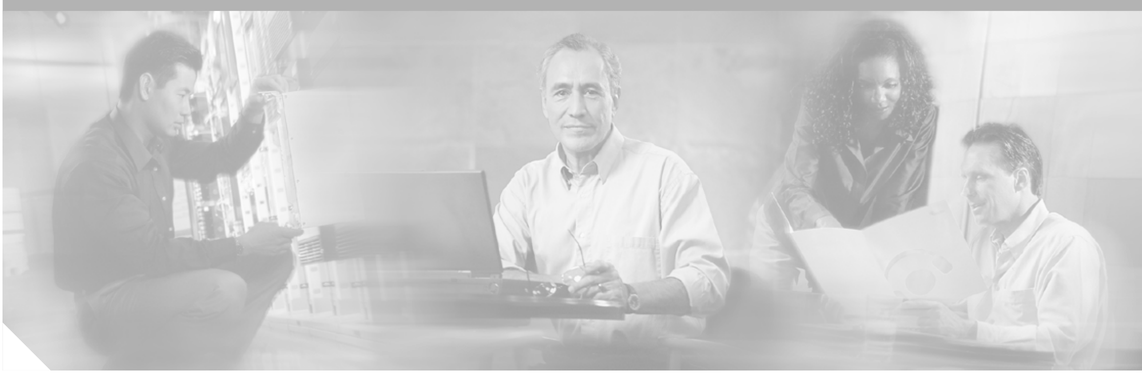

Systems Test Architecture Reference Manual for North America IPT

IP Communications Systems Test Release 3.0

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CONTENTS

Preface

Overview

Systems Test Architecture Reference Manual for North America IPT describes the components and configurations that have been tested and verified as part of IP Communications Systems Test Release 3.0 for North America 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 Very Large Site model
Chapter 3, "Cisco Unity Configuration"	Provides an overview of how Cisco Unity was set up for the Very Large Site model
Chapter 4, "Cisco CallManager Express and Cisco Unity Express Configurations"	Provides a sample configuration file for Cisco CallManager Express and Cisco Unity Express.
Chapter 5, "Cisco Personal Assistant System Configuration"	Provides an overview of how Cisco Personal Assistant was set up for the Very Large Site model
Chapter 6, "Cisco Emergency Responder Configuration"	Provides an overview of how Cisco Emergency Responder was set up for the Very Large Site model
Chapter 7, "Cisco Customer Response Applications Configuration"	Provides an overview of how Cisco Customer Response Applications was set up for the Very Large Site model
Chapter 8, "Cisco MeetingPlace Configuration"	Provides an overview of how Cisco MeetingPlace was set up for the Very Large Site model
Chapter 9, "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 10, "IP Video Telephony Configuration"	Provides an overview of how IP Video Telephony was configured
Chapter 11, "Using Microsoft Active Directory 2003 with an IPT Solution"	Provides guidelines and references for using Microsoft Active Directory 2003 with an IPT solution
Chapter 12, "Hardware Configurations"	Provides sample configuration information for a gatekeeper and a gateway
Chapter 13, "Fax, Modem, and TTY/TDD Configurations"	Provides an overview of how gateway devices were configured for fax/modem pass-through and fax relay modes
Chapter 14, "Quality of Service Configuration"	Provides sample Quality of Service configuration files
Chapter 15, "Call Flow"	Describes a selected call flow
Chapter 16, "Troubleshooting and Technical Tips"	Provides guidance and resources for diagnosing and correcting errors
Chapter 17, "Cisco CallManager Failure, Failover, and Recovery"	Provides an overview of failover testing
Chapter 18, "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

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

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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

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CHAPTER 1

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 North America IPT:

- Single Site scenario
- Multi-Site Centralized scenario
- Multi-site Distributed scenario

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

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 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 IPT:

- Single Site Scenario, page 1-3
- Multi-Site Centralized Scenario, page 1-5
- Multi-Site Distributed Scenario, page 1-7

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 any one of these site models:

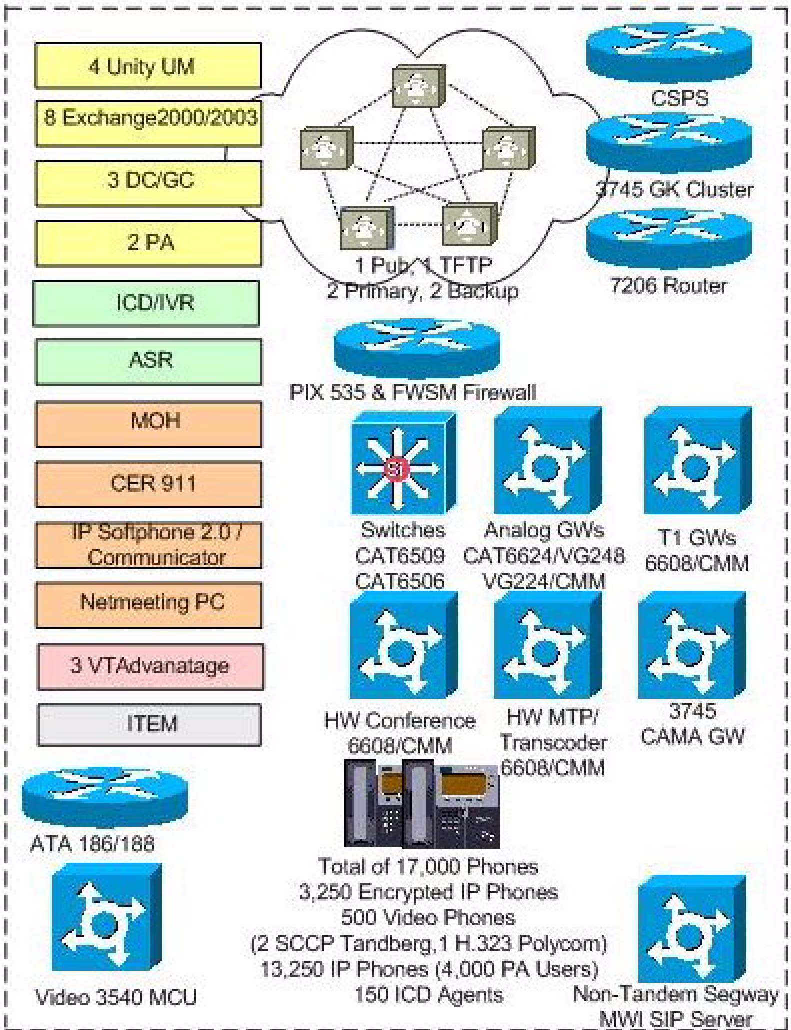
- Very Large Site model. For more information, see the "Very Large Site Model" section on page 1-10.
- Large Site model. For more information, see the "Large Site Model" section on page 1-13.
- Medium Site model. For more information, see the "Medium Site Model" section on page 1-15.
- Small Site Model. For more information, see the "Small Site Model" section on page 1-17.
- Small Site Model with Cisco CallManager Express. For more information, see the "Small Site with Cisco CallManager Express Model" section on page 1-19.

The tested Single Site scenarios have the following design characteristics:

- Support for up to 17,000 phones at a very large site, 3,000 phones at a large site, 1,000 phones at a medium site, 500 phones at a small site, or 500 phones at a small site with Cisco CallManager Express
- Cisco CallManager cluster for redundancy and system scaling (except in the small site with Cisco CallManager Express)

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

Figure 1-1 Single Site Scenario (Very Large Site Model)



Multi-Site Centralized Scenario

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

In this scenario, the central site was split into two sub-sites. One site included a publisher server, a TFTP server, and two subscriber servers. The other site included two subscriber servers. These six servers composed a single Cisco CallManager cluster. The sub-sites were connected over a T3 WAN, which was used for intra-cluster communication between Cisco CallManager servers. Each Cisco CallManager server used a corresponding Cisco CallManager server at the other site for failover.

If the central site or the IP 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 IP WAN becomes temporarily over-subscribed.

The tested Multi-Site Centralized scenario is composed of one Central Site model and 503 Remote Site models. (For testing purposes, approximately 400 of the Remote Site models were simulated.)

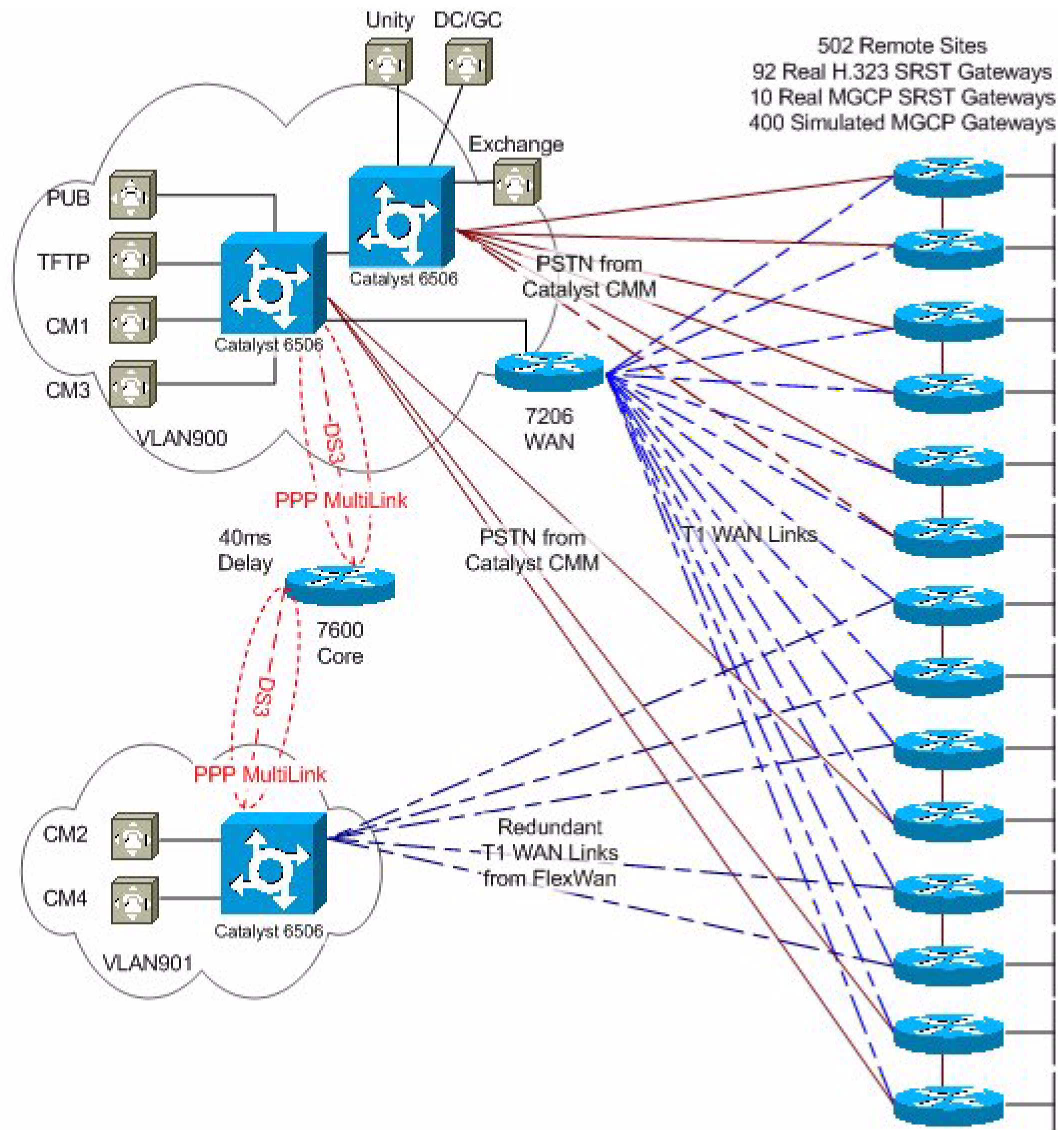
For more information about the Multi-Site Centralized scenario site models, see the "Central Site Model" section on page 1-21 and the "Remote Site Models" section on page 1-23.

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

- Cisco CallManager cluster and Cisco Unity Unified Messaging reside at the central site
- Cisco Unity, in a deployment where Microsoft Exchange 5.5 is migrated to Microsoft Exchange 2003
- Cisco Unity ViewMail for Outlook (VMO) in remote sites
- Centralized dial plan and administration
- Call admission control based on locations (to protect voice quality of IP WAN calls)
- The cluster was split across a T3 WAN
- 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 Distributed Scenario

A Multi-Site Distributed scenario consists of multiple independent sites, each with its own Cisco CallManager cluster. An IP WAN carries only IP encapsulated voice traffic and intercluster trunk call signaling for calls between the two sites.

The Multi-Site Distributed scenario is composed of various Very Large Site models, Large Site models, Medium Site models, and Small Site models.

For more information about the Multi-Site Distributed scenario site models, see the following sections:

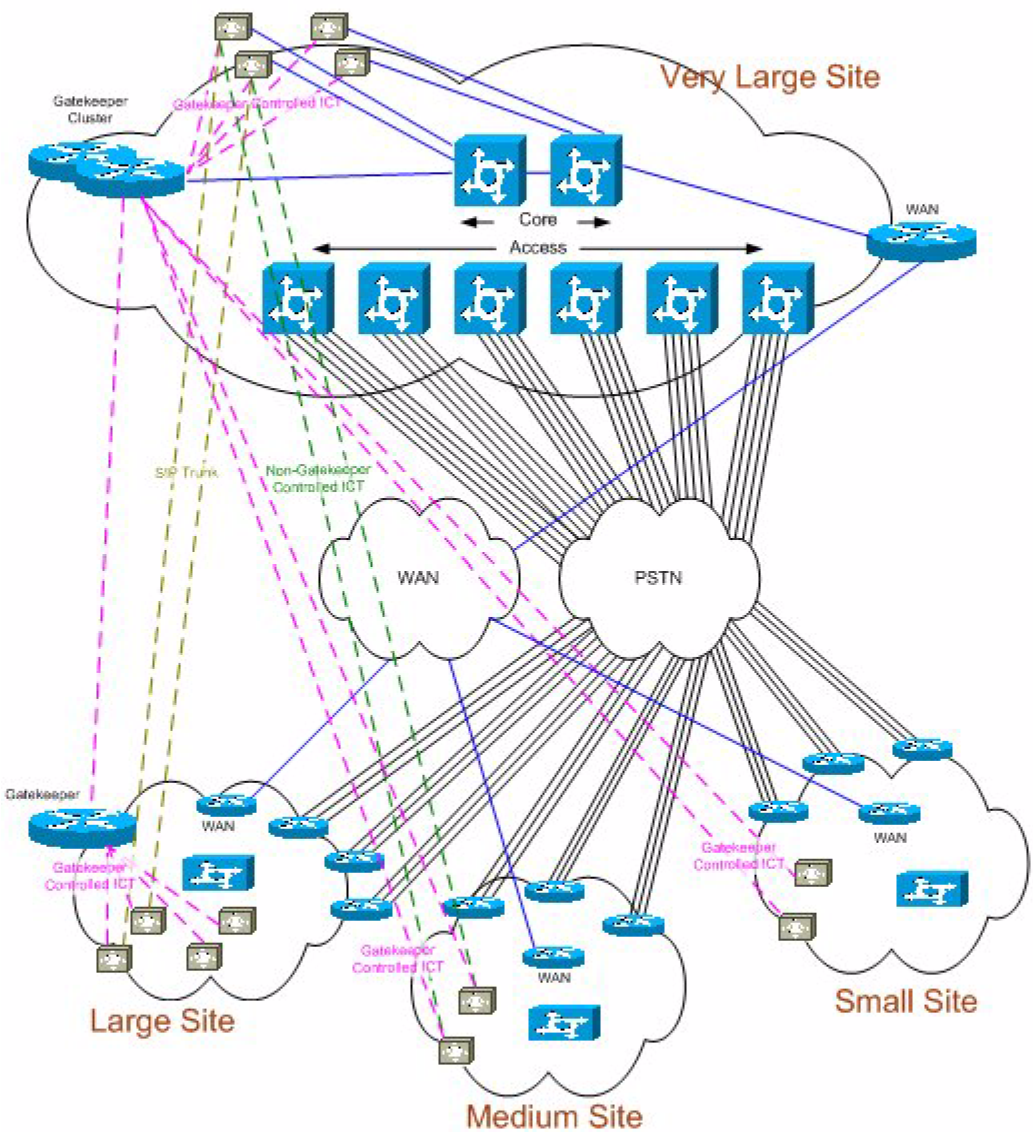
- Very Large Site Model, page 1-10
- Large Site Model, page 1-13
- Medium Site Model, page 1-15
- Small Site Model, page 1-17

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

- Separate Cisco CallManager clusters reside at each site
- Cisco Unity Unified Messaging resides at each site
- Distributed dial plan and administration
- Call admission control (CAC) based on gatekeeper CAC (to protect voice quality of IP WAN calls)

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

Figure 1-3 Multi-Site 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
Very Large Site	See the "Very Large Site Model" section on page 1-10	Can stand alone as a Single Site scenario or be one of the remote sites in the Multi-Site Distributed scenario
Large Site	See the "Large Site Model" section on page 1-13	
Medium Site	See the "Medium Site Model" section on page 1-15	
Small Site	See the "Small Site Model" section on page 1-17	
Small Site with Cisco CallManager Express	See the "Small Site with Cisco CallManager Express Model" section on page 1-19	Standalone single site
Central Site	See the "Central Site Model" section on page 1-21	Location of the Cisco CallManager cluster in the Multi-Site Centralized scenario
Remote Site A Remote Site B	See the "Remote Site Models" section on page 1-23	Remote sites in the Multi-Site Centralized scenario

Very Large Site Model

The Very Large Site model can stand alone as a Single Site scenario or, with the addition of a WAN router and gatekeepers, it can be one of the remote sites in the Multi-Site Distributed scenario. This model contains approximately 17,000 phones and approximately 150 agents.

Figure 1-4 shows the topology of the Very Large Site model.

Figure 1-4   Very Large Site Model Topology

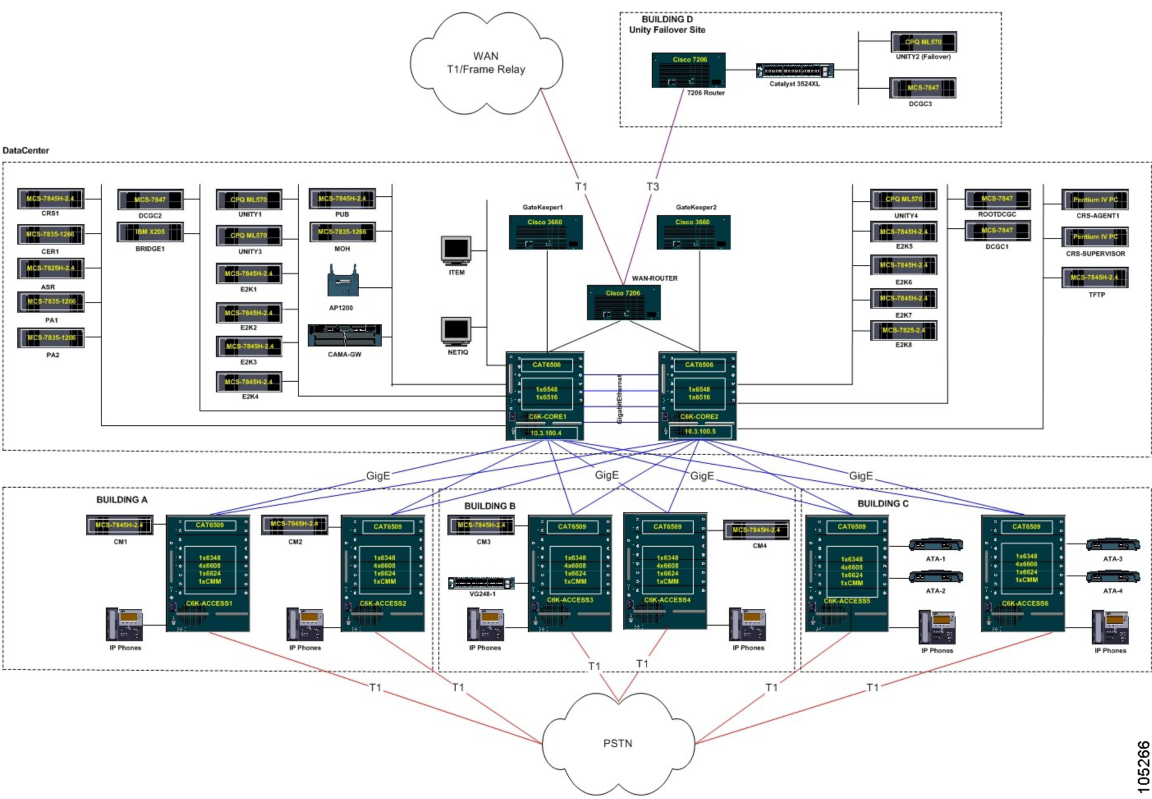


Table 1-2 lists the hardware and software components used in the Very Large Site model.

Table 1-2 Very Large Site Model Components

Component	Description	Qty.
Cisco Analog Telephone Adaptor (ATA	Cisco ATA 186 and Cisco ATA188	4
Cisco CallManager Cluster	Cisco CallManager installed on an MCS-7845H-2.4-EVV1	6
Cisco Customer Response Applications (Cisco CRA)	Cisco CRA, IPCC Express and IP IVR[1] installed on an MCS-7845H-2.4-CC1, Dedicated MCS-7825-1133 set up as an ICD Call Statistics, Recording, and Monitoring Server	1
Cisco Emergency Responder	Cisco Emergency Responder installed on an MCS-7835-1266	1
Cisco Emergency Responder CAMA[2] Gateway	CS3745 with CAMA (H.323) (NM-HD-1V and VIC2-2FXO)	1
Cisco IP Phone	Cisco IP Phone 7905G Cisco IP Phone 7910 Cisco IP Phone 7912G Cisco IP Phone 7935 Cisco IP Phone 7940G Cisco IP Phone 7960G Cisco IP Phone 7970G	50
Cisco IP/VC 3540	Cisco IP/VC 3540 Series Videoconferencing System	1
Cisco MeetingPlace	Cisco MeetingPlace Audio Server MP-8112 Cisco MeetingPlace IP Gateway installed on an MCS-7835	1
Cisco Personal Assistant	Cisco Personal Assistant installed on an MCS-7835-1266	2
Cisco Unity Unified Messaging	Cisco Unity installed on a Compaq ML570 with Quad Processor	4

Table 1-2 Very Large Site Model Components (continued)

Component	Description	Qty.
Distribution Switch	Cisco Catalyst 6509 with Supervisor II	2
Domain Controller/ Global Catalog	Installed on an MCS-7847	3
Exchange2000	Installed on an MCS-7845H-2.4-EVV1	5
Exchange2003	Installed on an MCS-7845H-2.4-ECS1	3
FXS Gateway	VG224	2
	VG248	2
	Cisco Catalyst 6624 (bundled)	4
Gatekeeper[3]	CS3745	2
ITEM Network Management	ITEM, bundle consisting of ITM 2.0 with IDU-4, GSU 2.0 with IDU-4, and WPU 2.0 with IDU-4	1
Music on Hold (MOH)	Installed on an MCS-7835-1266	1
Softphone	Cisco Softphone installed on a Pentium IV PC	1
IP Communicator	Cisco IP Communicator installed on a Pentium IV PC	1
Core Switch	Cisco Catalyst 6509 with Supervisor II	6
Core Switch MSFC	Multi-Switch Feature Card running	6
T1 Gateway	Cisco Catalyst Communication Media Module (CMM)	6
	Cisco Catalyst 6608 (bundled)	24
Video Endpoint	Cisco VT Advantage	3
	Tandberg 1000 (H.323)	1
	Tandberg 1000 (SCCP)	2
	VSX-7000 Polycom (H.323)	1
WAN Router[3]	7206 NPE400	1
Wireless Access Point	Cisco Aironet 1200 Series	1

1. IP IVR = IP Interactive Voice Response.

2. CAMA = Centralized Automatic Message Accounting.
3. Not used in a Single Site scenario. Used only if this model is part of a Multi-Site Distributed scenario.

Large Site Model

The Large Site model can stand alone as a Single Site scenario or, with the addition of a WAN router and a gatekeeper, it can be one of the remote sites in the Multi-Site Distributed scenario. This model contains approximately 3,000 phones and approximately 75 agents.

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

Figure 1-5 Large Site Model Topology

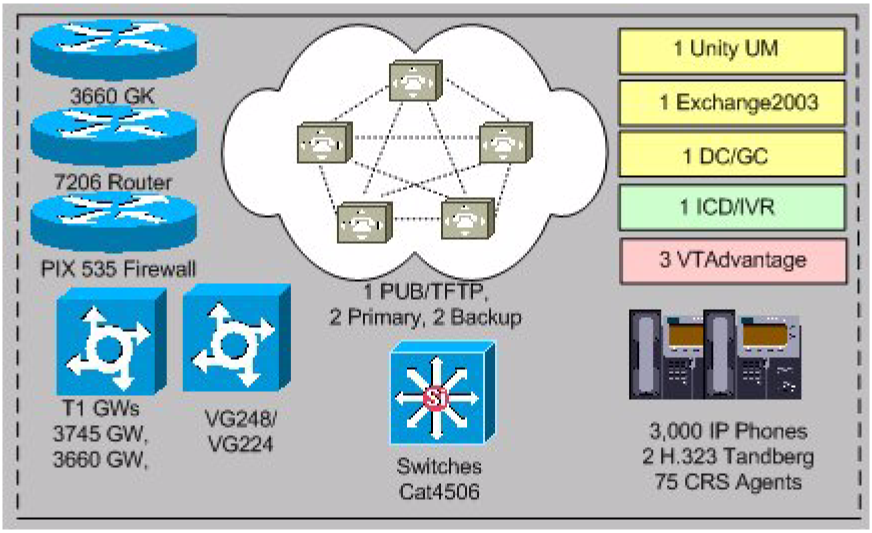


Table 1-3 lists the hardware and software components used in the Large Site model.

Table 1-3 Large Site Model Components

Component	Description	Qty.
Active Directory 2003 Domain Controller/Global Catalog	Installed on an MCS-7845H-2.4-EVV1 running the Windows 2003 Server operating system	1
Cisco CallManager Cluster	Cisco CallManager installed on an MCS-7845H-2.4-EVV1	5
Cisco Customer Response Applications (Cisco CRA)	Cisco CRA, IPCC Express and IP IVR[1] installed on an MCS-7845H-2.4-CC1	1
Cisco IP Phone	Cisco IP Phone 7940G Cisco IP Phone 7960G Cisco IP Phone 7970G	7
Cisco Unity Unified Messaging	Cisco Unity installed on a MCS-7845H-2.4-ECS1 with Quad Processor running the Windows 2003 Server operating system	1
Exchange2003	Installed on an MCS-7845H-2.4-ECS1 running the Windows 2003 Server operating system	1
FXS Gateway	VG224	2
Gatekeeper[2]	CS3660	1
NetMeeting	Windows Video Conferencing Applications (H.323)	1
Switch	Cisco Catalyst 4506 with Supervisor III	1
T1 Gateway	Cisco 3745 (H.323) (NM-HDV with VWIC-2MFT-T1)	1
	Cisco 3745 (MGCP) (NM-HDV with VWIC-2MFT-T1)	1
	Cisco 3660 (H.323) (NM-HDV with VWIC-2MFT-T1)	1
	Cisco 3660 (MGCP) (NM-HDV with VWIC-2MFT-T1)	1

Table 1-3 Large Site Model Components (continued)

Component	Description	Qty.
Video Endpoint	Cisco VT Advantage	3
	Tandberg 1000 (SCCP)	1
	VSX-7000 Polycom (H.323)	1
WAN Router	7206 NPE400	1

1. IP IVR = IP Interactive Voice Response
2. Not used in a Single Site scenario. Used only if this model is part of a Multi-Site Distributed scenario.

Medium Site Model

The Medium Site model can stand alone as a Single Site scenario or, with the addition of a WAN router, it can be one of the remote sites in the Multi-Site Distributed scenario. This model contains approximately 1,000 phones.

Figure 1-6 shows the topology of the Medium Site model.

Figure 1-6 Medium Site Model Topology

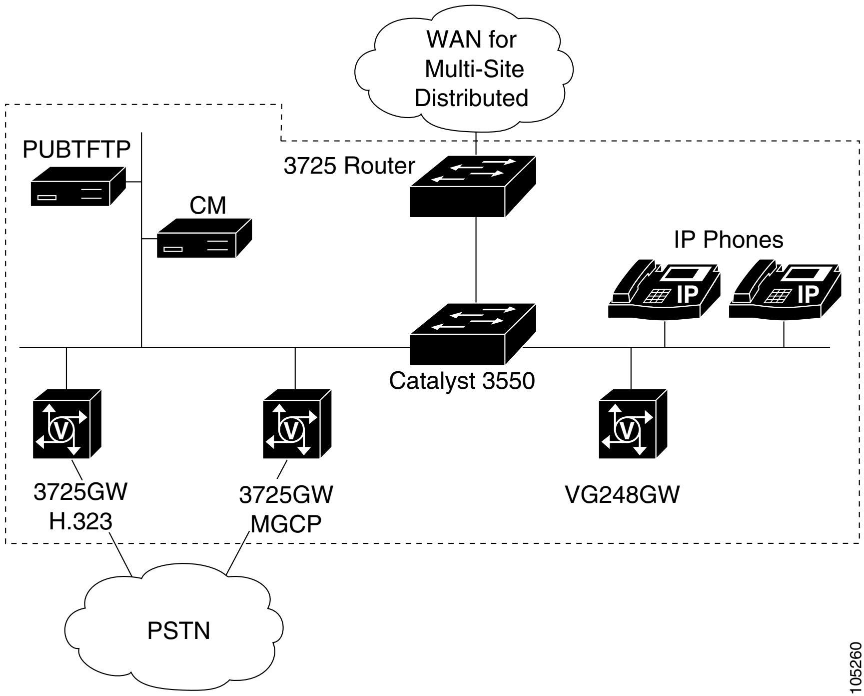


Table 1-4 lists the hardware and software components used in the Medium Site model.

Table 1-4 Medium SIte Model Components

Component	Description	Qty.
Cisco CallManager Cluster	Cisco CallManager installed on an MCS-7845H-2.4-EVV1	2
Cisco IP Phone	Cisco IP Phone 7910 Cisco IP Phone 7960G	4
FXS Gateway	VG224	2
	VG248	2
Switch	Cisco Catalyst 3550	1

Table 1-4 Medium SIte Model Components (continued)

Component	Description	Qty.
T1 Gateway	Cisco 3725 (H.323) (NM-HDV with VWIC-2MFT-T1)	1
	Cisco 3725 (MGCP) (NM-HDV with VWIC-2MFT-T1)	1
WAN Router[1]	Cisco 3725	1

1. Not used in a Single Site scenario. Used only if this model is part of a Multi-Site Distributed scenario.

Small Site Model

The Small Site model can stand alone as a Single Site scenario or, with the addition of a WAN router, it can be one of the remote sites in the Multi-Site Distributed scenario. This model contains approximately 500 phones.

Figure 1-7 shows the topology of the Small Site model.

Figure 1-7 Small Site Model Topology

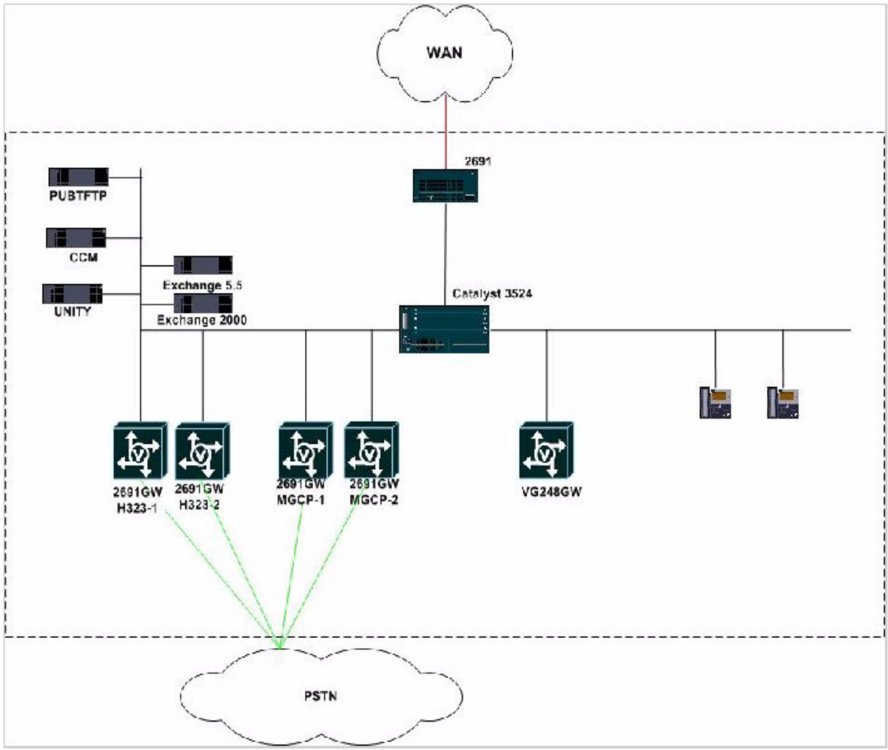


Table 1-5 lists the hardware and software components used in the Small Site model.

Table 1-5 Small Site Model Components

Component	Description	Qty.
Cisco CallManager Cluster	Cisco CallManager installed on an MCS-7845H-2.4-EVV1	2
Cisco Unity Unified Messaging	Cisco Unity installed on a Compaq MCS-7845H-2.4-ECS1	1
Domain Controller/ Global Catalog	Installed on an MCS-7845H-2.4-ECS1	1

Table 1-5 Small Site Model Components (continued)

Component	Description	Qty.
Exchange 5.5	Installed on an MCS-7845H-2.4-ECS1	1
Exchange2000	Installed on an MCS-7845H-2.4-ECS1	1
FXS Gateway	VG248	1
IP Phone	Cisco IP Phone 7910 Cisco IP Phone 7960G	4
Switch	Cisco Catalyst 3524	1
T1 Gateway	Cisco 2691 (H.323) (NM-HDV with VWIC-2MFT-T1)	1
	Cisco 2691 (MGCP) (NM-HDV with VWIC-2MFT-T1)	1
WAN Router[1]	Cisco 2691	1

1. Not used in a Single Site scenario. Used only if this model is part of a Multi-Site Distributed scenario.

Small Site with Cisco CallManager Express Model

The Small Site with Cisco CallManager Express model can stands alone as a Single Site scenario. This model includes Cisco Unity Express and contains approximately 120 phones.

Figure 1-8 shows the topology of the Small Site with Cisco CallManager Express model. This figure also shows how this model can interact with other sites through an IP-WAN and the PSTN.

Figure 1-8   Small Site with Cisco CallManager Express Model Topology

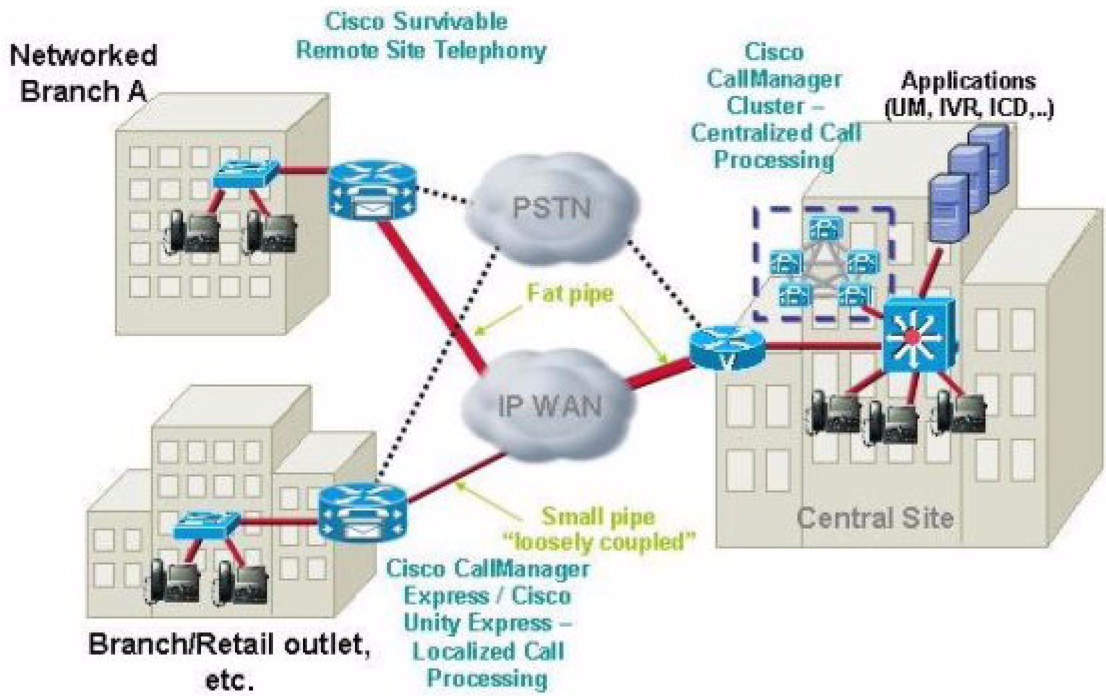


Table 1-6 lists the hardware and software components used in the Small Site with Cisco CallManager Express model.

Table 1-6   Small Site with Cisco CallManager Express Model Components

Component	Description	Qty.
Cisco CallManager Express	Cisco CallManager Express running on a 2800	1

Table 1-6 Small Site with Cisco CallManager Express Model Components (continued)

Component	Description	Qty.
Cisco CallManager Express with Cisco Unity Express	Cisco CallManager Express and Cisco Unity Express running on a 3745 with a NM-CUE card	1
	Cisco CallManager Express and Cisco Unity Express running on a 3725 with a NM-CUE card	1
	Cisco CallManager Express and Cisco Unity Express running on a 2650 with a NM-CUE card	1
	Cisco CallManager Express and Cisco Unity Express running on a 1760 with a NM-CUE card	1
Cisco IP Phone	Cisco IP Phone 7910, Cisco IP Phone 7912G Cisco IP Phone 7940G Cisco IP Phone 7960G	8
Switch	Cisco Catalyst 3550	1
T1 Gateway	Cisco 3745 (H.323) (NM-HDV with VWIC-2MFT-T1)	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 1-9 shows the topology of the Central Site model. This figure also includes topologies of typical Remote Site models.

Figure 1-9 Central Site and Remote Sites Topologies

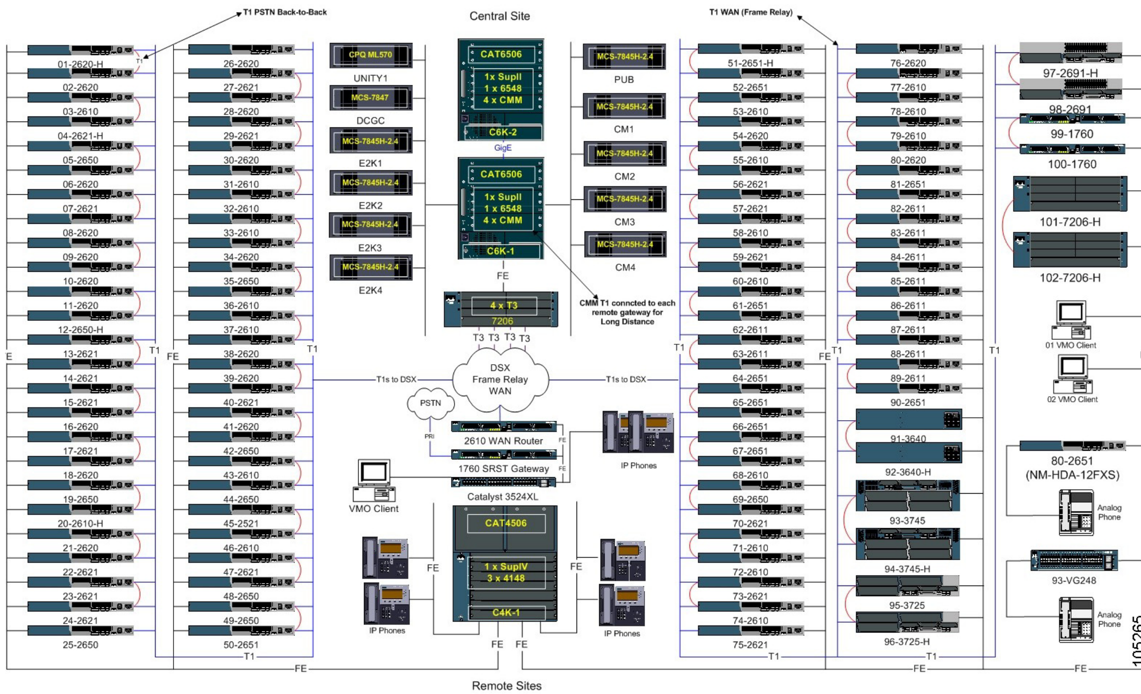


Table 1-7 lists the hardware and software components used in the Central Site model.

Table 1-7 Central Site Model Components

Component	Description	Qty.
Cisco CallManager	Cisco CallManager installed on an MCS-7845H-2.4-EVV1	6
Cisco IP Phone	Cisco IP Phone 7902G Cisco IP Phone 7905G Cisco IP Phone 7910 Cisco IP Phone 7912G Cisco IP Phone 7940G Cisco IP Phone 7960G Cisco IP Phone 7970G	26
Cisco Unity Unified Messaging	Cisco Unity installed on a Compaq MCS-7845H-2.4-ECS1	1
Domain Controller/ Global Catalog	Installed on an MCS-7845H-2.4-ECS1	1
Exchange2003	Installed on an MCS-7845H-2.4-ECS1	2
Exchange 5.5	Installed on an MCS-7845H-2.4-ECS1	2
Switch	Cisco Catalyst 6509 with Supervisor II	3
T1 Gateway	Cisco Catalyst Communication Media Module (CMM)	8
WAN Router	7206 NPE400	1
Core Router	7600	1
FlexWAN Module	Bundled	2
T1 Gateway	Cisco Catalyst 6608 (bundled)	2

Remote Site Models

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

There are two versions of the Remote Site model:

- Remote Site A—Router and gateway are on separate platforms
- Remote Site B—Router and gateway are on the same platform

Figure 1-9 shows topologies of typical Remote Site models. This figure also includes the topology of the Central Site model.

Table 1-8 lists the hardware and software components used in the Remote Site A model.

Table 1-9 lists the hardware and software components that can be used in the Remote Site B model.

Table 1-8 Remote Site A Model Components

Component	Description	Qty.
Cisco IP Phone	Cisco IP Phone 7960G	3
FXS Gateway	VG224	1
Gateway	Cisco 1760 (H.323) (VWIC-2MFT-T1)	1
Router	Cisco 2610	1
Switch	Catalyst 3524	1

Note Not all of the routers and gateways shown in Table 1-9 are used in a single Remote Site B model.

Table 1-9 Remote Site B Model Components

Component	Description	Qty.
Cisco IP Phone	Cisco IP Phone 7905G Cisco IP Phone 7910 Cisco IP Phone 7912G Cisco IP Phone 7935 Cisco IP Phone 7940G Cisco IP Phone 7960G Cisco IP Phone 7970	24
Router/Gateway	1760 (H.323) (VWIC-2MFT-T1)	1
	1760 (H.323) (VIC-VG-2FXS)	1
	2610XM (H.323) (NM-HDV with VWIC-2MFT-T1)	20
	2611XM (MGCP) (NM-HDV with VWIC-2MFT-T1)	5
	2611XM (H.323) (NM-HDV with VWIC-2MFT-T1)	5
	2620XM (H.323) (NM-HDV with VWIC-2MFT-T1)	20
	2621XM (H.323) (NM-HDV with VWIC-2MFT-T1)	20
	2650XM (H.323) (NM-HDV with VWIC-2MFT-T1)	10
	2651XM (MGCP) (NM-HDV with VWIC-2MFT-T1)	1
	2651XM (H.323) (NM-HDV with VWIC-2MFT-T1)	9
	2691 (H.323) (NM-HDV with VWIC-2MFT-T1)	2
	3725 (H.323) (NM-HDV with VWIC-2MFT-T1)	2

Table 1-9 Remote Site B Model Components (continued)

Component	Description	Qty.
Router/Gateway (continued)	3745 (MGCP) (NM-HDV with VWIC-2MFT-T1)	4
	7206VXR (H.323)	2
Switch	Cisco Catalyst 4506 with Supervisor IV	1

Cisco CallManager Configuration

This chapter provides an overview of how Cisco CallManager was set up for the Very Large Site model in IP Communications Systems Test Release 3.0 for North America 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 multiple Cisco MCS-7845H-2.4-EVV1 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 8-4.

For additional information about configuring Cisco CallManager for IP Video Telephony, see the "Configuring Cisco CallManager for Tandberg SCCP Video Endpoints" section on page 10-6.

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—Applies only to Multi-Site Centralized scenario
- Enterprise Parameters—Use default setting
- Location—Applies only to Multi-Site Centralized scenario
- SRST—Applies only to Multi-Site Centralized scenario

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 Very Large Site model:

- CM1
- CM2
- CM3
- CM4
- MOH
- Publisher
- TFTP

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, CM1 or Publisher
MAC Address	Blank
Description	Description of server For example, Cisco CallManager 1 Server or Publisher Server

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 Very Large Site model:

- CM1
- CM2
- CM3
- CM4

- MOH
- Publisher
- TFTP

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, CM1
Description	Description of the Cisco CallManager For example, Cisco CallManager 1 Server.
Starting Directory Number	21000
Ending Directory Number	22000
Partition	internal_p
External Phone Number Mask	91939XXXXX
Auto-registration Disabled on this Cisco CallManager	Unchecked
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 Very Large Site model. The four Standalone groups are for use in the Multi-Site Distributed scenario.

- CM1–CM2
- CM2–CM1
- CM3–CM4
- CM4–CM3
- Default
- Standalone–CM1
- Standalone–CM2
- Standalone–CM3
- Standalone–CM4

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 group CM1-CM2 Unchecked for all other groups
Selected Cisco CallManagers (ordered by highest priority)	For group CM1–CM2: CM1, CM2 For group CM2–CM1: CM2, CM1 For group CM3–CM4: CM3, CM4 For group CM4–CM3: CM4, CM3 For group Standalone–CM1: CM1 For group Standalone–CM2: CM2 For group Standalone–CM3: CM3 For group Standalone–CM4: CM4

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 Very Large Site model:

- Default
- Transcoder Region
- Video

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	384Kbps
Transcoder Region	G.729	384Kbps
Video Region	G.711	384Kbps

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 Very Large Site model:

- Default
- HQ–CM1–CM2
- HQ–CM2–CM1
- HQ–CM3–CM4
- HQ–CM4–CM3
- Standalone–CM1
- Standalone–CM2
- Standalone–CM3
- Standalone–CM4

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

Table 2-5 Device Pool Configuration

Field	**Setting**
Device Pool Name	Name of the device pool
Cisco CallManager Group	For HQ–CM1–CM2: CM1-CM2 For HQ–CM2–CM1: CM2-CM1 For HQ–CM3–CM4: CM3-CM4 For HQ–CM4–CM3: CM4-CM3 For Standalone–CM1: Standalone-CM1 For Standalone–CM2: Standalone-CM2 For Standalone–CM3: Standalone-CM3 For Standalone–CM4: Standalone-CM4
Date/Time Group	CMLocal
Region	For HQ– device pools: Default For Standalone– device pools: Transcoder Region
Softkey Template	Standard User
SRST Reference	Disable

Table 2-5 Device Pool Configuration (continued)

Field	Setting
Calling Search Space for Auto-registration	For HQ– device pools: internal_css For Standalone– device pools: <None>
Media Resource Group List	MRGL_1
Network Hold MOH Audio Source	Network_Hold_MOH
User Hold MOH Audio Source	SampleAudioSource
Network Locale	United States
User Locale	English United States
MLPP Indication	Default
MLPP Preemption	Default
MLPP Domain	blank

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 2-9
- Route Plan > Calling Search Space, page 2-9
- Route Plan > Route/Hunt > Line Group, page 2-10
- Route Plan > Route/Hunt > Route Group, page 2-11
- Route Plan > Route Hunt > Route/Hunt List, page 2-12
- Route Plan > Route Pattern/Hunt Pilot, page 2-13
- Route Plan > Translation Pattern, page 2-16

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 2-6 describes the partitions that were configured for the Very Large Site model.

Table 2-6 Partitions

Partition Name	Description
E911_p	Partition for CER911 CTI Route Point
E911_to_PSAP_p	Partition for E911 PSAP
Internal_p	Partition for phones with internal calling privileges
National_p	Partition for phones with national calling privileges
International_p	Partition for phones with international calling privileges
PA_p	Partition for PA CTI Route Point
PAManagedEmployee_p	Partition for PA Interceptor Ports
Unity_p	Partition for Cisco Unity Unified Messaging

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 2-7 describes the calling search spaces that were configured for the Very Large Site model.

Table 2-7 Calling Search Spaces

Calling Search Space Name	Description	Selected Partitions
E911_CSS	Calling Search Space for E911	E911_p, E911_to_PSAP_p
E911_to PSAP_CSS	Calling Search Space for E911 to PSAP	E911_to_PSAP_p
Internal_CSS	Calling Search Space for internal calling privileges	Internal_p
National_CSS	Calling Search Space for internal, national, PA, and Unity calling privileges	National_p
International_CSS	Calling Search Space for internal, national, international PA, and Unity calling privileges	Internal_p, National_p, International_p, PA_p, and Unity_p
PA_CSS	Calling Search Space for PA	PAManagedEmployee_p, National_p
Unity_CSS	Calling Search Space for Unity	Internal_p, PAManagedEmployee_p, Unity_p

Route Plan > Route/Hunt > Line Group

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

Table 2-8 shows one of the line groups that were configured for the Very Large Site model. Fourteen other line groups were configured but are not shown in this table.

Table 2-8 Line Groups

Field	Setting
Line Group Name	LG22100
RNA Reversion Timeout	10
Distribution Algorithm	Top Down
No Answer	Try next member; then, try next group in Hunt List
Busy	Try next member; then, try next group in Hunt List
Not Available	Try next member; then, try next group in Hunt List
Route Partition	<None>
Available DN/Route Partitions	*Includes a list of DNs or route partitions at your site*
Selected DN/Route Partitions	*Includes a list of DNs or route partitions at your site*

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 2-9 shows four of the route groups that were configured for the Very Large Site model. Approximately 120 other route groups were configured but are not shown in this table.

Table 2-9 Route Groups

Route Group Name	Route Group Members
E911_Call_to_PSAP_RG	Device=S0/DS1-0@SDA222222222201, Port=All, Order=1 Device=S0/DS1-0@SDA222222222202, Port=All, Order=2
Access_1_port_4_4_dms_trunk56	Device=S0/DS1-0@SDA0002FCE1A455, Port=All, Order=1
Access_1_port_9_1_Nortel_pbx	Device= S1/DS1-1@ACCESS1-CMM1.cisco.com CMM1.cisco.com, Port=All, Order=1
GK_ICT_RG[1]	Device=ict_to_cm1, Port=All, Order=1 Device=ict_to_cm2, Port=All, Order=2 Device=ict_to_cm3, Port=All, Order=3 Device=ict_to_cm4, Port=All, Order=4

1. Required only for Multi-Site Distributed scenario.

Route Plan > Route Hunt > Route/Hunt List

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

Table 2-10 describes five of the route/hunt lists that were configured for the Very Large Site model. Fifty-eight other route/hunt lists were configured but are not shown in this table.

Table 2-10 Route/Hunt Lists

Route/Hunt List Name	Description	Cisco CallManager Group	Selected Route Groups
E911call_to_PSAP_RL	Route List for E911 calls to PSAP	HQ-CM1-CM2	E911_Call_to_PSAP_RG
DMS_trunk56	Route List for DMS100 Trunk 56	HQ-CM2-CM1	Access_1_port_4_4_dms_ trunk56
Nortel_pbx	Route List for Nortel PBX	HQ-CM3-CM4	Access_1_port_9_1_ Nortel_pbx
GK_ICT_RL	Route List for Gatekeeper ICT	HQ-CM4-CM3	GK_ICT_RG
HL22201	Hunt group for voice messaging system line group	HQ-CM1-CM2	LG22100 LG22201

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.

Seventy route pattern/hunt pilots were configured for the Very Large Site model. This section describes the following route patterns/hunt pilots:

- 911
- DMS
- PBX
- Intercluster trunk
- Voice Messaging System

Table 2-11 describes the settings in the Route Pattern Configuration page for these selected route patterns.

Table 2-11 Route Pattern/Hunt Pilot Configuration

Field	911 Route Pattern/Hunt Pilot Settings	DMS Route Pattern/Hunt Pilot Settings	PBX Route Pattern/Hunt Pilot Settings	Intercluster Trunk Route Pattern/Hunt Pilot Settings	Voice Messaging System Route Pattern/Hunt Pilot Settings
Route Pattern	10.911	9.@	9.405x	8.xxxxxxx	22201
Partition	E911_p	International_p	National_p	Internal_p	Unity_p
Description	Route Pattern for E911	Route Pattern to DMS Trunk56	Route Pattern to Nortel PBX	Route Pattern for GK controlled ICT calls	Hunt pilot for voice messaging system
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>
MLPP Precedence	Default	Default	Default	Default	Default
Gateway or Route/Hunt List	E911_call_to_PSAP_RL	DMS_trunk56	Nortel_pbx_RL	GK_ICT_RL	HL22201
Route Option	Route This Pattern	Route This Pattern	Route This Pattern	Route This Pattern	Route This Pattern
Provide Outside Dial Tone	Checked	Checked	Checked	Checked	Unchecked
Allow Overlap Sending	Unchecked	Unchecked	Unchecked	Unchecked	Checked
Urgent Priority	Unchecked	Unchecked	Unchecked	Unchecked	Unchecked
Use Calling Party’s External Phone Number Mask	Checked	Unchecked	Unchecked	Checked	Unchecked

Table 2-11 Route Pattern/Hunt Pilot Configuration (continued)

Field	911 Route Pattern/Hunt Pilot Settings	DMS Route Pattern/Hunt Pilot Settings	PBX Route Pattern/Hunt Pilot Settings	Intercluster Trunk Route Pattern/Hunt Pilot Settings	Voice Messaging System Route Pattern/Hunt Pilot Settings
Calling Party Transform Mask	6013929911	91939xxxxx	blank	919392xxxx	blank
Prefix Digits (Outgoing Calls)	blank	blank	blank	blank	blank
Calling Line ID Presentation	Default	Default	Default	Default	Default
Calling Name Presentation	Default	Default	Default	Default	Default
Connected Line ID Presentation	Default	Default	Default	Default	Default
Connected Name Presentation	Default	Default	Default	Default	Default
Discard Digits	PreDot	PreAt	PreDot	PreDot	<None>
Called Party Transform Mask	9194725016	blank	blank	601xxxxxxx	blank
Prefix Digits (Outgoing Calls)	blank	blank	blank	blank	blank
ISDN Network-Specific Facilities Information Element fields	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 2-12 describes the settings in the Translation Pattern Configuration page for two of the translation patterns that were configured for the Very Large Site model. Thirteen other translation patterns were configured but are not shown in this table.

Table 2-12 Translation Pattern Configuration

Field	24xxx Translation Pattern Settings	9.11 Translation Pattern Settings
Translation Pattern	24xxx	9.11
Partition	National_p	Internal_p
Description	PA-Translation Pattern-24XXX Route Point	Translation Pattern for 911 calls
Numbering Plan	North American Numbering Plan	North American Numbering Plan
Route Filter	<None>	<None>
MLPP Precedence	Default	Default
Calling Search Space	PA_CSS	E911_CSS
Route Option	Route This Pattern	Route This Pattern
Provide Outside Dial Tone	Checked	Checked
Use Calling Party's External Phone Number Mask	Checked	Unchecked
Calling Party Transform Mask	919392xxxx	blank
Prefix Digits (Outgoing Calls)	blank	blank
Calling Party Presentation	Default	Default

Table 2-12 Translation Pattern Configuration (continued)

Field	24xxx Translation Pattern Settings	9.11 Translation Pattern Settings
Discard Digits	PreDot	PreDot
Called Party Transform Mask	601xxxxxxx	911
Prefix Digits (Outgoing Calls)	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 2-18
- Service > Media Resource > Media Termination Point, page 2-18
- Service > Media Resource > Music On Hold Audio Source, page 2-19
- Service > Media Resource > Music On Hold Server, page 2-20
- Service > Media Resource > Transcoder, page 2-20
- Service > Media Resource > Media Resource Group, page 2-21
- Service > Media Resource > Media Resource Group List, page 2-22
- Service > Service Parameters, page 2-23

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 2-13 describes the settings in the Conference Bridge Configuration page for one software conference bridge and one hardware conference bridge that were configured for the Very Large Site model. Five other software conference bridges and 26 other hardware conference bridges were configured but are not shown in this table.

Table 2-13 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	CM1	—
MAC Address	—	00027E38FF9
Conference Bridge Name	CFB_CM1	—
Description	SW Conference Bridge in CM1	Catalyst 6608 Access 1 port 7/5
Device Pool	HQ-CM1-CM2	HQ-CM1-CM2
Location	<None>	<None>
Special Load Information	—	blank

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 2-14 describes the settings in the Media Termination Point Configuration page for one media termination point that was configured for the Very Large Site model. Five other media termination points were configured but are not shown in this table.

Table 2-14 Media Termination Point Configuration

Field	Setting
Host Server	CM1
Media Termination Point Name	MTP_CM1
Description	Software MTP in CM1
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.

Two MOH audio sources were configured for the Very Large Site model.

Table 2-15 describes how the MOH audio sources were configured in the Music On Hold (MOH) Audio Source Configuration page.

Table 2-15 Music On Hold Audio Source Configuration

Field	MOH Audio Source 1 Settings	MOH Audio Source 2 Settings
MOH Audio Source File	SampleAudioSource	SampleAudioSource
MOH Audio Source Name	SampleAudioSource	MOH_MOH
Play Continuously	Checked	Checked
Allow Multicasting	Checked	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 2-16 describes the settings in the Music On Hold (MOH) Server Configuration page.

Table 2-16 Music On Hold Server Configuration

Field	Setting
Host Server	MOH
Music on Hold Server Name	MOH_MOH
Description	Music on Hold Server
Device Pool	HQ-CM1-CM2
Location	<None>
Maximum Half Duplex Streams	250
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
Sample Audio Source	2 Max Hops
MOH_MOH	2 Max Hops

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 Very Large Site model. Twenty-two other transcoders were configured but are not shown in this table.

Table 2-17 describes how two of the transcoders were configured in the Transcoder Configuration page.

Table 2-17 Transcoder Configuration

Field	Transcoder 1 Settings	Transcoder 2 Settings
Transcoder Type	Cisco Media Termination Point Hardware	Cisco Media Termination Point Hardware
Description	Hardware MTP	Hardware MTP
MAC Address	00027E38FF98	0002FCE1D030
Device Pool	HQ-CM1-CM2	HQ-CM3-CM4
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 2-18 describes the settings in the Media Resource Group Configuration page for the following media resource groups. Two other media resource groups were configured for the Very Large Site model but are not shown in this table.

- MRG_HWCFB
- MRG_MOH
- Video-MRG
- MRG_SWMTP

Table 2-18 Media Resource Group Configuration

Field	MRG_HWCFB Settings	MRG_MOH Settings	Video-MRG Settings	MRG_SWMTP Settings
Media Resource Group Name	MRG_HWCFB	MRG_MOH	Video-MRG	MRG_SWMTP
Description	Media Resource Group for Hardware Conference Bridge	Media Resource Group for MOH	Video MRG	Media Resource Group for Software MTP
Selected Media Resource	CFB00027E38FF9A (CFB)[1]	MOH_MOH (MOH) [Multicast]	VCB0003D6001C44 (CFB)	MTP_CM1 (MTP)
Use Multicast for MOH Audio	Not checked	Checked	Not checked	Not checked

1. Additional Selected Medial Resources were configured but are not shown in this example.

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 2-19 describes the settings in the Media Resource Group List Configuration page.

Table 2-19 Media Resource Group List Configuration

Field	Setting
Media Resource Group Name	MRGL_HQ
Selected Media Resource Groups	MRG_HWCFB MRG_HWMTP MRG_SWCFB MRG_MTP MRG_MOH

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 2-20 describes the settings in the Music On Hold (MOH) Server Configuration page for the CM1, CM2, CM3, and CM4 servers.

Table 2-20 Server Parameter Configuration Settings for Each Server

Service for CM1, CM2, CM3, and CM4 Servers	Settings
Cisco CallManager	All default settings except the following: • Call Diagnostics Enabled—True • CDR Enabled Flag—True • CDR Log Calls With Zero Duration Flag—True • SDL Trace File Path—F: drive • SDL Trace Total Number of Files—9000 • Maximum MeetMe Conference Unicast—10 • Maximum number of registered devices—7500 • Maximum Ad Hoc Conference—6 • Multiple Tenant MWI Modes—True • Forward No Answer Timer—18 • Automated Alternate Routing Enable—True • T302 Timer—4000 • Enable All User Search—False
Cisco CTIManager	default settings
Cisco Database Layer Monitor	default settings
Cisco Extended Functions	default settings
Cisco Extension Mobility Logout	All default settings except Call Diagnostics Enabled: True
Cisco IP Voice Media Streaming App	default settings

Table 2-20 Server Parameter Configuration Settings for Each Server

Service for CM1, CM2, CM3, and CM4 Servers	Settings
Cisco Messaging Interface	default settings
Cisco RIS Data Collector	default settings
Cisco Telephony Call Dispatcher	default settings

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 > Call Park, page 2-25
- Feature > Cisco IP Phone Services, page 2-26
- Feature > Voice Mail > Cisco Voice Mail Port, page 2-26
- Feature > Voice Mail > Message Waiting, page 2-27
- Feature > Voice Mail > Voice Mail Pilot, page 2-28
- Feature > Voice Mail > Voice Mail Profile, page 2-29

The following sections do not discuss these Feature menu options:

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

Feature > Call Park

To access the Cisco CallManager Administration web pages for adding and configuring call park numbers, choose **Feature > Call Park** from the Cisco CallManager Administration application.

Table 2-21 describes the settings in the Call Park Configuration page.

Table 2-21 Call Park Configuration

Field	Setting
Call Park Number/Range	55551
Description	Call park for CM1
Partition	Internal_p
Call Manager	CM1

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.

Table 2-22 describes the settings in the Cisco IP Phone Services Configuration page.

Table 2-22 Cisco IP Phone Services Configuration

Field	Setting
Service Name	Login Extension Mobility
Service Description	For Extension Mobility
Service URL	http://1.1.1.1/emapp/EMAppServlet?device=#DEVICENAME#
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 2-23 describes the settings in the Cisco Voice Mail Port Configuration page for one Cisco voice mail port that was configured for the Very Large Site model. Six hundred and eight other Cisco voice mail ports were configured but are not shown in this table.

Table 2-23 Cisco Voice Mail Port Configuration

Field	Setting
Port name	CiscoUM1-VI1
Description	UNITY1 Voicemail
Device Pool	HQ-CM1-CM2
Calling Search Space	unity_css
AAR Calling Search Space	<None>
Location	<None>
Directory Number	22100
Partition	Unity_p
Calling Search Space	Unity_css
AAR Group	<None>
Display (Internal Caller ID)	Unity1Voicemail
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 2-24 describes the settings in the Message Waiting Configuration page for two of the message waiting numbers that were configured for the Very Large Site model. Ten other message waiting numbers were configured but are not shown in this table.

Table 2-24 Message Waiting Configuration

Field	Message Waiting Number 1 Settings	Message Waiting Number 2 Settings
Message Waiting Number	22198	22199
Description	MWI number to turn the light on	MWI number to turn the light off
Message Waiting Indicator	On	Off
Partition	Unity_p	Unity_p
Calling Search Space	Unity_css	Unity_css

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 2-25 describes the settings in the Voice Mail Pilot Configuration page for one of the voice mail pilots that was configured for the Very Large Site model. Four other Cisco voice mail pilots were configured but are not shown in this table.

Table 2-25 Voice Mail Pilot Configuration

Field	Setting
Voice Mail Pilot Number	22100
Description	Unity1 Pilot Number
Calling Search Space	Unity_css
Make this the default Voice Mail Pilot for the system	Checked

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 2-26 describes the settings in the Voice Mail Profile Configuration page for one of the voice mail profiles that was configured for the Very Large Site model. Nine other Cisco voice mail profiles were configured but are not shown in this table.

Table 2-26 Voice Mail Profile Configuration

Field	Setting
Voice Mail Profile Name	Unity1
Description	Voice Mail Profile for Unity1
Voice mail Pilot	22100/unity_css
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 2-30
- Device > Gatekeeper, page 2-33
- Device > Gateway, page 2-33
- Device > Phone, page 2-39
- Device > Trunk, page 2-44

- Device > Device Settings > Device Profile, page 2-47

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 2-27 describes the settings on the CTI Route Point Configuration page for three CTI route points that were configured for the Very Large Site model. Table 2-28 shows how one of the directory numbers was configured for each example route point.

Fourteen 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 2-27 CTI Route Point Configuration

Field	CTI Route Point 1 Setting	CTI Route Point 2 Setting	CTI Route Point 3 Setting
Device Name	AA	PA24XXX	PAMainNumber
Description	x23110 Auto Attendant	PA24XXX	PAMainNumber
Device Pool	HQ-CM1-CM2	HQ-CM1-CM2	HQ-CM1-CM2
Calling Search Space	National_css	PA_CSS	PA_CSS
Location	<None>	<None>	<None>
Media Resource Group List	<None>	<None>	<None>

Table 2-27 CTI Route Point Configuration (continued)

Field	CTI Route Point 1 Setting	CTI Route Point 2 Setting	CTI Route Point 3 Setting
User Hold Audio Source	<None>	<None>	<None>
Network Hold Audio Source	<None>	<None>	<None>

Table 2-28 Directory Number Configuration for CTI Route Points

Field	CTI Route Point 1 Setting	CTI Route Point 2 Setting	CTI Route Point 3 Setting
Directory Number	23110	24XXX	24000
Partition	Internal_p	PA	National_p
Voice Mail Profile	<None>	<None>	<None>
Calling Search Space	National_css	PA_CSS	PA_CSS
AAR Group	<None>	<None>	<None>
User Hold Audio Source	<None>	<None>	<None>
Network Hold Audio Source	<None>	<None>	<None>
Auto Answer	Not available on this device	Not available on this device	Not available on this device
Forward All	default settings	default settings	default settings
Forward Busy	default settings	default settings	default settings, except Calling Search Space: PA_CSS
Forward No Answer	default settings	default settings	default settings, except Calling Search Space: PA_CSS
Forward On Failure	default settings	default settings	default settings

Table 2-28 Directory Number Configuration for CTI Route Points (continued)

Field	CTI Route Point 1 Setting	CTI Route Point 2 Setting	CTI Route Point 3 Setting
No Answer Ring Duration	Not available on this device	Not available on this device	Not available on this device
Call Pickup Group	<None>	<None>	<None>
Target (Destination)	blank	blank	blank
Calling Search Space	<None>	<None>	<None>
No Answer Ring Duration	blank	blank	blank
Display (Internal Caller ID)	blank	blank	blank
Line Text Label	Not available on this device	Not available on this device	Not available on this device
External Phone Number Mask	blank	blank	blank
Line Text Label	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
Ring Setting (Phone Idle)	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
Maximum Number of Calls	5000	5000	5000
Busy Trigger	4500	4500	4500
Caller Name	Checked	Checked	Checked
Redirected Number	Unchecked	Unchecked	Unchecked
Caller Number	Unchecked	Unchecked	Unchecked

Table 2-28 Directory Number Configuration for CTI Route Points (continued)

Field	CTI Route Point 1 Setting	CTI Route Point 2 Setting	CTI Route Point 3 Setting
Dialed Number	Unchecked	Unchecked	Unchecked
Character Set	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 primary gatekeeper and an alternate gatekeeper were configured for the Very Large Site model. Table 2-29 describes the settings in the Gatekeeper Configuration page for these gatekeepers.

Table 2-29 Gatekeeper Configuration

Field	Primary Gatekeeper Settings	Alternate Gatekeeper Settings
Host Name/IP Address	10.3.100.51	10.3.100.52
Description	Primary Gatekeeper	Alternate 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 Very Large Site model. One hundred and sixty-five other gateways were configured but are not described in this section.

Table 2-30 describes how the 6624 analog gateway was configured in the Gateway Configuration page. Table 2-31 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 2-32 describes how the CMM digital gateway was configured in the Gateway Configuration page. Table 2-33 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 2-30 Analog Gateway Configuration

Field	Setting
MAC Address	001007F338E5
Description	Analog 6624
Device Pool	HQ–CM1–CM2
Load Information	A00203030017
Network Locale	United States
Location	<None>
AAR Group	<None>
Calling Search Space	national_css
AAR Calling Search Space	<None>
Media Resource Group List	mrgl
Network Hold Audio Source	MOH_MOH

Table 2-30 Analog Gateway Configuration (continued)

Field	Setting
Port Selection Order	Top Down
MLPP Domain	blank
MLPP Indication	blank
MLPP Preemption	Not available on this device
SNMP Community String	Public
Disable SNMP Set operations	Unchecked

Table 2-31 Directory Number Configuration for 6624 Analog Gateway

Field	Setting
Directory Number	29948
Partition	internal_p
Voice Mail Profile	<None>
Calling Search Space	national_css
AAR Group	<None>
Auto Answer	Not available on this device
Network Hold Audio Source	<None>
Forward All	default settings
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)	blank

Table 2-31 Directory Number Configuration for 6624 Analog Gateway (continued)

Field	Setting
Line Text Label	Not available on this device
External Phone Number Mask	blank
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	Unchecked
Character Set	Western European (Latin1)

Table 2-32 Digital Gateway Configuration

Field	Setting
Domain Name	ACCESS1-CMM1.cisco.com
Description	Digital-CMM
Cisco CallManager Group	12
Module in Slot 1	WS-X6600
Subunit	WS-X6600-6T1
Module in Slot 2	<None>
Module in Slot 3	<None>
Module in Slot 4	<None>
Switchback Timing	Graceful
Global ISDN Switch Type	DMS-100

Table 2-32 Digital Gateway Configuration (continued)

Field	Setting
Switchback uptime-delay (min)	10
Switchback schedule (hh:mm)	12:00
Fax mode	Fax Relay

Table 2-33 End Point Configuration for Digital-CMM Digital Gateway

Field	Setting
End-Point Name	S1/DS1-0@ACCESS1-CMM1.cisco.com
Description	S1/DS1-0@ACCESS1-CMM1.cisco.com
Device Pool	HQ–CM2–CM1
Network Locale	United States
Media Resource Group List	<None>
Location	<None>
AAR Group	<None>
Load Information	blank
MLPP Domain	blank
MLPP Indication	On
MLPP Preemption	Forceful
PRI Protocol Type	PRI DMS-100
Protocol Side	User
Channel Selection Order	Bottom Up
Channel IE Type	Use Number when 1B
PCM Type	u-law
Delay for first restart (1/8 sec ticks)	32
Delay between restarts (1/8 sec ticks)	4
Inhibit restarts at PRI initialization	Unchecked

Table 2-33 End Point Configuration for Digital-CMM Digital Gateway (continued)

Field	Setting
Enable status poll	Checked
Significant Digits	All
Calling Search Space	internal_css
AAR Calling Search Space	<None>
Prefix DN	blank
Calling Line ID Presentation	Allowed
Calling Party Selection	Originator
Called party IE number type unknown	National
Calling party IE number type unknown	National
Called Numbering Plan	ISDN
Calling Numbering Plan	ISDN
Number of digits to strip	0
Caller ID DN	blank
SMDI Base Port	1
Display IE Delivery	Checked
Redirecting Number IE Delivery - Outbound	Unchecked
Redirecting Number IE Delivery - Inbound	Unchecked
Send Extra Leading Character In DisplayIE	Checked
Setup non-ISDN Progress Indicator IE Enable	Unchecked
MCDN Channel Number Extension Bit Set to Zero	Unchecked
Send Calling Name In Facility IE	Unchecked

Table 2-33 End Point Configuration for Digital-CMM Digital Gateway (continued)

Field	Setting
Interface Identifier Present	Checked
Interface Identifier Value	0
Connected Line ID Presentation (QSIG Inbound Call)	Default
Connected PBX Model	<None>s
Line Coding	B8ZS
Framing	ESF
Clock	External
Input Gain (-6..14 db)	0
Output Attenuation (-6..14 db)	0
Echo Cancellation Enable	Enable
Echo Cancel Coverage (ms)	64

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 Very Large Site model. More than 18,000 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 2-34 describes how the Cisco IP Phone 7960 device and the ATA 186 device were configured in the Phone Configuration page. Table 2-35 describes how one of the directory numbers for each of these devices was configured in the Directory Number Configuration page for that device.

Table 2-34 Phone Configuration

Field	Cisco IP Phone 7960 Setting	ATA 186 Setting
MAC Address	00097CEC8E33	000AB7447F84
Description	Cisco IP Phone 7960	ATA 186
Owner User ID	blank	blank
Device Pool	HQ-CM4-CM3	HQ-CM1-CM2
Calling Search Space	Internal_css	National_css
AAR Calling Search Space	<None>	<None>
Media Resource Group List	mrgl	mrgl
User Hold Audio Source	MOH_MOH	MOH_MOH
Network Hold Audio Source	MOH_MOH	MOH_MOH
Location	<None>	<None>
User Locale	English United States	<None>
Network Locale	United States	<None>
Device Security Mode	Use System Default	—
Built In Bridge	Default	—
Privacy	Default	—
Retry Video Call as Audio	Checked	—
Phone Button Template	Standard 7960	Standard ATA 186
Softkey Template	Standard User	—
Module 1	<None>	—
Module 2	<None>	—
Phone Load Name	blank	blank
Module 1 Load Name	blank	—
Module 2 Load Name	blank	—

Table 2-34 Phone Configuration (continued)

Field	Cisco IP Phone 7960 Setting	ATA 186 Setting
Information	blank	—
Directory	blank	—
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	—

Table 2-34 Phone Configuration (continued)

Field	Cisco IP Phone 7960 Setting	ATA 186 Setting
Auto Line Select	Disabled	—
Web Access	Enabled	—

Table 2-35 Directory Number Configuration for Phone and ATA

Field	Cisco IP Phone 7960 Setting	ATA 186 Setting
Directory Number	29009	29901
Partition	Internal_p	Internal_p
Voice Mail Profile	unity1	unity1
Calling Search Space	National_css	National_css
AAR Group	<None>	<None>
User Hold Audio Source	MOH_MOH	MOH_MOH
Network Hold Audio Source	MOH_MOH	MOH_MOH
Auto Answer	Auto Answer Off	Not available on this device
Forward All	Voice Mail: Unchecked Destination: blank Calling Search Space: <None>	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>

Table 2-35 Directory Number Configuration for Phone and ATA (continued)

Field	Cisco IP Phone 7960 Setting	ATA 186 Setting
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>
No Answer Ring Duration	blank	blank
Display (Internal Caller ID)	39-29009	blank
Line Text Label	39-29009	blank
External Phone Number Mask	9193929009	blank
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	4	2
Busy Trigger	2	2
Caller Name	Checked	Checked
Redirected Number	Unchecked	Unchecked
Caller Number	Unchecked	Unchecked

Table 2-35 Directory Number Configuration for Phone and ATA (continued)

Field	Cisco IP Phone 7960 Setting	ATA 186 Setting
Dialed Number	Unchecked	Unchecked
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.

The following four trunks were configured for the Very Large Site model. Each trunk was added with the characteristics shown. Eleven other trunks were configured but are not shown in this section.

- ICT-CM
  - Product: Intercluster Trunk (Non-Gatekeeper Controlled)
  - Device Protocol: Intercluster Trunk
- ICT_to_CM1
  - Product: Intercluster Trunk (Gatekeeper Controlled)
  - Device Protocol: Intercluster Trunk
- SIP-Trunk
  - Product: SIP Trunk
  - Device Protocol: SIP
- CME
  - Product: Intercluster Trunk (Gatekeeper Controlled)
  - Device Protocol: Intercluster Trunk

Table 2-36 describes the settings in the Trunk Configuration page for the trunks that were configured for the Very Large Site model.

Table 2-36 Trunk Configuration

Field	Trunk 1 Settings	Trunk 2 Settings	Trunk 3 Settings	Trunk 4 Settings
Device Name	ICT-CM	ICT_to_CM1	SIP-Trunk	CME
Description	Non-GK-ICT connected to PUBTFTP and CM1 at another site	ICT_to_CM	SIP trunk Very Large site to Large site	3745 CME
Device Pool	HQ-CM1-CM2	Standalone_ cm1	HQ-12-sip-soft-mt p	Standalone_ cm1
Media Resource Group List	mrgl	VIDEO-MRG_List	mrgl	mrgl
Location	<None>	Very Large Site	<None>	<None>
AAR Group	<None>	<None>	<None>	<None>
Media Termination Point Required	Unchecked	Unchecked	Checked	Checked
Destination Address	—	—	10.3.31.11	—
Destination Address is an SRV	—	—	Unchecked	—
Destination Port	—	—	5060	—
Incoming Port	—	—	5063	—
Outgoing Transport Type	—	—	TCP	—
Preferred Originating Codec	—	—	711ulaw	—
Retry Video Call as Audio	Checked	Checked	—	Checked
Significant Digits	All	All	5	5
Connected Line ID Presentation	—	—	Allowed	—

Table 2-36 Trunk Configuration (continued)

Field	Trunk 1 Settings	Trunk 2 Settings	Trunk 3 Settings	Trunk 4 Settings
Connected Name Presentation	—	—	Allowed	—
Calling Search Space	Internal_css	International_ css	National_css	International_ css
AAR Calling Search Space	<None>	<None>	<None>	<None>
Prefix DN	blank	blank	blank	blank
Redirecting Number IE Delivery - Inbound	Checked	Checked	Unchecked	Checked
Calling Party Selection	Originator	Originator	Originator	Originator
Calling Line ID Presentation	Default	Default	Allowed	Allowed
Calling Name Presentation	—	—	Allowed	—
Called party IE number type unknown	Cisco CallManager	Cisco CallManager	—	Cisco CallManager
Calling party IE number type unknown	Cisco CallManager	Cisco CallManager	—	Cisco CallManager
Called Numbering Plan	Cisco CallManager	Cisco CallManager	—	Cisco CallManager
Calling Numbering Plan	Cisco CallManager	Cisco CallManager	—	Cisco CallManager
Caller ID DN	blank	blank	blank	blank
Caller Name	—	—	blank	—

Table 2-36 Trunk Configuration (continued)

Field	Trunk 1 Settings	Trunk 2 Settings	Trunk 3 Settings	Trunk 4 Settings
Redirecting Number Delivery - Outbound	—	—	Unchecked	—
Display IE Delivery	checked	checked	—	checked
Redirecting Number IE Delivery - Outbound	checked	checked	—	checked
Gatekeeper Name	—	10.3.100.51	—	10.3.100.51
Terminal Type	—	Gateway	—	Gateway
Technology Prefix	—	1#*	—	1#*
Zone	—	GK1	—	GK1
MLPP Domain	blank	blank	blank	blank
MLPP Indication	Not available on this device	Not available on this device	Not available on this device	Not available on this device
MLPP Preemption	Not available on this device	Not available on this device	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 2-37 describes the settings in the Device Profile page for one of the device profiles that was configured for the Very Large Site model. Table 2-38 shows how one of the directory numbers was configured for the example device profile.

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

Table 2-37 Device Profile Configuration

Field	Setting
User Device Profile Name	em49903
Description	Extension Mobility for 49903
User Hold Audio Source	<None>
User Locale	<None>
Phone Button Template	5 line 7960
Softkey Template	<None>
Module 1	<None>
Module 2	<None>
MLPP Domain	blank
MLPP Indication	Default
MLPP Preemption	Default
Login User ID	blank

Table 2-38 Directory Number Configuration for em49903 Device Profile

Field	Setting
Directory Number	49903
Partition	Internal_p
Voice Mail Profile	<None>
Calling Search Space	national_css
AAR Group	<None>
User Hold Audio Source	<None>
Network Hold Audio Source	<None>
Auto Answer	Auto Answer Off

Table 2-38 Directory Number Configuration for em49903 Device Profile (continued)

Field	Setting
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: 22100 Calling Search Space: internal_css
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 49903
Line Text Label	em 49903
External Phone Number Mask	blank
Message Waiting Lamp Policy	Use System Policy
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

Table 2-38 Directory Number Configuration for em49903 Device Profile (continued)

Field	Setting
Dialed Number	Unchecked
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 4,000 other users were configured but are not shown in this table.

- PA Administrator—Example of a JTAPI user created in Cisco CallManager
- Kim Jones—Example of a Cisco Personal Assistant user

Table 2-39 describes the settings in the User Configuration page for these example users.

Table 2-39 User Configuration

Field	User 1 Settings	User 2 Settings
First Name	PA	Kim
Last Name	Administrator	Jones
User ID	PA	kjones
Telephone Number	blank	25000
Manager User ID	blank	blank
Department	blank	blank

Table 2-39 User Configuration (continued)

Field	User 1 Settings	User 2 Settings
User Locale	English United States	English United States
Enable CTI Application Use	Checked	Unchecked
Call Park Retrieval Allowed	Unchecked	Unchecked
Enable Calling Party Number Modification	Unchecked	Unchecked
Associated PC	Not Defined	Not Defined
Primary Extension	none	25000
ICD Extension	Not Defined	Not Defined
Controlled Devices	PA24XXX, PA25XXX, PaMainNumber	none
Enable Authentication Proxy Rights	False	False
Controlled Device Profiles	none	none

Cisco Unity Configuration

This chapter provides an overview of how Cisco Unity was set up in IP Communications Systems Test Release 3.0 for North America IPT. It also provides information about using Cisco Unity with Windows Server 2003 and 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 multiple Compaq ML570 with Quad Processor 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:

Cisco Unity Topology

The following figures show how Cisco Unity was set up in various site models models:

- Figure 3-1—Very Large Site model
- Figure 3-2—Large Site model
- Figure 3-3—Central Site model
- Figure 3-4—Small Site model

Figure 3-1 Cisco Unity Topology in Very Large Site Model

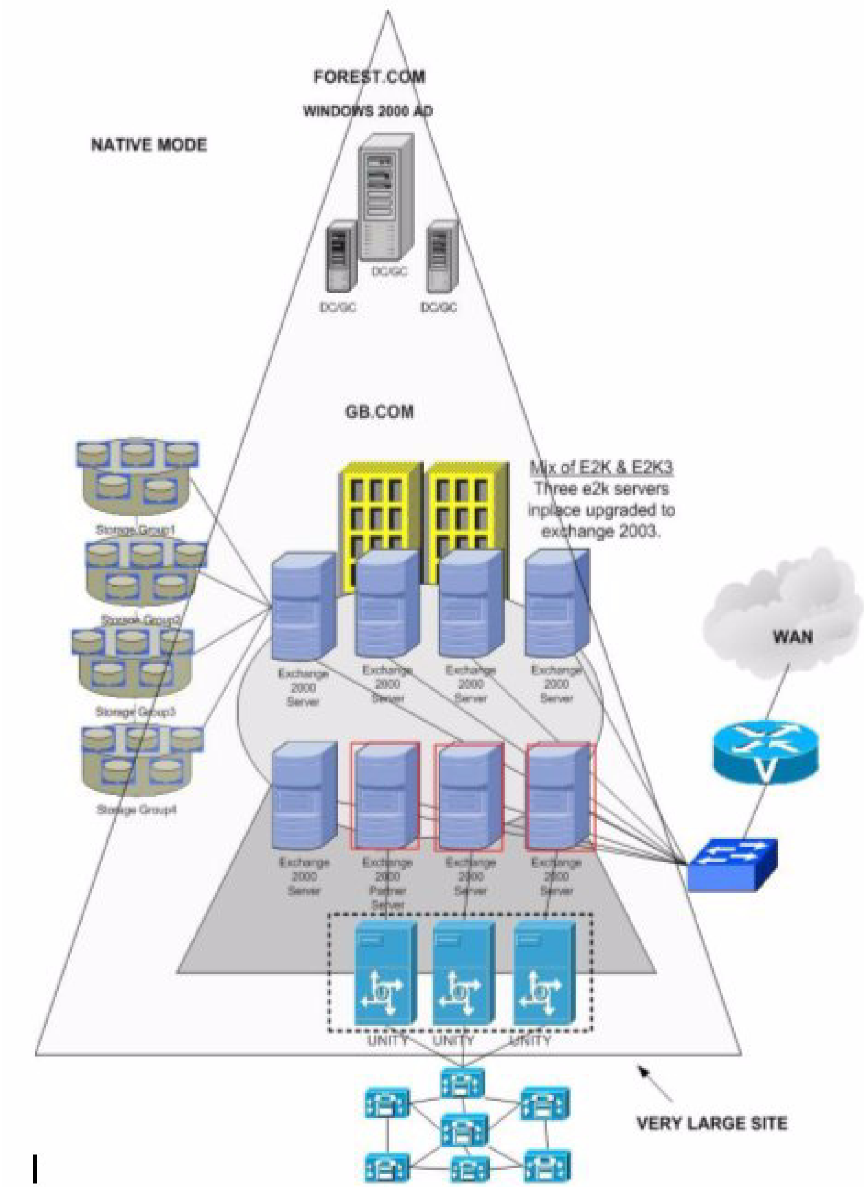


Figure 3-2 Cisco Unity Topology in Large Site Model

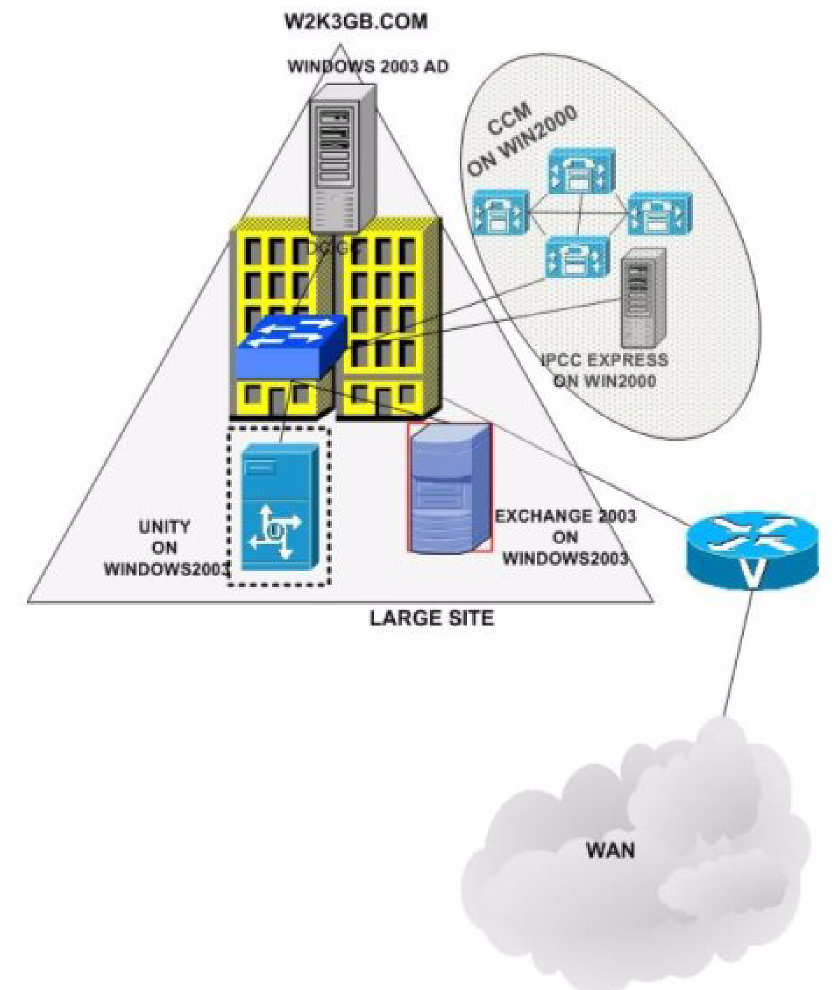


Figure 3-3 Cisco Unity Topology in Central Site Model

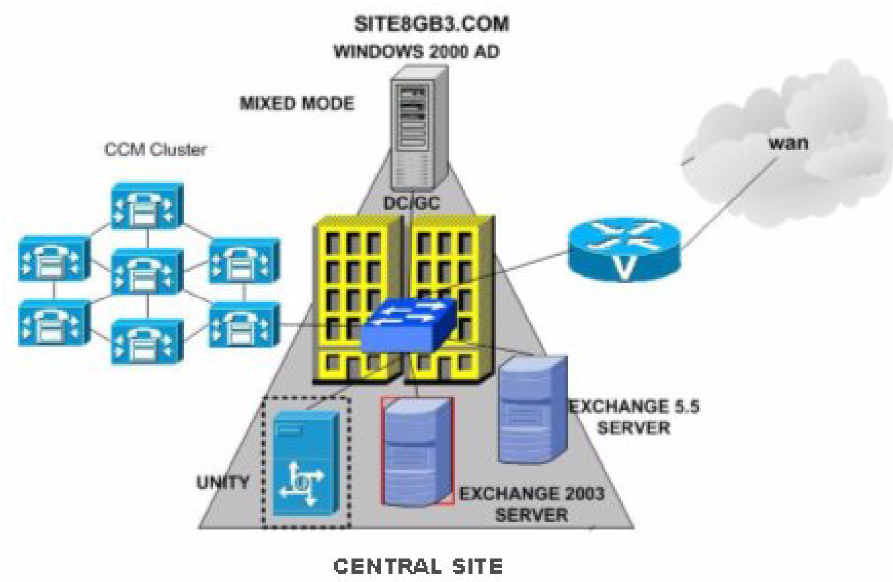
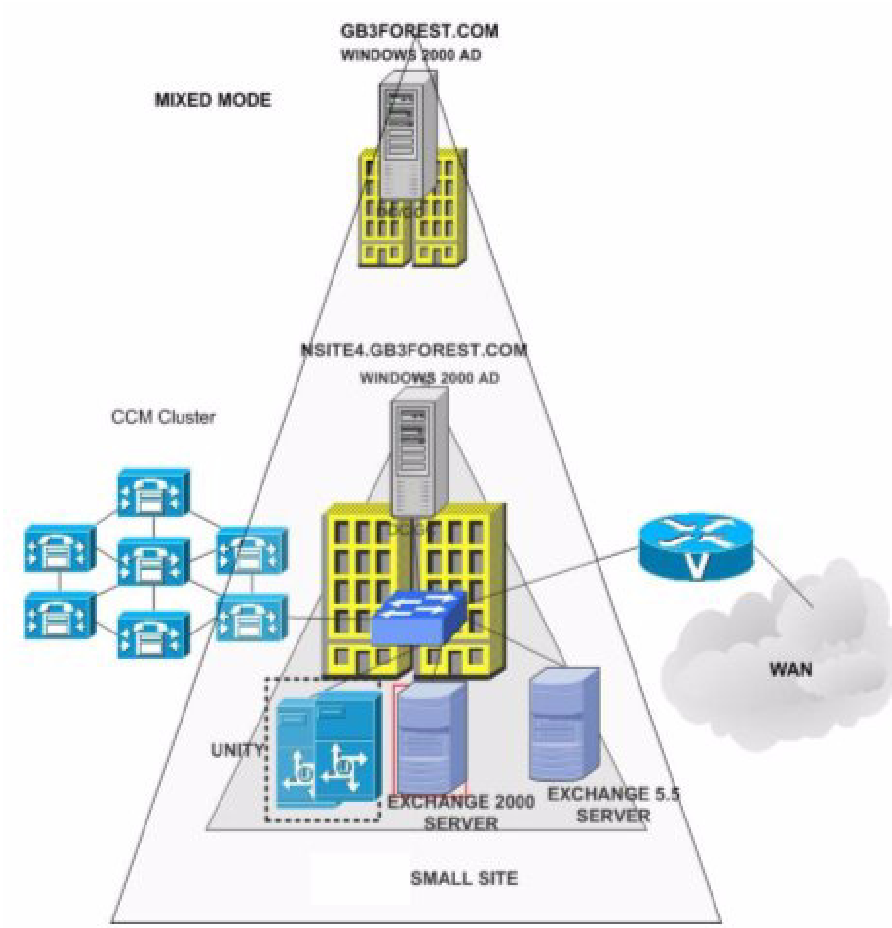


Figure 3-4 Cisco Unity Topology in Small Site Model



Upgrading From IP Communications Systems Test Release 2.0

If you are upgrading from IP Communications Systems Test Release 2.0 for IPT, refer to "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

Using Cisco Unity with Windows Server 2003

For information about installing 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

Using Cisco Unity with Microsoft Exchange

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

In the Large Site model, Cisco Unity and Microsoft Exchange 2003 were installed on servers running the Windows 2003 Server operating system for use with Microsoft Active Directory 2003. In addition, Cisco Unity was tested with Microsoft Exchange 2003 in the Central Site model.

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

The following reconfiguration procedures were tested for IP Communications Systems Test Release 3.0 for North America IPT:

- In the Very Large Site model, use the InPlace upgrade procedure to upgrade Microsoft Exchange 2000 servers to Microsoft Exchange 2003.

  Make sure to run the Exchange 2003 the forestprep and domainprep procedures before installing Exchange 2003.
- In the Small Site model, migrate Microsoft Exchange 5.5 users to Microsoft Exchange 2000 using the Move Mailbox method. Then, reconfigure Cisco Unity for Exchange 2000.
- In the central site model, migrate Microsoft Exchange 5.5 users to Microsoft Exchange 2003 using the Move Mailbox method. Then, reconfigure Cisco Unity for Exchange 2003. Make sure to run the Exchange 2003 forestprep and domainprep procedures before installing Exchange 2003.

Cisco CallManager Express and Cisco Unity Express Configurations

This chapter provides a sample configuration files for Cisco CallManager Express and Cisco Unity Express. These configuration files apply to the Small Site with Cisco CallManager Express model.

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

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

http://www.cisco.com/univercd/cc/td/doc/product/voice/unityexp/rel1_1/index.htm

This chapter includes these topics:

Configuration Files for Cisco CallManager Express and Cisco Unity Express in a Single-Server Deployment

The following sections show configuration files for use when Cisco CallManager Express and Cisco Unity Express reside on the same server:

Configuration File for Cisco CallManager Express

This section shows a configuration file for Cisco CallManager Express when it is deployed on the same server as Cisco Unity Express.

```
3725-CME-CUE-1#sh run
Building configuration...

version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname 3725-CME-CUE-1
!
boot-start-marker
boot system flash:c3725-ipvoice-mz.123-8.T.bin
boot-end-marker
!
enable password lab
!
username lab password 0 lab
username beda password 0 lab
username user1 password 0 lab
clock timezone est -5
clock summer-time etd recurring
network-clock-participate wic 1
no network-clock-participate aim 0
no network-clock-participate aim 1
voice-card 3
 dspfarm
```

```
!
no aaa new-model
ip subnet-zero
ip cef
!
ip dhcp excluded-address 10.3.60.1 10.3.60.20
!
ip dhcp pool ITS
   network 10.3.60.0 255.255.255.0
   option 150 ip 10.3.60.3
   default-router 10.3.60.3
!
ip ftp username beda
ip ftp password lab
no ip domain lookup
no ftp-server write-enable
isdn switch-type primary-ni
!
voice rtp send-recv
!
voice service voip
 allow-connections h323 to h323
 supplementary-service h450.12
 h323
!
voice class codec 1
 codec preference 1 g711ulaw
!
controller T1 0/1
 framing esf
 linecode b8zs
 pri-group timeslots 1-2,24
!
translation-rule 1
 Rule 1 80000 88888
!
translation-rule 2
!
interface FastEthernet0/0
 ip address 10.3.60.3 255.255.255.0
 speed 100
 full-duplex
 h323-gateway voip interface
 h323-gateway voip id GK1 ipaddr 10.3.100.51 1719
 h323-gateway voip h323-id 3725-CUE-1
 h323-gateway voip bind srcaddr 10.3.60.3
!
interface Serial0/0
```

```
 no ip address
 shutdown
 no fair-queue
 clockrate 2000000
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface Serial0/1:23
 no ip address
 no logging event link-status
 isdn switch-type primary-ni
 isdn incoming-voice voice
 isdn bchan-number-order ascending
 no cdp enable
!
interface Service-Engine1/0
 ip unnumbered FastEthernet0/0
 service-module ip address 10.3.60.4 255.255.255.0
 service-module ip default-gateway 10.3.60.3
!
ip classless
ip route 0.0.0.0 0.0.0.0 10.3.60.1
ip route 10.3.60.4 255.255.255.255 Service-Engine1/0
ip http server
ip http authentication local
ip http path flash:
!
tftp-server flash:P00303020214.bin
tftp-server flash:P00403020214.bin
tftp-server flash:music-on-hold.au
!
control-plane
!
voice-port 0/1:23
!
dial-peer cor custom
!
dial-peer voice 300 voip
 destination-pattern 88...
 voice-class codec 1
 session protocol sipv2
 session target ipv4:10.3.60.4
 dtmf-relay sip-notify
 no vad
```

```
!
dial-peer voice 100 pots
 destination-pattern 919472....
 direct-inward-dial
 port 0/1:23
 forward-digits all
!
dial-peer voice 502 voip
 destination-pattern .T
 voice-class codec 1
 session target ras
 dtmf-relay h245-alphanumeric
 no vad
!
num-exp 601640.... 7....
gateway
 timer receive-rtp 1200
!
telephony-service
 load 7910 P00403020214
 load 7960-7940 P00303020214
 max-ephones 144
 max-dn 288
 ip source-address 10.3.60.3 port 2000
 auto assign 1 to 100
 create cnf-files version-stamp 7960 Jul 07 2004 04:38:47
 dialplan-pattern 1 919397.... extension-length 5
 voicemail 88888
 max-conferences 8
 call-forward pattern .T
 moh music-on-hold.au
 web admin system name site password lab
 dn-webedit
 time-webedit
 transfer-system full-blind
 transfer-pattern 408.......
 transfer-pattern 408*
 transfer-pattern 919.......
 transfer-pattern 601.......
 transfer-pattern .T
!
ephone-dn  1
 number 77001
 call-forward busy 80000
 call-forward noan 80000 timeout 10
!
ephone-dn  2
 number 77002
```

```
 call-forward busy 80000
 call-forward noan 80000 timeout 10
!
ephone-dn  119
 number 80000
 loopback-dn 120
 preference 1
 no huntstop
!
ephone-dn  120
 number 22222
 loopback-dn 119
 translate called 1
!
ephone-dn  287
 number 8001.....
 mwi off
!
ephone-dn  288
 number 8000.....
 mwi on
!
ephone  1
 mac-address 0009.7C5F.CBA3
 type 7910
 button  1:101
!
line con 0
 exec-timeout 0 0
line 33
 password lab
 login
 no activation-character
 no exec
 transport preferred none
 transport input all
 transport output all
line aux 0
line vty 0 4
 exec-timeout 0 0
 password lab
 login
!
ntp clock-period 17185481
ntp server 10.3.60.1
end
```

Configuration File for Cisco Unity Express

This section shows a configuration file Cisco Unity Express when it is deployed on the same server as Cisco CallManager Express.

```
3725-CUE-1#service-module service-Engine 1/0 session
Trying 10.3.60.3, 2033 ... Open

User Access Verification

Password:
Password OK
se-10-3-60-4>
se-10-3-60-4> en
Password:
se-10-3-60-4# sh run
Generating configuration:

clock timezone America/New_York

hostname se-10-3-60-4

ip domain-name cisco.com

ntp server 10.3.60.1

groupname Administrators create

username admin create
username ph9 create
username ph99 create
username ph100 create
username ph101 create
username ph102 create
username beda create
username ph10 create
username ph11 create
username ph12 create
username ph13 create
username ph14 create
username ph15 create
username ph55 create
username ph69 create
username ph94 create
username ph1 create

username ph1 phonenumberE164 "4085557001"
```

```
username ph10 phonenumberE164 "4085557010"
username ph101 phonenumberE164 "4085557101"
username ph102 phonenumberE164 "4085557102"
username beda phonenumberE164 "111111"
username ph11 phonenumberE164 "4085557011"
username ph12 phonenumberE164 "4085557012"
username ph13 phonenumberE164 "4085557013"
username ph14 phonenumberE164 "4085557014"

username ph1 phonenumber "77001"
username ph10 phonenumber "77010"
username ph100 phonenumber "77100"
username ph101 phonenumber "77101"
username ph102 phonenumber "77102"
username ph11 phonenumber "77011"
username ph12 phonenumber "77012"

groupname Administrators member admin
groupname Administrators privilege superuser
groupname Administrators privilege ManagePrompts

backup server url "ftp://127.0.0.1/ftp" credentials hidden
"EWlTygcMhYmjazXhE/VNXHCkplVV4KjescbDaLa4fl4WLSPFvv1rWUnfGWTYHfmPSd8ZZ
Ngd+Y9J3xlk2B3
5jwAAAAA="

ccn application autoattendant
 description "autoattendant"
 enabled
 maxsessions 8
 script "aa.aef"
 parameter "MaxRetry" "3"
 parameter "operExtn" "0"
 parameter "welcomePrompt" "AAWelcome.wav"
 end application

ccn application ciscomwiapplication
 description "ciscomwiapplication"
 enabled
 maxsessions 8
 script "setmwi.aef"
 parameter "strMWI_OFF_DN" "8001"
 parameter "strMWI_ON_DN" "8000"
 parameter "CallControlGroupID" "0"
 end application

ccn application promptmgmt
 description "promptmgmt"
```

```
 enabled
 maxsessions 1
 script "promptmgmt.aef"
 end application

ccn application voicemail
 description "voicemail"
 enabled
 maxsessions 8
 script "voicebrowser.aef"
 parameter "logoutUri"
"http://localhost/voicemail/vxmlscripts/mbxLogout.jsp"
 parameter "uri" "http://localhost/voicemail/vxmlscripts/login.vxml"
 end application

ccn engine
 end engine

ccn subsystem jtapi
 ccm-manager address
 end subsystem

ccn subsystem sip
 gateway address "10.3.60.3"
 end subsystem

ccn trigger sip phonenumber 7801
 application "autoattendant"
 enabled
 locale "en_US"
 maxsessions 8
 end trigger

ccn trigger sip phonenumber 88888
 application "voicemail"
 enabled
 locale "en_US"
 maxsessions 8
 end trigger

log console warning

voicemail default expiration time 30
voicemail default language en_US
voicemail default mailboxsize 3000
voicemail recording time 900
voicemail default messagesize 60
voicemail operator telephone 0
```

```
voicemail capacity time 6000
voicemail mailbox owner "ph1" size 3000
 description "ph1 mailbox"
 end mailbox

voicemail mailbox owner "ph10" size 3000
 description "ph10 mailbox"
 end mailbox

voicemail mailbox owner "ph100" size 3000
 description "ph100 mailbox"
 end mailbox

voicemail mailbox owner "ph11" size 3000
 description "ph11 mailbox"
 end mailbox

voicemail mailbox owner "ph12" size 3000
 description "ph12 mailbox"
 end mailbox

end
```

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

The following sections show portions of configuration files to use If you deploy multiple Cisco CallManager Express systems with centralized Cisco Unity:

- Configuration File for MWI SIP Server, page 4-10
- Configuration File for MWI SIP Clients, page 4-11

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
```

Cisco Personal Assistant Configuration

This chapter provides an overview of how Cisco Personal Assistant was set up for the Very Large Site model in IP Communications Systems Test Release 3.0 for North America 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 Personal Assistant component of your IPT solution.

Cisco Personal Assistant was installed on a Cisco MCS-7835-1266 server. The Cisco Personal Assistant Speech Server was installed on the same server. These systems were configured according to the instructions in Cisco Personal Assistant documentation. In general, default or recommended configuration values were used.

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

http://www.cisco.com/univercd/cc/td/doc/product/voice/assist/assist14/index.htm

This chapter includes the following topics:

- Cisco Personal Assistant System Configuration, page 5-2
- Cisco Personal Assistant Server Configuration, page 5-6

Cisco Personal Assistant System Configuration

The following sections provide an overview of how Cisco Personal Assistant was configured on many of the System menu web pages that you access from Cisco Personal Assistant 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 Personal Assistant was set up to perform most effectively.

- System > Speech Services, page 5-2
- System > Telephony, page 5-3
- System > Messaging, page 5-4
- System > Enhanced TTS, page 5-5

The following sections do not discuss these System menu options:

- AA Prompt—Use default settings
- Dial Rules—Use default settings
- Directory Lookup Rules—Use default settings
- Corporate Directory Settings—Use default settings
- Directory Hierarchy—Use default settings
- Miscellaneous Settings—Use default settings

System > Speech Services

To access the Cisco Personal Assistant Administration web pages for configuring speech services, choose **System > Speech Services** from the Cisco Personal Assistant Administration application.

Table 5-1 describes the settings in the Speech Services Configuration page.

Table 5-1 Speech Services Configuration

Field	Setting
Daily Automatic Refresh	Checked
Refresh Schedule	02:00

Table 5-1 Speech Services Configuration (continued)

Field	Setting
Send Refresh Status	Unchecked
Administrator E-mail Address	blank
License Key	appropriate license key
Number of Licenses	8 Speech Ports (Multiple Locales)
Speech Recognition Server Hosts	10.3.201.64 10.3.201.65
Speech Recognition License Manager Hosts	10.3.201.64 10.3.201.65
Supported Locales	American English
Default Locale	American English
Maximum Number for Disambiguation	3
Allow Barge-in	Checked
Rejection Confidence Level	45
Reconfirm Confidence Level	60
Max Error Count per Dialog	3
Max Error Count per Call	8
Max Help Count per Dialog	2
Max Help Count per Call	5

System > Telephony

To access the Cisco Personal Assistant Administration web pages for configuring telephony providers, choose **System > Telephony** from the Cisco Personal Assistant Administration application.

Table 5-2 describes two of the telephony providers that were configured in the Telephony Configuration web page. Two other telephony providers were configured but are not shown in this table.

Table 5-2 Telephony Configuration

Field	Telephony Provider 1 Settings	Telephony Provider 2 Settings
Provider Group Name	CallManager Publisher 1	skinny
Provider Type	Jtapi	Skinny
Telephony Providers	10.3.201.10	10.3.201.10
User Name	PA	blank
Password	user-entered password	blank

System > Messaging

To access the Cisco Personal Assistant Administration web pages for configuring messaging, choose **System > Messaging** from the Cisco Personal Assistant Administration application.

Table 5-3 describes the settings in the Messaging Configuration page.

Table 5-3 Messaging Configuration

Field	Setting
Calendar Server Name	Exchange Server 1
Calendar Mailbox Name	Unity Messaging System - Unity Server 1
Paging SMTP Server Name	blank
Paging SMTP Server Port	25
Paging SMTP Domain Name	blank
Mailbox Name	Unity Messaging System - Unity Server 1
Number of Cisco Unity Licenses	72
Redirection Delay (milliseconds)	4000
Voice Mail Server Name (e.g: VmServer)	blank

Table 5-3 Messaging Configuration (continued)

Field	Setting
Pilot Number	blank
Internal DTMF Redirection Sequence	#X#2
External DTMF Redirection Sequence	#X#2
Voice Mail Servers	Server Name: Unity Server 1 Pilot Number: 22100 DTMF Sequence (Internal): #X#2 DTMF Sequence (External): #X#2
Notify Users of PIN Change	Checked
Administrator E-mail Address	appropriate e-mail address
Unique Attribute for Corporate Directory	mail
Unique Attribute for Message Store	mail

System > Enhanced TTS

To access the Cisco Personal Assistant Administration web pages for configuring enhanced TTS, choose **System > Enhanced TTS** from the Cisco Personal Assistant Administration application.

Table 5-4 describes the settings in the Enhanced TTS Configuration page.

Table 5-4 Enhanced TTS Configuration

Field	Setting
TTS Server Name	PA Server 1
TTS Port Number	6666
TTS License Key	appropriate license key
Number of Licences	12 ports

Cisco Personal Assistant Server Configuration

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

To access the Server Configuration web page, choose **Server > Server Configuration** from the Cisco Personal Assistant Administration application.

Table 5-5 describes the settings in the Server Configuration page.

Table 5-5 Server Configuration

Field	Setting
Server Name	PA Server 1
Hostname or IP Address	10.3.201.64
Media Termination UDP Beginning Port	32000
Route Address Provider	CallManager Publisher Server 1
Route Address	24000
AA Route Address	blank
Media Port Provider	skinny
Media Port Beginning Address	29801
Number of Media Ports	40
Interceptor Port Provider	CallManager Publisher Server 1
Interceptor Ports	24XXX 25XXX
Fail-over Server Names	blank
Trace Package List	all options Checked
Debug Package List	all options checked

CHAPTER 6

Cisco Emergency Responder Configuration

This chapter provides an overview of how Cisco Emergency Responder was set up for the Very Large Site model in IP Communications Systems Test Release 3.0 for North America 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 Unity Emergency Responder in your IPT solution.

Cisco Emergency Responder was installed on a Cisco MCS-7845H-2.4-EVV1 server and configured according to the instructions in the Cisco Emergency Responder documentation. In general, default or recommended configuration values were used.

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

http://www.cisco.com/univercd/cc/td/doc/product/voice/respond/res12/index.htm

This chapter includes the following topics:

Cisco Emergency Responder CER Groups

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

- CER Groups > CER Group Settings, page 6-2
- CER Groups > Telephony Settings, page 6-3

The following sections do not discuss these CER Groups menu options:

- Server settings—Use recommended settings
- License Manager—Use recommended settings
- Control Center—Use recommended setting

CER Groups > CER Group Settings

To access the Cisco Emergency Responder Administration web pages for configuring group settings, choose **CER Groups > CER Group settings** from the Cisco Emergency Responder Administration application.

Table 6-1 describes the settings in the CER Group Settings page.

Table 6-1 CER Group Settings Configuration

Field	Setting
CER Group Name	10.3.201.10 **Note** This IP address is that of the Publisher
Peer TCP Port	17001
Heart beat Count	3
Heart beat Interval (in sec)	30
Active Call Time out (in min)	180
SMTP Mail Server	blank

Table 6-1 CER Group Settings Configuration (continued)

Field	Setting
Source Mail ID	mandatory if the SNMP server is configured
System Administrator Mail ID	blank
Calling Party Modification	disable
SysLog	disable
Syslog Server	mandatory if syslog enabled (example setting: logserver.cisco.com)

CER Groups > Telephony Settings

To access the Cisco Emergency Responder Administration web pages for configuring telephony settings, choose **CER Groups > Telephony settings** from the Cisco Emergency Responder Administration application.

Table 6-2 describes the settings in the Telephony Settings page.

Table 6-2 Telephony Settings Configuration

Field	Setting
UDP Port Begin	32000
Inter CER Group Route Pattern	1000.911
PSAP Callback Route Point Pattern	913XXXXXXXXXX
ELIN Digit Strip Pattern	913
Route Point for Primary CER Server	911
Route Point for Standby CER Server	912

Cisco Emergency Responder ERL

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

Four onsite alerts were set for the very large site model. These alerts were entered in the Onsite Alert Settings page (**ERL > Onsite Alert settings**) using site-appropriate values for Onsite Alert ID, Contact Name, Contact Number, and Email Address.

Table 6-3 describes how one of the onsite alerts was configured in the ERL Information page. You access this page from the Cisco Emergency Responder Administration application by choosing **ERL > ERL Details** and selecting the desired Onsite Alert.

Table 6-3 Onsite Alert Settings Configuration

Field	Setting
ERL Name	Default
ELIN Settings	Route/Translation pattern: 10.911 ELIN: 6013929911
Onsite Alert Settings	Available Onsite Alert IDs: 29001 29063 29061 Onsite Alert IDs for the ERL 29026

Cisco Emergency Responder Phone Tracking

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

This section does not discuss these Phone Tracking menu web pages:

- SNMP Settings—Use recommended settings
- Schedule—Use recommended settings
- LAN Switch Details—Use recommended settings

To access the Cisco Emergency Responder Administration web pages for configuring Cisco CallManager details, choose **Phone Tracking > Cisco CallManager Details** from the Cisco Emergency Responder Administration application.

Table 6-4 describes how Cisco CallManager Details were configured in the Cisco CallManager Details page.

Table 6-4 Cisco CallManager Details

Field	Setting
Cisco CallManager	10.3.211.10 **Note** This IP address is that of Cisco CallManager 1, which is the Subscriber
CTI Manager	10.3.211.10 **Note** This IP address is that of Cisco CallManager 1, which is the Subscriber
CTI Manager User Name	cer **Note** This user name is defined in Cisco CallManager and is associated with all CTI route points and the CTI port

Table 6-4 Cisco CallManager Details (continued)

Field	Setting
CTI Manager Password	Password specified for the user “cer” in Cisco CallManager
BackUp CTI Manager 1	10.3.213.10
BackUp CTI Manager 2	blank
Telephony Port Begin Address	50001
Number of Telephony Ports	3

Cisco Customer Response Applications Configuration

This chapter provides an overview of how Cisco Customer Response Applications (Cisco CRA) was set up for the Very Large Site model in IP Communications Systems Test Release 3.0 for North America IPT.

Note Effective with release 3.0, Cisco Customer Response Applications (CRA) has been renamed Cisco Customer Response Solutions (CRS) and, effective with release 3.1, is marketed under the names IPCC Express and IP IVR. The Cisco website and packaging materials have been updated to reflect the new name, but the user interface, and therefore the documentation, have not.

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

The following servers were used for Cisco CRA:

- Cisco CRA Server—Cisco MCS-7845H-2.4-CC1
- ICD Call Statistics, Recording, and Monitoring Server—Cisco MCS-7825-1133
- Automatic speech recognition (ASR) Server—Cisco MCS-7825-1133

In general, default or recommended configuration values were used during installation and setup. For detailed information about installing, configuring, and administering Cisco CRA, refer to the Cisco CRA documentation at this URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/sw_ap_to/apps_3_1/index.htm

The this chapter includes the following topics:

Overview

For the Very Large Site model in IP Communications Systems Test Release 3.0 for North America IPT, a call center with the following characteristics was configured:

- Operational 24 hours a day, 7 days a week.
- 150 Agents.
- Agents grouped by their support function. The Cisco CRA system routes calls to the contact service queue (CSQ) for the appropriate group.
- Automatic speech recognition (ASR) support for accessing menu choices.
- Callers to the call center access a menu that offers the following choices:
  - Press 1 for Computer Hardware and Operating System Support.
  - Press 2 for Standard Desktop Software Support.
  - Press 3 for Financial and Trading Applications Support.
  - Press 4 for Reporting Applications Support.
  - Press 5 for Network and Password Support.
  - Press 8 for the Executive System Support.

Cisco CallManager Configuration for Cisco CRA System

Table 7-1 describes how Cisco CallManager was configured to work with the sample Cisco CRA system that is described in this chapter.

For additional information about Cisco CallManager configuration for the Very Large Site model, see Chapter 2, "Cisco CallManager Configuration."

Table 7-1 Cisco CallManager Configuration for Cisco CRA

Configuration Item	Quantity Configured	Associated Item
Agent	150	Phone
CTI Port	145	JTAPI User
JTAPI USER	1	—
Phone	150	Agent and RM User
RM USER	1	—
Route Point	8	JTAPI User
TELECASTER USER	1	Phone

In addition, the following CTI Managers were configured in Cisco CallManager:

- CM4—Primary CTI Manager
- CM2—Backup CTI Manager for redundancy

CSQ Configuration

Table 7-2 describes the resource groups that were configured in Cisco CRA Administration. Each resource group has an associated CSQ, which was also configured in Cisco CRA Administration.

Table 7-2 Cisco CRA Resource Groups

Resource Group Name	Description	Number of Agents
CHOS_RG	Computing Hardware and Operating System Support	21
SDSS_RG	Standard Desktop Software Support	21
FAT_RG	Financial and Trading Applications Support	44
RAS_RG	Reporting Applications Support	18
NPS_RG	Network and Password Support	47
EIS_RG	Executive System Support	5

Table 7-3 describes the skill groups that were configured in Cisco CRA Administration. This example assumes that some agents have more than one skill.

Table 7-3 Cisco CRA Skill Groups

SKill Group Name	Description	Number of Agents
S_SDSS	Agents cross-trained in Standard Desktop Software Support	6
S_EIS	Agents cross-trained in Executive Information Support	3
S_FAT	Agents cross-trained in Financial and trading Applications Support	4
S_NPS	Agents cross-trained in Network and Password support	2

Table 7-4 describes how a resource-based CSQ (named CHOS) and a skills-based CSQ (named S_SDSS) were configured in Cisco CRA Administration. These skills were configured in the Contact Service Queue Configuration area in the ICD Configuration web page.

Table 7-4 Cisco CRA CSQ Configuration

Field	CHOS Setting	S_SDSS Setting
Contact Service Queue Name	CHOS	S_SDSS
Contact Queuing Criteria	FIFO	FIFO
Automatic Work	Disabled	Disabled
Resource Pool Selection Model	Resource Group	Resource Skill
Service Level	20	20
Service Level Percentage	80	80
Resource Selection Criteria	Longest Available	Longest Available
Assigned Skills	—	S_SDSS(5)
Resource Group	CHOS_RG	—
Show Resources	19 agents configured	6 agents configured

Scripts for IVR Menu Choices

The following sections describe the scripts that were associated with each of the IVR menu choices in Cisco CRA Administration:

- Computing Hardware and Operating System Support, page 7-6
- Standard Desktop Software Support, page 7-6
- Financial and Trading Application Support, page 7-6
- Reporting Application Support, page 7-7
- Network and Password Support, page 7-7
- Executive System Support, page 7-7

Computing Hardware and Operating System Support

A caller who selects the option for Computing Hardware and Operating System Support is queued for an agent with the longest time available in the CHOS resource group.

A caller who is in queue for more than 2 minutes is given the options to continue to hold or to be transferred to voice mail.

Standard Desktop Software Support

A caller who selects the option for Standard Desktop Software Support is queued for an agent with the longest time available in the SDSS resource group. If an agent is not available within 90 seconds, the system considers other agents who are in the S_SDSS skill group.

An Enterprise Data pop-up window provides an agent with information about which resource group the call was queued for.

A caller who is in queue for more than 4 minutes is given the options to continue to hold or to be transferred to voice mail.

Financial and Trading Application Support

A caller who selects the option for Financial and Trading Application Support is queued for an agent with the longest time available in the FAT resource group. If an agent is not available within 60 seconds, the system considers other agents who are in the S_FAT skill group.

An Enterprise Data pop-up window provides an agent with information about which resource group the call was queued for.

A caller who is in queue for more than 2 minutes is given the options to continue to hold or to be transferred to voice mail.

Reporting Application Support

A caller who selects the option for Reporting Application Support before 10:00 a.m. (1000) hears a message that daily reports may not have been generated. The message asks the caller to try again later. Then the call is terminated.

A caller who selects the option for Reporting Application Support after 10:00 a.m. (1000) is queued for an agent with the longest time available in the RAS resource group.

A caller who is in queue for more than 3 minutes is the given options to continue to hold or to be transferred to voice mail.

Network and Password Support

A caller who selects the option for Network and Password Support is queued for an agent with the longest time available in the NPS resource group.

A caller who is in queue for more than 5 minutes is the given options to continue to hold or to be transferred to voice mail.

Executive System Support

A caller who selects the option for Executive System Support is queued for an agent with the longest time available in the EIS resource group. If an agent is not available within 60 seconds, the system considers agents from other groups with the S_EIS skill.

CHAPTER 8

Cisco MeetingPlace Configuration

This chapter provides an overview of how Cisco MeetingPlace was set up for the Very Large Site model in IP Communications Systems Test Release 3.0 for North America 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 MeetingPlace component of your IPT solution.

In IP Communications Systems Test Release 3.0 for North America IPT, Cisco MeetingPlace consists of these components:

- Cisco MeetingPlace Audio Server MP-8112
- Cisco MeetingPlace IP Gateway installed on an MCS-7835

Cisco MeetingPlace was configured according to the instructions in the Cisco MeetingPlace documentation. In general, default or recommended configuration values were used.

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

http://www.cisco.com/univercd/cc/td/doc/product/conf/mtgplace/index.htm

This chapter includes the following topics:

Cisco Audio Server Configuration

The following sections show how the Net command and the Blade command were use to configure the Cisco MeetingPlace Audio Server MP-8112:

Net Command

Table 8-1 shows the Net command parameters that were used to configure the Cisco MeetingPlace Audio Server MP-8112.

Table 8-1 Net Command Parameters

Field	Setting
IP Address	10.0.5.201
NTP Servers	172.10.0.110
Site	#0 (Home Site)
Site Subnetmask	255.255.255.0
Site Broadcast address	10.0.5.255
Site default Gateway	10.0.5.1
Route daemon	disabled

Blade Command

Table 8-2 shows the Blade command parameters that were used to configure the DSP cards in the Cisco MeetingPlace Audio Server MP-8112.

Table 8-2 Blade Command Parameters

Slot	Card	Type	CardID	Ports
1–14	CG6000C	SB	0-9	—
15	TP1610	IP	1	480–959 (10.0.112.204, 10.0.112.205)
16	TP1610	IP	0	0–479 (10.0.112.202, 10.0.112.203)

Cisco MeetingPlace IP Gateway Configuration

This section shows how Cisco MeetingPlace IP Gateway was configured for H.323 and SIP.

Table 8-3 shows the settings that were made in the MeetingPlace IP Gateway Management Console for H.323.

Table 8-4 shows the settings that were made in the MeetingPlace IP Gateway Management Console for SIP.

Table 8-3 Cisco MeetingPlace IP Gateway Configuration for H.323

Field	Setting
E.164	52000
Gateway Address	10.0.211.10
H.323 Enabled	1
H323 ID	MeetingPlace

Table 8-4 Cisco MeetingPlace IP Gateway Configuration for SIP

Field	Setting
Display Name	MeetingPlace
Max Number of caller	960

Table 8-4 Cisco MeetingPlace IP Gateway Configuration for SIP (continued)

Field	Setting
Proxy Server IP address	10.0.211.10
Proxy server Port	5060
SIP enabled	1
User Name	52005

Cisco CallManager Configuration for Cisco MeetingPlace

Cisco CallManager routes IP calls to the MeetingPlace IP Gateway. When you dial a number from an IP phone, the call is directed to Cisco CallManager. Cisco CallManager associates the dialed number with a route pattern that points to the appropriate gateway.

To configure Cisco CallManager for MeetingPlace IP Gateway, you add a gateway and then assign it to a route pattern. You also must configure a SIP trunk.

The following sections describes how Cisco CallManager was configured for Cisco MeetingPlace IP Gateway:

For additional information about how Cisco CallManager was configured for the Very Large Site model, see Chapter 2, "Cisco CallManager Configuration."

For additional information about 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/3_3/index.htm

Gateway Configuration

When you use Cisco CallManager Administration to add a new gateway for Cisco MeetingPlace, choose **H.323 Gateway** from the Gateway Type drop-down list in the Add a New Gateway page.

Table 8-5 describes how the gateway was configured in the Gateway Configuration page. Default values were used for fields that are not shown in this table.

Table 8-5 Cisco CallManager Gateway Configuration for MeetingPlace

Field	Setting
Device Name	Enter the host name or the IP address of the MeetingPlace IP Gateway server
Device Pool	Default
Locations	Choose the location of the MeetingPlace IP Gateway server
Calling Party Selection	Originator
Presentation Bit	None
Gatekeeper Registration	None
Media Termination Point Required	Unchecked

Route Pattern Configuration

After you add a gateway, you must use Cisco CallManager Administration to assign a route pattern to the gateway.

Table 8-6 describes how the route pattern was configured in the Route Pattern Configuration page. Default values were used for fields that are not shown in this table.

Table 8-6 Cisco CallManager Route Pattern Configuration for MeetingPlace

Field	Setting
Route Pattern	Enter the dialable number for the configured MeetingPlace IP Gateway. MeetingPlace IP users use this number to connect to the MeetingPlace server.
Numbering Plan	Choose the appropriate option.
Gateway/Route List	Enter the host name or the IP addre.ss of the MeetingPlace IP Gateway.
Route Option Dial Tone	Choose **Route this pattern** and uncheck the **Provide Outside** check box.

SIP Trunk Configuration

When you use Cisco CallManager Administration to add a SIP trunk for Cisco MeetingPlace, choose **Trunk** from the Device Type drop-down list in the Add a New Device page.

Table 8-7 describes how the SIP trunk was configured in the Trunk Configuration page. Default values were used for fields that are not shown in this table.

Table 8-7 Cisco CallManager SIP Trunk Configuration for MeetingPlace

Field	Setting
Device Name	Enter the dialable number for the MeetingPlace IP Gateway
Device Pool	Choose the appropriate device pool
Locations	Choose the location of the MeetingPlace IP Gateway server
Media Termination Point Required	Checked

Wireless Configuration

This chapter 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) 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 North America 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

The wireless portion of the IP Communications Systems Test Release 3.0 for North America IPT was configured based on the recommendations and configurations described in the documents listed in Table 9-1.

Table 9-1 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 North America 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 1231. In addition, a Cisco Aironet AP 1231 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 1231 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 1231 to encrypt traffic. Both signaling (SCCP) and media (RTP) streams are encrypted between the Cisco IP Phone 7920 and the Cisco Aironet AP 1231. 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 1231 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 1231. The recommended maximum number of users is 15 to 25.
- No more than 16 VLANs were used per Cisco Aironet AP 1231. 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 1231. Cisco recommends that you limit the Cisco Aironet AP 1231 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 9-1.

Cisco Aironet 1231 Access Point Configuration File

This section shows a configuration file for the Cisco Aironet AP 1231 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 1231 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 9-1.

```
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
1443435E59220F7D767D60667030214750510 37C0902062F533D400A7904057377
  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
055D535B02196A5F3B5036342D285C7A0A74706466064150475559730C0C05055F
```

```
  user cisco8 nthash 7
143645292A50737C750D64637B3153375B2200010F75052F564935017D03010507
  user cisco9 nthash 7
06512D076F185C4E5035462859560E0A7570156401435530202705OB0104755E52
  user cisco10 nthash 7
091A185F3A5635375C5D510B080178606D7531574656570701 7C700059534A300E
  user cisco11 nthash 7
05535129716F6A5B4C563645582A220B73017E17117B4254435025020B0A70765B
  user cisco12 nthash 7
01472756785920590 71B68583D5346425E2D530809067A6A6D0445574454250408
  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 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 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 9-1).

CHAPTER 10

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 North America IPT. 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.

For additional information and guidelines for implementing Cisco IP Video Telephony, refer to the following documents, which are available at the URLs shown. (You must be a registered user of Cisco Connection Online to access some of these URLs.)

- *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

- *Understanding Video Telephony*

  http://www.cisco.com/en/US/partner/products/sw/voicesw/ps556/products_administration_guide_chapter09186a00801ec5cf.html

- Deploying video telephony

  http://cco/en/US/partner/about/ac123/ac114/ac173/Q3-04/tech_videotel.html

- Cisco IP/VC 3500 Series video conferencing products:

  http://www.cisco.com/en/US/partner/products/hw/video/ps1870/index.html

- Cisco VT Advantage:

  http://www.cisco.com/en/US/partner/products/sw/voicesw/ps5662/index.html

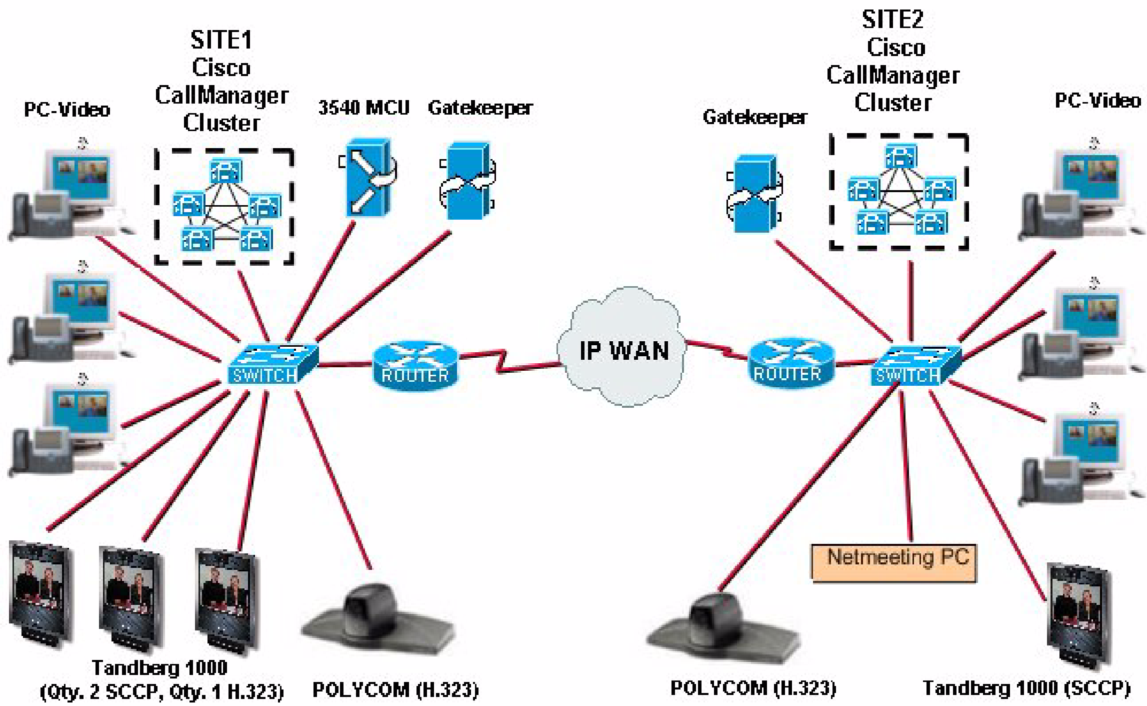
You can also refer to documentation provided by the vendors of Tandberg video endpoints, Polycom video endpoints, and NetMeeting.

This chapter includes the following topics:

IP Video Telephony Topology

Figure 10-1 shows how Cisco IP Video Telephony was deployed in IP Communications Systems Test Release 3.0.

Figure 10-1 IP Video Telephony Topology



Supported Call Types

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

- Calls between Tandberg SCCP video endpoints
- Calls between Tandberg SCCP and Tandberg H.323 video endpoints
- Calls between a Cisco IP Phone associated with Cisco VT Advantage and Tandberg SCCP video endpoint
- Calls between a Cisco IP Phone associated with Cisco VT Advantage and Tandberg H.323 video endpoint

- Calls between a Tandberg SCCP video endpoint and a Polycom H.323
- Calls between a Tandberg H.323 video endpoint and a Polycom H.323
- Calls between a Cisco IP Phone associated with Cisco VT Advantage a Polycom
- Calls between NetMeeting H.323 and SCCP video endpoints
- SCCP ad-hoc video conference call between Tandberg SCCP, Tandberg H.323, Cisco IP Phone 7970, Polycom H.323, and NetMeeting H.323 using a Cisco IP/VC 3540 MCU

Call Routing

The IP Video Telephony deployment supports the following call routings:

- SCCP endpoint > Cisco CallManager > SCCP endpoint
- H.323 endpoint > Gatekeeper > Cisco CallManager > SCCP endpoint
- SCCP endpoint > Cisco CallManager > Gatekeeper > H.323 endpoint
- H.323 endpoint > Gatekeeper > Cisco CallManager > Gatekeeper > H.323 endpoint

The call routing for video calls functions just as the call routing for audio calls. For more information, see the "Understanding Video Telephony" chapter in *Cisco CallManager System Guide*, which is available at this URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_callmg/4_0/sys_ad/4_0_1/ccmsys/index.htm

General Procedures for Configuring IP Video Telephony

This section provides an overview of the steps that you take to configure IP Video Telephone. The Reference column in the following table refers to *Cisco CallManager Administration Guide* and to *Cisco CallManager System Guide*, which are available at this URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_callmg/4_0/sys_ad/4_0_1/index.htm

	Procedure	Reference
Step 1	If you use regions for call admission control, configure regions for video call bandwidth **Note** All devices have a default region, which defaults to 384 kbps for video.	• Refer to the "Region Configuration" chapter in *Cisco CallManager Administration Guide*. • Refer to the "Call Admission Control" chapter in *Cisco CallManager System Guide*.
Step 2	If you use locations for call admission control, configure locations for video call bandwidth.	• Refer to the "Location Configuration" chapter in *Cisco CallManager Administration Guide*. • Refer to the "Call Admission Control" chapter in *Cisco CallManager System Guide*.
Step 3	To use a Cisco video conference bridge, configure the appropriate conference bridge for your network.	• Refer to the "Conference Bridge Configuration" chapter in *Cisco CallManager Administration Guide*.
Step 4	To configure a user to use the video conference bridge instead of other conference bridges, configure the media resource groups and media resource group lists for the user.	• Refer to the "Media Resource Group Configuration" chapter in *Cisco CallManager Administration Guide*. • Refer to the "Media Resource Group List Configuration" chapter in *Cisco CallManager Administration Guide*.

Configuring Cisco CallManager for Tandberg SCCP Video Endpoints

The following sections describes how Cisco CallManager was configured for Tandberg video endpoints:

- Phone Configuration for Tandberg SCCP Video Endpoints, page 10-6
- Directory Number Configuration for Tandberg SCCP Video Endpoints, page 10-7

For additional information about how Cisco CallManager was configured for the Very Large Site model, see Chapter 2, "Cisco CallManager Configuration."

For additional information about 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/3_3/index.htm

Phone Configuration for Tandberg SCCP Video Endpoints

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

Two Tandberg SCCP video endpoints were configured for the Very Large Site model. Table 10-1 shows how one of these devices was configured.

One Tandberg H.323 video endpoint was also used in the Very Large Site model. This device does not need to be added or configured in Cisco CallManager Administration.

Table 10-1 Phone Configuration for SCCP Video Endpoints

Field	Tandberg SCCP Video Endpoint Setting
MAC Address	00506000D6B9
Description	Auto 51001
Owner User ID	blank
Device Pool	HQ-21
Calling Search Space	Internal_css

Table 10-1 Phone Configuration for SCCP Video Endpoints (continued)

Field	Tandberg SCCP Video Endpoint Setting
AAR Calling Search Space	<None>
Media Resource Group List	Video-MRG_list
User Hold Audio Source	50-MOH_SITE1-MOH3
Location	site1
Retry Video Call as Audio	Checked
Phone Button Template	Standard Tandberg Video
Softkey Template	<None>
Information	blank
Directory	blank
Messages	blank
Services	blank
Authentication Server	blank
Proxy Server	blank
Idle Timer (seconds)	blank
MLPP Domain	blank
MLPP Indication	Not available on this device
MLPP Preemption	Not available on this device

Directory Number Configuration for Tandberg SCCP Video Endpoints

Table 10-2 shows how the directory number was configured in Cisco CallManager Administation for the Tandberg SCCP video endpoints in the Very Large Site model.

Table 10-2 Directory Number Configuration for SCCP Video Endponts

Field	Tandberg SCCP Video Endpoint Setting
Directory Number	51001
Partition	Internal_p
Voice Mail Profile	site1Unity1
Calling Search Space	International_css
AAR Group	<None>
Auto Answer	Auto Answer Off
Forward All	default settings
Forward Busy	Voice Mail—Checked Destination—blank Calling Search Space—International_css
Forward No Answer	default settings
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)	site1-51001
Line Text Label	site1-51001
External Phone Number Mask	blank
Message Waiting Lamp Policy	Use System Policy
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

Table 10-2 Directory Number Configuration for SCCP Video Endponts (continued)

Field	Tandberg SCCP Video Endpoint Setting
Redirect Number	Unchecked
Caller Number	Checked
Dialed Number	Checked
Character Set	Western European (Latin 1)

Gatekeeper Configuration for H.323 Video Endpoints

The following sections provide sample configurations for gatekeepers used with H.323 video endpoints in IP Video Telephony:

- Gatekeeper Configuration for the Very Large Site Model, page 10-9
- Gatekeeper Configuration for the Large Site Model, page 10-10

Gatekeeper Configuration for the Very Large Site Model

This section provides a sample configuration for a gatekeeper in the Very Large Site model.

```
gatekeeper
zone local SITE1-GK1 cisco.com 10.3.100.51
zone remote SITE2-GK1 cisco.com 10.3.30.3 1719
zone cluster local East-Cluster SITE1-GK1
element SITE1-GK2 10.3.100.52 1719
!
zone cluster remote west-cluster via
element SITE2-GK1 10.3.30.3 1719
!
zone prefix SITE2-GK1 20001*
zone prefix SITE1-GK1 2* gw-priority 10 SITE1_ict_to_cm1_2
zone prefix SITE1-GK1 58888* gw-priority 10 Video-MCU-ICT-Trk_3
Video-MCU-ICT-Trk_2
zone prefix SITE1-GK1 60139* gw-priority 10 SITE1_ict_to_cm3_4
SITE1_ict_to_cm4_5 SITE1_ict_to_cm2_3 SITE1_ict_to_cm1_2
zone prefix SITE1-GK1 60139* gw-default-priority 0
zone prefix SITE1-GK1 60144* gw-priority 10 SITE4_CallManager_2
```

```
zone prefix SITE1-GK1 60144* gw-priority 8 SITE4_CallManager_1
zone prefix SITE1-GK1 60144* gw-default-priority 0
zone prefix SITE1-GK1 60155* gw-priority 10 SITE5_CallManager_2
zone prefix SITE1-GK1 60155* gw-priority 8 SITE5_CallManager_1
zone prefix SITE1-GK1 60155* gw-default-priority 0
zone prefix SITE1-GK1 601640* gw-priority 10 SITE5_CallManager_2
zone prefix SITE1-GK1 601640* gw-priority 8 SITE5_CallManager_1
zone prefix SITE1-GK1 601640* gw-default-priority 0
zone prefix SITE1-GK1 601649* gw-priority 10 SITE5_CallManager_2
zone prefix SITE1-GK1 601649* gw-priority 8 SITE5_CallManager_1
zone prefix SITE1-GK1 601649* gw-default-priority 0
zone prefix SITE1-GK1 60172* gw-priority 10 SITE4_CallManager_2
zone prefix SITE1-GK1 60172* gw-priority 8 SITE4_CallManager_1
zone prefix SITE1-GK1 60172* gw-default-priority 0
zone prefix SITE1-GK1 601811* gw-priority 10 SITE1_ict_to_cm1_2
SITE1_ict_to_cm2_3 SITE1_ict_to_cm4_5 SITE1_ict_to_cm3_4
zone prefix SITE1-GK1 601833* gw-priority 10 SITE3_CallManager_2
gw-priority 8 SITE3_CallManager_1
zone prefix SITE1-GK1 60193* gw-priority 10 SITE3_CallManager_2
zone prefix SITE1-GK1 60193* gw-priority 8 SITE3_CallManager_1
zone prefix SITE1-GK1 60193* gw-default-priority 0
zone prefix SITE1-GK1 6037111111* gw-priority 10 Tandberg-01144232
zone prefix SITE1-GK1 6037777777* gw-priority 10 SITE_1_h323
zone prefix SITE1-GK1 70139* gw-priority 10 ICT-G711_2 ICT-G711_3
zone prefix SITE1-GK1 70139* gw-default-priority 0
zone prefix SITE2-GK1 *
gw-type-prefix 1#* default-technology
gw-type-prefix 3#*
lrq forward-queries
no use-proxy SITE1-GK1 default inbound-to terminal
no use-proxy SITE1-GK1 default outbound-from terminal
no shutdown
```

Gatekeeper Configuration for the Large Site Model

This section provides a sample configuration for the gatekeeper in the Large Site model.

```
gatekeeper
zone local SITE2-GK1 cisco.com 10.3.30.3
zone remote SITE1-GK1 cisco.com 10.3.100.51 1719
zone remote SITE1-GK2 cisco.com 10.3.100.52 1719
zone cluster local west-cluster SITE2-GK1
zone cluster remote East-Cluster via
element SITE1-GK1 10.3.100.51 1719
```

```
element SITE1-GK2 10.3.100.52 1719
!
zone prefix SITE2-GK1 2000*
zone prefix SITE2-GK1 6011* gw-priority 10 SITE8_ict_hq12_3
SITE8_ict_hq12_2
zone prefix SITE2-GK1 6011* gw-default-priority 0
zone prefix SITE2-GK1 60120* gw-priority 10 SITE8_ict_hq12_2
SITE8_ict_hq12_3
zone prefix SITE2-GK1 60128* gw-priority 8 SITE2_CallManager_2
SITE2_CallManager_3
zone prefix SITE2-GK1 60128* gw-default-priority 0
zone prefix SITE2-GK1 60152* gw-priority 8 SITE2_CallManager_3
SITE2_CallManager_2
zone prefix SITE2-GK1 601822* gw-priority 10 SITE2_CallManager_1 8
gw-priority SITE2_CallManager_2 SITE2_CallManager_3
zone prefix SITE2-GK1 70152* gw-priority 10 G711_ICT_3
zone prefix SITE2-GK1 7055261011* gw-priority 10 SITE2-Netmeeting
zone prefix SITE2-GK1 7055261012* gw-priority 10 SITE2-polycom
gw-type-prefix 1#* default-technology
lrq forward-queries
no use-proxy SITE2-GK1 default inbound-to terminal
no use-proxy SITE2-GK1 default outbound-from terminal
no shutdown
```

IP Video Telephony Video Conferencing

This section provides basic concepts and configuration for the Cisco IP/VC 3540 used in the IP Communications Systems Test Release 3.0 for North America IPT.

For related information, refer to *Setting Up Cisco CallManager to Use the IP/VC MCU and Placing Calls* at the following URL. You must be a registered user of Cisco.com to access this document.

http://cisco.com/en/US/partner/products/hw/video/ps1870/products_administration_guide_chapter09186a00801ea69c.html

The IP/VC MCU was used in the following ways to provide video conferencing for IP Communications Systems Test Release 3.0:

- The IP/VC MCU was deployed as a SCCP device in the Cisco CallManager environment.
- Cisco CallManager was connected to an H.323 environment in which the IP/VC MCU was deployed.

When deployed in a Cisco CallManager SCCP environment, the primary function of the IP/VC MCU is to provide media processing for conferences. In this capacity, the IP/VC MCU negotiates parameters with the terminals participating in a conference, and it provides media processing. Cisco CallManager manages the call flow. Terminals use Cisco CallManager processes to initiate conferences, and Cisco CallManager manages the allocation of IP/VC MCU resources.

When deployed in an H.323 environment, the IP/VC MCU manages its own resources and provides media processing. Calls are initiated utilizing the processes the IP/VC MCU uses for H.323 terminals. In this scenario, you must configure Cisco CallManager to communicate with an H.323 gatekeeper, which controls call flows. This arrangement allows SCCP terminals to participate in conferences that include other SCCP terminals, H.323 terminals, or terminals of both types.

The following tables show how the Cisco IP/VC 3540 MCU was configured using the IP/VC Administrator web page:

- Table 10-3 on page 10-12—IP/VC Board > Basics
- Table 10-4 on page 10-13—IP/VC Board > Addressing
- Table 10-5 on page 10-13—IP/VC MCU > Settings > Basics
- Table 10-6 on page 10-13—IP/VC MCU > Protocols > H.323 Protocol Configuration
- Table 10-7 on page 10-14—IP/VC MCU > Protocols > SCCP Protocol Configuration

Table 10-3 IP/VC MCU Configuration: IP/VC Board > Basics

Field	Value
Board name	MC06A
Location	Site1
Serial number	*Appropriate value*
Hardware version	*Appropriate value*
Date/Time	*Appropriate value*
Slot number	*Appropriate value*
Software version	3.2

Table 10-4 IP/VC MCU Configuration: IP/VC Board > Addressing

Field	Value
IP Address	*Appropriate value*
Subnet Mask	*Appropriate value*
Router IP	*Appropriate value*
DNS Suffix	XX.com
Preferred DNS server	*Appropriate value*
Alternate DNS server	*Appropriate value*
Port type	Ethernet-CSMA/CD
MAC Address	*Appropriate value*
Port settings	100/Mbps / Full Duplex
Port status	100/Mbps / Full Duplex

Table 10-5 IP/VC MCU Configuration: IP/VC MCU > Settings > Basics

Field	Value
MCU Mode	MCU
Number of SCCP ports	30

Table 10-6 IP/VC MCU Configuration: IP/VC MCU > Protocols > H.323 Protocol Configuration

Field	Value
Activate protocol settings	Checked
Description	H.323 Protocol Configuration
Registration Name	blank
Gatekeeper Address	10.3.100.51
Gatekeeper Port	1719
Strip zone prefix	Unchecked

Table 10-6 IP/VC MCU Configuration: IP/VC MCU > Protocols > H.323 Protocol Configuration (continued)

Field	Value
Enable H.329	Unchecked
Enable Fast Start	Unchecked
Enable generic audio capabilities	Unchecked
Enable alternate Gatekeeper	Unchecked
Enable H.245 tunneling	Unchecked

Table 10-7 IP/VC MCU Configuration: IP/VC MCU > Protocols > SCCP Protocol Configuration

Field	Value
TFTP Servers: IP Address / Port	*Appropriate values*
CallManagers: IP Address / Port	*Appropriate values*
Perform MCU reset on CallManager Reset message	Checked
Local port base	11000
Priority	24
Registration: Retries	3
Initial timeout	30
Consequent timeout	10
Keep Alive: Retries	3
Timeout	10
Recovery mode	Not applicable
Change configuration locally	Unchecked

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.

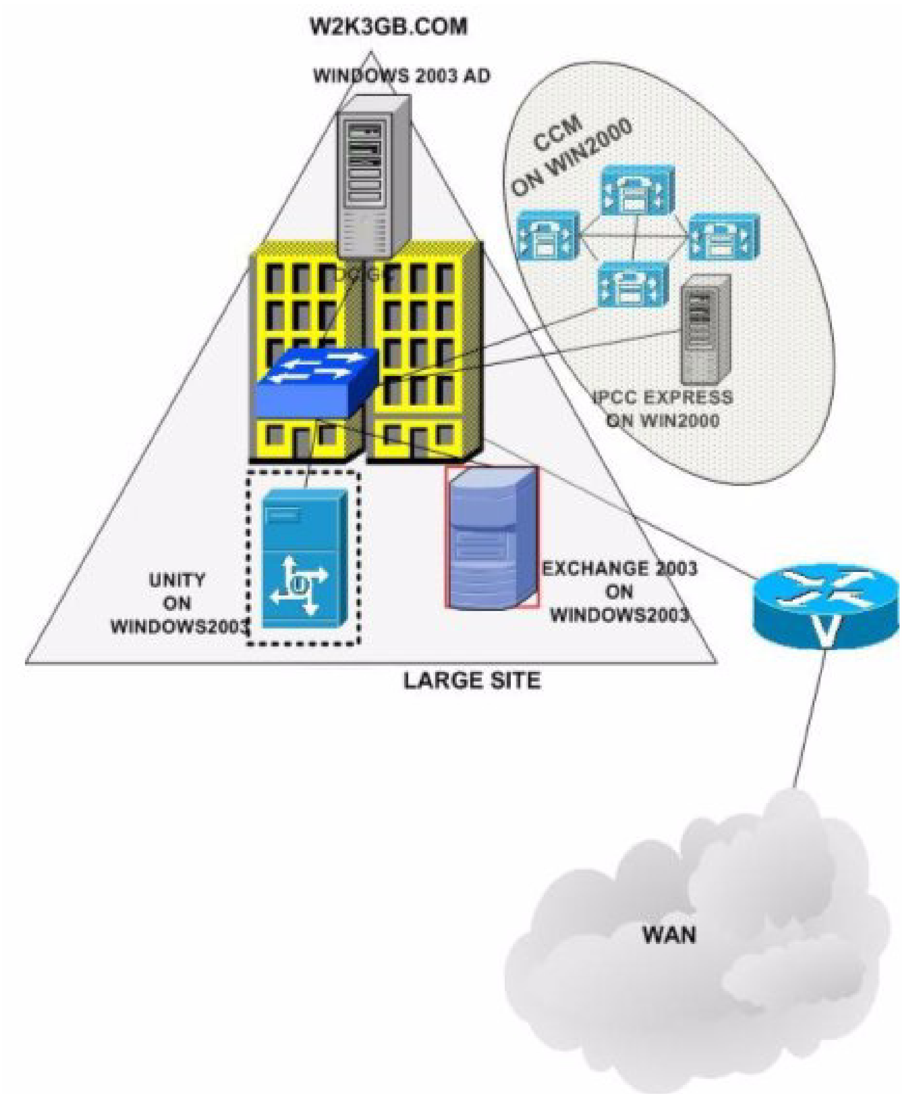
If you will use Microsoft Active Directory 2003, review the guidelines and references that are provided in this chapter.

This chapter includes the following topics:

Microsoft Active Directory 2003 Topology in a Large Site Model

Figure 11-1 shows the topology of Microsoft Active Directory 2003 when integrated with a Large Site model.

Figure 11-1 Microsoft Active Directory 2003 Topology in a Large 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 in the Large Site model.
- 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

In the Large Site model, a Windows 2003 server was promoted to Domain Controller. Cisco Unity and Microsoft Exchange were tested in that Microsoft Active Directory 2003 domain.

Hardware Configurations

This chapter provides sample configuration files and configuration information for selected hardware components in the Very Large Site model. The configuration files and information in this chapter apply to Multi-Site Distributed scenarios.

The this chapter includes the following topics:

Cisco 3660 and 3745 Gatekeeper Configuration

This section shows configuration files for the Cisco 3745 gatekeepers used in the Very Large Site model and the Cisco 3660 used in the Large Site model.

Two gatekeepers were configured:

- GK1—Primary gatekeeper. See the “GK1 Configuration” section on page 12-2
- GK2—Alternate gatekeeper See the “GK2 Configuration” section on page 12-4

For related information, refer to the Cisco High-Performance Gatekeeper documentation at this URL:

http://www.cisco.com/en/US/products/sw/iosswrel/ps5013/products_feature_guide09186a0080080e92.html

GK1 Configuration

This section shows a configuration file for the Cisco 3660 gatekeeper that was configured as the primary gatekeeper in the Very Large Site model.

```
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Very_Large_Site-GK1
!
logging queue-limit 100
logging buffered 1000000 debugging
enable password password
!
ip subnet-zero
!
voice call carrier capacity active
!
interface FastEthernet0/0
  no ip address
  shutdown
  duplex auto
  speed auto
!
interface FastEthernet0/1
  description Connection to CORE1 Port 3/38
  ip address 10.3.100.51 255.255.255.0
  speed 100
  full-duplex
!
ip http server
ip classless
ip route 0.0.0.0 0.0.0.0 10.3.100.1
!
access-list 101 deny   ip host 10.3.221.10 host 10.3.100.51
access-list 101 permit ip any any
snmp-server community public RO
snmp-server enable traps tty
```

```
!
dial-peer cor custom
!
gatekeeper
  zone local Very_Large_Site-GK1 cisco.com 10.3.100.51
  zone remote Large_Site-GK1 cisco.com 10.3.30.3 1719
  zone cluster local west-cluster Very_Large_Site-GK1
   element Very_Large_Site-GK2 10.3.100.52 1719
  !
  zone prefix Very_Large_Site-GK1 6011* gw-default-priority 0
  zone prefix Very_Large_Site-GK1 60120* gw-default-priority 0
  zone prefix Very_Large_Site-GK1 60139* gw-priority 10
Very_Large_Site_ict_to_cm3_4
Very_Large_Site_ict_to_cm4_5 Very_Large_Site_ict_to_cm2_3
Very_Large_Site_ict_to_cm1_2
  zone prefix Very_Large_Site-GK1 60139* gw-default-priority 0
  zone prefix Very_Large_Site-GK1 60144* gw-priority 10
Smal_Site_CallManager_2
  zone prefix Very_Large_Site-GK1 60144* gw-default-priority 0
  zone prefix Very_Large_Site-GK1 60172* gw-priority 10
Small_Site_CallManager_2
  zone prefix Very_Large_Site-GK1 60172* gw-default-priority 0
  zone prefix Very_Large_Site-GK1 60193* gw-priority 10
Medium_Site_CallManager_2
  zone prefix Very_Large_Site-GK1 60193* gw-priority 8
Medium_Site__CallManager_1
  zone prefix Very_Large_Site-GK1 60193* gw-default-priority 0
  zone prefix LArge_Site-GK1 *
  gw-type-prefix 1#* default-technology
  lrq forward-queries
  bandwidth total zone Very_Large_Site-GK1 4632000
  no shutdown
!
line con 0
line aux 0
line vty 0 4
  password password
  login
!
ntp clock-period 17180299
ntp server 172.18.137.110
!
end
```

GK2 Configuration

This section shows a configuration file for the Cisco 3660 gatekeeper that was configured as the alternate gatekeeper in the Very Large Site model.

```
version 12.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Very_Large_Site-GK2
!
logging queue-limit 100
enable password password
!
ip subnet-zero
!
voice call carrier capacity active
!
interface FastEthernet0/0
  no ip address
  shutdown
  duplex auto
  speed auto
!
interface FastEthernet0/1
  ip address 10.3.100.52 255.255.255.0
  ip access-group 101 in
  speed 100
  full-duplex
!
ip http server
ip classless
ip route 0.0.0.0 0.0.0.0 10.3.100.1
!
access-list 101 deny   ip host 10.3.221.10 host 10.3.100.52
access-list 101 permit ip any any
snmp-server community public RO
snmp-server enable traps tty
!
dial-peer cor custom
!
gatekeeper
  zone local Very_Large_Site-GK2 cisco.com 10.3.100.52
  zone cluster local west-cluster Very_Large_Site-GK2
   element Very_Large_Site-GK1 10.3.100.51 1719
  !
```

```
  zone prefix Very_Large_Site-GK2 6011* gw-default-priority 0
  zone prefix Very_Large_Site-GK2 60120* gw-default-priority 0
  zone prefix Very_Large_Site-GK2 60139* gw-priority 10
Very_Large_Site_ict_to_cm3_4
Very_Large_Site_ict_to_cm1_2 Very_Large_Site_ict_to_cm2_3
Very_Large_Site_ict_to_cm4_5
  zone prefix Very_Large_Site-GK2 60139* gw-default-priority 0
  zone prefix Very_Large_Site-GK2 60144* gw-priority 10
Small_Site__CallManager_2
  zone prefix Very_Large_Site-GK2 60144* gw-default-priority 0
  zone prefix Very_Large_Site-GK2 60172* gw-priority 10
Small_SIte_CallManager_2
  zone prefix Very_Large_Site-GK2 60172* gw-default-priority 0
  zone prefix Very_Large_Site-GK2 60193* gw-priority 10
Medium_Site__CallManager_2
  zone prefix Very_Large_Site-GK2 60193* gw-priority 8
Medium_Site__CallManager_1
  zone prefix Very_Large_Site-GK2 60193* gw-default-priority 0
  gw-type-prefix 1#* default-technology
  lrq forward-queries
  bandwidth total zone Very_Large_Site-GK2 4632000
  shutdown
!
line con 0
line aux 0
line vty 0 4
  password password
  login
!
ntp clock-period 17180297
ntp server 172.18.137.110
!
end
```

Standard MGCP Gateway Configuration File

This section shows a standard MGCP gateway configuration file.

```
version 12.3
service timestamps debug datetime msec localtime show-timezone
service timestamps log datetime msec localtime show-timezone
no service password-encryption
!
hostname SITE185-2611
!
boot-start-marker
```

```
boot system flash:c2600-ipvoice-mz.123-8.T5
boot-end-marker
!
no logging buffered
no logging console
enable password lab
!
memory-size iomem 20
clock timezone EDT -4
no network-clock-participate slot 1
no network-clock-participate wic 0
voice-card 1
!
no aaa new-model
ip subnet-zero
ip cef
ip tcp synwait-time 13
!
ip domain name hq.com
ip name-server 10.3.197.20
no ftp-server write-enable
isdn switch-type primary-ni
!
<note: mgcp configuration>
ccm-manager redundant-host SITE8-CM2
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server 10.3.197.11
ccm-manager config
!
controller T1 0/0
 framing esf
 linecode b8zs
 channel-group 1 timeslots 1-24 speed 64
!
controller T1 0/1
 framing esf
 clock source internal
 linecode b8zs
 channel-group 1 timeslots 1-24 speed 64
!
controller T1 0/2
 shutdown
 framing sf
 linecode ami
!
controller T1 0/3
 shutdown
```

```
 framing sf
 linecode ami
!
controller T1 1/0
 framing esf
 clock source internal
 linecode b8zs
 pri-group timeslots 1-24 service mgcp
!
controller T1 1/1
 framing esf
 clock source internal
 linecode b8zs
 pri-group timeslots 1-24 service mgcp
!
interface Multilink2
 ip address 10.3.181.162 255.255.255.252
 no cdp enable
 ppp multilink
 ppp multilink fragment delay 10
 ppp multilink interleave
 ppp multilink group 2
!
interface FastEthernet0/0
 ip address 10.3.195.162 255.255.255.248
 speed 100
 full-duplex
 no cdp enable
!
interface Serial0/0:1
 no ip address
 encapsulation frame-relay
 no keepalive
!
interface Serial0/0:1.1 point-to-point
 ip address 10.3.181.81 255.255.255.252
 frame-relay interface-dlci 185
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
 no cdp enable
!
interface Serial0/1:1
 bandwidth 64
 no ip address
```

```
 encapsulation ppp
 ppp multilink
 ppp multilink group 2
!
interface Serial1/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-ni
 isdn protocol-emulate network
 isdn incoming-voice voice
 isdn bind-l3 ccm-manager
 no cdp enable
!
interface Serial1/1:23
 no ip address
 no logging event link-status
 isdn switch-type primary-ni
 isdn protocol-emulate network
 isdn incoming-voice voice
 isdn bind-l3 ccm-manager
 no cdp enable
!
router ospf 8
 log-adjacency-changes
 network 10.3.181.80 0.0.0.3 area 0
 network 10.3.181.160 0.0.0.3 area 0
 network 10.3.195.160 0.0.0.7 area 185
!
ip classless
ip route 0.0.0.0 0.0.0.0 10.3.181.82
ip route 10.3.128.0 255.255.192.0 10.3.195.161
ip route 172.18.137.110 255.255.255.255 10.3.195.161
ip http server
!
logging trap debugging
logging 172.18.137.110
snmp-server community public RO
snmp-server enable traps snmp authentication linkdown linkup coldstart
warmstart
snmp-server enable traps tty
snmp-server enable traps xgcp
snmp-server enable traps envmon
snmp-server enable traps isdn call-information
snmp-server enable traps isdn layer2
snmp-server enable traps isdn chan-not-avail
snmp-server enable traps isdn ietf
snmp-server enable traps cnpd
snmp-server enable traps config
```

```
snmp-server enable traps dial
snmp-server enable traps dsp card-status
snmp-server enable traps entity
snmp-server enable traps frame-relay
snmp-server enable traps frame-relay subif
snmp-server enable traps hsrp
snmp-server enable traps ipmobile
snmp-server enable traps ipmulticast
snmp-server enable traps pim neighbor-change rp-mapping-change
invalid-pim-message
snmp-server enable traps pppoe
snmp-server enable traps rsvp
snmp-server enable traps rtr
snmp-server enable traps syslog
snmp-server enable traps vtp
snmp-server enable traps voice poor-qov
snmp-server enable traps dnis
snmp-server host 10.3.111.100 public
!
control-plane
!
voice-port 1/0:23
!
voice-port 1/1:23
!
<mgcp config>
mgcp
mgcp call-agent SITE8-CM1 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000 action notify
mgcp package-capability rtp-package
no mgcp package-capability res-package
mgcp package-capability sst-package
no mgcp package-capability fxr-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 inhibit
mgcp rtp payload-type g726r16 static
!
mgcp profile default
!
dial-peer cor custom
!
dial-peer voice 1 pots
 destination-pattern 9T
 incoming called-number T
 direct-inward-dial
 port 1/0:23
```

```
=dial-peer voice 2 pots
 destination-pattern 9T
 incoming called-number T
 direct-inward-dial
 port 1/1:23
!
<srst config>
call-manager-fallback
 max-conferences 4
 ip source-address 10.3.195.162 port 2000
 max-ephones 48
 max-dn 192
 dialplan-pattern 1 485185.... extension-length 4
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 4
 password lab
 login
!
ntp clock-period 17208177
ntp server 172.18.137.110
!
end
```

Standard H.323 Gateway Configuration File

This section shows a standard H.323 gateway configuration file.

```
version 12.3
service timestamps debug datetime msec localtime show-timezone
service timestamps log datetime msec localtime show-timezone
no service password-encryption
!
hostname SITE160-2610
!
boot-start-marker
boot system flash:c2600-ipvoice-mz.123-8.T5
boot-end-marker
!
no logging buffered
no logging console
enable password lab
!
memory-size iomem 20
```

```
clock timezone EDT -4
no network-clock-participate slot 1
no network-clock-participate wic 0
voice-card 1
!
no aaa new-model
ip subnet-zero
ip cef
ip tcp synwait-time 13
!
ip domain name hq.com
ip name-server 10.3.197.20
no ftp-server write-enable
isdn switch-type primary-ni
!
voice service voip
 h323
!
voice class h323 1
!
controller T1 0/0
 framing esf
 linecode b8zs
 channel-group 1 timeslots 1-24 speed 64
!
controller T1 0/1
 shutdown
 framing sf
 linecode ami
!
controller T1 1/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
!
controller T1 1/1
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
!
<h.323 gw config>
!
interface FastEthernet0/0
 ip address 10.3.194.218 255.255.255.248
 speed 100
 full-duplex
 no cdp enable
 h323-gateway voip bind srcaddr 10.3.194.218
```

```
!
interface Serial0/0:1
 no ip address
 encapsulation frame-relay
 no keepalive
!
interface Serial0/0:1.1 point-to-point
 ip address 10.3.180.237 255.255.255.252
 frame-relay interface-dlci 160
!
interface Serial1/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-ni
 isdn incoming-voice voice
 no isdn outgoing ie redirecting-number
 no isdn incoming alerting add-PI
 no cdp enable
!
interface Serial1/1:23
 no ip address
 no logging event link-status
 isdn switch-type primary-ni
 isdn incoming-voice voice
 no isdn outgoing ie redirecting-number
 no isdn incoming alerting add-PI
 no cdp enable
!
router ospf 8
 log-adjacency-changes
 network 10.3.180.236 0.0.0.3 area 0
 network 10.3.194.216 0.0.0.7 area 160
!
ip classless
ip route 0.0.0.0 0.0.0.0 10.3.180.238
ip route 10.3.128.0 255.255.192.0 10.3.194.217
ip route 10.3.197.10 255.255.255.255 10.3.180.236
ip route 172.18.137.110 255.255.255.255 10.3.194.217
ip http server
!
logging trap debugging
logging 172.18.137.110
snmp-server community public RO
snmp-server enable traps snmp authentication linkdown linkup coldstart
warmstart
snmp-server enable traps tty
snmp-server enable traps xgcp
snmp-server enable traps envmon
```

```
snmp-server enable traps isdn call-information
snmp-server enable traps isdn layer2
snmp-server enable traps isdn chan-not-avail
snmp-server enable traps isdn ietf
snmp-server enable traps cnpd
snmp-server enable traps config
snmp-server enable traps dial
snmp-server enable traps dsp card-status
snmp-server enable traps entity
snmp-server enable traps frame-relay
snmp-server enable traps frame-relay subif
snmp-server enable traps hsrp
snmp-server enable traps ipmobile
snmp-server enable traps ipmulticast
snmp-server enable traps pim neighbor-change rp-mapping-change
invalid-pim-message
snmp-server enable traps pppoe
snmp-server enable traps rsvp
snmp-server enable traps rtr
snmp-server enable traps syslog
snmp-server enable traps vtp
snmp-server enable traps voice poor-qov
snmp-server enable traps dnis
snmp-server host 10.3.111.100 public
no cdp run
!
control-plane
!
voice-port 1/0:23
!
voice-port 1/1:23
!
no mgcp timer receive-rtcp
!
dial-peer cor custom
!
<h.323 gw config>
!
dial-peer voice 3 voip
 preference 1
 destination-pattern T
 progress_ind setup enable 3
 session target ipv4:10.3.197.67
!
dial-peer voice 4 voip
 preference 1
 destination-pattern T
 progress_ind setup enable 3
```

```
 session target ipv4:10.3.197.14
!
dial-peer voice 1 pots
 preference 2
 destination-pattern T
 no digit-strip
 direct-inward-dial
 port 1/0:23
 forward-digits all
!
dial-peer voice 2 pots
 preference 2
 incoming called-number T
 no digit-strip
 direct-inward-dial
 port 1/1:23
 forward-digits all
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 4
 password lab
 login
!
ntp clock-period 17208361
ntp server 172.18.137.110
!
end
```

Cisco 3745 CAMA Gateway Configuration

For the Cisco 3745 gateway used in the Very Large Site model, the standard H.323 gateway configuration file was used, with the following additions. These lines are required for CAMA (Centralized Automatic Message Accounting) operation.

```
voice-port 4/1/0
signal cama KP-0-NPA-NXX-XXXX-ST
```

For related information, refer to the Analog Centralized Automatic Message Accounting E911 Trunk documentation at this URL:

http://www.cisco.com/en/US/products/sw/iosswrel/ps1839/products_feature_guide09186a00800b5d63.html

Fax, Modem, and TTY/TDD Configurations

This chapter provides an overview of how gateway devices were configured for fax/modem pass-through and fax relay modes for IP Communications Systems Test Release 3.0 for North America IPT. These configurations were designed to provide the following:

- Interoperability between Media Gateway Control Protocol (MGCP) and H.323 gateway devices.
- Fax/modem call flows that support IP-to-IP, IP-to-PSTN, and PSTN-to-IP calls. PSTN calls may be either T1 or E1 in origination and termination.

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

This chapter includes the following topics:

Overview

Gateway devices were manually configured using a command line interface (CLI), with the exception of the VG248, Digital-CMM, Digital-6608, and Analog-6624. These four devices were configured through a CLI or from

Cisco CallManager Administration. When you configure these devices for fax using Cisco CallManager Administration, you perform this configuration on a per-port basis. See the "Cisco CallManager Device Configuration" section on page 2-29 for more information.

Port-specific parameters for the VG248, such as MWI method, Input/Output gain, and fax relay are configurable from the VG248 console and from Cisco CallManager Administration. (These parameters are called Product Specific Configuration in Cisco CallManager Administration.) However, unlike it does for the IOS MGCP gateways, Cisco CallManager does not download the Product Specific Configuration parameters (port specific parameters) to the VG248 gateway. You must manually configure these parameters on the VG248 console for them to be effective.

Note For bandwidth consumption for fax up speed (G.729 to G.711), Cisco CallManager is unaware of the increase in bandwidth use after the negotiated codec switches to G.711. This situation can reduce the efficiency of the Call Admission Control (CAC) mechanisms (location-based CAC or Gatekeeper-CAC). In this case, the bandwidth calculation for CAC is no longer a simple function of the Codec bandwidth usage times the number of concurrent calls to support, and some buffer should be allowed for fax up speed.

There are no special configuration commands on the gateway devices for TTY/TDD, except that all call flows must be G.711 codec regions and have a less than 0.1% RTP packet drop rate in the network. Packet drop rates above this threshold could result in TTY/TDD calls having error rate above the comfortable threshold for meaningful conversations.

Fax/Modem Pass-Through/Up Speed Configuration

The following sections show configuration files for H.323 and MGCP gateway devices in fax/modem pass-through mode:

- H.323 Fax/Modem Pass-Through Configuration, page 13-3
- MGCP Fax/Modem Pass-Through Configuration, page 13-8

H.323 Fax/Modem Pass-Through Configuration

This section shows a H.323 fax/modem pass-through configuration file.

The following commands must be configured for the H.323 gateway:

- `modem passthrough nse codec g711ulaw`
- `fax protocol pass-through g711 ulaw`

If the `fax protocol pass-through g711 ulaw` command is missing, the gateway device will always perform fax relay, even if modem pass-through is configured.

```
version 12.2
service nagle
no service pad
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
!
hostname ap1-2611-h
!
logging queue-limit 100
!
clock timezone SG 8
voice-card 1
!
ip subnet-zero
!
no ip domain lookup
ip domain name lab.com
!
isdn switch-type primary-ni
!
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g711alaw
 codec preference 3 g729br8
 codec preference 4 g729r8
!
voice class h323 1
 h225 timeout tcp establish 5
!
no voice hpi capture buffer
no voice hpi capture destination
!
mta receive maximum-recipients 0
!
```

```
controller T1 1/0
 framing esf
 linecode b8zs
 cablelength short 133
 pri-group timeslots 1-24
!
controller T1 1/1
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
!
interface FastEthernet0/0
 ip address 10.0.0.0 192.168.0.0
 speed 100
 full-duplex
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface Serial1/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-ni
 isdn incoming-voice voice
 no cdp enable
!
interface Serial1/1:23
 no ip address
 no logging event link-status
 isdn switch-type primary-ni
 isdn protocol-emulate network
 isdn incoming-voice voice
 no cdp enable
!
no ip http server
ip classless
ip route 0.0.0.0 0.0.0.0 10.0.0.0
!
snmp-server community private RW
snmp-server community public RO
snmp-server enable traps tty
snmp-server host 10.0.0.0 version 2c private
call rsvp-sync
!
voice-port 1/0:23
```

```
!
voice-port 1/1:23
!
mgcp profile default
!
dial-peer cor custom
!
dial-peer voice 1 pots
 description Incoming POTS
 incoming called-number .T
 direct-inward-dial
 forward-digits all
!
dial-peer voice 2 voip
 description Outgoing VOIP to primary CCM ap1-cm1
 preference 10
 destination-pattern 1.......
 progress_ind setup enable 3
 modem passthrough nse codec g711ulaw
 voice-class codec 1
 voice-class h323 1
 session target ipv4:10.0.0.0
 dtmf-relay h245-alphanumeric
 fax protocol pass-through g711ulaw
!
dial-peer voice 3 voip
 description Outgoing VOIP to secondary CCM ap1-pub
 preference 5
 destination-pattern 1.......
 progress_ind setup enable 3
 modem passthrough nse codec g711ulaw
 voice-class codec 1
 voice-class h323 1
 session target ipv4:10.0.0.0
 dtmf-relay h245-alphanumeric
 fax protocol pass-through g711ulaw
!
dial-peer voice 26 pots
 description Outgoing POTS to other gateway via PSTN
 destination-pattern [2-6].........
 port 1/0:23
 forward-digits all
!
dial-peer voice 98 pots
 description Outgoing POTS to DCOSS
 destination-pattern 98........
 port 1/1:23
 forward-digits all
```

```
!
dial-peer voice 99 pots
 description Outgoing POTS to PSTN
 destination-pattern 99........
 port 1/0:23
 forward-digits all
!
dial-peer voice 71 voip
 description outgoing VOIP to secondary CCM ap1-pub
 preference 10
 destination-pattern 7.............
 progress_ind setup enable 3
 modem passthrough nse codec g711ulaw
 voice-class codec 1
 voice-class h323 1
 session target ipv4:10.0.0.0
 fax protocol pass-through g711ulaw
!
dial-peer voice 72 voip
 description outgoing VOIP to primary CCM ap1-cm1
 preference 5
 destination-pattern 7.............
 progress_ind setup enable 3
 modem passthrough nse codec g711ulaw
 voice-class codec 1
 voice-class h323 1
 session target ipv4:10.0.0.0
 fax protocol pass-through g711ulaw
!
dial-peer voice 81 voip
 description outgoing VOIP to secondary CCM ap1-pub
 preference 10
 destination-pattern 8........
 progress_ind setup enable 3
 modem passthrough nse codec g711ulaw
 voice-class codec 1
 voice-class h323 1
 session target ipv4:10.0.0.0
 fax protocol pass-through g711ulaw
!
dial-peer voice 82 voip
 description outgoing VOIP to primary CCM ap1-cm1
 preference 5
 destination-pattern 8........
 progress_ind setup enable 3
 modem passthrough nse codec g711ulaw
 voice-class codec 1
 voice-class h323 1
```

```
 session target ipv4:10.0.0.0
 fax protocol pass-through g711ulaw
!
dial-peer voice 91 voip
 description outgoing VOIP to secondary CCM ap1-pub
 preference 10
 destination-pattern 9...........
 progress_ind setup enable 3
 modem passthrough nse codec g711ulaw
 voice-class codec 1
 voice-class h323 1
 session target ipv4:10.0.0.0
 fax protocol pass-through g711ulaw
!
dial-peer voice 92 voip
 description outgoing VOIP to primary CCM ap1-cm1
 preference 5
 destination-pattern 9...........
 progress_ind setup enable 3
 modem passthrough nse codec g711ulaw
 voice-class codec 1
 voice-class h323 1
 session target ipv4:10.0.0.0
 fax protocol pass-through g711ulaw
!
dial-peer voice 100 voip
 incoming called-number T
 modem passthrough nse codec g711ulaw
 voice-class codec 1
 voice-class h323 1
 fax protocol pass-through g711ulaw
!
dial-peer voice 90 pots
 description Outgoing Modem to PSTN
 destination-pattern 90........
 port 1/0:23
 forward-digits all
!
line con 0
line aux 0
line vty 0 4
 login
!
ntp clock-period 17208230
ntp server 10.0.0.0
!
end
```

MGCP Fax/Modem Pass-Through Configuration

This section shows a MGCP fax/modem pass-through configuration file.

The following commands must be configured for the MGCP gateway:

- `no ccm-manager fax protocol cisco`
- `mgcp modem passthrough voip mode nse`

If the `no ccm-manager fax protocol cisco` command is missing, the gateway device will always perform fax relay, even if modem pass-through is configured

```
version 12.2
service nagle
no service pad
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
!
hostname ap1-2691-m
!
logging queue-limit 100
logging buffered 3000000 debugging
!
clock timezone SG 8
voice-card 1
 dspfarm
!
ip subnet-zero
ip tcp synwait-time 13
!
!
no ip domain lookup
ip domain name lab.com
!
isdn switch-type primary-ni
!
no voice hpi capture buffer
no voice hpi capture destination
!
ccm-manager redundant-host 10.0.0.0
ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager music-on-hold
ccm-manager config server 10.0.0.0
ccm-manager config
mta receive maximum-recipients 0
```

```
!
controller T1 1/0
 framing esf
 linecode b8zs
 cablelength short 133
 pri-group timeslots 1-24 service mgcp
!
controller T1 1/1
 framing esf
 clock source internal
 linecode b8zs
 pri-group timeslots 1-24 service mgcp
!
interface FastEthernet0/0
 ip address 10.0.0.0 192.168.0.0
 speed 100
 full-duplex
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface Serial1/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-ni
 isdn incoming-voice voice
 isdn bind-l3 ccm-manager
 no cdp enable
!
interface Serial1/1:23
 no ip address
 no logging event link-status
 isdn switch-type primary-ni
 isdn protocol-emulate network
 isdn incoming-voice voice
 isdn bind-l3 ccm-manager
 no cdp enable
!
no ip http server
ip classless
ip route 0.0.0.0 0.0.0.0 10.0.0.0
!
snmp-server community private RW
snmp-server community public RO
snmp-server enable traps tty
```

```
snmp-server host 10.0.0.0 version 2c private
!
call rsvp-sync
!
voice-port 1/0:23
!
voice-port 1/1:23
!
mgcp
mgcp call-agent 110.0.0.0 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp package-capability rtp-package
mgcp package-capability sst-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 inhibit
mgcp rtp payload-type g726r16 static
!
mgcp profile default
!
dial-peer cor custom
!
line con 0
line aux 0
line vty 0 4
 login
!
ntp clock-period 17180383
ntp server 10.0.0.0
end
```

Fax Relay Configuration

The following sections show configuration files for H.323 and MGCP gateway devices in fax relay mode:

- H.323 Fax Relay Configuration, page 13-11
- MGCP Fax Relay Configuration, page 13-16

H.323 Fax Relay Configuration

This section shows an H.323 fax relay configuration file.

The following commands must be configured for the H.323 gateway:

- `no modem passthrough nse codec g711ulaw`
- `no fax protocol pass-through g711 ulaw`

Cisco also recommends that you add the `fax rate` command to limit the bandwidth.

```
version 12.2
service nagle
no service pad
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
!
hostname ap2-3745-h
!
logging queue-limit 100
logging buffered 1000000 debugging
no logging console
!
clock timezone SG 8
voice-card 1
 no dspfarm
!
ip subnet-zero
!
no ip domain lookup
ip domain name lab.com
!
isdn switch-type primary-net5
!
voice call carrier capacity active
!
voice service voip
!
voice class codec 1
 codec preference 1 g711alaw
 codec preference 2 g711ulaw
 codec preference 3 g729r8
!
voice class h323 1
 h225 timeout tcp establish 5
```

```
!
no voice hpi capture buffer
no voice hpi capture destination
!
mta receive maximum-recipients 0
!
controller E1 1/0
 framing NO-CRC4
 pri-group timeslots 1-31
!
controller E1 1/1
 framing NO-CRC4
 pri-group timeslots 1-31
!
interface FastEthernet0/0
 ip address 10.0.0.0 192.168.0.0
 speed 100
 full-duplex
!
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 incoming-voice voice
 no cdp enable
!
interface Serial1/1:15
 no ip address
 no logging event link-status
 isdn switch-type primary-net5
 isdn protocol-emulate network
 isdn incoming-voice voice
 no isdn outgoing display-ie
 no cdp enable
!
no ip http server
ip classless
ip route 0.0.0.0 0.0.0.0 10.0.0.0
!
snmp-server community private RW
snmp-server community public RO
snmp-server enable traps tty
```

```
snmp-server host 10.0.0.0 version 2c private
!
call rsvp-sync
!
voice-port 1/0:15
!
voice-port 1/1:15
!
voice-port 3/0/0
!
voice-port 3/0/1
!
voice-port 3/1/0
!
voice-port 3/1/1
 no battery-reversal
!
mgcp profile default
!
dial-peer cor custom
!
dial-peer voice 1 pots
 description incoming POTS
 incoming called-number .T
 direct-inward-dial
 forward-digits all
!
dial-peer voice 2 voip
 description outgoing VOIP to secondary CCM ap2-pub
 preference 10
 destination-pattern 2.......
 progress_ind setup enable 3
 no modem passthrough
 voice-class codec 1
 voice-class h323 1
 session target ipv4:10.0.0.0
 fax rate 14400
!
dial-peer voice 3 voip
 description outgoing VOIP to primary CCM ap2-cm1
 preference 5
 destination-pattern 2.......
 progress_ind setup enable 3
 no modem passthrough
 voice-class codec 1
 voice-class h323 1
 session target ipv4:10.0.0.0
 fax rate 14400
```

```
!
dial-peer voice 98 pots
 description Outgoing POTS to DCOSS
 destination-pattern 98........
 port 1/1:15
 forward-digits all
!
dial-peer voice 99 pots
 description Outgoing POTS to PSTN
 destination-pattern 99........
 port 1/0:15
 forward-digits all
!
dial-peer voice 26 pots
 description Outgoing POTS to other gateway (via PSTN)
 destination-pattern [2-6].........
 port 1/0:15
!
dial-peer voice 20500200 pots
 description FXS Port
 destination-pattern 20500200
 port 3/1/0
!
dial-peer voice 20500201 pots
 description FXS Port
 destination-pattern 20500201
 port 3/1/1
!
dial-peer voice 71 voip
 description outgoing VOIP to secondary CCM ap2-pub
 preference 10
 destination-pattern 7.............
 progress_ind setup enable 3
 no modem passthrough
 voice-class codec 1
 voice-class h323 1
 session target ipv4:10.0.0.0
 fax rate 14400
!
dial-peer voice 72 voip
 description outgoing VOIP to primary CCM ap2-cm1
 preference 5
 destination-pattern 7.............
 progress_ind setup enable 3
 no modem passthrough
 voice-class codec 1
 voice-class h323 1
 session target ipv4:10.0.0.0
```

```
 fax rate 14400
!
dial-peer voice 81 voip
 description outgoing VOIP to secondary CCM ap2-pub
 preference 10
 destination-pattern 8........
 progress_ind setup enable 3
 no modem passthrough
 voice-class codec 1
 voice-class h323 1
 session target ipv4:10.0.0.0
 fax rate 14400
!
dial-peer voice 82 voip
 description outgoing VOIP to primary CCM ap2-cm1
 preference 5
 destination-pattern 8........
 progress_ind setup enable 3
 no modem passthrough
 voice-class codec 1
 voice-class h323 1
 session target ipv4:10.0.0.0
 fax rate 14400
!
dial-peer voice 91 voip
 description outgoing VOIP to secondary CCM ap2-pub
 preference 10
 destination-pattern 9............
 progress_ind setup enable 3
 no modem passthrough
 voice-class codec 1
 voice-class h323 1
 session target ipv4:10.0.0.0
 fax rate 14400
!
dial-peer voice 92 voip
 description outgoing VOIP to primary CCM ap2-cm1
 preference 5
 destination-pattern 9............
 progress_ind setup enable 3
 no modem passthrough
 voice-class codec 1
 voice-class h323 1
 session target ipv4:10.0.0.0
 fax rate 14400
!
dial-peer voice 100 voip
 incoming called-number T
```

```
 no modem passthrough
 voice-class codec 1
 voice-class h323 1
 fax rate 14400
!
dial-peer voice 20500210 pots
 description FXS port
 destination-pattern 20500210
 port 3/0/0
!
dial-peer voice 20500211 pots
 description FXS port
 destination-pattern 20500211
 port 3/0/1
!
dial-peer voice 90 pots
 description Outgoing Modem to PSTN
 destination-pattern 90........
 port 1/0:15
 forward-digits all
!
line con 0
line aux 0
line vty 0 4
 exec-timeout 0 0
 login
!
ntp clock-period 17174990
ntp server 10.0.0.0
end
```

MGCP Fax Relay Configuration

This section shows an MGCP fax relay configuration file.

The following command must be configured for the MGCP gateway:

```
ccm-manager fax protocol cisco
```

> **Note** This command may not appear in the running config output from the show run command. To verify the fax mode is Cisco Fax Relay, enter **show ccm** on the MGCP gateway. The output will show `FAX mode: cisco` for fax relay mode.

```
version 12.2
```

```
service nagle
no service pad
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
!
hostname ap2-3640-m
!
logging queue-limit 100
logging buffered 4000000 debugging
!
clock timezone SG 8
voice-card 1
!
ip subnet-zero
ip tcp synwait-time 13
!
no ip domain lookup
ip domain name lab.com
!
isdn switch-type primary-net5
!
voice call carrier capacity active
!
voice class codec 1
 codec preference 1 g711alaw
 codec preference 2 g711ulaw
 codec preference 3 g729r8
!
no voice hpi capture buffer
no voice hpi capture destination
!
mta receive maximum-recipients 0
ccm-manager redundant-host 10.0.0.0
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server 10.0.0.0
ccm-manager config
!
controller E1 1/0
 framing NO-CRC4
 pri-group timeslots 1-31 service mgcp
!
controller E1 1/1
 framing NO-CRC4
 pri-group timeslots 1-31 service mgcp
!
interface FastEthernet0/0
```

```
 ip address 10.0.0.0 192.168.0.0
 speed 100
 full-duplex
!
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 incoming-voice voice
 isdn bind-l3 ccm-manager
 no cdp enable
!
interface Serial1/1:15
 no ip address
 no logging event link-status
 isdn switch-type primary-net5
 isdn protocol-emulate network
 isdn incoming-voice voice
 isdn bind-l3 ccm-manager
 no cdp enable
!
no ip http server
ip classless
ip route 0.0.0.0 0.0.0.0 10.0.0.0
!
snmp-server community private RW
snmp-server community public RO
snmp-server enable traps tty
snmp-server host 10.0.0.0 version 2c private
call rsvp-sync
!
voice-port 1/0:15
 cptone SG
!
voice-port 1/1:15
 cptone SG
!
voice-port 2/0/0
 cptone SG
!
voice-port 2/0/1
 cptone SG
```

```
!
voice-port 2/1/0
 cptone SG
!
voice-port 2/1/1
 cptone SG
!
mgcp
mgcp call-agent 10.0.0.0 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000 action notify
mgcp package-capability rtp-package
mgcp package-capability sst-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 inhibit
mgcp rtp payload-type g726r16 static
!
mgcp profile default
!
dial-peer cor custom
!
dial-peer voice 999200 pots
 application mgcpapp
 port 2/0/0
!
dial-peer voice 999201 pots
 application mgcpapp
 port 2/0/1
!
dial-peer voice 999210 pots
 application mgcpapp
 port 2/1/0
!
dial-peer voice 999211 pots
 application mgcpapp
 port 2/1/1
!
line con 0
line aux 0
line vty 0 4
 exec-timeout 67 6
 login
!
ntp clock-period 17179708
ntp server 10.0.0.0
end
```

Quality of Service Configuration

This chapter provides sample configuration files that show how Quality of Service (QoS) settings were configured on selected switches and WAN routers for IP Communications System Test Release 3.0 for North America IPT.

This chapter includes the following topics:

Cisco Catalyst 6506 Switch Configuration

This section shows a configuration file for the Cisco Catalyst 6506 switch that was used in the QoS testing.

For additional installation and configuration information, refer to the Cisco Catalyst 6500 series documentation at this URL:

http://www.cisco.com/univercd/cc/td/doc/product/lan/cat6000/index.htm

```
#dot1x
set dot1x system-auth-control disable
set dot1x shutdown-timeout 0
set feature dot1x-radius-keepalive disable
!
#system
set system name  ap1-6509
set system location Lab
```

```
set system contact IPT@cisco.com
set system highavailability enable
set system highavailability versioning enable
set system core-dump enable
!
#Default Inlinepower
set inlinepower defaultallocation 7000
!
#stp mode
set spantree mode pvst+
!
#vtp
set vtp domain lab
set vtp mode transparent vlan
set vtp version 2
set vlan 102 name Gatekeepers type ethernet mtu 1500 said 100102 state
active
set vlan 103 name CallManagers type ethernet mtu 1500 said 100103
state active
set vlan 104 name Digital_Gateways type ethernet mtu 1500 said 100104
state active
set vlan 105 name Analog_Gateways type ethernet mtu 1500 said 100105
state active
set vlan 106 name Server_Farm type ethernet mtu 1500 said 100106 state
active
set vlan 107 name Tools type ethernet mtu 1500 said 100107 state
active
set vlan 113 name Lab_Backbone type ethernet mtu 1500 said 100113
state active
set vlan 610 name Data1 type ethernet mtu 1500 said 100610 state
active
set vlan 611 name Voice1 type ethernet mtu 1500 said 100611 state
active
set vlan 710 name Data2 type ethernet mtu 1500 said 100710 state
active
set vlan 711 name Voice2 type ethernet mtu 1500 said 100711 state
active
set vlan 810 name Data3 type ethernet mtu 1500 said 100810 state
active
set vlan 811 name Voice3 type ethernet mtu 1500 said 100811 state
active
set vlan 910 name Data4 type ethernet mtu 1500 said 100910 state
active
set vlan 911 name Voice4 type ethernet mtu 1500 said 100911 state
active
set vlan 1002 name fddi-default type fddi mtu 1500 said 101002 state
active
```

```
set vlan 1004 name fddinet-default type fddinet mtu 1500 said 101004
state active bridge 0x0 stp ieee
set vlan 1005 name trbrf-default type trbrf mtu 4472 said 101005 state
active bridge 0xf stp ibm
set vlan 1,888,999
set vlan 1025 name
set vlan 1026 name
set vlan 1027 name
set vlan 1028 name
set vlan 1029 name
set vlan 1003 name trcrf-default type trcrf mtu 4472 said 101003 state
active parent 1005 ring 0xccc mode srb aremaxhop 7 stemaxhop 7
backupcrf off
!
set interface sc0 113 10.0.0.0/255.255.255.192 10.0.0.1
set interface sc1 0 0.0.0.0/0.0.0.0 0.0.0.0
set interface sc1 down
set ip route 0.0.0.0/0.0.0.0         10.0.0.2
set ip alias default         0.0.0.0
!
#ntp
set ntp client enable
set timezone SG 8 0
!
#set boot command
set boot config-register 0x102
set boot config-register auto-config sync enable
set boot system flash bootflash:cat6000-sup2cvk8.8-1-1.bin
set boot system flash bootflash:cat6000-sup2cvk8.7-6-2.bin
!
#mls
set mls agingtime long-duration 1920
set mls agingtime 256
set mls agingtime ipx 256
!
#qos
set qos enable
set qos map 2q2t tx 2 1 cos 1
set qos map 2q2t tx 2 1 cos 2
set qos map 2q2t tx 2 1 cos 3
set qos map 2q2t tx 2 2 cos 5
set qos drop-threshold 2q2t tx queue 1 100 100
set qos map 1p1q4t rx 1 3 cos 1
set qos map 1p1q4t rx 1 3 cos 2
set qos map 1p1q4t rx 1 3 cos 3
set qos map 1p1q4t rx 1 4 cos 6
set qos map 1p2q2t tx 2 1 cos 1
set qos map 1p2q2t tx 2 1 cos 2
```

```
set qos map 1p2q2t tx 2 1 cos 3
set qos map 1p2q2t tx 2 2 cos 6
set qos wrr 1p2q2t 50 255
set qos wred 1p2q2t tx queue 1 70:100 70:100
set qos wred 1p2q2t tx queue 2 70:90 100:100
set qos map 1p3q1t tx 2 1 cos 1
set qos map 1p3q1t tx 3 1 cos 3
set qos map 1p3q1t tx 3 1 cos 4
set qos map 1p3q1t tx 3 cos 6
set qos map 1p3q1t tx 3 cos 7
set qos wrr 1p3q1t 20 100 200
set qos wred 1p3q1t tx queue 3 70:90
set qos map 1p1q0t rx 2 cos 6
set qos map 1p1q0t rx 2 cos 7
set qos map 1p2q1t tx 2 1 cos 1
set qos map 1p2q1t tx 2 1 cos 2
set qos map 1p2q1t tx 2 1 cos 3
set qos map 1p2q1t tx 2 cos 6
set qos map 1p2q1t tx 2 cos 7
set qos wrr 1p2q1t 50 255
set qos txq-ratio 1p2q1t 70 15 15
set qos wred 1p2q1t tx queue 2 70:90
set qos cos-dscp-map 0 10 18 26 34 46 48 56
set qos ipprec-dscp-map 0 10 18 26 34 46 48 56
set qos policed-dscp-map 0,24,46:0
set qos policed-dscp-map 1:1
set qos policed-dscp-map 2:2
set qos policed-dscp-map 3:3
set qos policed-dscp-map 4:4
set qos policed-dscp-map 5:5
set qos policed-dscp-map 6:6
set qos policed-dscp-map 7:7
set qos policed-dscp-map 8:8
set qos policed-dscp-map 9:9
set qos policed-dscp-map 10:10
set qos policed-dscp-map 11:11
set qos policed-dscp-map 12:12
set qos policed-dscp-map 13:13
set qos policed-dscp-map 14:14
set qos policed-dscp-map 15:15
set qos policed-dscp-map 16:16
set qos policed-dscp-map 17:17
set qos policed-dscp-map 18:18
set qos policed-dscp-map 19:19
set qos policed-dscp-map 20:20
set qos policed-dscp-map 21:21
set qos policed-dscp-map 22:22
set qos policed-dscp-map 23:23
```

```
set qos policed-dscp-map 25:25
set qos policed-dscp-map 26:26
set qos policed-dscp-map 27:27
set qos policed-dscp-map 28:28
set qos policed-dscp-map 29:29
set qos policed-dscp-map 30:30
set qos policed-dscp-map 31:31
set qos policed-dscp-map 32:32
set qos policed-dscp-map 33:33
set qos policed-dscp-map 34:34
set qos policed-dscp-map 35:35
set qos policed-dscp-map 36:36
set qos policed-dscp-map 37:37
set qos policed-dscp-map 38:38
set qos policed-dscp-map 39:39
set qos policed-dscp-map 40:40
set qos policed-dscp-map 41:41
set qos policed-dscp-map 42:42
set qos policed-dscp-map 43:43
set qos policed-dscp-map 44:44
set qos policed-dscp-map 45:45
set qos policed-dscp-map 47:47
set qos policed-dscp-map 48:48
set qos policed-dscp-map 49:49
set qos policed-dscp-map 50:50
set qos policed-dscp-map 51:51
set qos policed-dscp-map 52:52
set qos policed-dscp-map 53:53
set qos policed-dscp-map 54:54
set qos policed-dscp-map 55:55
set qos policed-dscp-map 56:56
set qos policed-dscp-map 57:57
set qos policed-dscp-map 58:58
set qos policed-dscp-map 59:59
set qos policed-dscp-map 60:60
set qos policed-dscp-map 61:61
set qos policed-dscp-map 62:62
set qos policed-dscp-map 63:63
clear qos acl all
#ACL_TRUST-COS
set qos acl ip ACL_TRUST-COS trust-cos ip any any
#ACL_TRUST-DSCP
set qos acl ip ACL_TRUST-DSCP trust-dscp ip any any
#ACL_TRUST-IPPREC
set qos acl ip ACL_TRUST-IPPREC trust-ipprec ip any any
#ACL_VOIP-ALL
set qos acl ip ACL_VOIP-ALL dscp 46 udp any any  range 16384 32767
set qos acl ip ACL_VOIP-ALL dscp 26 tcp any any  range 2000 2002
```

```
set qos acl ip ACL_VOIP-ALL dscp 26 tcp any any  eq 1720
set qos acl ip ACL_VOIP-ALL dscp 26 tcp any any  range 11000 11999
set qos acl ip ACL_VOIP-ALL dscp 26 udp any any  eq 2427
#ACL_IP-TRUSTDSCP
set qos acl ip ACL_IP-TRUSTDSCP trust-dscp ip any any
#
commit qos acl all
set qos acl map ACL_TRUST-COS 810,910
!
#port channel
set port channel 3/28 586
set port channel 3/25-27 592
!
#qos statistics data export
set qos statistics export enable
!
# default port status is enable
!
#module 1 : 2-port 1000BaseX Supervisor
set vlan 113  1/1
set port name       1/1  6509_1 3/6
set port dot1x 1/1 guest-vlan 0
!
#module 2 : 2-port 1000BaseX Supervisor
set port dot1x 2/1 guest-vlan 0
!
#module 3 : 48-port 10/100BaseTX Ethernet
set vlan 102  3/2
set vlan 103  3/3-4
set vlan 104  3/5-13,3/21,3/29
set vlan 105  3/14-15
set vlan 106  3/16
set vlan 107  3/17-20,3/23-25
set vlan 113  3/37
set vlan 610  3/39-40
set vlan 710  3/41-42
set vlan 810  3/43-44
set vlan 910  3/27-28,3/45-46
set vlan 999  3/30
set port auxiliaryvlan 3/39 611
set port auxiliaryvlan 3/40 611
set port auxiliaryvlan 3/41 711
set port auxiliaryvlan 3/42 711
set port auxiliaryvlan 3/43 811
set port auxiliaryvlan 3/44 811
set port auxiliaryvlan 3/45 911
set port auxiliaryvlan 3/46 911
set port qos 3/15 trust-ext trust-cos
```

```
set port speed
3/2-3,3/5-13,3/15-16,3/18-19,3/25,3/27-28,3/39,3/45  100
set port duplex     3/2-3,3/5-13,3/15-16,3/18-19,3/25,3/27-28  full
set port name       3/2  ap1-gk
set port name       3/3  ap1-pub
set port name       3/4  ap1-cm1
set port name       3/5  ap1-1760-m
set port name       3/6  ap1-2611-h
set port name       3/7  ap1-2621-m
set port name       3/8  ap1-2651-h
set port name       3/9  ap1-2691-m
set port name       3/10 ap1-3640-h
set port name       3/11 ap1-3662-m
set port name       3/12 ap1-3725-h
set port name       3/13 ap1-1760-h
set port name       3/14 ap1-ata186
set port name       3/15 ap1-ata188
set port name       3/16 ap1-dc1
set port name       3/17 ap-ciscoworks
set port name       3/18 ap-fr-switch2
set port name       3/19 ap-fr-switch1
set port name       3/20 ap-pstn1
set port name       3/23 ap-snooper1-qfe3
set port name       3/24 ixia-4-4
set port name       3/25 ap-irec1-eth1
set port name       3/27 Ixia port 5/1
set port name       3/29 PAGENT-fe0/1
set trunk 3/28 off negotiate 1-1005,1025-4094
set trunk 3/39 auto negotiate 1-1005,1025-4094
set trunk 3/41 auto negotiate 1-1005,1025-4094
set trunk 3/43 auto negotiate 1-1005,1025-4094
set trunk 3/45 auto negotiate 1-1005,1025-4094
set spantree portfast    3/28 enable
set port qos 3/6,3/11 cos 5
set port qos 3/39-46 trust trust-cos
set port qos 3/43-46 vlan-based
set qos acl map ACL_TRUST-COS 3/15,3/39-42
set qos acl map ACL_TRUST-DSCP 3/2-12,3/27-28
set qos acl map ACL_TRUST-IPPREC 3/29
set port qos 3/1-48 policy-source local
set port channel 3/27-28 mode off
!
#module 4 : 8-port E1
set port voice interface  4/1 dhcp enable vlan 104
set port voice interface  4/2 dhcp enable vlan 104
set port voice interface  4/3 dhcp enable vlan 104
set port voice interface  4/4 dhcp enable vlan 104
set port voice interface  4/5 dhcp enable vlan 104
```

```
set port voice interface  4/6 dhcp enable vlan 104
set port voice interface  4/7 dhcp enable vlan 104
set port voice interface  4/8 dhcp enable vlan 104
!
#module 5 : 8-port T1
set port voice interface  5/1 dhcp enable vlan 104
set port voice interface  5/2 dhcp enable vlan 104
set port voice interface  5/3 dhcp enable vlan 104
set port voice interface  5/4 dhcp enable vlan 104
set port voice interface  5/5 dhcp enable vlan 104
set port voice interface  5/6 dhcp enable vlan 104
set port voice interface  5/7 dhcp enable vlan 104
set port voice interface  5/8 dhcp disable 10.0.0.2 255.255.255.192
vlan 104
 tftp 10.0.0.3 gateway 10.0.0.4 dns 10.0.0.5 lab.com
!
#module 6 : 5-port Communication Media Mod.
set vlan 104  6/1
!
#module 7 : 5-port Communication Media Mod.
set vlan 104  7/1
!
#module 8 : 24-port FXS
set port voice interface  8/1-24 dhcp enable vlan 105
!
#module 9 : 0-port FlexWAN Module
!
#module 15 : 1-port Multilayer Switch Feature Card
!
#module 16 empty
end
```

Cisco 3725 Router Configuration

This section shows a serial interface configuration for the Cisco 3725 router that was used in the QoS testing.

For additional installation and configuration information, refer to the Cisco 3725 router documentation at this URL:

http://www.cisco.com/univercd/cc/td/doc/product/access/acs_mod/cis3700

```
version 12.2
service nagle
no service pad
```

```
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
!
hostname ap3-1-3725-hda-m
!
logging queue-limit 100
logging buffered 20000000 debugging
enable password Lab
!
clock timezone SG 8
voice-card 1
 no dspfarm
!
ip subnet-zero
ip tcp synwait-time 13
!
!
ip cef
no ip domain lookup
ip domain name lab.com
!
voice class codec 1
 codec preference 1 g711alaw
 codec preference 2 g711ulaw
 codec preference 3 g729r8
!
no voice hpi capture buffer
no voice hpi capture destination
!
ccm-manager redundant-host 10.0.0.1
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server 10.0.0.1
ccm-manager config
mta receive maximum-recipients 0
!
controller E1 0/0
 framing NO-CRC4
 channel-group 0 timeslots 1-12
!
controller E1 0/1
 framing NO-CRC4
!
 class-map match-all WAN-VOICE
  match ip dscp ef
 class-map match-all PRIORITY-DATA
   description ftp http tftp etc
```

```
  match access-group name PRIORITY-DATA
 class-map match-all WAN-VOICE-CONTROL
  match ip dscp af31
!
 policy-map WAN-EDGE-LFI
  class WAN-VOICE
   priority 84
  class WAN-VOICE-CONTROL
   bandwidth percent 5
  class PRIORITY-DATA
   bandwidth percent 15
  class class-default
   fair-queue
 policy-map WAN-EDGE-1PCT-LLQ
  class WAN-VOICE
   priority percent 1
  class WAN-VOICE-CONTROL
   bandwidth percent 2
  class PRIORITY-DATA
   bandwidth percent 20
  class class-default
   fair-queue
 policy-map WAN-EDGE
  description %35 for 252 Kbps = 3 G711 calls
  class WAN-VOICE
   priority percent 35
  class WAN-VOICE-CONTROL
   bandwidth percent 2
  class PRIORITY-DATA
   bandwidth percent 20
  class class-default
   fair-queue
!
interface Loopback0
 ip address 10.0.0.2 255.255.255.255
 h323-gateway voip bind srcaddr 10.0.0.2
!
interface FastEthernet0/0
 no ip address
 speed 100
 full-duplex
!
interface FastEthernet0/0.630
 encapsulation dot1Q 630
 ip address 10.0.0.3 255.255.255.240
 ip helper-address 10.0.0.4
 no cdp enable
!
```

```
interface FastEthernet0/0.631
 encapsulation dot1Q 631
 ip address 10.0.0.5 255.255.255.240
 ip helper-address 10.0.0.4
 no cdp enable
!
interface FastEthernet0/0.730
 encapsulation dot1Q 730
 ip address 10.0.0.6 255.255.255.240
 ip helper-address 10.0.0.4
 no cdp enable
!
interface FastEthernet0/0.731
 encapsulation dot1Q 731
 ip address 10.0.0.7 255.255.255.240
 ip helper-address 10.0.0.4
 no cdp enable
!
interface FastEthernet0/0.830
 encapsulation dot1Q 830
 ip address 10.0.0.8 255.255.255.240
 ip helper-address 10.0.0.4
 no cdp enable
!
interface FastEthernet0/0.831
 encapsulation dot1Q 831
 ip address 10.0.0.9 255.255.255.240
 ip helper-address 10.0.0.4
 no cdp enable
!
interface FastEthernet0/0.930
 encapsulation dot1Q 930
 ip address 10.0.0.10 255.255.255.240
 ip helper-address 10.0.0.4
 no cdp enable
!
interface FastEthernet0/0.931
 encapsulation dot1Q 931
 ip address 10.0.0.11 255.255.255.240
 ip helper-address 10.0.0.4
 no cdp enable
!
interface Serial0/0:0
 bandwidth 768
 ip address 10.0.0.12 255.255.255.252
 encapsulation frame-relay
 ip ospf network point-to-point
 ip ospf dead-interval 65000
```

```
 load-interval 30
 cdp enable
 frame-relay class FRTS-720k
 frame-relay traffic-shaping
 frame-relay interface-dlci 503
 frame-relay ip rtp header-compression
 frame-relay ip rtp compression-connections 3
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
router ospf 3
 router-id 10.0.0.2
 log-adjacency-changes
 network 10.0.0.13 0.0.255.255 area 113
!
no ip http server
ip classless
!
ip access-list extended PRIORITY-DATA
 permit tcp any eq www any
 permit tcp any any eq www
 permit tcp any range ftp-data telnet any
 permit tcp any any range ftp-data telnet
 permit udp any any eq tftp
 permit udp any eq tftp any
 permit tcp any range 135 139 any
 permit tcp any any range 135 139
 permit udp any any range 135 netbios-ss
!
map-class frame-relay FRTS-128k
 frame-relay cir 124872
 frame-relay bc 1250
 frame-relay be 0
 frame-relay mincir 124872
 service-policy output WAN-EDGE-LFI
!
map-class frame-relay FRTS-1896k
 frame-relay cir 1896000
 frame-relay bc 18960
 frame-relay be 0
 frame-relay mincir 1896000
 service-policy output WAN-EDGE
!
```

```
map-class frame-relay FRTS-112k
 frame-relay cir 109264
 frame-relay bc 1092
 frame-relay be 0
 frame-relay mincir 109264
 service-policy output WAN-EDGE-LFI
!
map-class frame-relay FRTS-720k
 frame-relay fragment 960
 frame-relay cir 720000
 frame-relay bc 7200
 frame-relay be 0
 frame-relay mincir 720000
 service-policy output WAN-EDGE
!
map-class frame-relay one-percent
 frame-relay cir 100000
 frame-relay bc 1000
 frame-relay be 0
 frame-relay mincir 100000
 service-policy output WAN-EDGE-1PCT-LLQ
!
snmp-server community private RW
snmp-server community public RO
snmp-server enable traps tty
snmp-server host 10.0.0.14 version 2c private
!
call rsvp-sync
!
voice-port 1/0/0
 no battery-reversal
!
voice-port 1/0/1
!
voice-port 1/0/2
!
voice-port 1/0/3
!
voice-port 1/0/4
!
voice-port 1/0/5
!
voice-port 1/0/6
!
voice-port 1/0/7
!
voice-port 1/0/8
!
```

```
voice-port 1/0/9
!
voice-port 1/0/10
!
voice-port 1/0/11
!
mgcp
mgcp call-agent 10.0.0.15 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000 action notify
mgcp package-capability rtp-package
mgcp package-capability sst-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 inhibit
mgcp rtp payload-type g726r16 static
!
mgcp profile default
!
dial-peer cor custom
!
dial-peer voice 999100 pots
 application mgcpapp
 port 1/0/0
!
dial-peer voice 999101 pots
 application mgcpapp
 port 1/0/1
!
dial-peer voice 999102 pots
 application mgcpapp
 port 1/0/2
!
dial-peer voice 999103 pots
 application mgcpapp
 port 1/0/3
!
dial-peer voice 999104 pots
 application mgcpapp
 port 1/0/4
!
dial-peer voice 999105 pots
 application mgcpapp
 port 1/0/5
!
dial-peer voice 999106 pots
 application mgcpapp
 port 1/0/6
```

```
!
dial-peer voice 999107 pots
 application mgcpapp
 port 1/0/7
!
dial-peer voice 999108 pots
 application mgcpapp
 port 1/0/8
!
dial-peer voice 999109 pots
 application mgcpapp
 port 1/0/9
!
dial-peer voice 9991010 pots
 application mgcpapp
 port 1/0/10
!
dial-peer voice 9991011 pots
 application mgcpapp
 port 1/0/11
!
line con 0
 exec-timeout 30 30
line aux 0
line vty 0 4
 exec-timeout 0 0
 password Lab
 login
!
ntp clock-period 17180586
ntp server 10.0.0.16

end
```

Cisco 7206 Router Configuration

This section shows an HSSI configuration for the Cisco 7206 router NPE400 that was used in the QoS testing.

For additional installation and configuration information, refer to the Cisco 7206 router documentation at this URL:

http://www.cisco.com/univercd/cc/td/doc/product/core/7206/index.htm

```
version 12.2
no parser cache
service nagle
no service pad
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
!
hostname ap2-7206
!
boot system disk0:c7200-is-mz.122-15.T8.bin
logging queue-limit 100
logging buffered 10000000 debugging
no logging console
enable password Lab
!
clock timezone SG 8
ip subnet-zero
!
ip cef
no ip domain lookup
ip domain name lab.com
!
no voice hpi capture buffer
no voice hpi capture destination
!
mta receive maximum-recipients 0
!
 class-map match-any WAN-VOICE
  match ip dscp ef
  match ip dscp cs5
 class-map match-all PRIORITY-DATA
   description ftp http tftp etc
  match access-group name PRIORITY-DATA
 class-map match-all WAN-VOICE-CONTROL
  match ip dscp af31
!
 policy-map WAN-EDGE-LFI
  class WAN-VOICE
   priority percent 50
  class WAN-VOICE-CONTROL
   bandwidth percent 5
  class PRIORITY-DATA
   bandwidth percent 20
  class class-default
   fair-queue
 policy-map WAN-EDGE
  class WAN-VOICE
```

```
   priority percent 33
  class WAN-VOICE-CONTROL
   bandwidth percent 2
  class PRIORITY-DATA
   bandwidth percent 20
  class class-default
   fair-queue
!
interface Loopback0
 ip address 10.0.0.1 255.255.255.255
!
interface FastEthernet0/0
 description ap2-4506 2/1
 ip address 10.0.0.2 255.255.255.252
 duplex full
 speed 100
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface Hssi1/0
 mtu 1500
 bandwidth 12959
 no ip address
 encapsulation frame-relay
 load-interval 30
 hssi internal-clock
 serial restart_delay 0
 frame-relay traffic-shaping
 frame-relay ip rtp header-compression
!
interface Hssi1/0.201 point-to-point
 description pvc to LAB AP1
 bandwidth 10368
 ip address 10.0.0.3 255.255.255.252
 frame-relay class FRTS-10368k
 frame-relay interface-dlci 201
 frame-relay ip rtp header-compression
 frame-relay ip rtp compression-connections 3
!
interface Hssi1/0.206 point-to-point
 description pvc to LAB AP1-2
 bandwidth 512
 ip address 10.0.0.4 255.255.255.252
 ip ospf dead-interval 65000
```

```
 frame-relay class FRTS-512k
 frame-relay interface-dlci 206
 frame-relay ip rtp header-compression
 frame-relay ip rtp compression-connections 3
!
interface Hssi1/1
 no ip address
 shutdown
 serial restart_delay 0
!
router ospf 3
 router-id 10.0.0.1
 log-adjacency-changes
 network 10.0.0.5 0.0.255.255 area 113
!
ip classless
no ip http server
!
ip access-list extended PRIORITY-DATA
 permit tcp any eq www any
 permit tcp any any eq www
 permit tcp any range ftp-data telnet any
 permit tcp any any range ftp-data telnet
 permit udp any any eq tftp
 permit udp any eq tftp any
 permit tcp any range 135 139 any
 permit tcp any any range 135 139
 permit udp any any range 135 netbios-ss
!
map-class frame-relay FRTS-10368k
 frame-relay cir 10368000
 frame-relay bc 103680
 frame-relay be 0
 frame-relay mincir 10368000
 service-policy output WAN-EDGE
!
map-class frame-relay FRTS-512k
 frame-relay cir 508816
 frame-relay bc 5090
 frame-relay be 0
 frame-relay mincir 508816
 service-policy output WAN-EDGE
 frame-relay fragment 640
!
map-class frame-relay FRTS-256k
 frame-relay cir 252832
 frame-relay bc 2530
 frame-relay be 0
```

```
 frame-relay mincir 252832
 service-policy output WAN-EDGE-LFI
 frame-relay fragment 320
!
map-class frame-relay FRTS-45000k
 frame-relay cir 45000000
 frame-relay bc 450000
 frame-relay be 0
 frame-relay mincir 45000000
 service-policy output WAN-EDGE
!
map-class frame-relay FRTS-1896k
 frame-relay cir 1896000
 frame-relay bc 18960
 frame-relay be 0
 frame-relay mincir 1896000
 service-policy output WAN-EDGE
!
map-class frame-relay FRTS-230k
 frame-relay cir 227548
 frame-relay bc 2275
 frame-relay be 0
 frame-relay mincir 227548
 service-policy output WAN-EDGE-LFI
 frame-relay fragment 320
!
snmp-server community private RW
snmp-server community public RO
snmp-server enable traps tty
snmp-server host 10.0.0.6 version 2c private
!
call rsvp-sync
!
mgcp profile default
!
dial-peer cor custom
!
gatekeeper
 shutdown
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password Lab
 login
```

```
line vty 5 15
 exec-timeout 30 0
 password Lab
 login
!
ntp clock-period 17180615
ntp server 10.0.0.7
!
end
```

Call Flow

This chapter provides information about a typical call flow in an IPT environment.

Figure 15-1 shows a call flow that involves the following devices and locations:

- Cisco IP Phone at a Very Large Site
- Cisco CallManager at Very Large Site
- Gatekeeper at a Very Large Site
- Cisco IP Phone at a Large Site
- Cisco CallManager at Large Site
- Gatekeeper at a Large Site

This call flow provides an overview of the actions involved in a typical call.

For related information and for additional call flow examples, refer to *Troubleshooting Cisco IP Telephony* (published by Cisco Press, ISBN 1-58705-075-7).

Figure 15-1 Call Flow in an IPT Environment

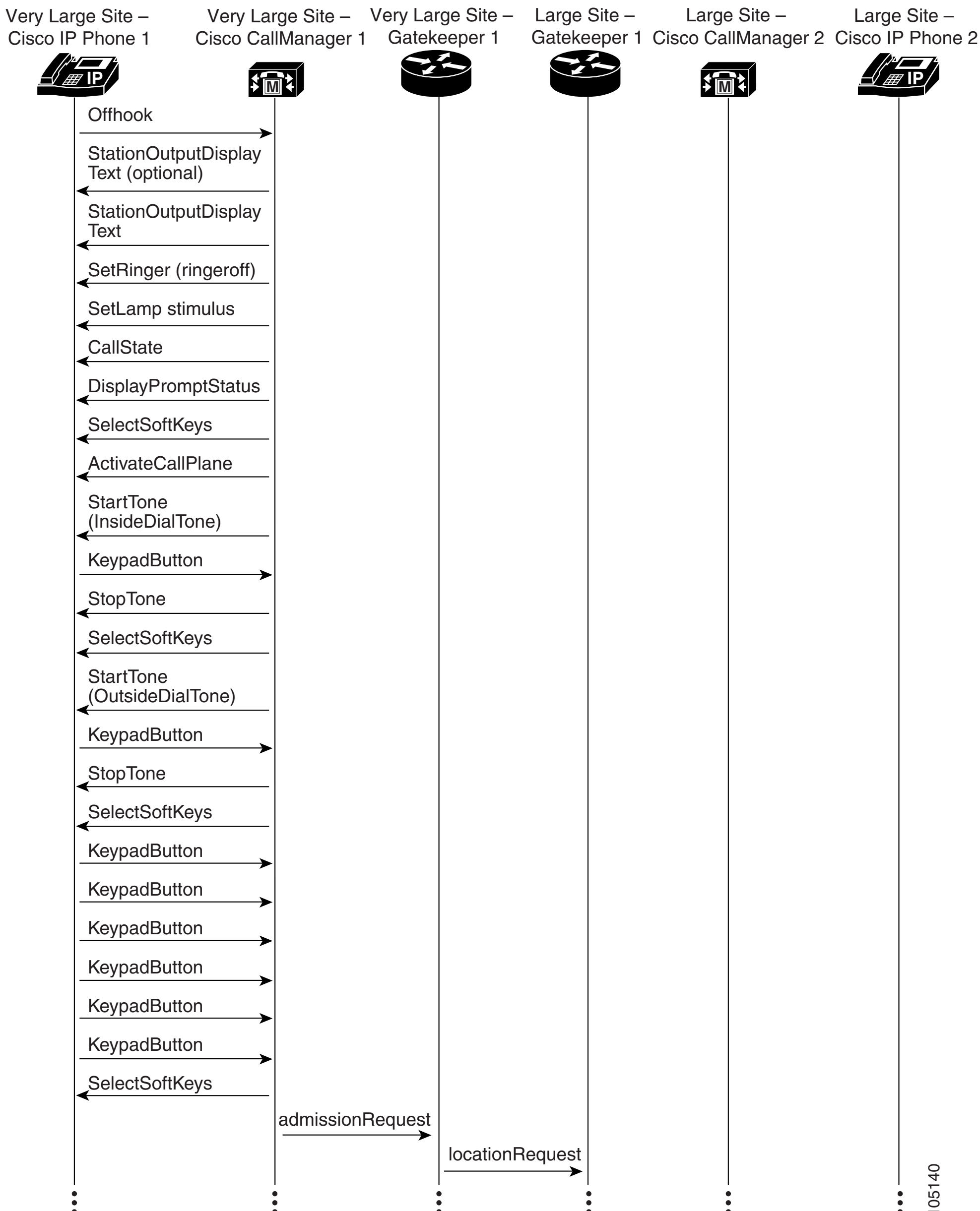


Figure 15-1 Call Flow in an IPT Environment (continued)

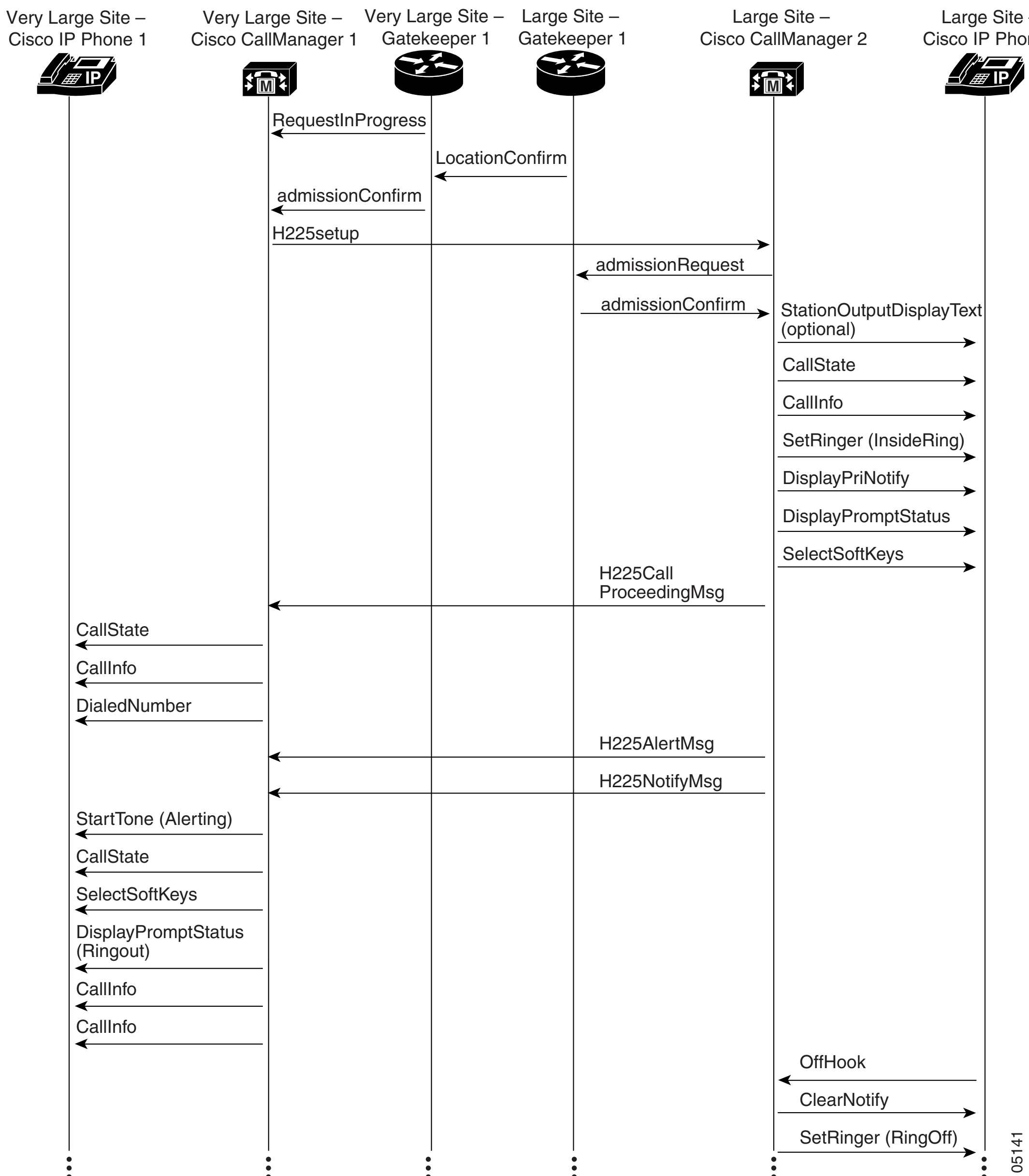


Figure 15-1 Call Flow in an IPT Environment (continued)

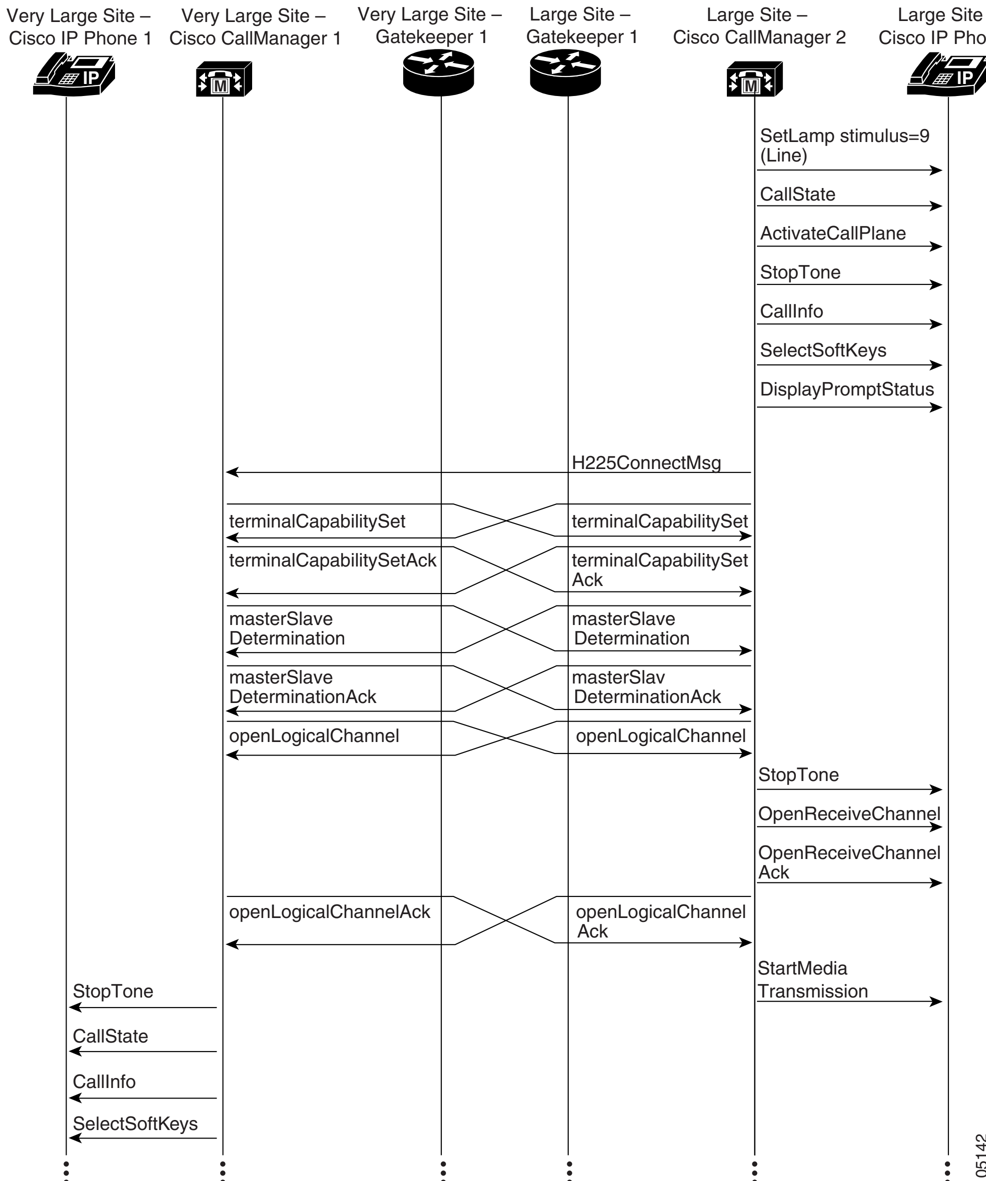


Figure 15-1 Call Flow in an IPT Environment (continued)

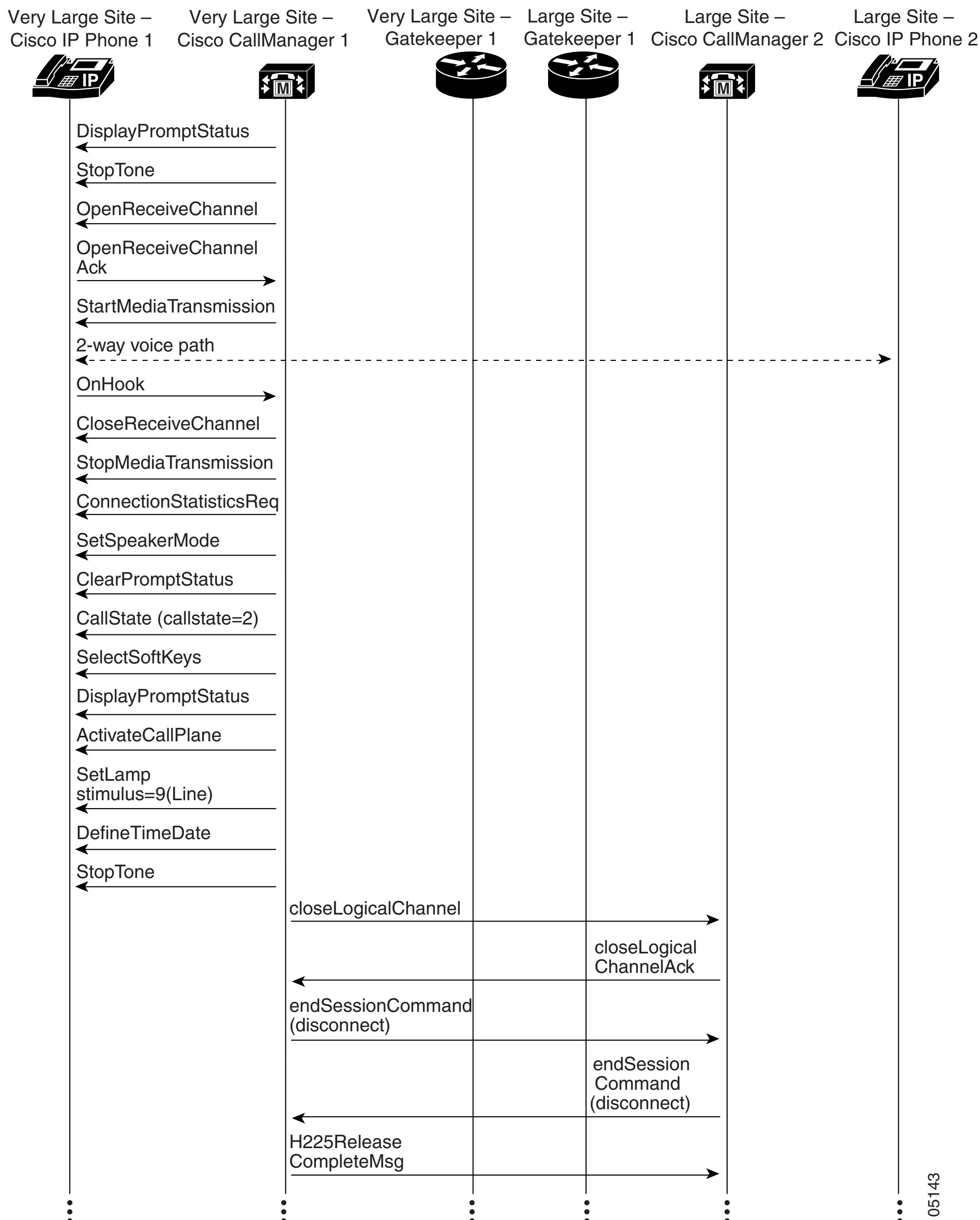


Figure 15-1 Call Flow in an IPT Environment (continued)

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:

Troubleshooting IPT Scenarios

This section provides troubleshooting information for the following situations:

Intercluster Trunk Calls with Gatekeepers Fail

Symptom: When the Primary Cisco CallManager fails and there is a secondary Cisco CallManager configured in the device pool, intercluster trunk calls with gatekeepers do not work.

Workaround: Perform the following steps. This procedure will allow intercluster trunk calls to be made through the backup Cisco CallManager if the primary Cisco CallManager fails.

1. Create a Cisco CallManager group with only one Cisco CallManager. Name this group Standalone_CM1.
2. Create a device pool called Standalone_CM1 and associate the Standalone_CM1 Cisco CallManager group with this device pool.
3. Associate the Standalone_CM1 device pool with the intercluster trunk configuration.
4. Create another Cisco CallManager group with another Cisco CallManager. Name this group Standalone_CM2.
5. Create a device pool called Standalone_CM2 and associate the Standalone_CM2 Cisco CallManager group with this device pool.
6. Associate the Standalone_CM2 device pool with the intercluster trunk configuration.

Site-to-Site IP Calls Fail

Symptom: A call fails in the following situation:

- The call is made from a Cisco IP Phone in a site with a Cisco CallManager Cluster to another Cisco IP Phone in another site with a Cisco CallManager cluster
- The call is made through a gatekeeper controlled intercluster trunk

Workaround: Perform the following steps:

1. Verify that each originating and terminating IP Phone has the proper calling search space and partition to make and receive intercluster trunk calls.
2. Verify that the route group is configured with the intercluster trunks that are listed as route group members.
3. Verify that the route list has the proper route group associated with it.
4. Verify that the route pattern has the proper route list associated with it.
5. Verify that the Gatekeeper web page in Cisco CallManager Administration has the proper host name and IP address of the gatekeeper.

6. Verify that the intercluster trunk is registered with the gatekeeper by using the **show gatekeeper endpoints** command on the gatekeeper.
7. If the intercluster trunk is not registered with the gatekeeper, turn on debug h225 asn1 on the gatekeeper and verify that there is no configuration mismatch.
8. If the call still fails, review the admissionRequest message in the debug h225 asn1 output and verify that e164 for destinationInfo and srcInfo is correct.
9. Verify that Current total bandwidth does not exceed the Maximum total bandwidth by using the **show gatekeeper zone status** command.

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 16-1 lists some of these documents. You can find additional troubleshooting information in other product-specific documentation.

Table 16-1 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
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

Table 16-1 Troubleshooting References (continued)

Troubleshooting Topic	Document	URL
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

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 North America 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, CM1 and CM3
  - Backup Cisco CallManagers, CM2 and CM4
  - Publisher
  - TFTP server
- Cisco CallManager servers: MCS-7845H-EVV1 with 4 GB RAM
- For Test 1 and Test 2:
  - 3,750 phones and additional gateways (Cisco Catalyst Communication Media Module, Cisco Catalyst 6608, Cisco Catalyst 6624, Cisco 3745, Cisco VG248, Cisco Analog Telephone Adaptor) registered to CM1
  - 3,750 phones and additional gateways (Cisco Catalyst Communication Media Module, Cisco Catalyst 6608, Cisco Catalyst 6624, Cisco 3745, Cisco VG248, Cisco Analog Telephone Adaptor) registered to CM2
- For Test 3:
  - No phones or gateways registered to CM1
  - 7,500 phones and additional gateways registered to CM2 after CM1 failed

Test 1: Shut Down Primary Cisco CallManager Server

Test

Made calls with 100 Cisco IP Phones, which were registered to CM1, and sustained the calls for long durations.

Failed CM1 by shutting down the server.

Results

Verified that 3,750 phones and additional gateways failed over to CM2. 100 sustained calls did not fail over. Verified voice path for 100 phones. After calls were disconnected, the 100 phones successfully registered with CM2. Total number of phones registered with CM2 was 7,500.

Test 2: Disconnected Cable from Primary Cisco CallManager Server

Test

Made calls with 100 Cisco IP Phones, which were registered to CM1, and sustained the calls for long durations.

Failed CM1 by disconnecting its Ethernet cable.

Results

Verified that 3,750 phones and additional gateways failed over to CM2. 100 sustained calls did not fail over. Verified voice path for 100 phones. After calls were disconnected, the 100 phones successfully registered with CM2. Total number of phones registered with CM2 was 7,500.

Test 3: Failback

Test

Failed of CM1, made new calls with 100 Cisco IP phones, and then brought CM1 back into service.

Results

Verified that 3,750 phones and additional gateways failed back to CM1. 100 sustained calls did not fail back. Verified voice path for 100 phones. After calls were disconnected, the 100 phones successfully registered with CM1. Total number of phones registered with CM1 was 3,750. Total number of phones registered with CM2 was also 3,750.

Call Load Testing

This chapter provides an overview of the call loads that were tested with the various site models in IP Communications Systems Test Release 3.0 for North America IPT. It includes information for calls made within various site models and for calls made between site models. Calls between sites were made in a Multi-Site Distributed scenario.

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

This chapter includes the following topics:

Very Large Site Call Load Testing

This section describes call load testing for calls originating in the Very Large Site model.

Table 18-1 shows the number of busy hour call attempts (BHCA) that were made from the Very Large Site to the same site and to other sites. This table shows BHCA for various call types and call activities.

Table 18-1 Load Testing for Very Large Site

Call Type	Call Activity	BHCA[1] to Very Large Site	BHCA to Large Site	BHCA to Medium Site	BHCA to Small Site	Total BHCA
IP to IP	Basic call	7,000	—	—	—	7,000
	Blind transfer	393	—	—	—	393
	Consult transfer	376	—	—	—	376
	Hold	371	—	—	—	371
	Conference	1,009	—	—	—	1,009
	Encrypted	7,200	—	—	—	7,200
	Video	500	—	—	—	500
IP to PSTN[2]	Basic call	19,000	—	—	—	19,000
	Blind transfer	802	—	—	—	802
	Consult transfer	722	—	—	—	722
	Hold	592	—	—	—	592
	Conference	1,651	—	—	—	1,651
PSTN to IP	Basic call	19,000	—	—	—	19,000
	Blind transfer	802	—	—	—	802
	Consult transfer	722	—	—	—	722
	Hold	592	—	—	—	592
	Conference	1,651	—	—	—	1,651
IP to ICT[3]	Basic call	—	3,960	640	960	5,560

Table 18-1 Load Testing for Very Large Site (continued)

Call Type	Call Activity	BHCA[1] to Very Large Site	BHCA to Large Site	BHCA to Medium Site	BHCA to Small Site	Total BHCA
ICT to IP	Basic call	—	3,000	—	—	3,000
	Blind transfer	—	248	533	200	1,400
	Consult transfer	—	361	771	189	1,321
	Hold	—	116	240	180	536
	Conference	—	338	675	171	1,184
IP to Cisco Catalyst 6624	Basic call	2,240	—	—	—	2,240
IP to Cisco VG248	Basic call	2,880	—	—	—	2,880
IP to Cisco Unity	Basic call	9,969	—	—	—	9,969
Total BHCA	—	**77,472**	**8,023**	**2,879**	**1,700**	**90,074**

1. BHCA = busy hour call attempts.
2. PSTN = public switched telephone network.
3. ICT = intercluster trunk.

Large Site Call Load Testing

This section describes call load testing for calls originating in the Large Site model.

Table 18-2 shows the number of busy hour call attempts (BHCA) that were made from the Large Site to the same site and to other sites. This table shows BHCA for various call types and call activities.

Table 18-2 Load Testing for Large Site

Call Type	Call Activity	BHCA[1] to Very Large Site	BHCA to Large Site	BHCA to Medium Site	BHCA to Small Site	Total BHCA
IP to IP	Basic call	—	2,000	—	—	2,000
	Blind transfer	—	187	—	—	187
	Consult transfer	—	165	—	—	165
	Hold	—	143	—	—	143
	Conference	—	411	—	—	411
IP to PSTN[2]	Basic call	—	2,400	—	—	2,400
	Blind transfer	—	320	—	—	320
	Consult transfer	—	318	—	—	318
	Hold	—	252	—	—	252
	Conference	—	727	—	—	727
PSTN to IP	Basic call	—	2,400	—	—	2,400
	Blind transfer	—	320	—	—	320
	Consult transfer	—	318	—	—	318
	Hold	—	252	—	—	252
	Conference	—	728	—	—	728
IP to ICT[3]	Basic call	1,473	—	—	—	1,473
ICT to IP	Basic call	1,000	—	—	—	1,000
ICT to PSTN	Basic call	2,000	—	—	—	2,000
PSTN to ICT	Basic call	2,000	—	—	—	2,000

Table 18-2 Load Testing for Large Site (continued)

Call Type	Call Activity	BHCA[1] to Very Large Site	BHCA to Large Site	BHCA to Medium Site	BHCA to Small Site	Total BHCA
IP to Cisco VG248	Basic call	960	—	—	—	960
Total BHCA	—	**7,433**	**10,941**	—	—	**18,374**

1. BHCA = busy hour call attempts.
2. PSTN = public switched telephone network.
3. ICT = intercluster trunk.

Medium Site Call Load Testing

This section describes call load testing for calls originating in the Medium Site model.

Table 18-3 shows the number of busy hour call attempts (BHCA) that were made from the Medium Site to the same site and to other sites. This table shows BHCA for various call types and call activities.

Table 18-3 Load Testing for Medium Site

Call Type	Call Activity	BHCA[1] to Very Large Site	BHCA to Large Site	BHCA to Medium Site	BHCA to Small Site	Total BHCA
IP to IP	Basic call	—	—	940	—	940
	Blind transfer	—	—	111	—	111
	Consult transfer	—	—	110	—	110
	Hold	—	—	72	—	72
	Conference	—	—	212	—	212

Table 18-3 Load Testing for Medium Site (continued)

Call Type	Call Activity	BHCA[1] to Very Large Site	BHCA to Large Site	BHCA to Medium Site	BHCA to Small Site	Total BHCA
IP to PSTN[2]	Basic call	—	—	620	—	620
	Blind transfer	—	—	158	—	158
	Consult transfer	—	—	116	—	116
	Hold	—	—	118	—	118
	Conference	—	—	346	—	346
PSTN to IP	Basic call	—	—	620	—	620
	Blind transfer	—	—	158	—	158
	Consult transfer	—	—	116	—	116
	Hold	—	—	118	—	118
	Conference	—	—	346	—	346
IP to ICT[3]	Basic call	999	—	—	—	999
IP to Cisco VG248	Basic call	640	—	—	—	—
Total BHCA	—	**1,639**	—	**4,161**	—	**5,800**

1. BHCA = busy hour call attempts.
2. PSTN = public switched telephone network.
3. ICT = intercluster trunk.

Small Site Call Load Testing

This section describes call load testing for calls originating in the Small Site model.

Table 18-4 shows the number of busy hour call attempts (BHCA) that were made from the Small Site to the same site and to other sites. This table shows BHCA for various call types and call activities.

Table 18-4 Load Testing for Small Site

Call Type	Call Activity	BHCA[1] to Very Large Site	BHCA to Large Site	BHCA to Medium Site	BHCA to Small Site	Total BHCA
IP to IP	Basic call	—	—	—	620	620
	Blind transfer	—	—	—	77	77
	Consult transfer	—	—	—	57	57
	Hold	—	—	—	57	57
	Conference	—	—	—	168	168
IP to PSTN[2]	Basic call	—	—	—	620	620
	Blind transfer	—	—	—	80	80
	Consult transfer	—	—	—	117	117
	Hold	—	—	—	60	60
	Conference	—	—	—	177	177

Table 18-4 Load Testing for Small Site (continued)

Call Type	Call Activity	BHCA[1] to Very Large Site	BHCA to Large Site	BHCA to Medium Site	BHCA to Small Site	Total BHCA
PSTN to IP	Basic call	—	—	—	620	620
	Blind transfer	—	—	—	80	80
	Consult transfer	—	—	—	117	117
	Hold	—	—	—	60	60
	Conference	—	—	—	177	177
IP to ICT[3]	Basic call	740	—	—	—	740
IP to Cisco VG248	Basic call	960	—	—	—	—
Total BHCA	—	**1,700**	**—**	**—**	**3,087**	**4,787**

1. BHCA = busy hour call attempts.
2. PSTN = public switched telephone network.
3. ICT = intercluster trunk.

Central Site and Remote Sites Call Load Testing

This section describes call load testing for calls in the Central Site model and in the Remote Site models.

Table 18-5 shows the number of busy hour call attempts (BHCA) that were made within and among the Central Site model and in the Remote Site models. This table shows BHCA for various call types and call activities.

Table 18-5 Load Testing for Central Site and Remote Sites

Call Type	Call Activity	Total BHCA[1]
IP to IP	4-digit call (call within same site)	927
	8-digit call (site-to-site call)	2,204
IP to PSTN[2] and PSTN to IP (long distance calls)	11-digit basic call	28,975
IP to PSTN and PSTN to IP (10-digit local calls)	Basic call	8,378
	Blind transfer	2,782
	Consult transfer	3,709
	Hold	3,709
	Conference	5,564
IP to IP	4-digit call (call within same site)	109
	8-digit call (site-to-site call)	2,291
IP to PSTN (Cisco Catalyst 6608) and PSTN (Cisco Catalyst 6608) to IP (11-digit local calls)	Basic call	2,182
	Blind transfer	327
	Consult transfer	436
	Hold	436
	Conference	436
IP to Cisco Unity	Basic call	1,400
Total	—	**63,865**

1. BHCA = busy hour call attempts.
2. PSTN = public switched telephone network.

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 IPT:

- Table A-1 on page A-1—Software release versions of components
- Table A-2 on page A-3—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)
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)

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

Component	Release Version
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
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

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

Component	Release Version
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 Firmware Release Versions of Cisco IP Phones

Phone Model	Firmware Version
Cisco IP Phone 7902G	CP7902050000SCCP041007A
Cisco IP Phone 7905G	CP7905050000SCCP041022A

Table A-2 Firmware Release Versions of Cisco IP Phones (continued)

Phone Model	Firmware Version
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 7940G	P00306000500
Cisco IP Phone 7960G	P00306000500
Cisco IP Phone 7970G	TERM70.6-0-1-0sr1s

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