin1)

Table 2-34 Digital Gateway Configuration

Field	Setting
Domain Name	GB1a_CAT_6K_CMM
Description	GB1a_CAT_6K_CMM1
Cisco CallManager Group	Central_1a
Module in Slot 1	WS-X6600-6E1
Module in Slot 2	WS-X6600-24FXS
Module in Slot 3	<None>
Module in Slot 4	<None>
Switchback Timing	Graceful
Global ISDN Switch Type	EURO
Switchback uptime-delay (min)	10

Table 2-34 Digital Gateway Configuration (continued)

Field	Setting
Switchback schedule (hh:mm)	12:00
Fax mode	Fax Relay

Table 2-35 End Point Configuration for Digital-CMM Digital Gateway

Field	Setting
End-Point Name	S1/DS1-0@GB1a_CAT_6K_CMM
Description	S1/DS1-0@GB1a_CAT_6K_CMM
Device Pool	Central_1a
Network Locale	United Kingdom
Media Resource Group List	MRGL_Central
Location	Central_1a
AAR Group	<None>
Load Information	blank
MLPP Domain	blank
MLPP Indication	Not available on this device
MLPP Preemption	Not available on this device
PRI Protocol Type	PRI EURO
Protocol Side	User
Channel Selection Order	Bottom Up
Channel IE Type	Use Number when 1B
PCM Type	A-law
Delay for first restart (1/8 sec ticks)	32
Delay between restarts (1/8 sec ticks)	4
Inhibit restarts at PRI initialization	Checked
Enable status poll	Unchecked

Table 2-35 End Point Configuration for Digital-CMM Digital Gateway (continued)

Field	Setting
Significant Digits	4
Calling Search Space	PSTN1aCSS
AAR Calling Search Space	PSTN1aCSS
Prefix DN	blank
Calling Party Presentation	DefaulT
Calling Party Selection	Originator
Called party IE number type unknown	Cisco CallManager
Calling party IE number type unknown	Cisco CallManager
Called Numbering Plan	Cisco CallManager
Calling Numbering Plan	Cisco CallManager
Number of digits to strip	0
Caller ID DN	blank
SMDI Base Port	0
Display IE Delivery	Checked
Redirecting Number IE Delivery - Outbound	Checked
Redirecting Number IE Delivery - Inbound	Checked
Send Extra Leading Character In DisplayIE	Checked
Setup non-ISDN Progress Indicator IE Enable	Checked
MCDN Channel Number Extension Bit Set to Zero	Unchecked
Send Calling Name In Facility IE	Unchecked
Interface Identifier Present	Unchecked
Interface Identifier Value	0
Line Coding	HDB3

Table 2-35 End Point Configuration for Digital-CMM Digital Gateway (continued)

Field	Setting
Framing	Non CRC4
Clock	External Primary
Input Gain (-6..14 db)	0
Output Attenuation (-6..14 db)	0
Echo Cancellation Enable	Enable
Echo Cancel Coverage (ms)	128

Device > Phone

To access the Cisco CallManager Administration web pages for adding and configuring Cisco IP Phones and Cisco Analog Telephone Adaptors (ATAs), choose **Device > Phone** from the Cisco CallManager Administration application.

This section shows how one phone (Cisco IP Phone 7960) and one ATA (ATA 186) were configured for the Multi-Site Single-Cluster Distributed scenario. Two thousand other such devices were configured but are not described in this section.

Note CTI ports were configured by adding a new phone with a Phone Type of CTI Port.

Table 2-36 describes how the Cisco IP Phone 7960 device and the ATA 186 device were configured in the Phone Configuration page. Table 2-37 describes how one of the directory numbers for each of these devices was configured in the Directory Number Configuration page for that device.

Table 2-36 Phone Configuration

Field	Cisco IP Phone 7960 Setting	ATA 186 Setting
MAC Address	000B5FAA9E17	0008A3D31127
Description	SEP000B5FAA9E17	ATA0008A3D31127
Owner User ID	blank	blank

Table 2-36 Phone Configuration (continued)

Field	Cisco IP Phone 7960 Setting	ATA 186 Setting
Device Pool	Default	Default
Calling Search Space	Central1CSS	Central1CSS
AAR Calling Search Space	<None>	<None>
Media Resource Group List	MRGL_Central	MRGL_Central
User Hold Audio Source	1 - SampleAudioSource	1 - SampleAudioSource
Network Hold Audio Source	1 - SampleAudioSource	1 - SampleAudioSource
Location	<None>	<None>
User Locale	English United States	English United States
Network Locale	United Kingdom	United Kingdom
Device Security Mode	Use System Default	—
Built In Bridge	On	—
Privacy	Off	—
Retry Video Call as Audio	Checked	—
Phone Button Template	Standard 7960	Standard ATA 186
Softkey Template	Standard User with CallBack	—
Module 1	<None>	—
Module 2	<None>	—
Phone Load Name	blank	blank
Module 1 Load Name	blank	—
Module 2 Load Name	blank	—
Information	blank	—
Directory	blank	—

Table 2-36 Phone Configuration (continued)

Field	Cisco IP Phone 7960 Setting	ATA 186 Setting
Device Pool	Default	Default
Calling Search Space	Central1CSS	Central1CSS
AAR Calling Search Space	<None>	<None>
Media Resource Group List	MRGL_Central	MRGL_Central
User Hold Audio Source	1 - SampleAudioSource	1 - SampleAudioSource
Network Hold Audio Source	1 - SampleAudioSource	1 - SampleAudioSource
Location	<None>	<None>
User Locale	English United States	English United States
Network Locale	United Kingdom	United Kingdom
Device Security Mode	Use System Default	—
Built In Bridge	On	—
Privacy	Off	—
Retry Video Call as Audio	Checked	—
Phone Button Template	Standard 7960	Standard ATA 186
Softkey Template	Standard User with CallBack	—
Module 1	<None>	—
Module 2	<None>	—
Phone Load Name	blank	blank
Module 1 Load Name	blank	—
Module 2 Load Name	blank	—
Information	blank	—
Directory	blank	—

Table 2-36 Phone Configuration (continued)

Field	Cisco IP Phone 7960 Setting	ATA 186 Setting
Messages	blank	—
Services	blank	—
Authentication Server	blank	—
Proxy Server	blank	—
Idle	blank	—
Idle Timer (seconds)	blank	—
Enable Extension Mobility Feature	Checked	—
Log Out Profile	<Use Current Device Settings>	—
MLPP Domain	blank	blank
MLPP Indication	Default	Not available on this device
MLPP Preemption	Default	Not available on this device
Disable Speakerphone	Unchecked	—
Disable Speakerphone and Headset	Unchecked	—
Forwarding Delay	Disabled	—
PC Port	Enabled	—
Settings Access	Enabled	—
Gratuitous ARP	Enabled	Enabled
PC Voice VLAN Access	Enabled	—
Video Capabilities	Disabled	—
Auto Line Select	Disabled	—
Web Access	Enabled	—

Table 2-37 Directory Number Configuration for Phone and ATA

Field	Cisco IP Phone 7960 Setting	ATA 186 Setting
Directory Number	2130	2125
Partition	Central1	Central1
Voice Mail Profile	Default	Default
Calling Search Space	Central1CSS	Central1CSS
AAR Group	<None>	<None>
User Hold Audio Source	1 - SampleAudioSource	1 - SampleAudioSource
Network Hold Audio Source	1 - SampleAudioSource	1 - SampleAudioSource
Auto Answer	Auto Answer Off	Not available on this device
Forward All	Voice Mail: Unchecked Destination: blank Calling Search Space: CforwardCSS	Voice Mail: Unchecked Destination: blank Calling Search Space: <None>
Forward Busy	Voice Mail: Checked Destination: blank Calling Search Space: <None>	Voice Mail: Unchecked Destination: blank Calling Search Space: <None>
Forward No Answer	Voice Mail: Checked Destination: blank Calling Search Space: <None>	Voice Mail: Unchecked Destination: blank Calling Search Space: <None>
No Answer Ring Duration	blank	blank
Call Pickup Group	<None>	<None>
Target (Destination)	blank	blank
Calling Search Space	<None>	<None>

Table 2-37 Directory Number Configuration for Phone and ATA (continued)

Field	Cisco IP Phone 7960 Setting	ATA 186 Setting
No Answer Ring Duration	blank	blank
Display (Internal Caller ID)	Site A 7960	Site A ATA 1
Line Text Label	blank	Not available on this device
External Phone Number Mask	0162845XXXX	0162845XXXX
Message Waiting Lamp Policy	Use System Policy	Not available on this device
Ring Setting (Phone Idle)	Use System Default	Not available on this device
Ring Setting (Phone Active)	Use System Default	Not available on this device
Maximum Number of Calls	2	2
Busy Trigger	1	1
Caller Name	Checked	Checked
Redirected Number	Unchecked	Unchecked
Caller Number	Unchecked	Unchecked
Dialed Number	Checked	Checked
Character Set	Western European (Latin 1)	Western European (Latin 1)

Device > Trunk

To access the Cisco CallManager Administration web pages for trunks, choose **Device > Trunk** from the Cisco CallManager Administration application.

Sixteen four trunks were configured for the Multi-Site Single-Cluster Distributed scenario. Table 2-38 describes the settings in the Trunk Configuration page for the following trunks, which were added with the characteristics shown:

- gb1-level2—Connects via a gatekeeper to the Cisco EGW
  - Product: H.225 Trunk (Gatekeeper Controlled)
  - Device Protocol: H.225
- gb1a-gk-trk4—Connects via a gatekeeper to Cisco CallManager Express nodes
  - Product: H.225 Trunk (Gatekeeper Controlled)
  - Device Protocol: H.225

Table 2-38 Trunk Configuration

Field	Trunk 1 Settings	Trunk 2 Settings
Device Name	SiteA-level2	SiteB-gk-trk4
Description	Site A Level 2 access	SiteB-gk-trk for 4xxx
Device Pool	Default	CME
Media Resource Group List	MRGL_Central	MRGL_Remote
Location	Central_1	<None>
AAR Group	<None>	<None>
Media Termination Point Required	Unchecked	Unchecked
Retry Video Call as Audio	Checked	Checked
Destination Address	—	—
Wait for Far End H.245 Terminal Capability Set	Checked	Checked
Significant Digits	All	All
Calling Search Space	—	Central1aCSS
AAR Calling Search Space	<None>	Central1aCSS

Table 2-38 Trunk Configuration (continued)

Field	Trunk 1 Settings	Trunk 2 Settings
Prefix DN	blank	blank
Redirecting Number IE Delivery - Inbound	Checked	Checked
Calling Party Selection	Originator	Originator
Calling Line ID Presentation	Default	Default
Called party IE number type unknown	Cisco CallManager	Cisco CallManager
Calling party IE number type unknown	Cisco CallManager	Cisco CallManager
Called Numbering Plan	Cisco CallManager	Cisco CallManager
Calling Numbering Plan	Cisco CallManager	Cisco CallManager
Caller ID DN	blank	blank
Display IE Delivery	checked	checked
Redirecting Number IE Delivery - Outbound	checked	Unchecked
Gatekeeper Name	10.10.50.73	10.10.50.73
Terminal Type	Gateway	Gateway
Technology Prefix	2	2
Zone	egw1	egw1
MLPP Domain	blank	blank
MLPP Indication	Not available on this device	Not available on this device
MLPP Preemption	Not available on this device	Not available on this device

Device > Device Settings > Device Profile

To access the Cisco CallManager Administration web pages for adding and configuring device profiles, choose **Device > Device Settings > Device Profile** from the Cisco CallManager Administration application.

Table 2-39 describes the settings in the Device Profile page for one of the device profiles that was configured for the Multi-Site Single-Cluster Distributed scenario. Table 2-40 shows how one of the directory numbers was configured for the example device profile.

Six other device profiles were configured but are not shown in these tables.

Table 2-39 Device Profile Configuration

Field	Setting
Device Type	Cisco 7960
User Device Profile Name	2847
Description	2847-Extension Mobility
User Hold Audio Source	1 - SampleAudioSource
User Locale	English United States
Phone Button Template	Standard 7960
Softkey Template	Standard User with CallBack
Module 1	<None>
Module 2	<None>
MLPP Domain	blank
MLPP Indication	Default
MLPP Preemption	Default
Login User ID	blank

Table 2-40 Directory Number Configuration for 2487 Device Profile

Field	Setting
Directory Number	2847
Partition	Central1
Voice Mail Profile	<None>
Calling Search Space	Central1CSS
AAR Group	<None>
User Hold Audio Source	1 - SampleAudioSource
Network Hold Audio Source	1 - SampleAudioSource
Auto Answer	Auto Answer Off
Forward All	Voice Mail: Unchecked Destination: blank Calling Search Space: <None>
Forward Busy	Voice Mail: Unchecked Destination: blank Calling Search Space: <None>
Forward No Answer	Voice Mail: Checked Destination: blank Calling Search Space: CforwardCSS
No Answer Ring Duration	blank
Call Pickup Group	<None>
Target (Destination)	blank
Calling Search Space	<None>
No Answer Ring Duration	blank
Display (Internal Caller ID)	EM phone 2847
Line Text Label	blank
External Phone Number Mask	0162845XXXX
Message Waiting Lamp Policy	Use System Policy

Table 2-40 Directory Number Configuration for 2487 Device Profile (continued)

Field	Setting
Ring Setting (Phone Idle)	Use System Default
Ring Setting (Phone Active)	Use System Default
Maximum Number of Calls	4
Busy Trigger	2
Caller Name	Checked
Redirected Number	Unchecked
Caller Number	Unchecked
Dialed Number	Checked
Character Set	Western European (Latin 1)

Cisco CallManager User Configuration

This section provides an overview of how Cisco CallManager was configured in the User web pages that you access from Cisco CallManager Administration. It points out selected configuration information that will help you understand how Cisco CallManager was set up to perform most effectively.

To access the Cisco CallManager Administration web pages for configuring users, choose **User > Add a New User** (to add a new user) or choose **User > Global Directory** (to update an existing user) from the Cisco CallManager Administration application.

This section provides information for the following users. Approximately 2,000 other users were configured but are not shown in this table.

- Operator console—Example of a Cisco CallManager Global Directory User (such as for TAPI/JTAPI applications)
- IP communicator—Example of a Cisco IP Communicator user configured in Cisco CallManager

Table 2-41 describes the settings in the User Configuration page for these example users.

Table 2-41 User Configuration

Field	User 1 Settings	User 2 Settings
First Name	Attendant	IP
Last Name	Console	Communicator
User ID	AC	IP-Communicator
Telephone Number	blank	blank
Manager User ID	blank	blank
Department	blank	blank
User Locale	English United States	English United States
Enable CTI Application Use	Unchecked	Unchecked
Call Park Retrieval Allowed	Unchecked	Unchecked
Enable Calling Party Number Modification	Unchecked	Unchecked
Associated PC	Not Defined	Not Defined
Primary Extension	Not Defined	8142
Controlled Devices	2135	SEP000D608BC21
Enable Authentication Proxy Rights	False	False
Controlled Device Profiles	none	none

CHAPTER 3

Cisco Unity Configuration

This chapter provides an overview of how Cisco Unity was set up in IP Communications Systems Test Release 3.0 for EMEA IPT. It also provides information about using Cisco Unity with Microsoft Exchange.

This chapter does not include detailed installation and configuration instructions. Rather, it is intended to provide you with guidance as you set up the Cisco Unity component of your IPT solution.

Cisco Unity was installed on MCS-7837, MCS-7847 or MCS-7845H-2.4-ECS1 servers and configured according to the instructions in the Cisco Unity documentation. In general, default or recommended configuration values were used.

For detailed information about installing, configuring, and administering Cisco Unity, refer to the documentation at this URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_unity/index.htm

The this chapter includes the following topics:

Using Cisco Unity with Lotus Domino

The following sections provide information about how Cisco Unity was used in two of the test scenarios:

- Cisco Unity with Domino in the Single Site Scenario, page 3-2
- Cisco Unity with Domino in the Multi-Site Single-Cluster Distributed Scenario, page 3-3

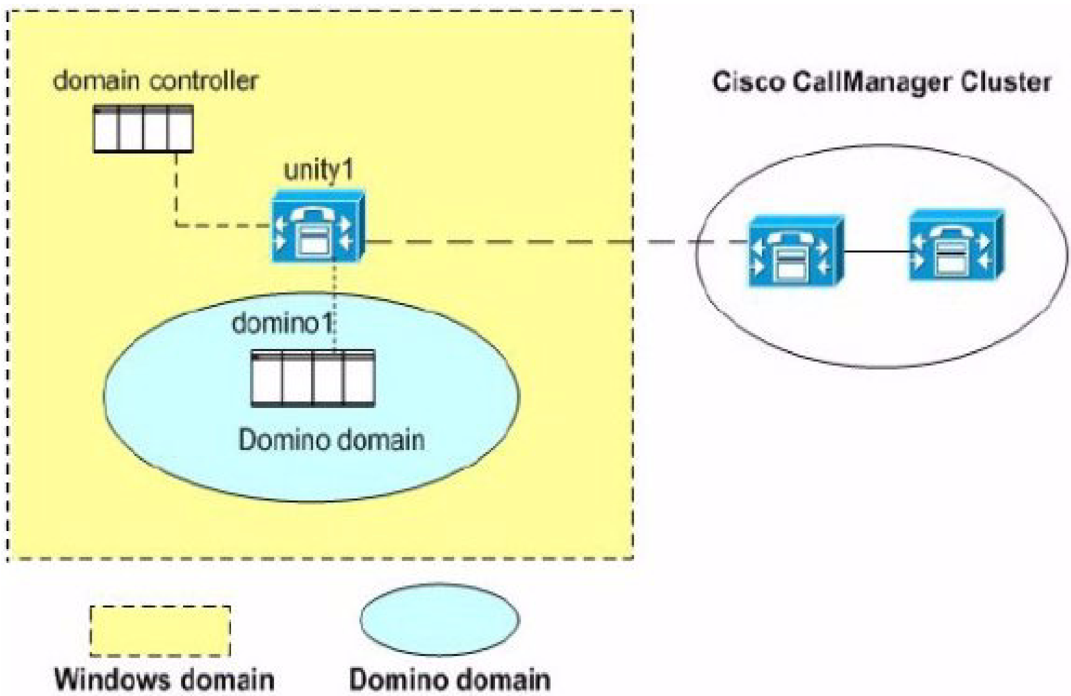
Cisco Unity with Domino in the Single Site Scenario

When setting up Cisco Unity with Lotus Domino in the Single Site scenario:

- The Windows domain and Domino domain are completely separate
- Cisco Unity should register to the Windows domain
- Lotus Notes users access only the Domino domain
- Cisco Unity does not support redundancy when using Domino as the message store

Figure 3-1 shows how Cisco Unity was deployed with Domino in the Single Site scenario.

Figure 3-1 Cisco Unity with Domino Topology in Single Site Scenario



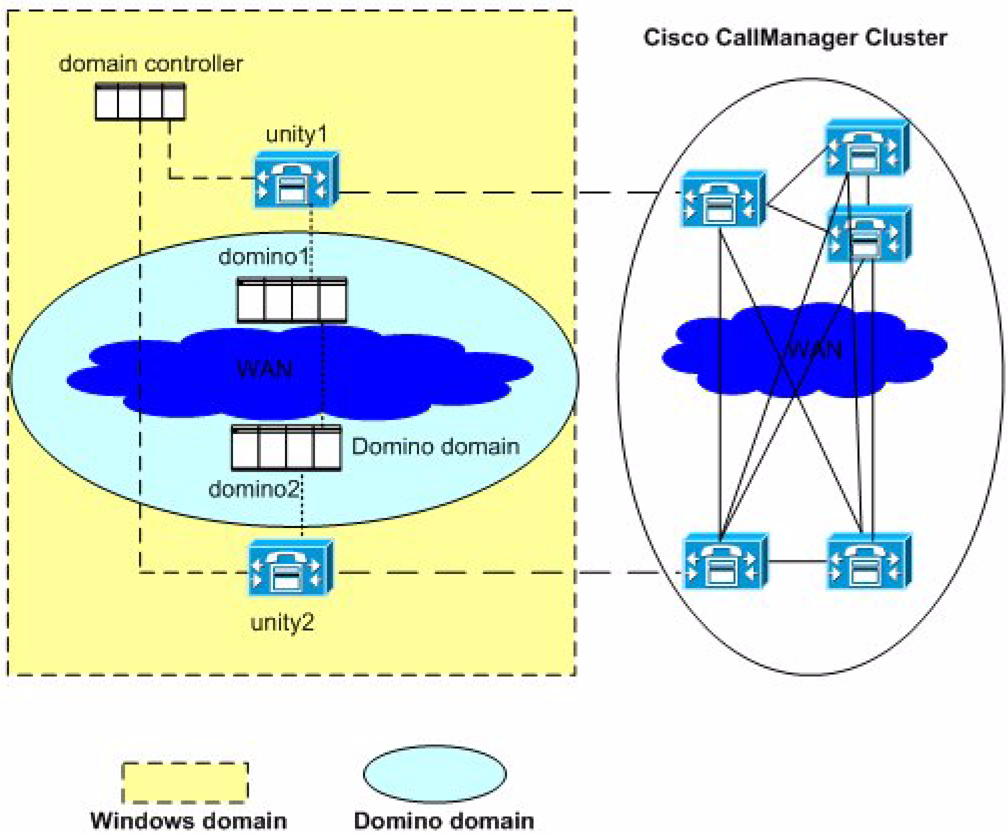
Cisco Unity with Domino in the Multi-Site Single-Cluster Distributed Scenario

When setting up Cisco Unity with Lotus Domino in the Multi-Site Single-Cluster Distributed scenario:

- All Cisco Unity servers belong to the same Windows domain
- All Domino servers belong to the same Domino domain
- A Lotus Notes account for a user should be stored on the local Domino server
- Cisco Unity does not support redundancy when using Domino as the message store

Figure 3-2 shows how Cisco Unity was deployed with Domino in the Multi-Site Single-Cluster Distributed scenario.

Figure 3-2 Cisco Unity with Domino Topology in Multi-Site Single-Cluster Distributed Scenario



Using Cisco Unity with Microsoft Exchange

This section provides information about using Cisco Unity with Microsoft Exchange in various site models.

In the Single Site scenario, Cisco Unity and Microsoft Exchange 2003 were installed on servers running the Windows 2003 Server operating system for use with Microsoft Active Directory 2003.

For detailed configuration information, refer to *Cisco Unity Reconfiguration and Upgrade Guide (With Microsoft Exchange)*. If you are a registered Cisco.com user, you can access this document at this URL:

http://www.cisco.com/en/US/partner/products/sw/voicesw/ps2237/products_upgrade_guides_book09186a0080222fdf.html

For information about using Cisco Unity with Windows 2003 Server, refer to *White Paper: Using Microsoft Windows Server 2003 with Cisco Unity 4.0(4)*, which is available at this URL:

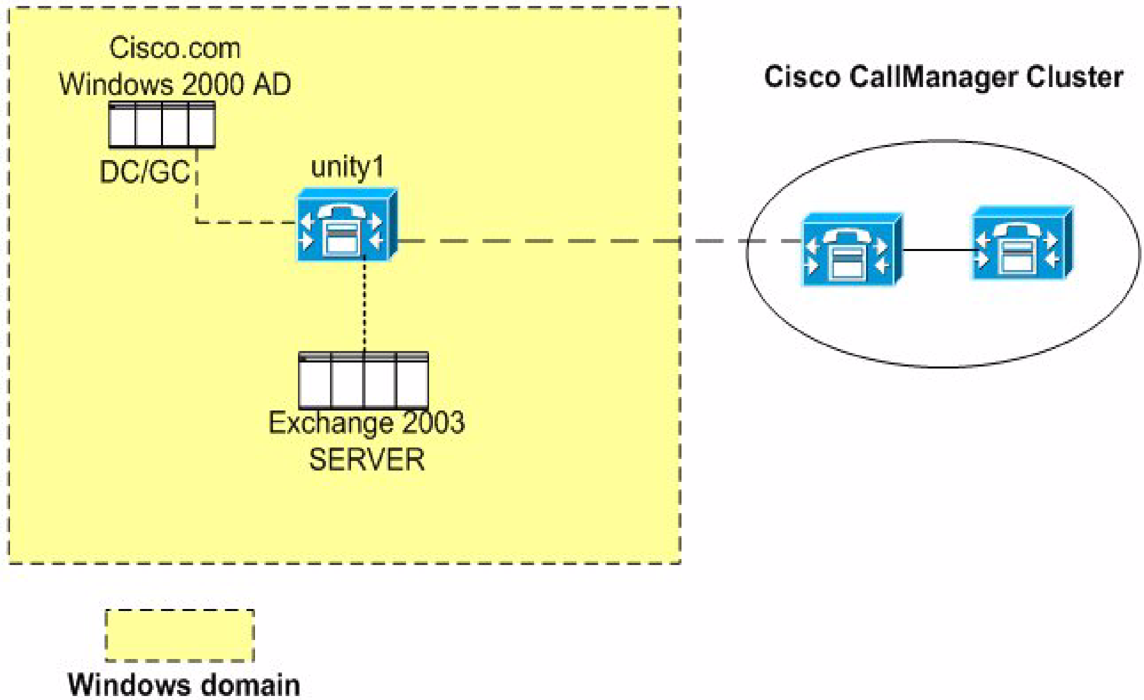
http://www.cisco.com/univercd/cc/td/doc/product/voice/c_unity/whitpapr/404win03.htm

The following reconfiguration procedure was tested for IP Communications Systems Test Release 3.0 for EMEA IPT:

1. In the Medium Site model, migrate Microsoft Exchange 5.5 users to Microsoft Exchange 2003 using the Move Mailbox method.
2. Then, reconfigure Cisco Unity for Exchange 2003. Make sure to run the Exchange 2003 forestprep and domainprep procedures before installing Exchange 2003.

Figure 3-1 shows how Cisco Unity was deployed with Microsoft Exchange in the Single Site scenario.

Figure 3-3 Cisco Unity with Microsoft Exchange Topology in Small Site Scenario



Integrating Cisco Unity with Cisco Enterprise Gateway

In IP Communications Systems Test Release 3.0 for EMEA IPT, Cisco Unity was integrated with Cisco CallManager and with Cisco EGW 2200 Enterprise Gateway (EGW) using the following protocols:

- For Cisco CallManager—SCCP
- For EGW—Session Initiation Protocol (SIP)

The integration of Cisco Unity with Cisco CallManager uses the standard procedures that are described in *Cisco CallManager 4.0 Integration Guide for Cisco Unity 4.0*, which is available at this URL:

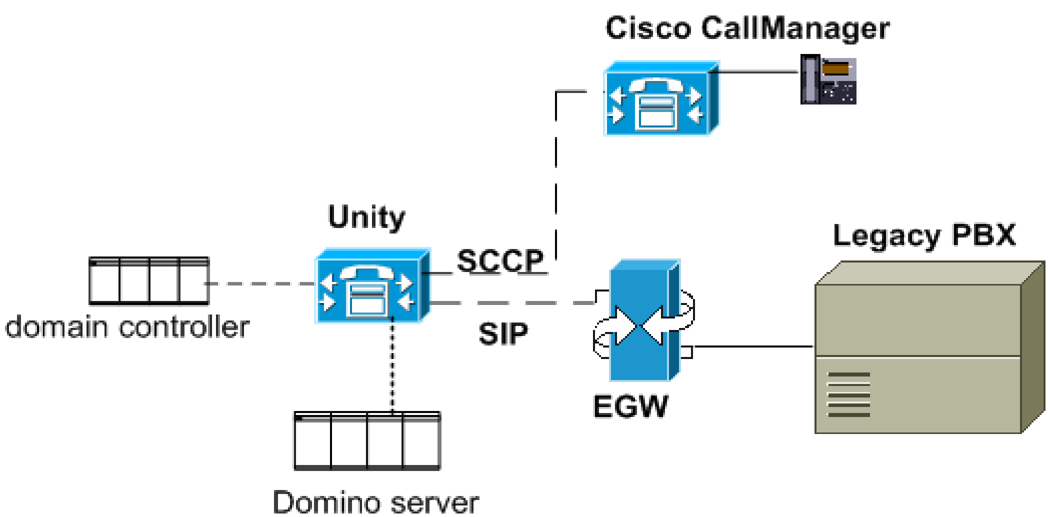
http://www.cisco.com/univercd/cc/td/doc/product/voice/c_unity/integuid/calma40/index.htm

For additional information, refer to *Dual Phone System Integration Guide for Cisco Unity 4.0*. This document is available to registered Cisco.com users at this URL:

http://www.cisco.com/en/US/partner/products/sw/voicesw/ps2237/prod_configuration_guide09186a0080211b2e.html

Figure 3-4 shows how Cisco Unity was integrated with Cisco CallManager and EGW.

Figure 3-4 Cisco Unity Integration with Cisco CallManager and Cisco EGW



Using Cisco Unity with Cisco IPMA

If Cisco IP Manager Assistant (IPMA) is running and a caller leaves a message for a manager, the message waiting indicator (MWI) on the phone of the manager will not light. To avoid this situation, make sure that the Calling Search Space field in the Cisco CallManager Administration Message Waiting Configuration page for the MWI contains the manager partition as the highest priority.

Localizing Cisco Unity

Cisco Unity supports English, French, German, and Japanese versions of the third-party software shown in Table 3-1. Cisco Unity does not support mixing locales in third-party software products. For example, using Windows 2000 Server (ENU) with Microsoft Exchange 2000 Server (FRA) is not supported.

If you will localize Cisco Unity, see Table 3-1, which provides the following information:

- Desired Language—Language that you want to use
- Telephony User Interface—Language in which users hear prompts when accessing voice messages from a phones
- System Administration—Language in which the Cisco Unity administration user interface will appear
- Utilities—Language in which the Cisco Unity utilities user interface will appear
- Text-to-Speech—Language in which Cisco Unity will play text that it converts to speech

Table 3-1 Languages for Cisco Unity

Desired Language	Telephony User Interface	System Administration	Utilities	Text-to-Speech
ENU (US English)	ENU	ENU	ENU	ENU
ENG (UK English)	ENG	ENU	ENU	ENG
ENA (Australian English)	ENA	ENU	ENU	—
ENZ (New Zealand English)	ENZ	ENU	ENU	—

Table 3-1 Languages for Cisco Unity (continued)

Desired Language	Telephony User Interface	System Administration	Utilities	Text-to-Speech
DEU (German)	DEU	DEU	DEU	DEU
FRA (French)	FRA	FRA	FRA	FRA
ESO (Colombian Spanish)	ESO	ENU	ENU	ESP
ESP (European Spanish)	ESP	ENU	ENU	ESP
NLD (Dutch)	NLD	ENU	ENU	NLD
NOR (Norwegian)	NOR	ENU	ENU	NOR
ITA (Italian)	ITA	ENU	ENU	ITA
SVE (Swedish)	SVE	ENU	ENU	SVE
CHS (Mainland Mandarin)	CHS	ENU	ENU	CHS
CHT (Taiwan Mandarin)	CHT	ENU	ENU	CHT
JPN (Japanese)	JPN	JPN	JPN	JPN
KOR (Korean)	KOR	ENU	ENU	—
DAN (Danish)	DAN	ENU	ENU	DAN
PTB (Brazilian Portuguese)	PTB	ENU	ENU	PTB
PTG (Standard Portuguese)	PTG	ENU	ENU	PTG
CSY (Czech)	CSY	ENU	ENU	—
ZHH (Cantonese)	ZHH	ENU	ENU	—
ENX (TTY/TDD)	ENX	ENU	ENU	—
ARA (Arabic)	ARA	ENU	ENU	—

Upgrading From IP Communications Systems Test Release 2.0 when Using Cisco Unity

If you are upgrading from IP Communications Systems Test Release 2.0 for IPT and you are using Cisco Unity with Domino, refer to C*isco Unity Reconfiguration and Upgrade Guide (With IBM Lotus Domino)*. This document provides important upgrade and is available at this URL:

http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products_upgrade_guides_book09186a0080222c7d.html

If you are upgrading from IP Communications Systems Test Release 2.0 for IPT and you are using Cisco Unity with Microsoft Exchange, refer to the "Cisco Unity Voice-Mail Port Changes" in *Release Notes for Cisco CallManager Release 4.0(1)*. This section provides information about configuring voice mail ports and failover when integrating Cisco Unity with Cisco Callmanager 4.0.

The release notes are available at this URL:

http://cco/en/US/products/sw/voicesw/ps556/prod_release_note09186a00801e87a5.html

Cisco CallManager Express Configuration

This chapter provides configuration information for Cisco CallManager Express in IP Communications Systems Test Release 3.0 for EMEA IPT.

For related information about Cisco CallManager Express, refer to the documentation at this URL:

http://www.cisco.com/univercd/cc/td/doc/product/access/ip_ph/ip_ks/cme31/index.htm

This chapter includes the following topics:

Cisco CallManager Express Overview

Cisco CallManager Express was configured with a DHCP pool for IP phones. With this configuration, Cisco CallManager Express will automatically discover up to 30 phones and will automatically assign extension numbers to these devices. Cisco CallManager Express also will automatically configure each phone to have two lines and to forward on busy or no answer to a voice messaging system.

Cisco CallManager Express registers each phone to a gatekeeper, which allows the phones to make extension-to-extension calls to phones connected to remote Cisco CallManager Express and Cisco CallManager systems.

The following sections provide information about Cisco CallManager Express for H.323 and for use with Primary Rate Interface (PRI), Basic Rate Interface (BRI), and Foreign Exchange Office (FXO).

Cisco CallManager Express Configuration for PRI

This section shows a Cisco CallManager Express configuration file that can be used when Cisco CallManager Express uses PRI.

In the configuration file below:

- Cisco CallManager Express uses the Dial-plan pattern command in Telephony-service to convert extension numbers to full PSTN E.164 numbers.
- Phones use the Description command to display the full E.164 number.
- Because incoming calls can have a local, national, or unknown number type, which can cause incoming DDI/DID calls to fail, a Translation rule is applied on the Voice port to translate any type of incoming called number to a single format to be routed by the dial-plan pattern configured.

```
version 12.3
service nagle
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
!
hostname TB3CME5-2621XM
!
boot-start-marker
```

```
boot-end-marker
!
logging buffered 10000 debugging
logging rate-limit all 10
!
clock timezone GMT 0
clock summer-time BST date Mar 28 2004 1:00 Oct 31 2004 2:00
network-clock-participate slot 1
network-clock-participate wic 0
network-clock-select 2 E1 1/0
voice-card 1
!
no aaa new-model
ip subnet-zero
ip cef
ip tcp synwait-time 5
!
ip dhcp excluded-address 10.10.90.1 10.10.90.10
ip dhcp excluded-address 10.10.90.21 10.10.90.253
!
ip dhcp pool ITS
   network 10.10.90.0 255.255.255.0
   default-router 10.10.90.1
   domain-name site3.com
   option 150 ip 10.10.90.1
   dns-server 10.10.50.138
!
ip ftp username Administrator
ip ftp password cisco
ip domain list site3.com
ip domain timeout 5
no ip domain lookup
ip name-server 10.10.50.138
no ftp-server write-enable
isdn switch-type primary-net5
!
voice hunt user-busy
voice call send-alert
voice rtp send-recv
!
voice service voip
 redirect ip2ip
 h323
!
voice class codec 1
 codec preference 1 g711alaw
 codec preference 2 g711ulaw
 codec preference 3 g729r8 bytes 30
```

```
!
voice class codec 2
 codec preference 1 g729r8 bytes 30
 codec preference 2 g711alaw
 codec preference 3 g711ulaw
!
voice class codec 3
 codec preference 1 g729r8 bytes 30
!
controller E1 0/0
 framing NO-CRC4
 channel-group 0 timeslots 1-31 speed 64
!
controller E1 1/0
 framing NO-CRC4
 pri-group timeslots 1-31
 description PGW-8260 0191 731XXXX
!
translation-rule 1
 Rule 1 ^1......... 01
 Rule 2 ^731.... 0191731
!
interface FastEthernet0/0
 ip address 10.10.90.1 255.255.255.0
 speed 100
 full-duplex
!
interface Serial0/0:0
 no ip address
 encapsulation frame-relay IETF
 no fair-queue
 frame-relay class TB3CME5
 frame-relay traffic-shaping
 frame-relay lmi-type q933a
!
interface Serial0/0:0.1 point-to-point
 ip address 192.168.254.102 255.255.255.252
 frame-relay interface-dlci 100
 h323-gateway voip interface
 h323-gateway voip id TB3 ipaddr 10.10.50.203 1718
 h323-gateway voip h323-id TB3CME5-2621XM
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
```

```
interface Serial1/0:15
 no ip address
 no logging event link-status
 isdn switch-type primary-net5
 isdn overlap-receiving T302 5000
 isdn not-end-to-end 64
 isdn incoming-voice voice
 isdn send-alerting
 isdn sending-complete
 no cdp enable
!
router eigrp 2
 passive-interface FastEthernet0/0
 passive-interface FastEthernet0/1
 network 10.0.0.0
 network 192.168.254.0
 no auto-summary
!
ip classless
ip http server
!
map-class frame-relay TB3CME5
 frame-relay traffic-rate 192000 256000
!
tftp-server flash:P00303020214.bin
tftp-server flash:P00305000301.sbn
tftp-server flash:P00403020214.bin
tftp-server flash:CP7912010200SCCP031023A.sbin
tftp-server flash:CP79050101SCCP030530B31.zup
tftp-server flash:CP7905010200SCCP031023A.sbin
tftp-server flash:P00503010100.bin
tftp-server flash:S00103020002.bin
tftp-server flash:ata18x-v2-16-ms-030327b.zup
tftp-server flash:cmterm_7920.3.3-01-06.bin
tftp-server flash:CP7902010200SCCP031023A.sbin
!
control-plane
!
voice-port 1/0:15
 translate called 1
 echo-cancel coverage 32
 cptone GB
 timeouts interdigit 6
 timeouts ringing 600
 music-threshold -70
!
dial-peer cor custom
!
```

```
dial-peer voice 1 pots
 application session
 destination-pattern 9[1-9].....
 direct-inward-dial
 port 1/0:15
 no register e164
!
dial-peer voice 2 pots
 application session
 destination-pattern 9[1-9].....
 direct-inward-dial
 port 1/0:15
 no register e164
!
dial-peer voice 31 voip
 preference 9
 application session
 destination-pattern 90..........
 voice-class codec 2
 session protocol sipv2
 session target ipv4:10.10.50.148
 dtmf-relay sip-notify
 no vad
!
dial-peer voice 30 voip
 preference 1
 destination-pattern [0-6]...
 voice-class codec 2
 session target ras
 dtmf-relay h245-alphanumeric
 no vad
!
dial-peer voice 32 voip
 preference 2
 destination-pattern 90..........
 voice-class codec 2
 session target ras
 dtmf-relay h245-alphanumeric
 no vad
!
dial-peer voice 33 voip
 preference 2
 destination-pattern 9[1-9].....
 voice-class codec 2
 session target ras
 dtmf-relay h245-alphanumeric
 no vad
!
```

```
gateway
 timer receive-rtp 1200
!
telephony-service
 load 7910 P00403020214
 load 7960-7940 P00305000301
 max-ephones 30
 max-dn 30
 ip source-address 10.10.90.1 port 2000
 auto assign 1 to 30
 caller-id block code *67
 system message TB3CME5-2621XMSIP
 network-locale GB
 create cnf-files version-stamp 7960 May 26 2004 22:11:20
 dialplan-pattern 1 01917315... extension-length 4 no-reg
 voicemail 1100
 max-conferences 4
 moh music-on-hold.au
 time-format 24
 date-format dd-mm-yy
 web admin system name Administrator password cisco
 dn-webedit
 time-webedit
 transfer-system full-consult
!
ephone-dn  1  dual-line
 number 5830
 description 0191 7315830
 call-forward busy 1100
 call-forward noan 1100 timeout 90
!
ephone-dn  2  dual-line
 number 5831
 description 0191 7315831
 call-forward busy 1100
 call-forward noan 1100 timeout 90
!
ephone-dn  3  dual-line
 number 5832
 description 0191 7315832
 call-forward busy 1100
 call-forward noan 1100 timeout 90
!
ephone-dn  4  dual-line
 number 5833
 description 0191 7315833
 call-forward busy 1100
 call-forward noan 1100 timeout 90
```

```
!
ephone  1
 mac-address 0005.5E7C.D4C5
 type 7960
 button  1:1
!
ephone  2
 mac-address 0002.FD06.E2E0
 type 7960
 button  1:4
!
ephone  3
 mac-address 0007.EBF0.EEB8
 type 7960
 button  1:3
!
ephone  4
 mac-address 0003.6BF2.4611
 type 7960
 button  1:2
!
alias exec c conf t
alias exec sib sh ip int brief
alias exec ipr sh ip route
alias exec cdp sh cdp nei
alias exec sip sh ip protocols
!
line con 0
 exec-timeout 30 0
 privilege level 15
 logging synchronous
line aux 0
line vty 0 4
 exec-timeout 30 0
 privilege level 15
 password cisco
 logging synchronous
 login
!
exception core-file TB3CME5-2621XMSIP
exception protocol ftp
exception dump 10.10.50.138
ntp clock-period 17180147
ntp peer 192.168.254.101
!
end
```

Cisco CallManager Express Configuration for BRI

This section shows portions of a Cisco CallManager Express configuration file that should be used if Cisco CallManager Express uses multiple BRIs instead of a single PRI. The configuration file lines in this section should replace similar lines in the configuration file shown in the "Cisco CallManager Express Configuration for PRI" section on page 4-2.

It is possible for a carrier to provide multiple BRIs with a single DID/DDI (direct inbound dialing/direct dialing inbound) range. However, for the example shown here, the PSTN provided two BRI lines with MSN (multiple subscriber numbering) in each.

Appropriate translation is required to ensure that incoming calls on either BRI can be received and that outgoing calls are modified depending on the BRI used to pass CLI screening.

In the configuration file sections below:

- Translation rule 1 is applied to the incoming calling number on all the voice ports, and provides a full number display on IP phones for all incoming calls. It also converts each incoming called number on the main BRI port to a number compatible with the dial-plan pattern, as in the PRI configuration file.
- Translation rule2 converts incoming calls on the second BRI port to a number compatible with the dial-plan pattern. (Each additional BRIs would require its own translation rule.)
- Translation rules 11 and 12 are applied on the POTS dial peers, and translate the outgoing calling number on each BRI port to a format that is appropriate for the PSTN and that will pass carrier screening. In this case the PSTN switch requires only the last 7 digits.
- In this example, each Cisco CallManager Express uses four-digit extension numbers. However only the last three digits are user defined from the PSTN on the BRI MSN range for each line. In this case the dial-plan pattern shows the full E.164 number mask, and the extension pattern provides the conversion from the three-digit MSN to the four-digit extension.

```
translation-rule 1
 Rule 1 ^1........ 01
 Rule 2 ^266... 01628266
!
translation-rule 2
```

```
 Rule 1 ^1628267... 01628266
 Rule 2 ^267... 01628266
 Rule 3 ^01628267... 01628266
!
translation-rule 11
 Rule 1 ^01628266... 8266
!
translation-rule 12
 Rule 1 ^01628266... 8267
!
interface BRI1/0
 description 01628 266xxx
 no ip address
 no ip mroute-cache
 isdn switch-type basic-net3
 isdn overlap-receiving
 isdn not-end-to-end 64
 isdn incoming-voice voice
 isdn send-alerting
 isdn sending-complete
 isdn skipsend-idverify
!
interface BRI1/1
 description 01628 267xxx
 no ip address
 no ip mroute-cache
 isdn switch-type basic-net3
 isdn overlap-receiving
 isdn not-end-to-end 64
 isdn incoming-voice voice
 isdn send-alerting
 isdn sending-complete
 isdn skipsend-idverify
!
voice-port 1/0
 translate calling 1
 translate called 1
 echo-cancel coverage 32
 compand-type a-law
 cptone GB
 timeouts interdigit 6
 timeouts call-disconnect 3
 timeouts ringing 600
 timeouts wait-release 15
 music-threshold -70
!
voice-port 1/1
 translate calling 1
```

```
 translate called 2
 echo-cancel coverage 32
 compand-type a-law
 cptone GB
 timeouts interdigit 6
 timeouts call-disconnect 3
 timeouts ringing 600
 timeouts wait-release 15
 music-threshold -70
!
dial-peer voice 1 pots
 preference 1
 application session
 destination-pattern 9[1-9].....
 progress_ind setup enable 3
 progress_ind alert enable 8
 translate-outgoing calling 11
 direct-inward-dial
 port 1/0
 no register e164
!
dial-peer voice 2 pots
 preference 1
 application session
 destination-pattern 90..........
 progress_ind setup enable 3
 progress_ind alert enable 8
 translate-outgoing calling 11
 direct-inward-dial
 port 1/0
 prefix 0
 no register e164
!
dial-peer voice 3 pots
 application session
 destination-pattern 9[1-9].....
 progress_ind setup enable 3
 progress_ind alert enable 8
 translate-outgoing calling 12
 direct-inward-dial
 port 1/1
 no register e164
!
dial-peer voice 4 pots
 application session
 destination-pattern 90..........
 progress_ind setup enable 3
 progress_ind alert enable 8
```

```
 translate-outgoing calling 12
 direct-inward-dial
 port 1/1
 prefix 0
 no register e164
!
dial-peer voice 30 voip
 destination-pattern [0-6]...
 voice-class codec 2
 session target ras
 dtmf-relay h245-alphanumeric
 no vad
!
dial-peer voice 32 voip
 preference 2
 destination-pattern 90..........
 voice-class codec 2
 session target ras
 dtmf-relay h245-alphanumeric
 no vad
!
dial-peer voice 33 voip
 preference 2
 destination-pattern 9[1-9].....
 voice-class codec 2
 session target ras
 dtmf-relay h245-alphanumeric
 no vad
!
gateway
 timer receive-rtp 1200
!
telephony-service
 load 7910 P00403020214
 load 7960-7940 P00305000301
 max-ephones 30
 max-dn 30
 ip source-address 10.10.89.1 port 2000
 auto assign 1 to 30
 caller-id block code *67
 system message TB3CME4-1760SIP
 network-locale GB
 create cnf-files version-stamp 7960 May 26 2004 22:05:04
 dialplan-pattern 1 01628266... extension-length 4 extension-pattern
4... no-reg
 voicemail 1100
 max-conferences 4
 moh music-on-hold.au
```

```
 time-format 24
 date-format dd-mm-yy
 web admin system name Administrator password cisco
 dn-webedit
 time-webedit
 transfer-system full-consult
!
```

Cisco CallManager Express Configuration for FXO

This section shows portions of a Cisco CallManager Express configuration file that should be used if Cisco CallManager Express uses multiple FXOs instead of a single PRI or multiple BRIs. The configuration file lines in this section should replace similar lines in the configuration file shown in the "Cisco CallManager Express Configuration for PRI" section on page 4-2.

For the example shown here, IPSTN provided four analogue lines. These lines are basic loop start lines, so there is no DID/DDI for incoming calls. The voice-port connection plar opx commands for each voice port forwards incoming calls via H.323 to a remote operator console on the central Cisco CallManager. An operator then transfers calls to Cisco CallManager Express users.

This situation does not require translation rules or dial-plan patterns. The following configuration file lines show the voice-port configuration and dial-peers that force incoming calls to the central Cisco CallManager.

```
voice-port 1/0
 no battery-reversal
 echo-cancel coverage 32
 cptone GB
 timeouts interdigit 6
 timeouts call-disconnect 3
 timeouts ringing 600
 timeouts wait-release 15
 timing percentbreak 60
 connection plar opx 0000
 impedance complex2
 description 01628-100072
 music-threshold -70
!
voice-port 1/1
 no battery-reversal
 echo-cancel coverage 32
 cptone GB
```

```
 timeouts interdigit 6
 timeouts call-disconnect 3
 timeouts ringing 600
 timeouts wait-release 15
 timing percentbreak 60
 connection plar opx 0000
 impedance complex2
 description 01628-100073
 music-threshold -70
!
voice-port 1/2
 no battery-reversal
 echo-cancel coverage 32
 cptone GB
 timeouts interdigit 6
 timeouts call-disconnect 3
 timeouts ringing 600
 timeouts wait-release 15
 timing percentbreak 60
 connection plar opx 0000
 impedance complex2
 description 01628-100074
 music-threshold -70
!
voice-port 1/3
 no battery-reversal
 echo-cancel coverage 32
 cptone GB
 timeouts interdigit 6
 timeouts call-disconnect 3
 timeouts ringing 600
 timeouts wait-release 15
 timing percentbreak 60
 connection plar opx 0000
 impedance complex2
 description 01628-100075
 music-threshold -70
!
dial-peer cor custom
!
dial-peer voice 1 pots
 preference 3
 application session
 destination-pattern 9[1-9].....
 direct-inward-dial
 port 1/0
 no register e164
!
```

```
dial-peer voice 2 pots
 preference 2
 application session
 destination-pattern 9[1-9].....
 direct-inward-dial
 port 1/1
 no register e164
!
dial-peer voice 3 pots
 preference 1
 application session
 destination-pattern 9[1-9].....
 direct-inward-dial
 port 1/2
 no register e164
!
dial-peer voice 4 pots
 application session
 destination-pattern 9[1-9].....
 direct-inward-dial
 port 1/3
 no register e164
!
dial-peer voice 5 pots
 preference 3
 application session
 destination-pattern 90..........
 direct-inward-dial
 port 1/0
 prefix 0
 no register e164
!
dial-peer voice 6 pots
 preference 2
 application session
 destination-pattern 90..........
 direct-inward-dial
 port 1/1
 prefix 0
 no register e164
!
dial-peer voice 7 pots
 preference 1
 application session
 destination-pattern 90..........
 direct-inward-dial
 port 1/2
 prefix 0
```

```
 no register e164
!
dial-peer voice 8 pots
 application session
 destination-pattern 90..........
 direct-inward-dial
 port 1/3
 prefix 0
 no register e164
!
dial-peer voice 31 voip
 preference 9
 application session
 destination-pattern 90..........
 voice-class codec 2
 session protocol sipv2
 session target ipv4:10.10.50.148
 dtmf-relay sip-notify
 no vad
!
dial-peer voice 30 voip
 preference 1
 destination-pattern [0-6]...
 voice-class codec 2
 session target ras
 dtmf-relay h245-alphanumeric
 no vad
!
dial-peer voice 32 voip
 preference 2
 destination-pattern 90..........
 voice-class codec 2
 session target ras
 dtmf-relay h245-alphanumeric
 no vad
!
dial-peer voice 33 voip
 preference 2
 destination-pattern 9[1-9].....
 voice-class codec 2
 session target ras
 dtmf-relay h245-alphanumeric
 no vad
!
gateway
 timer receive-rtp 1200
!
```

```
telephony-service
 load 7910 P00403020214
 load 7960-7940 P00305000301
 max-ephones 30
 max-dn 30
 ip source-address 10.10.86.1 port 2000
 auto assign 1 to 30
 caller-id block code *67
 system message TB3CME1-1760SIP
 network-locale GB
 create cnf-files version-stamp 7960 Jun 14 2004 11:35:45
 voicemail 1600
 max-conferences 4
 moh music-on-hold.au
 time-format 24
 date-format dd-mm-yy
 web admin system name admin password cisco
 dn-webedit
 time-webedit
 transfer-system full-consult
!
```

Configuration Files for Multiple Cisco CallManager Express Systems Deployed with Centralized Cisco Unity

The following portions of configuration file provide examples that you can use if you deploy multiple Cisco CallManager Express systems with centralized Cisco Unity:

- Configuration File for MWI SIP Server, page 4-17
- Configuration File for MWI SIP Clients, page 4-18

Configuration File for MWI SIP Server

This section shows a portion of a configuration file for the MWI SIP server.

```
telephony-service
load 7910 P00403020214
```

```
load 7960-7940 P00305000301
max-ephones 124
max-dn 288
ip source-address 10.3.84.3 port 2000
auto assign 1 to 124
create cnf-files version-stamp 7960 Jun 17 2004 07:59:23
dialplan-pattern 1 408394.... extension-length 5
voicemail 4085551234
mwi relay
mwi expires 99999
max-conferences 8
transfer-system full-consult
!
ephone 3
vm-device-id CiscoUM1-VI6
button 1:203
!
ephone 4
vm-device-id CiscoUM1-VI7
button 1:204
!
ephone 5
vm-device-id CiscoUM1-VI5
button 1:205
!
ephone 6
vm-device-id CiscoUM1-VI8
button 1:206
ephone-dn 201
number 10001 secondary 10002
mwi on-off
!
```

Configuration File for MWI SIP Clients

This section shows a portion of a configuration file for MWI SIP clients.

```
telephony-service
load 7910 P00403020214
load 7960-7940 P00305000301
max-ephones 144
max-dn 288
ip source-address 10.3.80.3 port 2000
auto assign 1 to 100
timeouts interdigit 3
timeouts ringing 10
```

```
create cnf-files version-stamp 7960 Jun 22 2004 05:19:28
dialplan-pattern 1 408391.... extension-length 5
voicemail 4083940001
mwi sip-server 10.3.84.3 transport tcp
mwi expires 86400
max-conferences 8
transfer-system full-consult
!
ephone-dn 1
number 11001
call-forward busy 4085551234
call-forward noan 4085551234 timeout 18
mwi sip
!
ephone-dn 2
number 11002
call-forward busy 4085551234
call-forward noan 10102 timeout 10
mwi sip
!
ephone-dn 3
number 11003
call-forward busy 4085551234
call-forward noan 4085551234 timeout 18
mwi sip
```

CHAPTER 5

Cisco IP Manager Assistant Configuration

This chapter provides an overview of how Cisco IP Manager Assistant (IPMA) was set up for the Multi-Site Single-Cluster Distributed scenario in IP Communications Systems Test Release 3.0 for EMEA IPT. This chapter does not include detailed installation and configuration instructions. Rather, it is intended to provide you with guidance as you set up the Cisco IPMA component of your IPT solution.

Cisco IPMA was configured for proxy line support and for shared line support using Cisco CallManager Administration. In general, default or recommended configuration values were used.

For detailed information about installing, configuring, and administering Cisco IPMA, refer to Cisco CallManager documentation at this URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_callmg/4_0/index.htm

In addition, registered Cisco.com users can refer to these documents:

- *Cisco IP Manager Assistant with Proxy Line Support*, which is available at this URL:

  http://cco/en/US/partner/products/sw/voicesw/ps556/products_administration_guide_chapter09186a00801ed116.html

- *Cisco IP Manager Assistant with Shared Line Support*, which is available at this URL:

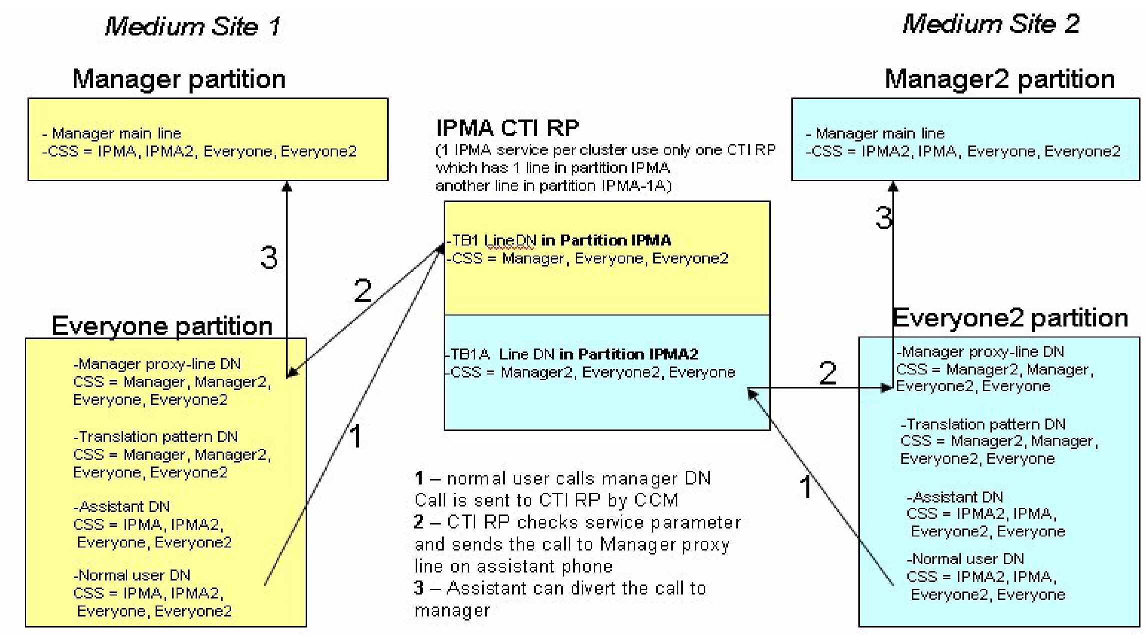
  http://cco/en/US/partner/products/sw/voicesw/ps556/products_administration_guide_chapter09186a00801ed11d.html

This chapter includes the following topics:

Cisco IPMA Configuration

Figure 5-1 shows how Cisco IPMA was configured in the Multi-Site Single-Cluster Distributed scenario.

Figure 5-1 Cisco IPMA Configuration



Using Translation Patterns with IPMA in Proxy Line Mode

When Using IPMA in proxy-line mode, translation patterns will not reroute a call to a manager when the IPMA service stops. To avoid this issue, configure the IPMA CTI route point to forward calls to the manager line when calls are not answered.

Localizing Cisco IPMA

To localize Cisco IPMA, follow these general steps from Cisco CallManager Administration. For more detailed information, Refer to the Cisco CallManager documentation.

1. In the Phone Configuration page for the desired phone, enter the desired language in the User Locale field.
2. In the User Configuration page, enter the same value in the User Locale field.
3. Create a new server in Cisco CallManager Administration. Enter the following URL in the Service URL field:

   http://*ip_address*/ma/servlet/MAService?cmd=doPhoneService&Name=#DEVICENAME#&locale=*language*

   where:

   ip_address = IP address of the IPMA server

   language = desired language

Wireless Configuration

This chapter provides an overview of how the Cisco Aironet Access Point (AP) 1120, the Cisco IP Phone 7920, and the Cisco Secure Access Control Server (ACS) were configured for wireless operation between IP phone devices registered to Cisco CallManager or to Cisco CallManager Express. This configuration supports:

- Calls between Cisco IP Phone 7920s
- Calls between the Cisco IP Phone 7920 and other Cisco IP Phone 79xx models supported by IP Communications Systems Test Release 3.0 for IPT
- Intercluster and intracluster Cisco CallManager and Cisco CallManager Express sites.

This chapter does not include detailed installation and configuration instructions. Rather, it is intended to provide you with guidance as you set up wireless devices in your IPT solution.

This chapter includes the following topics:

- Overview, page 6-2
- Cisco IP Phone 7920 Configuration, page 6-4
- Cisco Aironet 1121 Access Point Configuration File, page 6-4
- Cisco Access Control Server for LEAP Configuration, page 6-8

Overview

The wireless portion of the IP Communications Systems Test Release 3.0 for IPT was configured based on the recommendations and configurations described in the documents listed in Table 6-1.

Table 6-1 Wireless Configuration References

Document	Reference
Cisco 7920 Wireless IP Phone Design and Deployment Guide	http://www.cisco.com/en/US/products/hw/phones/ps379/products_implementation_design_guide_book09186a00802a029a.html
Cisco Aironet 1200 Series Access Point Installation and Configuration Guide	http://www.cisco.com/en/US/products/hw/wireless/ps430/products_installation_and_configuration_guide_book09186a0080147d69.html
Cisco AVVID Wireless LAN Design	http://www.cisco.com/application/pdf/en/us/guest/netsol/ns178/c649/ccmigration_09186a00800d67eb.pdf
Wireless Virtual LAN Deployment Guide	http://www.cisco.com/en/US/products/hw/wireless/ps430/prod_technical_reference09186a00801444a1.html
Cisco IOS Software Configuration Guide for Cisco Aironet Access Points	http://www.cisco.com/en/US/products/hw/wireless/ps4570/products_configuration_guide_book09186a00801ea410.html
Cisco Wireless IP Phone 7920 Administrator Guide	http://www.cisco.com/en/US/products/hw/phones/ps379/products_administration_guid_book09186a0080183c50.html
Cisco Wireless IP Phone 7920 for Cisco CallManager	http://www.cisco.com/en/US/products/hw/phones/ps379/products_user_guide_book09186a00802358c6.html
Configuring the Cisco 7920 Wireless IP Phone with WEP Keys, VLANs, and LEAP	http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_configuration_example09186a00801a90d3.shtml

IP Communications Systems Test Release 3.0 for IPT used a centralized Cisco Secure ACS with LEAP-compliant RADUIS authentication for all users of the Cisco IP Phone 7920 and the Cisco Aironet AP 1121. In addition, a Cisco Aironet AP 1121 was configured as the backup LEAP authentication local RADIUS server to be used if the WAN connection to the ACS becomes lost.

LEAP allows devices such as the Cisco Aironet AP 1121 and the Cisco IP Phone 7920 to be mutually authenticated based on username and password. Upon authentication, a dynamic key is used between the Cisco IP Phone 7920 and the Cisco Aironet AP 1121 to encrypt traffic. Both signaling (SCCP) and media (RTP) streams are encrypted between the Cisco IP Phone 7920 and the Cisco Aironet AP 1121. The Cisco IP Phone 7920 supports static WEP and EAP-Cisco (LEAP) for data encryption and authentication. 802.1x/LEAP was used with a central Cisco Secure ACS.

The wireless configuration followed these guidelines:

- To ensure the best voice quality, VAD was disabled for the Cisco IP Phone 7920. VAD is a Cisco CallManager parameter that applies to all phones registered to a specific cluster.
- The RSSI level in the RF network is at least 20 throughout the network.
- The QBSS level on the Cisco Aironet AP 1121 is maintained below 40.
- The Cisco Aironet AP 1231s were configured to support both 802.11b and 802.11b/g WANs.
- No more than 20 users were used for any single Cisco Aironet AP 1121. The recommended maximum number of users is 15 to 25.
- No more than 16 VLANs were used per Cisco Aironet AP 1121. Each wireless VLAN was represented with a unique SSID name.
- Distance between Cisco Aironet AP 1231s can cause throughput variations for clients based on distance from the Cisco Aironet AP 1121. Cisco recommends that you limit the Cisco Aironet AP 1121 data rate to the higher data rates of 11 Mbps and 5.5 Mbps.
- The number of Cisco Aironet AP 1231s that you will require depends on your coverage and throughput requirements.
- EAP-Cisco (Network EAP or LEAP) was used as the security mechanism.
- The Cisco Secure ACS local database was utilized to store the username and password. Remote databases can affect response times, which can affect overall quality of service (QoS) during L2 roaming.

Cisco IP Phone 7920 Configuration

The Cisco IP Phone 7920 was implemented with Open and LEAP authentication types. WEP encryption was not configured or used. The phones were installed and configured as described in the Cisco IP Phone 7920 documentation. For detailed information about installing, configuring, and administering the Cisco IP Phone 7920, refer to the phone documentation listed in Table 6-1.

Cisco Aironet 1121 Access Point Configuration File

This section shows a configuration file for the Cisco Aironet AP 1121 that was used for wireless testing. This example includes settings for the Cisco Secure ACS and the local RADIUS server hosts. In this way, this Cisco Aironet AP 1121 can be used as a backup LEAP authentication sever when the Cisco Secure ACS is unavailable.

For related information, refer to the Access Point documentation listed in Table 6-1.

```
version 12.2
no service pad
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
!
hostname s10-ap1200-2
!
logging queue-limit 10000000
enable password 7 130B181C0E
!
username Cisco password 7 094241071C
ip subnet-zero
!
aaa new-model
!
aaa group server radius rad_eap
 server 10.0.0.30 auth-port 1645 acct-port 1646
 server 10.0.0.61 auth-port 1812 acct-port 1813
!
aaa group server radius rad_mac
!
aaa group server radius rad_acct
```

```
!
aaa group server radius rad_admin
!
aaa group server tacacs+ tac_admin
!
aaa group server radius rad_pmip
!
aaa group server radius dummy
 server 10.0.0.61 auth-port 1812 acct-port 1813
!
aaa group server radius rad_eap1
 server 10.0.0.30 auth-port 1645 acct-port 1646
 server 10.0.0.61 auth-port 1812 acct-port 1813
!
aaa group server radius rad_eap2
 server 10.0.0.61 auth-port 1812 acct-port 1813
!
aaa authentication login eap_methods group rad_eap
aaa authentication login mac_methods local
aaa authentication login eap_methods1 group rad_eap1
aaa authentication login eap_methods2 group rad_eap2
aaa authorization exec default local
aaa authorization ipmobile default group rad_pmip
aaa accounting network acct_methods start-stop group rad_acct
aaa session-id common
dot11 phone
dot11 arp-cache
!
bridge irb
!
interface Dot11Radio0
 no ip address
 no ip route-cache
!
broadcast-key vlan 102 change 300
!
 broadcast-key vlan 120 change 300
!
 broadcast-key vlan 121 change 300
!
ssid s10-open
    vlan 121
    authentication network-eap eap_methods1
!
 ssid s10-wdata
    vlan 120
    authentication network-eap eap_methods2
!
```

```
 speed basic-11.0
 rts threshold 2312
 power client 30
 channel 2412
 antenna transmit right
 station-role root
!
interface Dot11Radio0.102
 encapsulation dot1Q 102 native
 no ip route-cache
 bridge-group 1
 bridge-group 1 subscriber-loop-control
 bridge-group 1 block-unknown-source
 no bridge-group 1 source-learning
 no bridge-group 1 unicast-flooding
 bridge-group 1 spanning-disabled
!
interface Dot11Radio0.120
 encapsulation dot1Q 120
 no ip route-cache
 bridge-group 120
 bridge-group 120 subscriber-loop-control
 bridge-group 120 block-unknown-source
 no bridge-group 120 source-learning
 no bridge-group 120 unicast-flooding
 bridge-group 120 spanning-disabled
!
interface Dot11Radio0.121
 encapsulation dot1Q 121
 no ip route-cache
 bridge-group 121
 bridge-group 121 subscriber-loop-control
 bridge-group 121 block-unknown-source
 no bridge-group 121 source-learning
 no bridge-group 121 unicast-flooding
 bridge-group 121 spanning-disabled
!
interface FastEthernet0
 no ip address
 no ip route-cache
 duplex auto
 speed auto
!
interface FastEthernet0.102
 encapsulation dot1Q 102 native
 no ip route-cache
 bridge-group 1
 no bridge-group 1 source-learning
```

```
 bridge-group 1 spanning-disabled
!
interface FastEthernet0.120
 encapsulation dot1Q 120
 no ip route-cache
 bridge-group 120
 no bridge-group 120 source-learning
 bridge-group 120 spanning-disabled
!
interface FastEthernet0.121
 encapsulation dot1Q 121
 no ip route-cache
 bridge-group 121
 no bridge-group 121 source-learning
 bridge-group 121 spanning-disabled
!
interface BVI1
 ip address 10.0.0.61 255.255.255.240
 no ip route-cache
!
ip default-gateway 10.0.0.49
ip http server
ip http help-path
http://www.cisco.com/warp/public/779/smbiz/prodconfig/help/eag/ivory/1
100
ip radius source-interface BVI1
logging 10.105.254.1
logging 10.0.0.62
snmp-server community private RW
snmp-server enable traps tty
radius-server local
  nas 10.0.0.61 key 7 030752180500701E1D
  user cisco nthash 7
1443435E59220F7D767D6066703021475051037C0902062F533D400A7904057377
  user cisco1 nthash 7
040A2F202E786E1B51405035475A5F5C087E007A6A64733221325555777A7A0701
  user cisco2 nthash 7
15442A5B5379787D0D66130135574627550 30F0F710C2F5A3C377C7D710574040C
  user cisco3 nthash 7
075C756E1C504B5146475D5D27720E760D6110724B243550250F7A0E0A065C264C
  user cisco4 nthash 7
013557560E5D222E716A17283A2041452A54277208070D12117730223453500601
  user cisco5 nthash 7
06252E751E68283A264745585A570E09757E1616704A2540562306787E77002F5A
  user cisco6 nthash 7
096D1751405C35372A5522787F727A60160644563725590701090A075A274F337F
  user cisco7 nthash 7
055D535B02196A5F3B5036342D285C7A0A747064660641504755597 30C0C05055F
```

```
  user cisco8 nthash 7
143645292A50737C750D64637B3153375B2200010F75052F564935017D03010507
  user cisco9 nthash 7
06512D076F185C4E5035462859560E0A75701564014355302027050B0104755E52
  user cisco10 nthash 7
091A185F3A5635375C5D510B080178606D75315746565707017C700059534A300E
  user cisco11 nthash 7
05535129716F6A5B4C563645582A220B73017E17117B4254435025020B0A70765B
  user cisco12 nthash 7
0147275678592059071B68583D5346425E2D530809067A6A6D0445574454250408
  user cisco13 nthash 7
0479532759721F6D2B4C2135405228507F08717C17630646534F5424007A7B000D
!
radius-server host 10.0.0.30 auth-port 1645 acct-port 1646 key 7
110A1016141D5A5E57
radius-server host 10.0.0.61 auth-port 1812 acct-port 1813 key 7
045802150C2E1D1C5A
radius-server deadtime 10
radius-server authorization permit missing Service-Type
bridge 1 route ip
!
line con 0
 password 7 082F43400C
line vty 0 4
 exec-timeout 60 0
 password 7 045504080A
line vty 5 15
 exec-timeout 60 0
 password 7 000A1C0801
!
end
```

Cisco Access Control Server for LEAP Configuration

The Cisco Secure ACS 3.2 was configured for LEAP authentication using RADIUS (Cisco Aironet). The local CiscoSecure user database was used.

The ACSs were installed and configured as described in the Cisco Secure ACS 3.2 documentation, which is available at this URL:

http://www.cisco.com/en/US/products/sw/secursw/ps2086/ps5340/index.html

For a detailed step-by-step configuration example, also refer to *Configuring the Cisco 7920 Wireless IP Phone with WEP Keys, VLANs and LEAP* (see Table 6-1).

CHAPTER 7

IP Video Telephony Configuration

This chapter provides an overview of how IP Video Telephony was set up and configured in IP Communications Systems Test Release 3.0 for EMEA IPT. The information in this chapter applies to calls between video-capable terminals. Calls between video and audio only endpoints and between video endpoints and CTI applications were not tested.

This chapter does not include detailed installation and configuration instructions. Rather, it is intended to provide you with guidance as you set up video devices in your IPT solution.

The Cisco CallManager and gatekeeper configuration information described in this chapter apply to conference services under control of both H.323 and SCCP protocols.

For additional information and guidelines for implementing Cisco IP Video Telephony, refer to *Cisco IP Video Telephony Solution Reference Network Design (SRND), Cisco CallManager Release 4.0* at this URL:

http://www.cisco.com/application/pdf/en/us/guest/netsol/ns268/c649/ccmigration_09186a008026c609.pdf

This chapter includes the following topics:

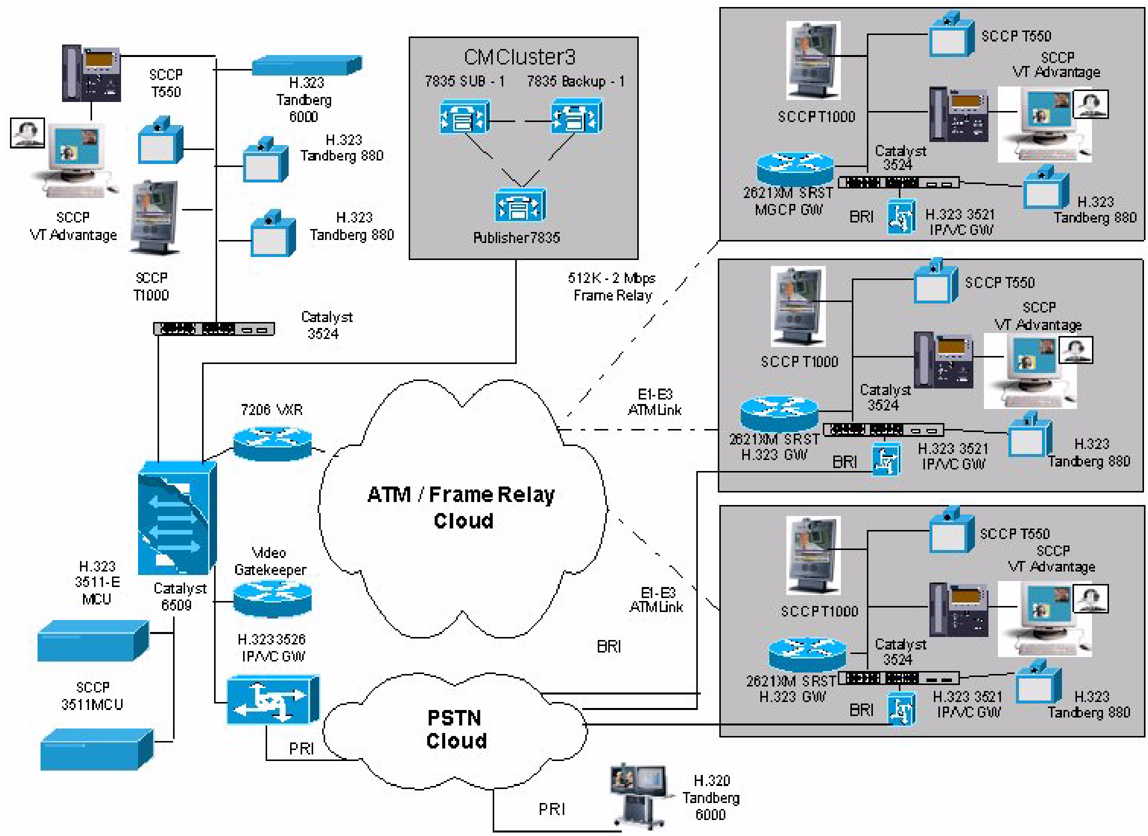
IP Video Telephony Components and Topology

The IP Video Telephony deployment included the following components:

- Tandberg PRI legacy H.320 ISDN video terminals
- Polycom 512 MultiPoint BRI H.320 ISDN video terminals
- Tandberg SCCP T1000 and T550
- Tandberg H.323 controlled T1000, T6000, T2500, and T880
- Polycom H.323 controlled VSX 7000
- Cisco VT Advantage
- IP-ISDN Video Conference Gateways
  - Cisco IP/VC 3526 gateway connecting central site to PSTN with H.320 ISDN over a PRI interface
  - Cisco IP/VC 3521 gateway connecting remote sites to PSTN with H.320 ISDN over a BRI interface
- SCCP/H.323 Conference bridges
  - Cisco IP/VC 3511 MCU with EMP for transrating
  - Cisco IP/VC 3540 MCU with EMP for transrating

Figure 7-1 shows how Cisco IP Video Telephony was deployed in IP Communications Systems Test Release 3.0.

Figure 7-1 IP Video Telephony Topology



Supported Call Types

The IP Video Telephony deployment supports video calls made between the following endpoints using the specified protocols:

- Tandberg SCCP video endpoint and Tandberg or Polycom 512 MultiPoint H.320 video terminals via Cisco IP/VC 3526 PRI gateways
- Cisco VT Advantage and Tandberg or Polycom 512 MultiPoint H.320 video terminals via Cisco IP/VC 3526 PRI gateways
- Tandberg SCCP video endpoint and Cisco VT Advantage via Cisco IP/VC 3526 PRI and Cisco IP/VC 3521 BRI gateways
- Tandberg SCCP video endpoint and Tandberg H.323 video endpoint via Cisco IP/VC 3526 PRI and Cisco IP/VC 3521 BRI gateways
- Tandberg SCCP video endpoint and Cisco VT Advantage via LAN and WAN
- Cisco VT Advantage and Tandberg H.323 video endpoints
- Tandberg H.323 and SCCP video endpoints
- Tandberg H.323 and Tandberg H.323 video endpoints
- Tandberg H.323 and Tandberg H.320 video endpoints via Cisco IP/VC 3526 PRI and 3521 BRI gateways
- SCCP ad-hoc video conference call between all the above endpoints

Call Routing

The IP Video Telephony deployment supports the following call routings.

Note In these routings, only calls from H.323 endpoints and inbound calls from IP/VC gateways are routed to Cisco CallManager through a gatekeeper.

- SCCP endpoint > Cisco CallManager > SCCP endpoint
- SCCP endpoint > Cisco CallManager > H.323 endpoint
- SCCP endpoint > Cisco CallManager > IP/VC gateway > PSTN > IP/VC gateway > gatekeeper > Cisco CallManager > SCCP endpoint

- SCCP endpoint > Cisco CallManager > IP/VC gateway > PSTN > IP/VC gateway > gatekeeper > Cisco CallManager > H.323 endpoint
- SCCP endpoint > Cisco CallManager > IP/VC gateway > PSTN > H.320 endpoint
- H.323 endpoint > gatekeeper > Cisco CallManager > H.323 endpoint
- H323 endpoint > gatekeeper > Cisco CallManager > SCCP endpoint
- H.323 endpoint > gatekeeper > Cisco CallManager > IP/VC gateway > PSTN > H.320 endpoint
- H323 endpoint > gatekeeper > Cisco CallManager > IP/VC gateway > PSTN > IP/VC gateway -> gatekeeper -> Cisco CallManager -> SCCP endpoint
- H.320 endpoint > IP/VC gateway -> gatekeeper -> Cisco CallManager > SCCP or H.323 endpoint

Configuring IP Video Telephony Components

The following sections provide information about configuring various components for IP Video Telephony. This information supplements the recommendations in *Cisco IP Video Telephony Solution Reference Network Design (SRND), Cisco CallManager Release 4.0*, which is available at this URL:

http://www.cisco.com/application/pdf/en/us/guest/netsol/ns268/c649/ccmigration_09186a008026c609.pdf

- Endpoint Gatekeeper Configuration for IP Video Telephony, page 7-6
- IP/VC Gateway Configuration for IP Video Telephony, page 7-8
- Configuring the Cisco IP/VC 3511 MCU Conference Bridge for IP Video Telephony, page 7-8
- Cisco Call Manager Configuration for IP Video Telephony, page 7-9

In this IP Video Telephony configuration, the gatekeeper routes all calls to Cisco CallManager, which performs these functions:

- Applies locations-based call admission control (CAC)
- Creates Call Detail Records
- Provides call control for monitoring by third-party voice recording applications

By default, the gatekeeper assigns all prefixes to Cisco CallManager for routing. Therefore, H.323 endpoints, IP/VC gateways, and H.323 MCUs must prefix each E.164 number with a # when registering with the gatekeeper. Cisco CallManager is configured strip the # from a calling party number.

Endpoint Gatekeeper Configuration for IP Video Telephony

Each Cisco CallManager cluster in an IP Video Telephony deployment requires a dedicated pair of Hot Standby Router Protocol (HSRP) controlled gatekeepers to force calls initiated by H.323 clients to route through the Cisco CallManager. In a Multi-Site Single-Cluster Distributed scenario, a dedicated pair of HSRP controlled gatekeepers must be configured independently for each site.

Two local zones must be created on the endpoint gatekeeper pair, one for Cisco Call Manager and one for the IP/VC Gateway and H.323 endpoints. (You may configure an additional zone for H.323 dial-in conference support on the MCU.)

Zone prefixes 0 through 9 are configured to route calls to the Cisco CallManager zone. In this way, all calls are routed to the Cisco CallManager zone, regardless of the digit with which the dialed number starts.

An H.225 Gatekeeper Controlled Trunk is registered in the Cisco CallManager zone. The Gatekeeper uses the Cisco CallManager technology prefix of #1 as the default prefix.

The H.225 gatekeeper controlled trunk device pool determines which Cisco CallManager servers should register with the gatekeeper, and which Cisco CallManager servers are primary and secondary. The Gatekeeper configuration uses gw-priority statements to identify primary and secondary Cisco CallManager servers.

Each H.323 video endpoint is defined in Cisco CallManager as an H.323 client with its own Calling Search Space, device pool, and so on. When the client originates a call, the gatekeeper forwards the call to Cisco CallManager. Cisco CallManager matches the IP address of the calling party to the static IP address configured in its database.

The device pool for the H.323 clients must each be the same and must match the gw-priority statements in the gatekeeper configuration. For example, in the following configuration, <ccm-subscriber-1> is the primary Cisco CallManager (priority 10), so each H.323 client also has that Cisco CallManager as its primary Cisco CallManager in its Cisco CallManager Redundancy Group.

Sample Gatekeeper Configuration

```
gatekeeper
zone local ccmzone site.com 10.10.50.150
zone local h323zone site.com
no zone subnet ccmzone default enable
zone subnet ccmzone 10.10.50.197/32 enable
zone subnet ccmzone 10.10.50.135/32 enable
zone prefix ccmzone 0* gw-priority 10 <ccm-subscriber-1>
zone prefix ccmzone 0* gw-priority 9 <ccm-subscriber-2>
zone prefix ccmzone 1* gw-priority 10 <ccm-subscriber-1>
zone prefix ccmzone 1* gw-priority 9 <ccm-subscriber-2>
zone prefix ccmzone 2* gw-priority 10 <ccm-subscriber-1>
zone prefix ccmzone 2* gw-priority 9 <ccm-subscriber-2>
zone prefix ccmzone 3* gw-priority 10 <ccm-subscriber-1>
zone prefix ccmzone 3* gw-priority 9 <ccm-subscriber-2>
zone prefix ccmzone 4* gw-priority 10 <ccm-subscriber-1>
zone prefix ccmzone 4* gw-priority 9 <ccm-subscriber-2>
zone prefix ccmzone 5* gw-priority 10 <ccm-subscriber-1>
zone prefix ccmzone 5* gw-priority 9 <ccm-subscriber-2>
zone prefix ccmzone 6* gw-priority 10 <ccm-subscriber-1>
zone prefix ccmzone 6* gw-priority 9 <ccm-subscriber-2>
zone prefix ccmzone 7* gw-priority 10 <ccm-subscriber-1>
zone prefix ccmzone 7* gw-priority 9 <ccm-subscriber-2>
zone prefix ccmzone 8* gw-priority 10 <ccm-subscriber-1>
zone prefix ccmzone 8* gw-priority 9 <ccm-subscriber-2>
zone prefix ccmzone 9* gw-priority 10 <ccm-subscriber-1>
zone prefix ccmzone 9* gw-priority 9 <ccm-subscriber-2>
//technology-prefix of CallManager. Use #1 to avoid clash with the
   zone prefixes above
gw-type-prefix #1* default-technology
// override default proxy-assisted call routing
no use-proxy h323zone default inbound-to terminal
no use-proxy h323zone default outbound-from terminal
```

```
no shutdown
endpoint ttl 60
```

IP/VC Gateway Configuration for IP Video Telephony

The following guidelines apply to Cisco IP/VC gateways for IP Video Telephony:

- For video calls, one service prefix was created with # followed by the PSTN outbound dialling prefix.
- A different prefix is used for outbound audio-only calls via voice gateways.
- The bandwidth associated with the prefix on the video gateway is set to 384 Kbps. The calculation for this setting includes the audio portion of video calls.
- Downspeeding was disabled (the default setting is enabled) to ensure logical channel clearing for calls disconnecting from the ISDN side of the IP-ISDN gateway.

BRI gateways located on SRST remote sites register with the endpoint gatekeeper on the central site. Note when the remote site gateway is in SRST mode, Cisco VT Advantage associated IP phones will be able to make audio only calls. All other remote video components are disabled.

Debug output for gateways is logged to the console. The default debug notify level is 3, which is generally sufficient for troubleshooting media channels failure. You can use Cisco Secure Telnet or the serial port to access logs and make test calls.

Configuring the Cisco IP/VC 3511 MCU Conference Bridge for IP Video Telephony

The Cisco IP/VC 3540 MCU can run both H.323 and SCCP protocols simultaneously. The Cisco IP/VC 3511 MCU cannot. For dual protocol support, two Cisco IP/VC 3511 MCUs must be deployed, one for each protocol.

The Enhanced Media Processing capability is required for transrating with H.323 controlled conferences.

Conference scheduling applications are not supported.

Cisco Call Manager Configuration for IP Video Telephony

The following sections provide an overview of how Cisco CallManager was configured to work with IP Video Telephony:

For additional information about Cisco CallManager configuration, see Chapter 2, "Cisco CallManager Configuration."

Regions

In Cisco CallManager Administration, the total bitrate was set for the combined video and audio codecs used in video calls. The regions for video calls between central and remote sites were configured to use 128 kbps across WAN links. A minimum capacity of 768 kbps must be available for a WAN link used for integrated voice, video and data. 384 kbps was used for video calls within the same site.

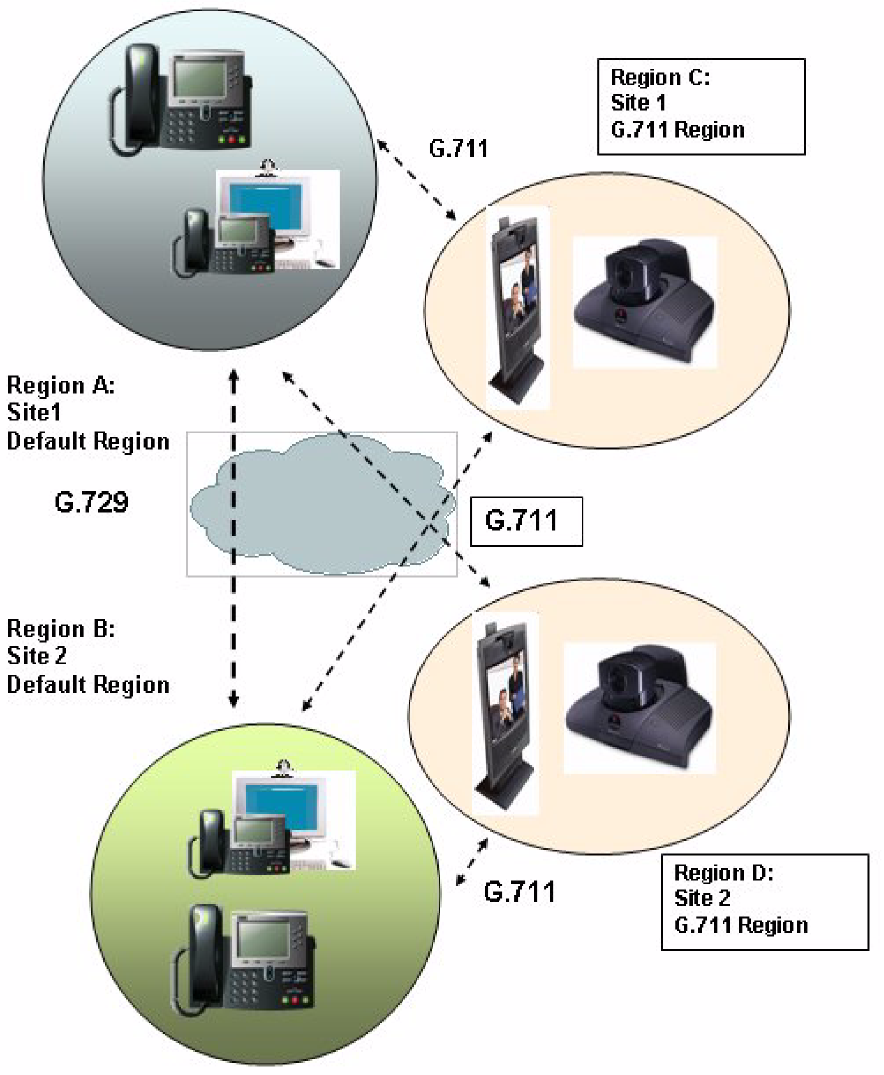
Cisco VT Advantage was connected through Cisco IP Phone Models 7940/7960/7970, which support G.711 and G.729 audio codecs. Tandberg SCCP and H.323 endpoints support the G.711 codec but not G.729. Cisco CallManager Administration was used to configure separate regions for Cisco VT Advantage and Tandberg video endpoints. These configurations were made to:

- Leave CVTA in the default region with the capability to use both G.711 and G.729
- Put Tandberg and H.323 video terminals into a "G711-only" region in each site
- Use G.729 for video calls between different VT Advantage endpoints across the WAN
- Use G.711 for all video calls between the G.711 only region and other default regions
- Place VT Advantage in a device pool within the default region

- Place Tandberg and H.323 video terminals in a device pool in the G.711 only region
- Use a video media resource group in which the video conference bridge appears first for both device pools

Figure 7-2 shows G.711-only regions per site.

Figure 7-2 G.711-Only Regions Per Site



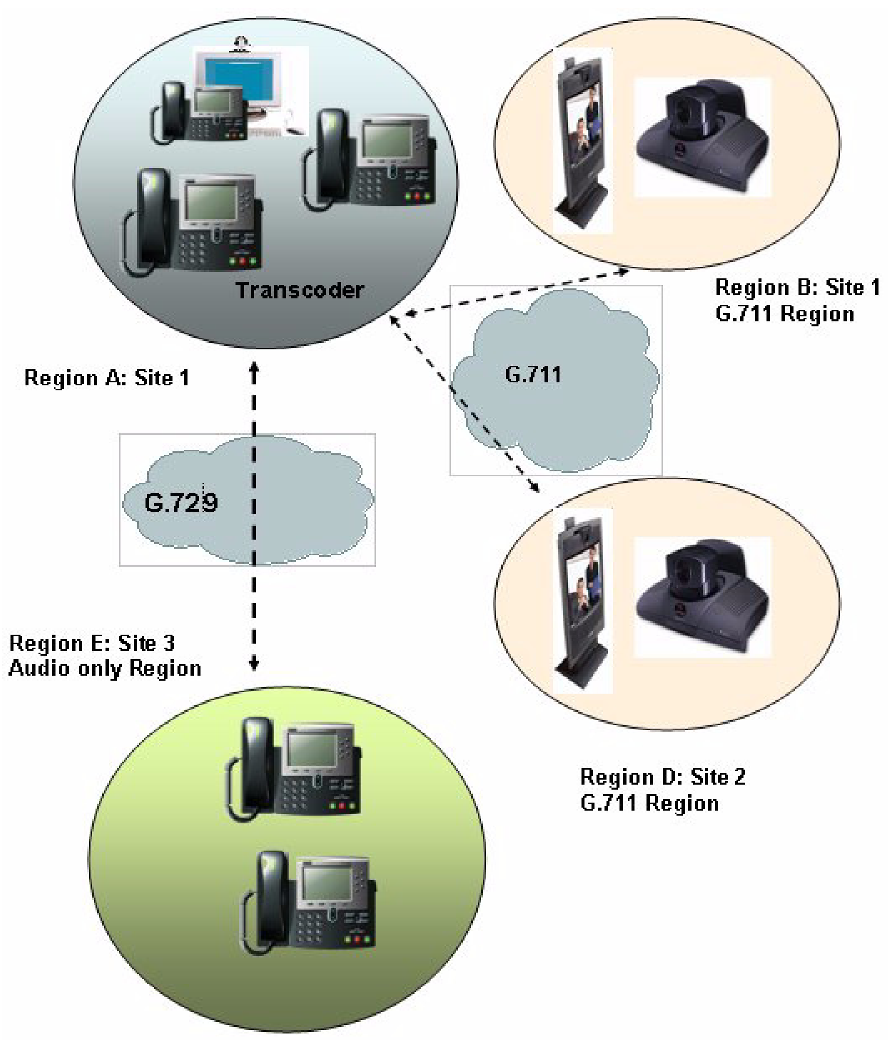
For audio calls and the audio portion of video calls, , Cisco CallManager Administration uses explicit codec selection for bandwidth determination.

When the intra-cluster WAN bandwidth is restricted, audio calls between an IP phone in an audio-only region using G.729 only for calls over the WAN and a video terminal in the region using G.711 are possible if calls are routed indirectly using an audio transcoder.

This scenario is a concern only when the call uses audio terminals across low bandwidth WAN links. G.729 was specified as the inter-region codec across the WAN to force use of G.729 only on the low bandwidth WAN link. G.711 was used as the default codec for all other audio and video calls that use the G.711 only region.

Figure 7-3 shows the use of an audio transcoder for audio calls between a G.711-only region and a region using G.729 for WAN calls.

Figure 7-3 G.711 Regions with G.729 Audio Calls



Stripping Prefixes of Calling Party Numbers from H.323 Clients

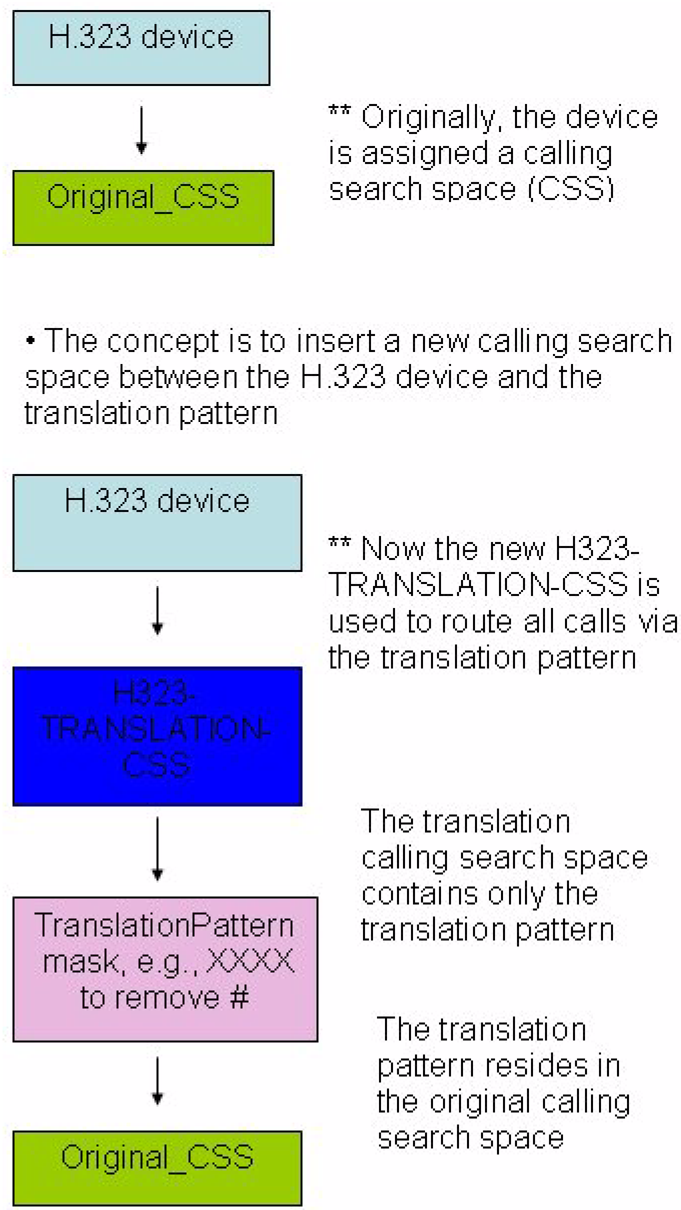
H.323 endpoints are statically configured with an E.164 number in Cisco CallManager, but register with a gatekeeper using a # prefix. Because H.323 endpoints prepend # to their own E.164 addresses, calls from H.323 endpoints include # in their Calling Line ID. Cisco CallManager uses a translation pattern to remove the # from the calling number.

The following example shows how the # prefix is stripped:

- #4321 sends SETUP to Cisco CallManager “00014089021234”
- Calling Search Space of #4321 recognizes translation pattern of !
- ! has Calling Party Transformation Mask of XXXX

Figure 7-4 illustrates this example.

Figure 7-4 Stripping the # Prefix



The H.323 endpoint is defined as an H.323 client (phone) in Cisco CallManager. The calling search space for this client contains a single partition with the sole purpose of prefix stripping via a translation pattern. This pattern includes ! to match all. The calling party external phone number mask is applied to remove the #. The calling search space of the translation pattern can access other partitions for call routing from the H.323 endpoint. The only partition directly visible to the H.323 client is the one containing the ! translation pattern.

When Cisco CallManager uses a dial plan with variable length numbers for the H.323 video clients, multiple translation patterns must be defined to handle the different lengths of calling party numbers.

The following lines show an example of the translation patterns for a dial plan that uses 2-digit, 3-digit, and 6-digit internal extensions:

```
#11
#222
#156787
CSS for Endpoints with 2-digit DNS -> trans.ptrn translates Calling
Party to XX
CSS for Endpoints with 3-digit DNS -> trans.ptrn translates Calling
Party to XXX
CSS for Endpoints with 6-digit DNS-> trans.ptrn translates Calling
Party to XXXXX
```

Route Pattern Prepends

Cisco CallManager route patterns should include the following configurations:

- Route patterns for video calls to the PSTN use prefix digit #.
- H.323 controlled IP/VC Gateways must register to the Gatekeeper with a # + service prefix. (See the "Endpoint Gatekeeper Configuration for IP Video Telephony" section on page 7-6 for related information.)
- Called destinations for outbound calls must begin with the string '# + service prefix'. This string is used by the gateway to apply associated bandwidth and call type settings.

Cisco CallManager routes outbound calls directly to the gateway, not through a Gatekeeper controlled trunk. The gateway strips off the # + dial prefix for outbound video calls for onward PSTN routing. Cisco CallManager routes calls to endpoints directly by using their client configurations.

H.245 Capabilities Exchange

Cisco VT Advantage does not support H.261. Tandberg SCCP endpoints support H.261, H.263, and H.263+. Tandberg ISDN endpoints support H.261, H.263, H.263+, and H.264, but will first attempt to open a video channel using H.261. Consequently, the Wait for Far End H.245 Terminal Capabilities Set option should be unchecked (disabled) in the Gateway Configuration page in Cisco CallManager Administration for calls to the PSTN via a video gateway. (By default, this option is enabled.)

Cisco Enterprise Gateway Configuration

This chapter provides configuration information for the Cisco EGW 2200 Enterprise Gateways used in IP Communications Systems Test Release 3.0 for EMEA IPT. It also provides additional configuration information for related components.

This chapter does not include detailed installation and configuration instructions for the Cisco EGW 2200 Enterprise Gateways. Rather, it is intended to provide you with guidance as you set up the Cisco EGWs to interconnect legacy DPNSS PABXs and Cisco CallManager-enabled VoIP networks.

Each Cisco EGW 2200 was installed on an MCS-7835H-3.0-CC1 server and configured according to the instructions in the Cisco Enterprise Gateway documentation

For detailed information about installing, configuring, and administering the Cisco EGW 2200 Enterprise Gateway, refer to the documentation at this URL:

http://www.cisco.com/univercd/cc/td/doc/product/access/sc/nirvdoc/index.htm

This chapter includes the following topics:

Cisco Enterprise Gateway Overview

The Cisco EGW 2200 Enterprise Gateway is a media gateway controller that serves as a migration tool for optimizing the transition from existing Digital Private Network Signaling System (DPNSS) PBX enterprises to Cisco CallManager. The Cisco EGW also introduces Cisco Unity into existing DPNSS or QSIG PBX enterprises and supports Toll Bypass for QSIG and DPNSS PBXs. In these deployments, the Cisco EGW acts as an adjunct to Cisco CallManager and Cisco Unity to complete a connection to the PSTN.

Figure 8-1 shows and overview of a deployment for DPNSS PBXs interworking with Cisco CallManager.

Figure 8-1 DPNSS PBXs Interworking with Cisco CallManager

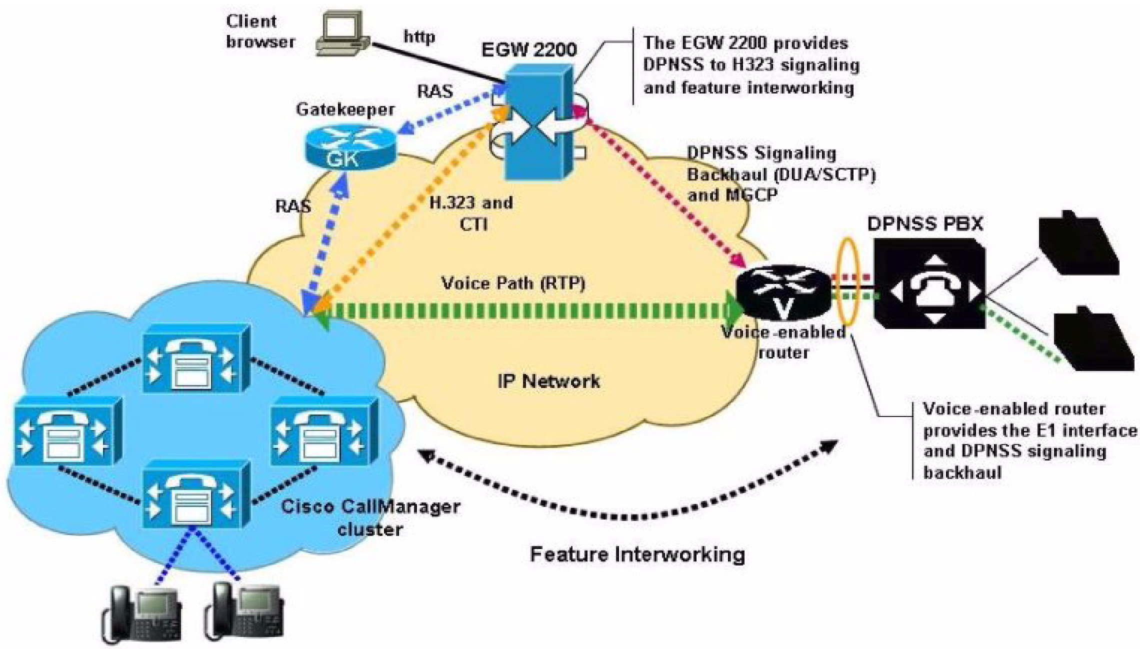
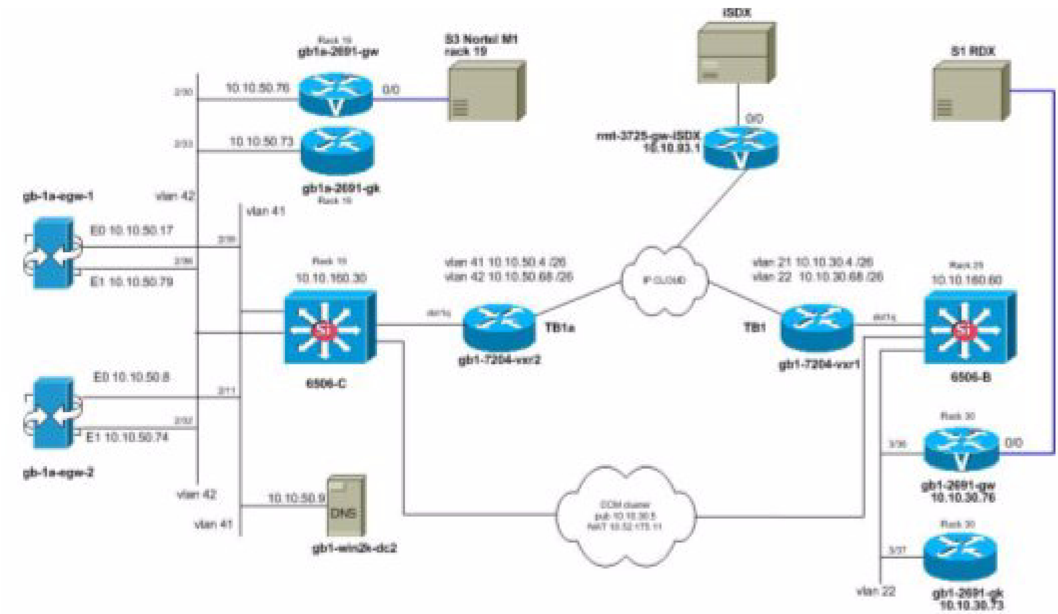


Figure 8-2 shows how the Cisco EGW was deployed in IP Communications Systems Test Release 3.0 for EMEA IPT.

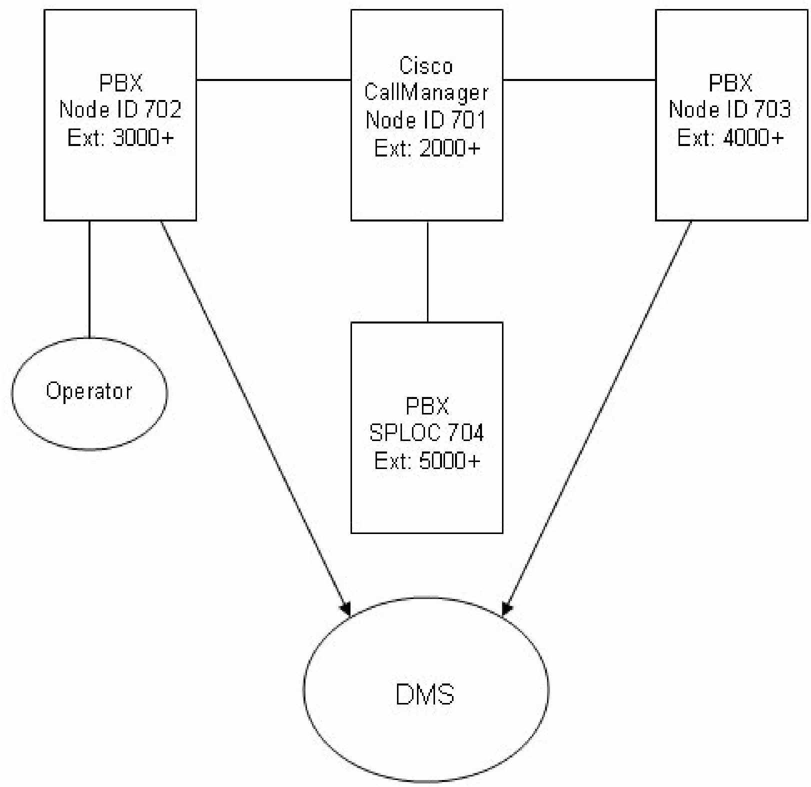
Figure 8-2 Cisco EGW Deployment



Legacy DPNSS PBX Configuration

Legacy PBXs were configured for the supplementary services, which are supported by the Cisco EGW gateway. Figure 8-3 shows an example of Legacy PBX topology.

Figure 8-3 Legacy PBX Topology



Configuration the Cisco EGW at Installation

Two Cisco EGW 2200s were installed for redundancy. Table 8-1 shows examples of the information that was provided to the Cisco EGW Installation Wizard when the Cisco EGWs were installed.

Table 8-1 Cisco EGW 2200 Enterprise Gateway Installation Information

Field	Setting for Active EGW	Setting for Standby EGW
Domain	site1.com	site1.com
DHCP	no	no
Host Name	Primary	Backup
IP Address	10.10.50.17	10.10.50.8
IP Mask	255.255.255.192	255.255.255.192
GW Address	10.10.50.5	10.10.50.5
Primary DNS Address	10.10.30.9	10.10.30.9
Secondary DNS Address (Optional)	—	—
Continent	Europe	Europe
City	London	London
SMTP location	*IP address or host name of mail server*	*IP address or host name of mail server*
Administrator ID	admin	admin
Password	*password*	*password*
Verify Password	*password*	*password*
2nd IP Addr	10.10.50.79	10.10.50.74
2nd IP Mask	255.255.255.192	255.255.255.192
SMTP location	cia.cisco.com	cia.cisco.com
Virtual IP Addr 1	blank	blank
Virtual IP Addr 2	blank	blank
NTP Client enabled	Yes	Yes
NTP Server 1	10.10.100.50	10.10.100.50
NTP Server 3	—	—
NTP Server 3	—	—
NTP Server 4	—	—
NTP Server 5	—	—

Configuring IP Routes, Media Gateways, and E1 Spans

The following sections provide an overview of how the IP routes, media gateways, and E1 spans were configured for the Cisco EGW 2200 Enterprise Gateways.

Configuring IP Routes

To access the Cisco EGW Administration web pages for adding and configuring IP routes, choose **Routes > IP Routes > Add New** from the Cisco EGW Administration application.

Table 8-2 describes the settings in the IP Route Page for the IP routes that were configured.

Table 8-2 IP Route Configuration

IPRoute Name	Priority	Dest. Network	Network Mask	Next Hop	Local Interface
gb1gwif1	1	10.10.30.64	255.255.255.192	10.10.50.4	IP_Addr1
gb1gwif2	2	10.10.30.64	255.255.255.192	10.10.50.68	IP_Addr2
gb1agwif1	1	10.10.50.64	255.255.255.192	10.10.50.4	IP_Addr1
gb1agwif2	2	10.10.50.64	255.255.255.192	10.10.50.68	IP_Addr2
rmtgwif1	1	10.10.93.0	255.255.255.0	10.10.50.4	IP_Addr1
rmtgwif2	2	10.10.93.0	255.255.255.0	10.10.50.68	IP_Addr2

Configuring Media Gateways

To access the Cisco EGW Administration web pages for adding and configuring media gateways, choose **Gateway Interfaces > Media Gateways > Add New** from the Cisco EGW Administration application.

Three media gateways were configured. Table 8-3 describes the settings in the Media Gateways page for these media gateways:

- gb1gw
- gb1agw
- rmtgw

Note You will not be able to enter settings in the Dial Plan field until you create dial plans.

Table 8-3 Media Gateway Configuration

Field	Settings for gb1gw	Settings for gb1agw	Settings for rmtgw
Name	gb1gw	gb1agw	rmtgw
Description	gb1-2691-gw-RDX	gb1-2691-gw-M1	rmt-3725-gw-iSDX
Gateway Type	c2600	c3600	c3725
Gateway Type	DPNSS_BTNR188	DPNSS_BTNR188	DPNSS_BTNR188
Dial Plan	gb03	gb03	gb03
First IP Address	10.10.30.76	10.10.50.76	10.10.93.1
IP Route	gb1gwif1	gb1agwif1	rmtgwif1
Second IP Address	10.10.30.76	10.10.50.76	10.10.93.1
IP Route	gb1gwif2	gb1agwif2	rmtgwif2
Gateway Host Name	gb1-2691-gw	gb1a-2691-gw	rmt-3725-gw
MGCP Port	2427	2427	2427
Session Set / Association Port	9900	9901	9902

Configuring Gateway Properties

To access the Cisco EGW Administration web pages for configuring gateway properties, choose **Gateway Interfaces > Media Gateways > *Gateway_name* > Properties** from the Cisco EGW Administration application.

Table 8-4 describes the settings in the Gateways Properties page for the configured media gateways.

Table 8-4 Gateway Properties Configuration

Field	Settings for gb1gw	Settings for gb1agw	Settings for rmtgw
Gateway Name	gb1gw	gb1agw	rmtgw
Audit On State Change to IS	Enable	Enable	Enable
Gateway Default Codec	Null	Null	Null
Gateway Protocol Version	MGCP 1.0	MGCP 1.0	MGCP 1.0
Initialize Endpoint	3	3	3
MGCP Retransmit Count	2000	2000	2000

Configuring E1 Spans

To access the Cisco EGW Administration web pages for adding and configuring E1 spans, choose **Gateway Interfaces > Media Gateways > *Gateway_name* > T1/E1** from the Cisco EGW Administration application.

Table 8-5 describes the settings in the T1/E1 Spans in the Gateways page for the configured media gateways.

Table 8-5 E1 Span Configuration

Field	Settings for gb1gw	Settings for gb1agw	Settings for rmtgw
Gateway Name	gb1gw	gb1agw	rmtgw
Signaling Slot	0	0	0

Table 8-5 E1 Span Configuration (continued)

Field	Settings for gb1gw	Settings for gb1agw	Settings for rmtgw
Signaling Port	0	0	0
E1/T1	E1	E1	E1
Trunk Selection Sequence	Ascending	Ascending	Ascending

Configuring Properties for E1 Spans

To access the Cisco EGW Administration web pages for configuring E1 span properties, choose **Gateway Interfaces > Media Gateways >** ***Gateway_name*** **> T1/E1 >** ***Span_name*** from the Cisco EGW Administration application.

Table 8-6 describes the settings in the E1 Span Properties page for the configured E1 Spans.

Table 8-6 E1 Span Properties Configuration

Field	Settings for gb1 E1 Span	Settings for gb1a E1 Span	Settings for rmtg E1 Span
Span Name	gb1gw-0-0-path	gb1agw-0-0-path	rmtgw-0-0-path
Compression	a-law	a-law	a-law
VPN Onnet Profile Index	5	5	5
T.38 Fax Support	Disable	Disable	Disable
Glare	X Side	X Side	X Side
Feature Transparency	Enable	Enable	Enable
Loop Avoidance Support	Enable	Enable	Enable
Loop Avoidance Counter	Enable	Enable	Enable
Incoming Calling name Display	Enable	Enable	Enable

Table 8-6 E1 Span Properties Configuration (continued)

Field	Settings for gb1 E1 Span	Settings for gb1a E1 Span	Settings for rmtg E1 Span
Outgoing Calling name Display	Enable	Enable	Enable
Incoming Connected Number Display	Enable	Enable	Enable
Outgoing Connected Number Display	Enable	Enable	Enable
Message Waiting Indicator Off String	*58B*AN*0#	*58B*AN*0#	*58B*AN*0#
Message Waiting Indicator On String	*58B*AN*1#	*58B*AN*1#	*58B*AN*1#
Call Origination Overlap Signaling	Disable	Disable	Disable
EGW Routing Number (Route Optimization)	701	701	701
Wait For Answer Timer (sec)	65	65	65
Trigger for SDP Transmit to H323	Answer	Answer	Answer

Media Gateway Configuration Files

The following sections show sample configuration files for the media gateways that were configured for the Cisco EGW 2200 Enterprise Gateway:

- Sample Configuration File for gb1gw Media Gateway, page 8-12
- Sample Configuration File for gb1agw Media Gateway, page 8-13
- Sample Configuration File for rmtgw Media Gateway, page 8-15

Sample Configuration File for gb1gw Media Gateway

```
version 12.3
service nagle
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname gb1-2691-gw
!
logging buffered 4096 debugging
logging rate-limit all 10
enable secret 5 $1$x8f0$QCQJ3aJIViXVaF2OdG1Zz1
!
clock timezone GMT 0
clock summer-time BST date Mar 28 2004 1:00 Oct 31 2004 2:00
!
network-clock-select 1 E1 0/0
ip subnet-zero
ip tcp synwait-time 5
ip cef
no ip domain lookup
!
ip host dpnss 10.10.50.17 10.10.50.79 10.10.50.8 10.10.50.74
!
isdn switch-type primary-dpnss
!
iua
AS dpnss-node 10.10.30.76 9900
ASP gwegw-1 AS dpnss-node 10.10.50.17 10.10.50.79 9900
ASP gwegw-2 AS dpnss-node 10.10.50.8 10.10.50.74 9900
!
controller E1 0/0
framing NO-CRC4
pri-group timeslots 1-31 service mgcp
!
interface FastEthernet0/0
description link to 6506-B 3/38
ip address 10.10.30.76 255.255.255.192
speed 100
full-duplex
!
interface Serial0/0:15
description link to S1 RDX
no ip address
no logging event link-status
isdn switch-type primary-dpnss
```

```
isdn incoming-voice voice
isdn bind-l3 iua-backhaul dpnss-node
no cdp enable
!
ip classless
ip route 0.0.0.0 0.0.0.0 10.10.30.68
ip http server
!
logging source-interface FastEthernet0/0
!
voice-port 0/0:15
cptone GB
!
mgcp
mgcp call-agent dpnss 2427 service-type mgcp version 1.0
mgcp dtmf-relay voip codec all mode nte-gw
mgcp rtp unreachable timeout 10000
mgcp codec g729r8 packetization-period 30
mgcp package-capability rtp-package
mgcp default-package gm-package
mgcp timer receive-rtcp 100
!
mgcp profile default
!
line con 0
line aux 0
line vty 0 4
exec-timeout 30 0
privilege level 15
password cisco
logging synchronous
login
!
ntp clock-period 17180430
ntp peer 10.10.100.52
ntp peer 10.10.100.53
ntp peer 10.10.100.50
ntp peer 10.10.100.51
!
end
```

Sample Configuration File for gb1agw Media Gateway

```
version 12.3
service nagle
service timestamps debug datetime msec localtime
```

```
service timestamps log datetime msec localtime
no service password-encryption
!
hostname gb1a-2691-gw
logging buffered 4096 debugging
!
clock timezone GMT 0
clock summer-time BST date Mar 28 2004 1:00 Oct 31 2004 2:00
!
network-clock-select 1 E1 0/0
ip subnet-zero
!
ip tcp synwait-time 5
!
ip cef
!
ip host dpnss 10.10.50.17 10.10.50.79 10.10.50.8 10.10.50.74
!
isdn switch-type primary-dpnss
!
iua
AS dpnss-node 10.10.50.76 9901
AS dpnss-node Fail-Over-Timer 10000
ASP gwegw-1 AS dpnss-node 10.10.50.17 10.10.50.79 9901
ASP gwegw-2 AS dpnss-node 10.10.50.8 10.10.50.74 9901
!
controller E1 0/0
framing NO-CRC4
pri-group timeslots 1-31 service mgcp
!
interface FastEthernet0/0
ip address 10.10.50.76 255.255.255.192
speed 100
full-duplex
!
interface Serial0/0:15
description link to S3 M1
no ip address
no logging event link-status
isdn switch-type primary-dpnss
isdn incoming-voice voice
isdn bind-l3 iua-backhaul dpnss-node
no cdp enable
!
ip classless
ip route 0.0.0.0 0.0.0.0 10.10.50.68
!
voice-port 0/0:15
```

```
cptone GB
!
mgcp
mgcp call-agent dpnss 2427 service-type mgcp version 1.0
mgcp dtmf-relay voip codec all mode nte-gw
mgcp codec g729r8 packetization-period 30
mgcp rtrcac
mgcp playout fixed 60
mgcp package-capability rtp-package
mgcp default-package gm-package
mgcp timer receive-rtcp 100
!
mgcp profile default
!
line con 0
line aux 0
line vty 0 4
privilege level 15
password cisco
logging synchronous
login
!
ntp clock-period 17180507
ntp peer 10.10.100.52
ntp peer 10.10.100.53
ntp peer 10.10.100.50
ntp peer 10.10.100.51
!
end
```

Sample Configuration File for rmtgw Media Gateway

```
version 12.3
service nagle
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
!
hostname rmt-3725-gw
!
logging buffered 100000 debugging
logging rate-limit all 10
!
clock timezone GMT 0
clock summer-time BST date Mar 28 2004 1:00 Oct 31 2004 2:00
network-clock-select 1 E1 0/0
```

```
ip subnet-zero
ip cef
!
ip tcp synwait-time 5
!
no ip domain lookup
ip host dpnss 10.10.50.17 10.10.50.79 10.10.50.8 10.10.50.74
!
isdn switch-type primary-dpnss
!
iua
AS dpnss-node 10.10.93.1 9902
ASP gwegw-rmt1 AS dpnss-node 10.10.50.17 10.10.50.79 9902
ASP gwegw-rmt2 AS dpnss-node 10.10.50.8 10.10.50.74 9902
!
controller E1 2/0
framing NO-CRC4
clock source internal
pri-group timeslots 1-31 service mgcp
!
interface FastEthernet0/0
ip address 10.10.93.1 255.255.255.0
load-interval 30
speed 100
full-duplex
!
interface Serial2/0:15
description link to ISDX
no ip address
load-interval 30
isdn switch-type primary-dpnss
isdn incoming-voice voice
isdn bind-l3 iua-backhaul dpnss-node
isdn dpnss pbxA
no cdp enable
!
ip classless
ip route 0.0.0.0 0.0.0.0 10.10.30.1
ip route 0.0.0.0 0.0.0.0 10.10.30.65
ip http server
!
voice-port 0/0:15
echo-cancel coverage 32
cptone GB
timeouts interdigit 6
timeouts ringing 600
music-threshold -70
!
```

```
mgcp
mgcp call-agent dpnss 2427 service-type mgcp version 1.0
mgcp rtp unreachable timeout 10000
mgcp codec g729br8 packetization-period 30
mgcp package-capability rtp-package
mgcp default-package gm-package
mgcp tse payload 102
mgcp timer receive-rtcp 100
mgcp bind control source-interface FastEthernet0/0
mgcp bind media source-interface FastEthernet0/0
!
mgcp profile default
!
line con 0
line aux 0
line vty 0 4
session-timeout 30
exec-timeout 30 0
privilege level 15
password cisco
logging synchronous
login
!
ntp clock-period 17180369
ntp peer 192.168.254.113
end
```

Configuring Cisco EGW Gatekeepers, Route Plans, and Dial Plans

The following sections provide an overview of how the gatekeeper, route plans, and dial plans were configured for the Cisco EGWs. These configurations must be performed on both the active and the standby Cisco EGW.

- Configuring Cisco EGW Gatekeepers, page 8-18
- Configuring the CTI Manager for the Cisco EGW, page 8-19
- Configuring the AXL Server for the Cisco EGW 2200, page 8-20
- Configuring Cisco EGW Route Plans, page 8-21
- Configuring Cisco EGW Dial Plans, page 8-22

Configuring Cisco EGW Gatekeepers

To access the Cisco EGW Administration web page for configuring gatekeepers, choose **CallManager Interfaces > H323** from the Cisco EGW Administration application.

Table 8-7 describes the settings in the H323 page for each gatekeeper.

Note You will not be able to enter settings in the Dial Plan field until you create dial plans.

Table 8-7 Gatekeeper Configuration

Field	Settings for Active Cisco EGW	Settings for Standby Cisco EGW
Gatekeeper Name	egw1	egw1
Gatekeeper IP Address	10.10.50.73	10.10.50.73
Gatekeeper Port	1719	1719
Terminal Alias	Active@site1.com	Standby@site1.com
Node ID	EGW-1	EGW-1
RAI Support	Enable	Enable
Notify Enabled	Enable	Enable
Screening/Presentation	Enable	Enable
T.38 Fax Support	Enable	Enable
Redirecting Number	Enable	Enable
DTMF Support Direction	Both	Both
DTMF Support Type	DTMF	DTMF
G.711 A-law Packetization	20 fpp	20 fpp
G.711 u-law Packetization	20 fpp	20 fpp
G.723.1 MaxAudioFrames	1 fpp	1 fpp

Table 8-7 Gatekeeper Configuration (continued)

Field	Settings for Active Cisco EGW	Settings for Standby Cisco EGW
G.723.1 SilenceSuppression	Disable	Disable
G.729 Packetization	3 fpp	3 fpp
G.729a Packetization	3 fpp	3 fpp
G.729b Packetization	3 fpp	3 fpp
Incoming Calling name Display	Enable	Enable
Outgoing Calling name Display	Enable	Enable
Incoming Connected Number Display	Enable	Enable
Outgoing Connected Number Display	Enable	Enable
Wait For Answer Timer (sec)	65	65
Dial Plan	gb03	gb03
Prefixes	3 4 5 702 703 704	3 4 5 702 703 704

Configuring the CTI Manager for the Cisco EGW

To access the Cisco EGW Administration web pages for configuring CTI managers, choose **CallManager Interfaces > CTI Managers** from the Cisco EGW Administration application.

Table 8-8 shows how the CTI manager was configured in the Add New CTI Manager page.

Table 8-8 CTI Manager Configuration

Field	Setting
CTI Manager	ctimgr1
Description	ctimgr1
CTI Manager Version	5.0
Local Interface	EGW_IP_Addr1
CTI Manager IP Address	10.10.30.69 **Note** This address is that of the Cisco CallManager server
IP Route	gb1gwif1
Port	2748
CallBack Timeout when Next Used (min)	60
CallBack Timeout when Next Free (min)	60
Maximum CallBacks queued	10

Configuring the AXL Server for the Cisco EGW 2200

To access the Cisco EGW Administration web page for configuring the AXL server, choose **CallManager Interfaces > AXL Servers** from the Cisco EGW Administration application.

Table 8-9 shows how the AXL server was configured in the Add New AXL Servers page.

Table 8-9 AXL Server Configuration

Field	Setting
AXL Server Name	axlsvr1
Description	axlsvr1
Local Interface	EGW_IP_Addr1

Table 8-9 AXL Server Configuration (continued)

Field	Setting
AXL Server IP Address	10.10.30.69
IP Route	gb1gwif1
Port	80
AXL Server Username	Administrator

Configuring Cisco EGW Route Plans

The route plan configuration identifies the trunks required for routing calls. The available routes are configured automatically when the E1 spans and the H.323 gatekeepers are configured.

To access the Cisco EGW Administration web page for configuring route plans, choose **Dial and Route Plans > Standard Route Plans > Add New** from the Cisco EGW Administration application.

The following route plans were added:

- ccm-std
- rdx-std
- m1-std
- isdx-std

Table 8-10 shows the route selections for each of these route plans.

The random selection check box is checked only for the ccm-std route plan on the Edit Route Plan page.

Table 8-10 Route Selections

Route Plan	Route Selection
ccm-std	rt-h323-a rt-h323-b
rdx-std	rt-gb1gw-0-0

Table 8-10 Route Selections (continued)

Route Plan	Route Selection
m1-std	rt-gb1agw-0-0
isdx-std	rt-rmtgw-0-0

Configuring Cisco EGW Dial Plans

To access the Cisco EGW Administration web page for configuring dial plans, choose **Dial and Route Plans > Dial Plans > Add New** from the Cisco EGW Administration application.

Table 8-11 shows how called numbers were configured in the Add Dial Plan page. In this configuration, the 70*X* numbers are necessary if route optimization is required.

Table 8-11 Dial Plan Configuration

Digit String	Digits to Remove	Prefix Digit String	CCM MWI	Min Digits	Max Digits	Route Plan
2	0	blank	Unchecked	2	4	ccm-std
3	0	blank	Unchecked	2	4	rdx-std
4	0	blank	Unchecked	2	4	m1-std
5	0	blank	Unchecked	2	4	isdx-std
701	0	blank	Unchecked	4	15	ccm-std
702	0	blank	Unchecked	4	15	rdx-std
703	0	blank	Unchecked	4	15	m1-std
704	0	blank	Unchecked	4	15	isdx-std

IOS Gatekeeper Configuration File

This section shows a sample IOS gatekeeper configuration file. This file was used on both Cisco EGWs.

```
version 12.3
service nagle
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
!
hostname gb1a-3725-gk
!
logging buffered 4096 debugging
!
clock timezone GMT 0
clock summer-time BST date Mar 28 2004 1:00 Oct 31 2004 2:00
ip subnet-zero
!
ip cef
!
interface FastEthernet0/0
ip address 10.10.50.73 255.255.255.192
speed 100
full-duplex
!
ip http server
ip classless
ip route 0.0.0.0 0.0.0.0 10.10.50.68
!
gatekeeper
zone local egw1 site1.com 10.10.50.73
no shutdown
line con 0
line aux 0
line vty 0 4
privilege level 15
password cisco
logging synchronous
login
!
ntp clock-period 17180759
ntp peer 10.10.100.52
ntp peer 10.10.100.53
ntp peer 10.10.100.50
ntp peer 10.10.100.51
!
end
```

Configuring Cisco CallManager for Cisco EGWs

The following sections describe how Cisco CallManager was configured for use with Cisco EGWs. These configuration settings were made in addition to the settings described in Chapter 2, "Cisco CallManager Configuration."

Configuring the Cisco CallManager Gatekeeper

To access the Cisco CallManager Administration web pages for configuring gatekeepers, choose **Device > Gatekeeper > Add a New Gatekeeper** from the Cisco CallManager Administration application.

Table 8-12 shows how the Cisco CallManager gatekeeper was configured for the Cisco EGW in the Gatekeeper Configuration page.

Table 8-12 Cisco CallManager Gatekeeper Configuration

Field	Setting
Host Name/IP Address	10.10.50.73
Description	gb1a-3725-gk
Registration Request Time to Live	60
Registration Retry Timeout	60
Enable Device	Checked

Configuring Cisco CallManager Trunks

To access the Cisco CallManager Administration web pages for configuring trunks, choose **Device > Trunk > Add a Trunk** from the Cisco CallManager Administration application.

The following H.255 gatekeeper controlled trunks were configured:

- 701_PBX_IC_call
- GB_CCM_2000+

Table 8-13 shows how the trunks were configured for the Cisco EGWs in the Trunk Configuration page.

Table 8-13 Cisco CallManager Trunk Configuration

Field	Setting for Trunk 1	Setting for Trunk 2
Device Name	701_PBX_IC_call	GB_CCM
Description	701_PBX_IC_call	GB_CCM_2000+
Device Pool	Default	Default
Media Resource Group List	MRGL_Central	MRGL_Central
Location	<None>	<None>
AAR Group	<None>	<None>
Media Termination Point Required	blank	blank
Retry Video Call as Audio	Checked	Checked
Wait for Far End H.245 Terminal Capability Set	Checked	Checked
Significant Digits	All	All
Calling Search Space	Central1CSS	Central1CSS
AAR Calling Search Space	<None>	<None
Prefix DN	blank	blank
Redirecting Number IE Delivery - Inbound	Checked	Checked
Calling Party Selection	Originator	Originator
Calling Line ID Presentation	Default	Default
Called party IE number type unknown	Cisco CallManager	Cisco CallManager
Calling party IE number type unknown	Cisco CallManager	Cisco CallManager

Table 8-13 Cisco CallManager Trunk Configuration (continued)

Field	Setting for Trunk 1	Setting for Trunk 2
Called Numbering Plan	Cisco CallManager	Cisco CallManager
Calling Numbering Plan	Cisco CallManager	Cisco CallManager
Caller ID DN	blank	blank
Display IE Delivery	Checked	Checked
Redirecting Number IE Delivery - Outbound	blank	blank
Gatekeeper Name	10.10.50.73	10.10.50.73
Technology Prefix	701	2
Zone	egw1	egw1
MLPP Domain	blank	blank
MLPP Indication	blank	blank
MLPP Preemption	blank	blank

Using Microsoft Active Directory 2003 with an IPT Solution

You can use Microsoft Active Directory 2003 instead of the default DC-Directory for directory services with your IPT solution. If you do so, make sure that you configure Cisco CallManager, Cisco Customer Response Applications, and Cisco Unity for use with Microsoft Active Directory 2003.

If you will use Microsoft Active Directory 2003, you must set it up before you make other configuration settings in Cisco CallManager.

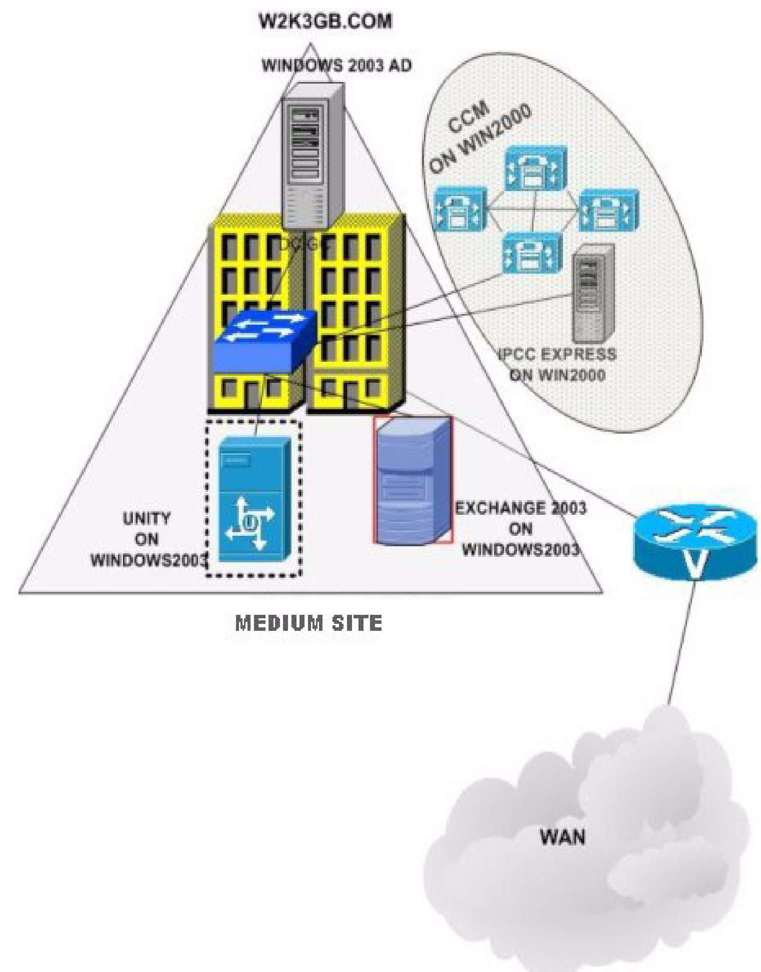
If you will use Microsoft Active Directory 2003, review the guidelines and references that are provided in this chapter.

This chapter includes the following topics:

Microsoft Active Directory 2003 Topology in a Medium Site Model

Figure 9-1 shows the topology of Microsoft Active Directory 2003 when integrated in IP Communications Systems Test Release 3.0 for EMEA IPT.

Figure 9-1 Microsoft Active Directory 2003 Topology in a Medium Site Model



Using Cisco CallManager with Microsoft Active Directory 2003

Review the following information and guidelines if you integrate Cisco CallManager with Microsoft Active Directory 2003:

- Cisco CallManager was tested with Microsoft Active Directory 2003.
- For detailed information about integrating refer to *Installing the Cisco Customer Directory Configuration Plugin for Cisco CallManager* at this URL:

  http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_installation_and_configuration_guide09186a00801ed28e.html
- Set the CCMAdministrator and CCMsysuser account passwords using the CCMPwdChanger utility.
- The Cisco CallManager servers do not need to be members of the domain. They can remain in a standalone workgroup.

Using Cisco Customer Response Applications with Microsoft Active Directory 2003

For detailed information about integrating Cisco CRA with Microsoft Active Directory 2003, refer to Getting Started with Cisco Customer Response Applications at this URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/sw_ap_to/apps_3_5/english/admn_app/gs35.pdf

Cisco CRA servers do not need to be members of the domain. They can remain in a standalone workgroup.

Using Cisco Unity with Microsoft Active Directory 2003

A Windows 2003 server was promoted to Domain Controller. Cisco Unity and Microsoft Exchange were tested in that Microsoft Active Directory 2003 domain.

Troubleshooting and Technical Tips

This chapter provides basic troubleshooting information and tips for IPT scenarios. It also provides references to other troubleshooting information.

This chapter includes the following topics:

- General Troubleshooting Tips, page 10-1
- Additional Troubleshooting Resources, page 10-2

General Troubleshooting Tips

- In an IPT solution, Cisco recommends that the Ethernet interface for Cisco IP Phones be set to Autodetection. Cisco recommends that other Ethernet interfaces, including those for switches, routers, gateways, and Cisco Media Convergence Servers (MCSs), be set to 100 Mbps/full duplex.
- The use of the Cisco CallManager parameter VoiceMailMaximumHopCount (accessed by choosing **Service Parameters > Call Manager > Cluster Wide Parameters** from Cisco CallManager Administration) substantially reduces the amount of time required for Cisco CallManager to discover and utilize an available voice mail port in a large system. To determine the appropriate value for this parameter, identify the number of voice mail ports within each voice mail profile and subtract 3 from the largest number.

  In addition, the AdvancedCallForwardHopFlag must be set to True to take advantage of this service parameter. If it is set to False the ForwardMaximumHopCount value will be used.

Note For the effects of this parameter to be realized, a system requires at least 75 voice mail ports in a single voice mail profile. For other systems, the default configuration should be sufficient.

Additional Troubleshooting Resources

There are many Cisco documents that provide troubleshooting information at the solutions level and for various hardware and software components. Table 10-1 lists some of these documents. You can find additional troubleshooting information in other product-specific documentation.

Table 10-1 Troubleshooting References

Troubleshooting Topic	Document	URL
General IPT issues	*Troubleshooting Cisco IP Telephony* (published by Cisco Press, ISBN 1-58705-075-7)	—
Cisco ATA 186 and ATA 188 issues	*Cisco ATA 186 and Cisco ATA 188 Analog Telephone Adaptor Administrator's Guide (SCCP)*, "Troubleshooting" chapter	http://www.cisco.com/univercd/cc/td/doc/product/voice/ata/ataadmn/sccp/sccpach5.htm
Cisco CallManager issues	• *Troubleshooting Guide for Cisco CallManager* • *System Error Messages for Cisco CallManager 3.3*	http://www.cisco.com/univercd/cc/td/doc/product/voice/c_callmg/4_0/trouble/4_0_1/index.htm
Cisco Emergency Responder issues	*Cisco Emergency Responder Administration Guide 1.2*, "Troubleshooting Cisco Emergency Responder" chapter	http://www.cisco.com/univercd/cc/td/doc/product/voice/respond/res12/admin12/e911trbl.htm

Table 10-1 Troubleshooting References (continued)

Troubleshooting Topic	Document	URL
Cisco Customer Response Applications issues	*Troubleshooting Cisco Customer Response Applications* **Note** Cisco Customer Response Applications Administration also provides on-line troubleshooting tips	http://www.cisco.com/univercd/cc/td/doc/product/voice/sw_ap_to/apps_3_5/english/admn_app/trbshoot.pdf
Cisco Personal Assistant issues	*Cisco Personal Assistant Installation and Administration Guide*, "Troubleshooting Personal Assistant" chapter	http://www.cisco.com/univercd/cc/td/doc/product/voice/assist/assist14/ag/ag141/patrbl.htm
Cisco SRS Telephony issues	*Cisco Survivable Remote Site Telephony Version 2.1*, "Cisco SRS Telephony Configuration" chapter	http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/taclinks.htm
Cisco Unity issues	*Cisco Unity Troubleshooting Guide*	http://www.cisco.com/univercd/cc/td/doc/product/voice/c_unity/unity40/tsg/tsg404/index.htm
Cisco Unity Bridge issues	*Cisco Unity Bridge Networking Guide*, "Cisco Unity Bridge Troubleshooting" chapter	http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products_installation_and_configuration_guide_chapter09186a00801187d7.html
Cisco VG248 issues	*Cisco VG248 Analog Phone Gateway Software Configuration Guide*, "Troubleshooting the VG248" chapter	http://www.cisco.com/univercd/cc/td/doc/product/voice/c_access/apg/vg248/v1_3/swcfg/vg248swt.htm

CHAPTER 11

Cisco CallManager Failure, Failover, and Recovery

This chapter provides an overview of the failover testing that was performed for the Cisco CallManager in IP Communications Systems Test Release 3.0 for EMEA IPT.

For more information about failover and recovery for a specific component, refer to the documentation for that component.

This chapter includes the following topics:

Test Conditions

The following conditions existed for the Cisco CallManager failover testing:

- Cisco CallManager cluster consisting of
  - Primary Cisco CallManagers, SUB1 and SUB2
  - Backup Cisco CallManagers, BACKUP1 and BACKUP2
  - Publisher/TFTP Server

- Cisco CallManager servers: MCS-7835-1266 with 1 GB RAM
- Tests were run on the Multi-Site Single-Cluster Distributed scenario, configured as follows:
  - Two sites
  - 1,000 phones and additional gateways at each site
  - In Medium Site 1: Cisco Catalyst 6608 and Cisco Catalyst 6624 registered to SUB1
  - In Medium Site 2: Cisco Catalyst Communication Media Module and Cisco VG248 registered to SUB2

Test 1: Disconnected Cable from Primary Cisco CallManager Server

Test

Made manual calls over the WAN, DPNSS, PSTN, and local LAN through multiple gateways that were registered to SUB2. Ensured that calls were maintained during failover. Also send simulated call traffic for load.

Failed SUB2 by disconnecting its Ethernet cable.

Results

Verified that 1,000 phones and additional gateways failed over to SUB2. Manual calls did register until calls cleared. Verified voice path for manual calls. After calls were disconnected, the 1,000 phones successfully registered with BACKUP2.

Test 2: Failback

Test

Made new manual calls as described in the "Test 1: Disconnected Cable from Primary Cisco CallManager Server" section on page 11-2. Reconnected Ethernet cable to SUB2.

Results

Verified that 1,000 phones and additional gateways failed back to SUB2. Manual calls did not fail back. Verified voice path for manual calls. After calls were disconnected, verified that manual calls successfully registered with SUB2.

Test 3: Failover, Failover, Failback

Test

Made new manual calls as described in the "Test 1: Disconnected Cable from Primary Cisco CallManager Server" section on page 11-2. Failed SUB2 and BACKUP2 by disconnecting their Ethernet cables.

Ensured that existing calls are maintained. After calls cleared, ensured that phones register to SUB1.

Then reconnected Ethernet cables to SUB2 and BACKUP2.

Results

Verified that phones failed over to highest priority Cisco CallManager. Verified that existing calls were maintained. After service restored to SUB2 and BACKUP2, verified that all phones re-registered to highest priority Cisco CallManager (SUB2).

Call Load Testing

This chapter provides an overview of the call loads that were tested in IP Communications Systems Test Release 3.0 for EMEA IPT. It includes information for calls made in a Multi-Site Single-Cluster Distributed scenario with two sites.

The information in this chapter represents tested call rates, not the maximum call rate capabilities of the system.

Table 12-1 shows the number of busy hour call attempts (BHCA) that were made. Calls were made from both sites in the scenario. Thirty percent of the IP phone-to-IP phone traffic was intersite, using the WAN.

Table 12-1 Load Testing for Multi-Site Single-Cluster Distributed Scenario

Call Type	Call Activity	Total BHCA
IP phone to IP phone	Basic call	1,680
	Blind transfer	100
	Consult transfer	60
	Hold	120
	Conference	20
	Operator	20

Table 12-1 Load Testing for Multi-Site Single-Cluster Distributed Scenario (continued)

Call Type	Call Activity	Total BHCA
PSTN to IP phone	Basic call	325
	Blind transfer	25
	Consult transfer	15
	Hold	30
	Conference	5
	Operator	100
IP phone to PSTN	Basic call	415
	Blind transfer	25
	Consult transfer	15
	Hold	30
	Conference	5
IP phone to DPNSS PBX	—	400
DPNSS PBX to IP phone	—	400
IP phone to Analog phone (on net)	—	54
Analog phone (on net) to IP phone	—	54
PSTN (Cisco 6608) to Analog phone (Cisco VG248/Cisco 6624)	—	36
Analog phone (Cisco VG248/Cisco 6624) to PSTN (Cisco 6608)	—	36
PSTN (Cisco 6608) to Fax (Cisco VG248/ Cisco 6624)	—	54
Fax (Cisco VG248/ Cisco 6624) to PSTN (Cisco 6608)	—	54

Table 12-1 Load Testing for Multi-Site Single-Cluster Distributed Scenario (continued)

Call Type	Call Activity	Total BHCA
IP Phone to Cisco Unity (message access)	—	400
IP Phone to Cisco Unity (message deposit)	—	1,000

Release Versions of Components

The following tables show the release versions of the hardware and software components used in IP Communications Systems Test Release 3.0 for North America and EMEA IPT:

- Table A-1 on page A-1—Software release versions of components used in IP Communications Systems Test Release 3.0 for North America IPT and EMEA IPT
- Table A-2 on page A-4—Software release versions of components used only in IP Communications Systems Test Release 3.0 for EMEA IPT
- Table A-3 on page A-4—Firmware release versions of Cisco IP Phones

Table A-1 Software Release Versions of Components

Component	Release Version
Cisco CallManager	4.0(2a) SR1a
Cisco CallManager—Cisco IP Telephony Operating System	2000.2.6SR5
Cisco Customer Response Solutions (IPCC Express / IP IVR)	3.5(2) SR1
Cisco Customer Response Solutions—Cisco IP Telephony Operating System	2000.2.6SR5
Cisco Emergency Responder	1.2(2)
Cisco Unity, TSP	4.0(4) SR1, 7.0(4)

Table A-1 Software Release Versions of Components (continued)

Component	Release Version
Cisco Unity–Microsoft Exchange	Exchange 2000 SP4
Cisco MeetingPlace MP8112	5.2.1.7
Cisco CallManager Express	3.1
Cisco Unity Express	1.1(2)
Cisco Personal Assistant	1.4(3)
Cisco IP Manager Assistant	1.3(4)
IP/VC (3511 MCU)	3.2.113
IP/VC (3521 BRI video gateway)	1.2.0.9.4
IP/VC (3526 PRI video gateway)	2.0.1.13
IP/VC (3540 MCU)	3.2.113
Cisco 3660 (gatekeeper)	12.3(8)T5
Cisco 3725, 3745 (gatekeeper)	12.3(8)T5
Cisco 1760 (voice/data gateway)	12.3(8)T5
Cisco 2610XM, 2611XM, 2620XM, 2621XM, 2650XM, 2651XM, 2691 (voice/data gateway)	12.3(8)T5
Cisco 3660 (voice/data gateway)	12.3(8)T5
Cisco 3725, 3745 (voice/data gateway)	12.3(8)T5
Cisco 7206 (voice/data gateway)	12.3(8)T5
Cisco Catalyst 3524 (access switch)	12.0(5)WC5
Cisco Catalyst 3550 (access switch)	12.1(19)EA1c
Cisco Catalyst 4506 (access switch)	12.1(19)EW1
Cisco Catalyst 6506, 6509 (voice access switch)	Cat 8.3(3)
Cisco Catalyst 6506, 6509 (core switch)	Cat 7.6(9)
Cisco Catalyst 6506, 6509 (MSFC)	12.1(23)E1

Table A-1 Software Release Versions of Components (continued)

Component	Release Version
Cisco Catalyst Communications Media Module (CMM)	12.3(8)XY
Cisco Catalyst 6608, 6624 (voice gateway)	Bundled with CatOS
Cisco VG224 (analog voice gateway)	12.3(8)T5
Cisco VG248 (analog voice gateway)	1.3(1)
Cisco ATA 186, 188 (analog telephony adaptor)	3.1(0)
Cisco Security Agent Management Center	4.0.2.629 with security policy 1.0.6
Cisco Security Agent Management Policy—Cisco CallManager	1.1(9)
Cisco Security Agent Management Policy—Cisco Customer Response Solutions	1.1(9)
Cisco Security Agent Management Policy—Cisco Personal Assistant	1.1(2)
Cisco Security Agent Management Policy—Cisco Unity	1.1(4)
Anti-virus—McAfee	Enterprise 7.1.0
CiscoWorks 2000 ITEM	2.0(2)
Cisco IP Phones models 7902G, 7905G, 7910, 7912G, 7920, 7935, 7936G, 7940G, 7960G, 7970G	Bundled with Cisco CallManager
Cisco IP Communicator	1.1(2)
Tandberg T550, T1000 (SCCP)	I1.3
Cisco VT Advantage	1.0(2)
Cisco Aironet Access Point (AP) 1100/1200	12.2(13)JA1

Table A-2 Software Release Versions of EMEA-Specific Components

Component	Release Version
Cisco EGW 2200 (enterprise gateway)	1.1(2)P4
Cisco 3725 (voice gateway used with EGW)	12.3(8)T5
Cisco 2691 (voice gateway used with EGW)	12.3(8)T5
Cisco Unity interworking with EGW	4.0(4)SR1 + ES39
Cisco Unity—IBM/Lotus Domino	6.0.x with DUC 1.2.2
Cisco IP Telephony Locale Installer	40010

Table A-3 Firmware Release Versions of Cisco IP Phones

Phone Model	Firmware Version
Cisco IP Phone 7902G	CP7902050000SCCP041007A
Cisco IP Phone 7905G	CP7905050000SCCP041022A
Cisco IP Phone 7910	P00405000600
Cisco IP Phone 7912G	CP7912050000SCCP041022A
Cisco IP Phone 7920	cmterm_7920.3.3-01-03
Cisco IP Phone 7935	P00503010800
Cisco IP Phone 7936G	cmterm_7936.3-3-2-0
Cisco IP Phone 7940G	P00306000500
Cisco IP Phone 7960G	P00306000500
Cisco IP Phone 7970G	TERM70.6-0-1-0sr1s


INDEX

P

R

S

T

V

W