Authorized Self-Study Guide

Cisco IP Telephony (CIPT)

Second Edition

Foundation Learning for CCVP IP Telephony

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Jeremy Cioara, CCIE® No. 11,727

Authorized Self-Study Guide
Cisco IP Telephony (CIPT)
Second Edition

Jeremy D. Cioara, CCIE No. 11727

Cisco Press

800 East 96th Street
Indianapolis, IN 46240 USA

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Published by:
Cisco Press
800 East 96th Street
Indianapolis, IN 46240 USA

Printed in the United States of America 1 2 3 4 5 6 7 8 9 0

First Printing October 2006

Library of Congress Cataloging-in-Publication Number: 2006926698

ISBN: 1-58705-261-x

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Printed in the USA

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Dedications

First and foremost, I'd like to thank Jesus Christ who continues to bless me in every way and keeps me from killing myself by doing something stupid. You will always be at the core of everything I do. Second, to my darling wife, Susan: Thank you for your support through all these projects that keep me glued to a computer screen through all hours of the day and night. You are a more wonderful companion than I could have ever hoped for. I love you! To the cricket chirping in the room right now: I hate you. If I ever find you, I'm going to feed you to my fish and dance merrily around the room as they eat you slowly. To my cat, Snuggles: Thanks for being soft. To the dog, Buttercup: I have nothing to say to you at this time, but at least I've acknowledged your existence. To all my readers: Be warned; this is what happens to you after months of writing hundreds of pages.

Acknowledgments

I'd like to give special recognition to Kevin Wallace, Larry Roberts, Michael Coffin, and Dennis Hartman for providing their expert technical knowledge in editing the book. As usual, they're not afraid to tell you when you're wrong.

A big "thank you" goes out to the production team for this book. Brett Bartow and Andrew Cupp have been incredibly professional and a pleasure to work with. I couldn't have asked for a finer team.

Contents at a Glance

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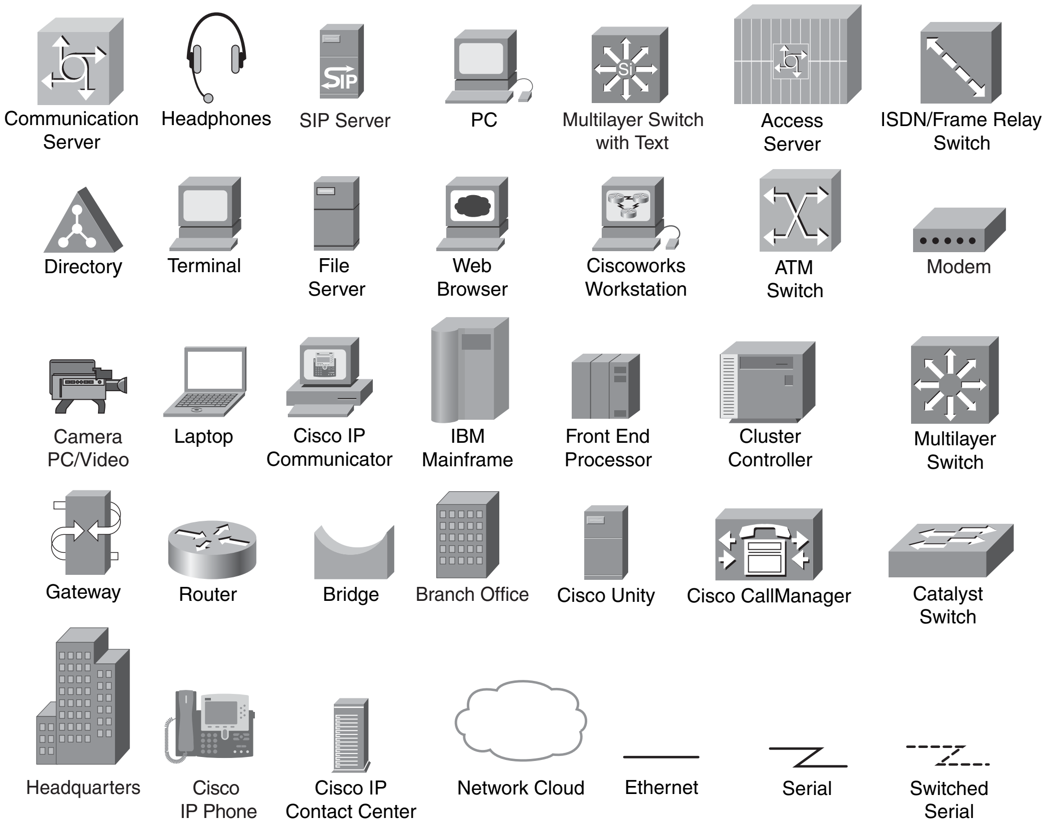

Command Syntax Conventions

The conventions used to present command syntax in this book are the same conventions used in the IOS Command Reference. The Command Reference describes these conventions as follows:

- **Boldface** indicates commands and keywords that are entered literally as shown. In actual configuration examples and output (not general command syntax), boldface indicates commands that are manually input by the user (such as a **show** command).
- *Italic* indicates arguments for which you supply actual values.
- Vertical bars (|) separate alternative, mutually exclusive elements.
- Square brackets [ ] indicate optional elements.
- Braces { } indicate a required choice.
- Braces within brackets [{ }] indicate a required choice within an optional element.

Foreword

Cisco IP Telephony (CIPT), Second Edition is an excellent self-study resource for the CCVP CIPT exam. Whether you are studying to become CCVP certified or are simply seeking to gain a better understanding of VoIP and PSTN components and technologies, you will benefit from the information presented in this book.

Cisco Press Self-Study Guide titles are designed to help educate, develop, and grow the community of Cisco networking professionals. As an early-stage exam preparation product, this book presents a detailed and comprehensive introduction to the technologies used to install, configure, and support Cisco CallManager 4.1 in a Cisco network, including such features as security and video. Developed in conjunction with the Cisco certifications team, Cisco Press books are the only self-study books authorized by Cisco Systems.

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I hope and expect that you'll find this guide to be an essential part of your exam preparation and a valuable addition to your personal library.

Don Field
Director, Certifications
Cisco Systems, Inc.
August 2006

Introduction

Professional certifications have been an important part of the computing industry for many years and will continue to become more important. Many reasons exist for these certifications, but the most popularly cited reason is that of credibility. All other considerations held equal, the certified employee/consultant/job candidate is considered more valuable than one who is not.

Goals and Methods

The most important and somewhat obvious goal of this book is to help you pass the Cisco IP Telephony (CIPT) exam (642-444). In fact, if the primary objective of this book was different, then the book's title would be misleading; however, the methods used in this book to help you pass the CCVP Cisco IP Telephony exam are designed to also make you much more knowledgeable about how to do your job. Although this book has more than enough questions to help you prepare for the actual exam, the method in which they are used is not to simply make you memorize as many questions and answers as you possibly can.

One key methodology used in this book is to help you discover the exam topics that you need to review in more depth, to help you fully understand and remember those details, and to help you prove to yourself that you have retained your knowledge of those topics. So, this book does not try to help you pass by memorization, but helps you truly learn and understand the topics. The Cisco IP Telephony exam is just one of the foundation topics in the CCVP certification and the knowledge contained within is vitally important to consider yourself a truly skilled voice engineer or specialist. This book would do you a disservice if it didn't attempt to help you learn the material.

Who Should Read This Book?

This book is not designed to be a general networking topics book, although it can be used for that purpose. This book is intended to tremendously increase your chances of passing the CCVP Cisco IP Telephony exam. Although you will achieve other objectives from using this book, the book is written with one goal in mind: to help you pass the exam.

The placement of CallManager in the CCVP series of exams is unique in that it could be one of the very first CCVP exams you take, or it could be one of the last. It stands as an isolated product in the CCVP series of exams. This means you could be getting into this book with an extensive knowledge of voice gateways, switches, and quality of service (from reading the CVOICE and/or QoS books) with a basic CCNA-level knowledge. Regardless, your learning curve will be the same; CallManager is configured differently, troubleshot uniquely, and managed distinctively from the rest of the voice network.

So why should you want to pass the CCVP Cisco IP Telephony exam? Because it is one of the milestones toward getting the CCVP certification; no small feat in itself. What would getting the

CCVP mean to you? A raise, a promotion, recognition? How about to enhance your résumé? To demonstrate that you are serious about continuing the learning process and that you are not content to rest on your laurels. To please your reseller-employer, who needs more certified employees for a higher discount from Cisco. Or one of many other reasons.

How This Book Is Organized

Although this book could be read from cover to cover, it is designed to be flexible and allow you to easily move between chapters and sections of chapters to cover just the material that you need more work with. If you do intend to read them all, the order in the book is an excellent sequence to use.

The chapters cover the following topics:

Chapter 1, "Introduction to Cisco Unified Communications and Cisco Unified CallManager"—This chapter provides an overview of the Cisco Unified Communications (AVVID) strategy, how the Cisco Unified CallManager product suite fits into the scheme, the interaction between CallManager and Cisco IP Phones, and the CallManager server platforms.

Chapter 2, "Cisco Unified CallManager Clustering and Deployment Options"—This chapter discusses the design strategies behind a Cisco CallManager cluster, cluster replication, and CallManager deployment models.

Chapter 3, "Cisco Unified CallManager Installation and Upgrades"—This chapter covers the requirements to perform a CallManager server installation, the CallManager installation and upgrade process, and postinstallation procedures.

Chapter 4, "Cisco IP Phones and Other User Devices"—This chapter examines the basic features of all Cisco IP Phones, the entry-level, midrange, and upper-end Cisco IP Phone models, the IP Phone startup process, and audio codec communication.

Chapter 5, "Configuring Cisco Unified CallManager to Support IP Phones"—This chapter examines the configuration of Cisco CallManager to support IP Phone registration and communication within a cluster, the creation of device pools, and manually or automatically registering Cisco IP Phones with the SQL Database.

Chapter 6, "Cisco IP Telephony Users"—This chapter covers the addition of IP telephony users to the CallManager LDAP database, associating users with devices, and allowing users to access the User Options web page to configure common features.

Chapter 7, "Cisco Bulk Administration Tool"—This chapter covers the features and components of the BAT application, the installation of BAT, and the configuration of BAT to apply bulk moves, adds, and changes to the Cisco CallManager cluster.

Chapter 8, "Cisco Catalyst Switches"—This chapter examines the functions that Catalyst switches perform in a Cisco IP telephony solution, the three options for powering Cisco IP Phones, the two types of power supplied by PoE switches, inline power configuration, dual VLAN configuration, and CoS configuration.

Chapter 9, "Configuring Cisco Gateways and Trunks"—This chapter discusses the role of the gateway in the IP telephony infrastructure, core gateway requirements, and gateway communication protocols; it also examines the configuration of analog and digital gateways, Cisco CallManager trunk configuration, and enabling the CallManager to use SIP capabilities.

Chapter 10, "Cisco Unified CallManager Route Plan Basics"—This chapter covers the fundamentals of the Cisco CallManager route plan, including the identification of the route plan building blocks, the configuration of route groups, route lists, and route patterns, and the design of a basic route plan.

Chapter 11, "Cisco Unified CallManager Advanced Route Plans"—This chapter covers the concepts and configurations behind route filters, digit discard instructions, transformation masks, translation patterns, and route plan reports.

Chapter 12, "Configuring Hunt Groups and Call Coverage"—This chapter examines the call distribution components and algorithms supported by CallManager; examines call hunting concepts; discusses the concepts and configurations of line groups, hunt lists, and hint pilots; and provides a scenario-based design discussion for hunting and forwarding calls.

Chapter 13, "Implementing Telephony Call Restrictions and Control"—This chapter discusses the concepts behind class of service as it relates to users of a phone system; it also covers the design and configuration of partitions, calling search spaces, and time-of-day routing.

Chapter 14, "Implementing Multiple-Site Deployments"—This chapter discusses why call admission control is important to maintain voice QoS across an IP WAN; describes the Cisco CallManager locations feature and how it provides the necessary call admission control for centralized call-processing environments; and explains the concepts and configurations behind locations, H.323 gatekeepers, and SRST.

Chapter 15, "Media Resources"—This chapter examines the necessary Cisco CallManager resources you should use as media resources, the configuration of conference bridges, media termination points, transcoders, and music on hold.

Chapter 16, "Configuring User Features, Part 1"—This chapter discusses the core Cisco IP Phone features supported by Cisco CallManager along with the enhanced IP Phone features configurable by an administrator; this chapter also examines the configuration of softkey templates, call park, call pickup, call back, barge, shared line appearances, and IP Phone services.

Chapter 17, "Configuring User Features, Part 2"—This chapter covers the configuration of the Cisco CallManager Extension Mobility service, FAC and CMC call accounting and restrictions, call display restrictions, malicious caller ID, and multilevel precedence and preemption.

Chapter 18, "Configuring Cisco Unified CallManager Attendant Console"—This chapter explains the functions and features of the Cisco CallManager Attendant Console application, and defines the key Attendant Console components and redundancy process. This chapter also covers the configuration of the Cisco CallManager server to support the Attendant Console and the client-side Attendant Console installation and configuration.

Chapter 19, "Configuring Cisco IP Manager Assistant"—This chapter covers the features of Cisco IPMA and the two modes of operation, the IPMA components, and the configuration of Cisco IPMA for shared-line mode installations.

Chapter 20, "Securing the Windows Operating System"—This chapter examines the security threats to the Windows operating system, the Cisco security and hotfix policy, the included enhanced security scripts, and the antivirus protection options for the CallManager server. This chapter also covers the suggested administrative password policy, the protection of the server from common exploits, the Cisco Security Agent, and the not-recommended security settings for the Cisco CallManager server.

Chapter 21, "Securing Cisco Unified CallManager Administration"—This chapter discusses the threats targeting remote Cisco CallManager Administration and other applications, the function of HTTPS in remote communication, the HTTP certificate operations, and the CallManager multilevel administration configuration.

Chapter 22, "Preventing Toll Fraud"—This chapter examines the vulnerability of legitimate devices to be used for fraudulent use in the IP telephony network, the use of partitions and calling search spaces to restrict call forwarding, the blocking of specific area codes, the configuration of CallManager to route calls based on time-of-day, the implementation of FAC to implement user authorization, and the restriction of conference call features to limit toll fraud.

Chapter 23, "Hardening the IP Phone"—This chapter covers the potential threats against IP Phones and the attack tools and methods a hacker can use, signed firmware images, CallManager IP Phone security techniques, and IP Phone authentication and encryption.

Chapter 24, "Understanding Cryptographic Fundamentals"—This chapter discusses the foundations of cryptography and the four cryptographic services, basic operation and uses for symmetric and asymmetric encryption algorithms, hashing, and digital signatures.

Chapter 25, "Understanding the Public Key Infrastructure"—This chapter examines the problem of secure, scalable distribution of public keys, the concepts behind a trusted introducer, certificates, CAs, certification paths, certificate enrollment, and certificate revocation.

Chapter 26, "Understanding Cisco IP Telephony Authentication and Encryption Fundamentals"—This chapter explains how file manipulation, tampering with call signaling, man-in-the-middle attacks, eavesdropping, and IP Phone theft can compromise a Cisco CallManager system. This chapter also discusses the authentication and encryption mechanisms supported by CallManager, the roll of CAPF, MICs, LSCs, CTLs, and the CTL Client. This chapter covers the processes and protocols used for signaling encryption and media encryption.

Chapter 27, "Configuring Cisco IP Telephony Authentication and Encryption"—This chapter examines the steps to configure a Cisco CallManager system for authentication and encryption, the activation of the CTL Provider and CAPF services, the installation of the CTL Client, and configuring the IP Phones for a hardened security structure.

Chapter 28, "Introducing IP Video Telephony"—This chapter covers the functions and components of the Cisco IP video telephony solution, the connection of a video call, H.323 versus SCCP video signaling, the two factors in determining the bandwidth requirement for video, and video call admission control.

Chapter 29, "Configuring Cisco VT Advantage"—This chapter discusses the features and function of the VT Advantage, placing and receiving calls with VT Advantage, configuring the Cisco CallManager to support video, and the VT Advantage client installation process.

Chapter 30, "Introducing Database Tools and Cisco Unified CallManager Serviceability"—This chapter examines the database structure and replication status of broken database connection, the services provided by CallManager Serviceability, the CallManager Control Center, the CallManager Service Activation window, and the various tools used to monitor Cisco CallManager listed by function.

Chapter 31, "Monitoring Performance"—This chapter defines performance objects, covers the Microsoft Event Viewer and Performance Monitor, and the Cisco Real-Time Monitoring Tool.

Chapter 32, "Configuring Alarms and Traces"—This chapter identifies the functions of the Cisco CallManager Alarm interface, and discusses the configuration of alarms and traces, the trace analysis process, and the Bulk Trace Analysis tool.

Chapter 33, "Configuring CAR"—This chapter discusses the usage, features, and operations of the CAR tool; the contents of CDR and CMR records; the three levels of CAR users and their reporting capabilities; the configuration of CAR system parameters, system schedule, and alerts; and the generation of user reports using CAR.

Chapter 34, "Using Additional Management and Monitoring Tools"—This chapter examines the use of SNMP, Syslog, and CiscoWorks ITEM in remotely managing and maintaining a Cisco CallManager system, the use of dependency records, the Password Changer tool, the Dialed Number Analyzer, and the Quality Reporting tool.

Finally, Appendix A, "Answers to Review Questions," contains the solutions to the review questions throughout the book.

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Part I: Cisco CallManager Fundamentals

This chapter covers the following topics:

- An introduction to the Cisco Unified Communications strategy (formerly Cisco AVVID) and Cisco Unified CallManager technology
- An understanding of the placement of Cisco Unified CallManager in an IP telephony design
- The interaction between Cisco Unified CallManager and Cisco IP Phones
- Cisco Unified CallManager server platforms

CHAPTER 1

Introduction to Cisco Unified Communications and Cisco Unified CallManager

Although IP telephony might seem like the new, emerging technology in IT environments, it has actually been around for many years. As WAN and LAN data connections became more stable and the total amount of available bandwidth increased, legacy PBX vendors rushed to add IP processing functions to their chassis-based management systems. Cisco Systems also saw the writing on the wall and in 1999 unveiled both the Cisco 7900 series IP Phones and the Cisco Architecture for Voice, Video, and Integrated Data (AVVID) strategy for the future. This was also the year that Cisco announced Cisco Unified CallManager version 2.4.

A Cisco IP telephony deployment relies on Cisco Unified CallManager for its call-processing and call-routing functions. Understanding the role that Cisco Unified CallManager plays in a converged network from a system, software, and hardware perspective is necessary to successfully install and configure Cisco Unified CallManager. This chapter discusses Cisco Unified Communications and the Cisco Unified CallManager functions, hardware requirements, software requirements, and installation and upgrade information.

NOTE With the recent launch of the new Cisco Unified Communications portfolio, Cisco AVVID has become an obsolete term because it no longer accurately describes the breadth of Cisco's converged, unified communications offerings. This chapter still occasionally makes reference to Cisco AVVID in historical contexts, but for the most part uses the term Cisco Unified Communications. You might find them used interchangeably during this transition period.

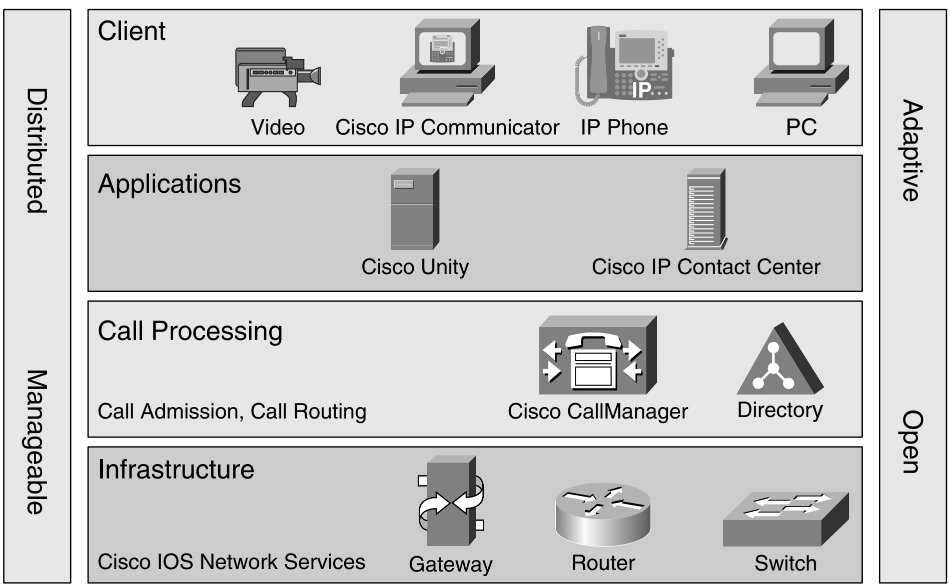
NOTE Cisco has recently changed the name of the Cisco CallManager product to Cisco Unified CallManager to reflect the Cisco Unified Communications campaign. For brevity, this book mainly uses the shortened name of Cisco CallManager or just CallManager.

Cisco Unified Communications

When Microsoft announced their new .NET platform, there was a huge amount of confusion in the industry. Systems administrators did not know if they were to expect a new operating system, programming language, or company direction. The same effect happened when Cisco announced the Architecture for Voice, Video, and Integrated Data (AVVID), now known as Cisco Unified Communications. Network administrators did not understand if this was a new router type, IOS feature set, or marketing campaign. Cisco Unified Communications is not any of these things. Instead, it is a company strategy that provides the foundation for converged networks. The goal of Cisco Unified Communications is to create network equipment that has the capability to handle voice, video, and data traffic within a single network infrastructure.

Figure 1-1 shows the four standard layers of the Cisco Unified Communications voice infrastructure model: the infrastructure layer, which lays the foundation for network components; the call-processing layer, which maintains PBX-like functions; the applications layer, where applications that provide additional network functionality reside; and the client layer, where end-user devices reside.

Figure 1-1 *The Cisco Unified Communications Model for Voice Networks*



Each of the major areas of the Cisco Unified Communications architecture has a similar model focused around the technologies of voice, video, and data. The key points about the four standard layers of the voice model are as follows:

- **Infrastructure layer**—The infrastructure carries data between all network devices and applications and consists of routers, switches, and voice gateways.
- **Call-processing layer**—Call processing is physically independent of the infrastructure. Thus, a Cisco CallManager in Chicago can provide call control for a bearer channel in Phoenix.
- **Applications layer**—Applications are physically independent of call-processing functions and the physical voice-processing infrastructure; that is, they can reside anywhere within the network.
- **Client layer**—The client layer makes the voice applications available to the user, whether the end device is a Cisco IP Phone, a PC using a Cisco IP Communicator, or a PC delivering converged messaging.

At first, this might seem like just another model to commit to memory; however, understanding Cisco's design of this model allows you to make the most of your voice network. The OSI model creates a standard for communication across networks. In addition, it allows vendors to isolate network functionality and specialize in equipment working at a specific layer. Likewise, one of the huge benefits of using an IP telephony system over a PBX system is the open standards for protocols and equipment. A vendor could specialize and design devices that work only at the client layer of the Cisco Unified Communications voice model. These devices could use an open standard protocol to communicate with the Cisco CallManager at the applications layer.

NOTE Because of the flexibility of the Cisco IP telephony network, many organizations have created their own custom applications for the voice network. Cisco has made a Cisco CallManager Software Development Kit (SDK) freely available to aid in the process of creating custom Extensible Markup Language (XML) applications.

Understanding Cisco Unified CallManager

The vast majority of this CIPT book is focused on the core product that controls a Cisco IP telephony network: Cisco Unified CallManager. Cisco CallManager brings enterprise telephony features and functions to packet telephony devices. These devices include Cisco IP Phones, media-processing devices, voice over IP (VoIP) gateways, and multimedia applications. Additional data, voice, and video services, such as unified messaging, multimedia conferencing, collaborative contact centers, and interactive multimedia response systems, interact with the IP telephony solution through the Cisco CallManager application programming interface (API).

Cisco CallManager provides the following functions:

- **Call processing**—Call processing refers to the complete process of originating, routing, and terminating calls, including any statistical collection processes.
- **Signaling and device control**—Cisco CallManager sets up all of the signaling connections between call endpoints and directs devices such as phones, gateways, and conference bridges to establish and tear down streaming connections.
- **Dial plan administration**—The dial plan is a set of configurable rules that Cisco CallManager uses to determine call routing. Cisco CallManager provides the ability to create flexible dial plans for users.
- **Phone feature administration**—Cisco CallManager extends services such as hold, transfer, forward, conference, speed dial, last-number redial, Call Park, and other features to IP Phones and gateways.
- **Directory services**—Cisco Unified CallManager uses DC Directory as an embedded Lightweight Directory Access Protocol (LDAP) directory. This directory stores authentication and authorization information about users and is a standard feature of Cisco CallManager (it does not require any special configuration or installation). However, Cisco CallManager can also be integrated with a corporate directory such as the Netscape Directory Server or Microsoft Active Directory.
- **Programming interface to external applications**—Cisco CallManager provides a programming interface to external applications such as Cisco IP SoftPhone, Cisco IP Communicator, Cisco IP Interactive Voice Response (IVR), Cisco Personal Assistant, and Cisco CallManager Attendant Console.

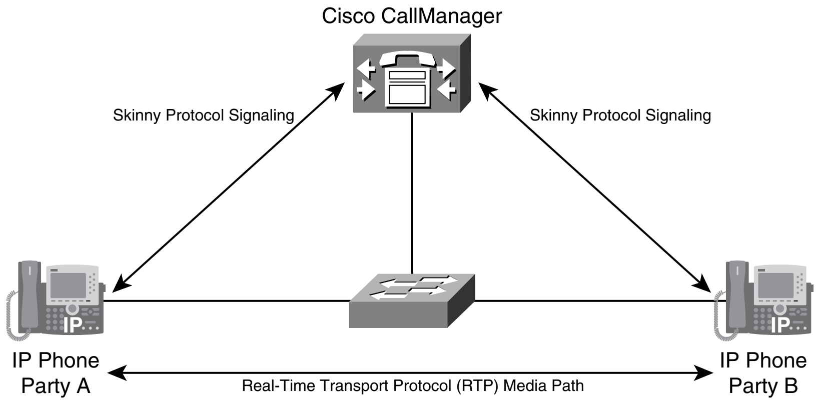
Cisco Unified CallManager and IP Phone Interaction

Cisco CallManager provides the intelligence behind call-processing functions when integrated with Cisco IP Phones. These IP Phones are virtually a paperweight without the Cisco CallManager instructing them with what they should do. The Cisco CallManager uses the Skinny Client Control Protocol (SCCP, or Skinny) signaling protocol over IP to communicate with Cisco IP Phones for call setup and maintenance tasks. When the call is set up, Cisco IP Phones communicate directly using Real-Time Transport Protocol (RTP) to carry the audio.

TIP Most Cisco 7900 IP Phones ship with a software image, allowing them to use the Skinny protocol to communicate with the Cisco CallManager. Cisco has also made a Session Initiation Protocol (SIP) image that allows Cisco IP Phones to be used with a third-party management system. Cisco CallManager does not support controlling the IP Phones using the SIP image as of Cisco CallManager 4.1. However, Cisco has added this support in the Cisco CallManager 5.0 release.

You can better understand how Cisco CallManager performs call processing and signaling functions by tracking a basic IP telephony call, as shown in Figure 1-2.

Figure 1-2 *Cisco CallManager Basic Call Processing and Signaling Capabilities*



In Figure 1-2, Party A (left IP Phone) wants to call Party B (right IP Phone). Party A picks up the handset and dials the number of Party B. In this environment, dialed digits are sent to Cisco CallManager, the call-processing engine. Cisco CallManager finds the address and determines where to route the call.

Using the Skinny protocol, Cisco CallManager signals the calling party over IP to initiate a ring back, and Party A hears ringing. Cisco CallManager also signals the destination phone to initiate ringing.

When Party B picks up the telephone, the RTP media path opens between the two stations. Party A or Party B can now initiate a conversation. Because the IP Phones manage this RTP media path themselves, the Cisco CallManager is able to move out of the call-processing functions for this call. The IP Phones require no further communication with Cisco CallManager until either Party A or Party B invokes a feature, such as call transfer, call conferencing, or call termination. Even if the Cisco CallManager were to fail during the course of the call, the RTP stream would continue until one of the parties involved in the call decided to disconnect the call.

TIP The Skinny (SCCP) protocol uses TCP port 2000, whereas the RTP bearer stream uses dynamically negotiated even UDP port numbers in the range from 16,384 to 32,767.

The Components of Cisco Unified CallManager

Cisco CallManager installs using an image-based procedure. The image expands and deploys the following foundation operating system components:

- Windows 2000 Server
- Microsoft SQL Server 2000
- DC Directory
- Cisco IP Telephony Backup and Restore System (BARS)

Cisco CallManager server relies on Microsoft Windows 2000 for its operating system and Microsoft Structured Query Language (SQL) Server 2000 for its database (both provided by Cisco Systems). The operating system version that Cisco provides is called the Cisco IP Telephony Operating System. For example, Cisco CallManager 4.1(2) requires Cisco IP Telephony Operating System Version 2000.2.6 (or later) and the latest Cisco IP Telephony Server Operating System service release. The latest operating system updates and service releases can be obtained from Cisco using an authorized CCO login.

Cisco CallManager uses DC Directory as an embedded LDAP directory. This directory stores authentication and authorization information about users and is standard with Cisco CallManager (it does not require any special configuration or installation). Maintaining a user database for your IP telephony system is optional, as the phone system will work just fine without requiring users to log in to the system. However, if you want to deploy any of the advanced features such as Extension Mobility or the Cisco IP SoftPhone, user authentication is necessary. Authentication establishes the right of the user to access the system, whereas authorization identifies the telephony resources that a user is permitted to use, such as a specific telephone extension.

The Cisco Customer Directory Plugin allows you to integrate Cisco CallManager with one of the following enterprise directories:

- Microsoft Active Directory, available with Microsoft Windows 2000
- Microsoft Active Directory, available with Microsoft Windows 2003
- Netscape Directory Server, Versions 4.1 and 4.2
- Sun ONE Directory Server 5.x

The Cisco IP Telephony Backup and Restore System (BARS) can be used to back up Cisco CallManager. Cisco BARS is installed separately from Cisco CallManager.

Cisco Unified CallManager Servers

In the original 2.4 version of Cisco CallManager, you were able to install the software on any platform that you desired. By allowing this freedom, a level of hardware-based instability plagued the original versions of the Cisco CallManager software. To compete with the legacy PBX-based voice network, VoIP networks should maintain an uptime of 99.999 percent. To achieve this, Cisco now requires you to install Cisco CallManager on a server that meets Cisco configuration standards. For this reason, Cisco has collaborated with two server hardware manufacturers, Hewlett-Packard and IBM, to create Cisco Media Convergence Servers (MCSs). Cisco chose these platforms because they have proven their reliability in the industry over time. Table 1-1 provides the current hardware specifications for the Cisco CallManager MCSs at the time of this writing.

Table 1-1 *Cisco Media Convergence Server Platforms*

Platform	Space	Processor	CPU Equipped	CPU Maximum	Maximum Phones Per Server
MCS 7815-I1	Tower	Pentium 4 3060 MHz	1	1	300
MCS 7825-I1	1U Rack Mount	Pentium 4 3400 MHz	1	1	1000
MCS 7835-I1	2U Rack Mount	Nocona Xeon 3400 MHz	1	2	2500
MCS 7825-H1	2U Rack Mount	Nocona Xeon 3400 MHz	2	2	7500
MCS 7835-H1	2U Rack Mount	Pentium 4 3400 MHz	1	1	1000
MCS 7845-H1	2U Rack Mount	Nocona Xeon 3400 MHz	1	2	2500
MCS 7845-H1	2U Rack Mount	Nocona Xeon 3400 MHz	2	2	7500

All of these servers, with the exception of the 7815-I1, are rack-mountable and do not include a monitor, mouse, or keyboard. Cisco designed the Cisco MCS for local setup, rack mounting, and remote administration.

NOTE The higher-end Cisco servers (beginning with the MCS 7835-I1) also include redundant power supplies and hard disk configurations.

Summary

At the turn of the millennium, Cisco moved forward with a new company direction described in the acronym of AVVID: Architecture for Voice, Video, and Integrated Data. Under this new banner, all three methods of communication would collapse under a single network infrastructure. The evolved term to describe this converged, IP-based environment is now Cisco Unified Communications. Standing at the forefront of the voice technology is the Cisco CallManager, the Cisco call-processing system that controls and manages converged voice over data networks.

Cisco CallManager runs on a foundation of Windows 2000 and SQL 2000 and can integrate with many popular directory services for user authentication purposes. Ever since the 3.x versions of Cisco CallManager, Cisco restricted the hardware platforms capable of running Cisco CallManager to the MCS series. This ensures a consistent hardware platform to provide stable support for the critical voice services of an organization.

Review Questions

You can find the solutions to these questions in Appendix A, "Answers to Review Questions."

1. The Cisco Unified Communications strategy is primarily focused around what goal?

   a. to create a new line of routers and switches equipped with increased processor and memory resources to handle a converged network environment

   b. to design network equipment equipped with the hardware resources and software features to handle a converged network environment

   c. to allow voice, video, and data to flow smoothly across the Internet

   d. to upgrade WAN links between locations to a speed capable of handling increased network traffic

2. In which layer of the voice infrastructure model does the Cisco Unified CallManager reside?

   a. infrastructure layer

   b. call-processing layer

   c. applications layer

   d. client layer

3. Which of the following functions are performed by the Cisco CallManager server? (Choose three.)
   a. call processing
   b. directory services
   c. PBX integration and signaling
   d. cial plan administration
   e. RTP audio processing
4. Which of the following protocols does the Cisco CallManager use to communicate with Cisco IP Phones?
   a. H.323
   b. MGCP
   c. Skinny
   d. SIP
5. Which of the following functions is the Cisco CallManager responsible for when setting up a phone call between Cisco IP Phones? (Choose three.)
   a. receiving dialed digits
   b. initiating ring tones on dialed IP Phone
   c. streaming RTP audio between devices
   d. handling call-processing functions when the calling party presses the Hold button
6. Which of the following software components constitute the foundation operating system of Cisco CallManager? (Choose two.)
   a. Windows 2000
   b. Windows 2003
   c. SQL 2000
   d. Exchange 2000
   e. Solaris
7. If you decide not to integrate Cisco CallManager into your company LDAP-compliant directory, what built-in option does Cisco CallManager provide to store user information?
   a. DC Directory
   b. Active Directory
   c. LW Directory
   d. Cisco LM Directory

8. You have purchased a Cisco MCS 7825 server. What is the maximum number of Cisco IP Phones you will be able to support?

   a. 250
   b. 500
   c. 1000
   d. 2500
   e. 7500

9. Maintaining a user directory is essential to the day-to-day operation of the IP telephony network. (True/False)

10. Cisco IP Phones can use which of the following signaling protocols? (Choose two.)

    a. H.323
    b. Skinny
    c. SIP
    d. MGCP

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This chapter covers the following topics:

- The concepts and design strategies behind a Cisco Unified CallManager cluster
- The types of replication that occur in a Cisco Unified CallManager cluster
- Properly designing the Cisco Unified CallManager cluster for redundancy
- The four Cisco Unified CallManager deployment models

CHAPTER 2

Cisco Unified CallManager Clustering and Deployment Options

Legacy PBX systems have quite the reputation for reliability. As a general track record, PBXs provide an uptime of 99.999 percent (known as the "five-nines" of reliability). This translates to an unbelievable 5.3 minutes of downtime per year! To match this standard of reliability, multiple Cisco CallManager servers are grouped into a cluster. Most administrators recognize the term "cluster" as an ability to allow multiple servers to act as one device, which is the type of clustering provided by Windows clustering services. The Cisco CallManager cluster operates quite differently.

After you understand the clustered relationship of Cisco CallManager, you can plan your deployment strategy using the four deployment models provided by Cisco CallManager. This chapter discusses the cluster relationship and deployment options provided by the Cisco CallManager and the options available to enterprises to deploy a highly available IP telephony network.

The Two Sides of the Cisco Unified CallManager Cluster

As a network administrator, you are accustomed to quite a bit of fiscal responsibility riding on your shoulders. Under the Cisco Unified Communications architecture, that responsibility has grown exponentially. Not only are you responsible for the operation of the data environment; you are now responsible for the company's voice network, which rides on top of this infrastructure. This voice network is critical to day-to-day business operations. Because of this, you should approach it just like any key network service: the more redundancy, the better.

A Cisco CallManager cluster is two or more servers grouped together to support a Cisco IP telephony network. The cluster relationship between Cisco CallManager servers provides redundancy and load balancing for the voice network. Cisco defines this cluster relationship in two ways: the SQL database structure and the intracluster run-time data.

The SQL Database Cluster

As discussed in Chapter 1, Cisco CallManager relies on Microsoft SQL 2000 as an information store for the data of the voice network. This data includes the phone extensions on the network, calling restrictions, route plan information, and so on. The database replication capability provided by Microsoft SQL Server makes clustering possible by allowing the same database to be on multiple machines. Database replication makes it appear as if a single machine is handling call processing along with other functions of the voice network and ensures that standby processors (Cisco CallManager servers) can seamlessly step in and fulfill the functions if the primary processor fails. This SQL database replication also ensures that all clustered Cisco CallManager servers have access to the same information.

You must have at least two Cisco CallManager servers to obtain this redundancy, and one of these servers must be a publisher database server. The publisher database server manages the only writable copy of the Microsoft SQL Server 2000 database. The subscriber database servers maintain read-only copies of the database. You can have only one publisher server and up to eight subscriber servers per cluster. It is these database servers that are able to actively participate in the call-processing functions of the Cisco voice network. This SQL limitation is also a key factor in determining the maximum size of a cluster, which is covered later in this chapter.

When you make changes to the Cisco CallManager configuration, these changes are made directly to the publisher server database. The publisher then replicates these changes to the subscriber servers. When the publisher server is offline, the Microsoft SQL Server 2000 database automatically locks, and thus prevents any database changes. The IP telephony network continues to operate, but you will not be able to add or configure any devices that are managed by Cisco CallManager. The only exception to this rule is the Call Detail Records (CDRs), which record information regarding the calls occurring within the cluster. When the publisher is down, the subscribers store CDRs until the publisher comes back online, and then the subscribers update the publisher with the CDRs.

In Cisco CallManager Release 3.3 and later, a single cluster is capable of handling approximately 30,000 Cisco IP Phones. This cluster limitation does not restrict the size of the voice over IP (VoIP) network. By creating additional clusters, you can increase the network size. However, the more clusters you create in a network, the more management the network requires to operate.

NOTE The 30,000 IP Phone maximum cluster size is only possible if you are using the MCS-7845 servers throughout your cluster deployment.

Intracluster Run-Time Data

The second communication method that defines the cluster relationship between Cisco CallManager servers is the intracluster run-time data, which is also called Intra-Cluster Communication Signaling (ICCS). This type of communication encompasses the "happenings" of the cluster. For example, when a new Cisco IP Phone connects to the network, it registers with its primary Cisco CallManager. That primary Cisco CallManager tells all the other servers in the cluster, "Hey everyone, a new phone just registered with me! I've given it the extension 4003." (IP address 10.5.5.1 in this example). All the other servers now know to send calls directed at extension 4003 to the Cisco CallManager at the IP address 10.5.5.1, which, in turn, makes the Cisco IP Phone ring.

After the initial phone registration, the Cisco IP Phone sends keepalive messages to the primary Cisco CallManager server every 30 seconds and sends a TCP connect message (which is technically a TCP three-way handshake) to its secondary Cisco CallManager server to ensure it is online and ready to accept a device failover, if necessary. When the Cisco IP Phone detects the failure of its TCP keepalive messages with the primary Cisco CallManager, the device attempts to register with a secondary Cisco CallManager server. The secondary Cisco CallManager server accepts the registration from the device and announces the new registration (using intracluster run-time data) to all of the Cisco CallManager servers in the cluster.

NOTE If the Cisco CallManager service is ever stopped manually (through the Services Windows 2000 Control Panel or a system shutdown), the Cisco CallManager server will clear all of its active TCP connections with the IP Phones. This causes them to failover immediately to their backup server rather than wait for the keepalive failure.

Cisco IP Phone registration is just one example of intracluster run-time data. You will discover many other types of intracluster communication as this book introduces new concepts in upcoming chapters.

Cluster Redundancy Designs

Because the IP telephony network is critical to business operation, you should never configure a single Cisco CallManager to support any organization. You should maintain at least two Cisco CallManager servers for the voice network to support the network should one server fail. Smaller organizations (240 users or less) might decide to use Cisco CallManager Express, which operates on an IOS-based router platform. The number of Cisco CallManager servers you use typically depends on the server platform you purchase and your cluster design strategy. Chapter 1, "Introduction to Cisco Unified Communications and Cisco Unified CallManager," discussed the server platforms and the number of phones that each server model supports. This section discusses the proper way to use those server platforms to build a Cisco CallManager cluster.

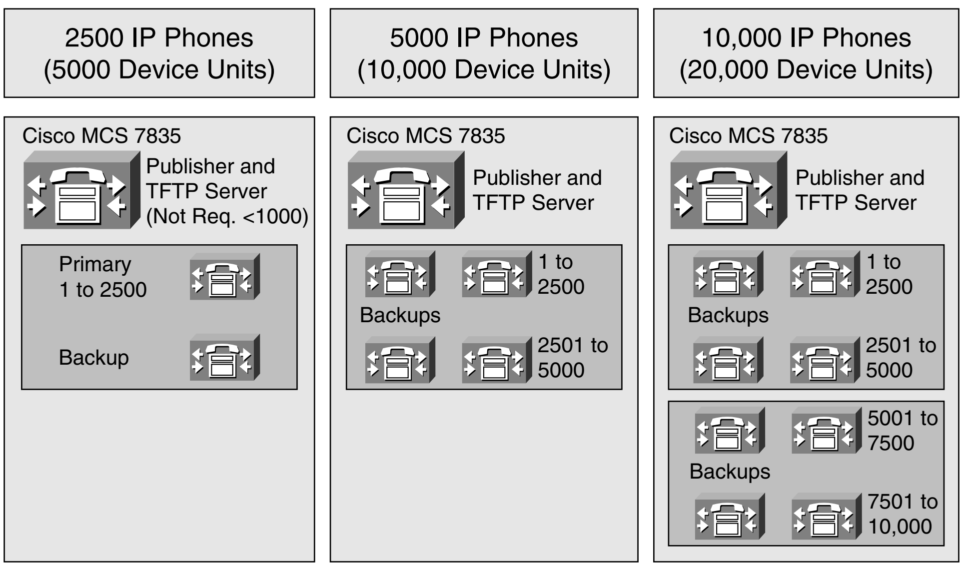
1:1 Redundancy Design

In a 1:1 Cisco CallManager redundancy deployment design, you can have a dedicated backup server for each primary server. This design guarantees that Cisco IP Phone registrations will never overwhelm the backup servers, even if multiple primary servers fail. However, the 1:1 redundancy design considerably limits the maximum cluster size and can be cost-intensive for the initial Cisco CallManager deployment.

Each cluster must also have a designated TFTP server. Depending on the number of devices that a server is supporting, you can combine this TFTP server functionality with the publisher or subscriber Cisco CallManager servers, or you can deploy the TFTP functionality on a separate, standalone server. The TFTP server is responsible for delivering IP Phone configuration files to each telephone, along with streamed media files, such as music on hold (MOH) and ring files; therefore, the TFTP server can experience a considerable network and processor load.

In the example shown in Figure 2-1, a Cisco 7835 Media Convergence Server (MCS) is used because each Cisco CallManager server installed on that platform supports a maximum of 2500 Cisco IP Phones.

Figure 2-1 *1:1 Server Redundancy Design*



The shading in the figure divides up the functions performed within a cluster. The lighter shading (outside the Primary/Backup server area) represents the server that is not participating in call processing. This is the SQL publisher server, which is also acting as a TFTP server. In cluster environments with less than 1000 IP Phones, the SQL publisher can also participate in the call-processing functions. However, in clusters larger than 1000 devices, you should isolate the SQL publisher because it will be quite busy handling updates to the SQL database and TFTP requests.

The darker shaded box (containing the Primary/Backup servers) represents Cisco CallManager servers handling the call-processing functions (intracluster run-time data) of the cluster. In the example showing cluster design for 2500 IP Phones, a single Cisco CallManager is the primary server, with a secondary server acting as a dedicated backup. The primary or backup server can also serve as the Microsoft SQL publisher and the TFTP server in smaller IP telephony deployments.

TIP Be sure you understand how the two types of intracluster communication apply to the Cisco CallManager cluster design. The publisher and subscriber relationship relates to the SQL database replication. The Primary and Secondary server roles (shown in Figure 2-1 as Primary and Backup) relate to the intracluster run-time relationship and device registration. The Primary and Backup servers will be SQL subscribers if you isolate the SQL publisher from call-processing functions.

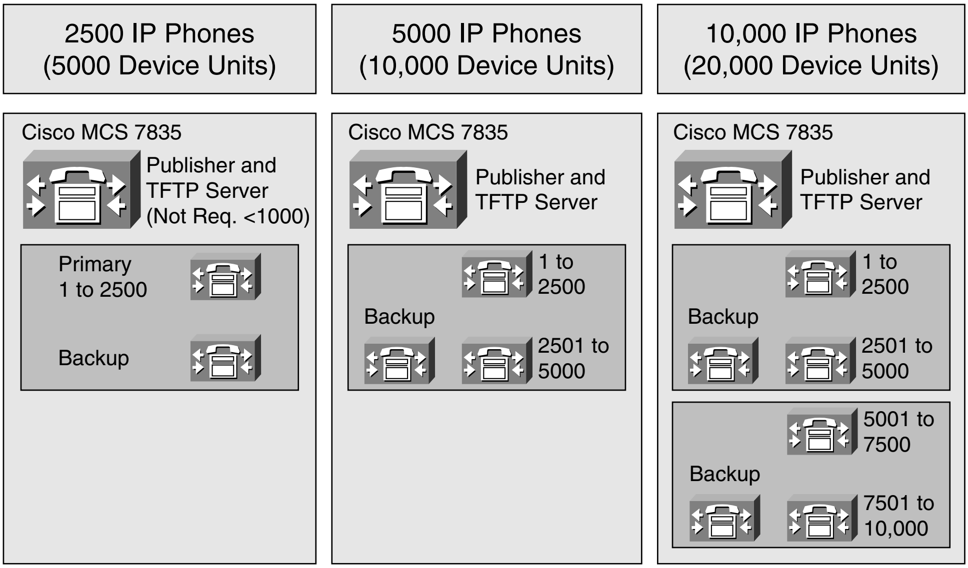
When you increase the number of IP Phones, you must increase the number of Cisco CallManager servers that are required to support the telephones. Some network engineers might consider the 1:1 redundancy design excessive because a well-designed network is unlikely to lose more than one primary server at a time. With the low possibility of server loss and the increased server cost, many network engineers elect to use a 2:1 redundancy design. However, other engineers choose to deploy the 1:1 redundancy to ensure maximum uptime of the IPT network. In addition, you can load-balance your IP Phones between the primary and backup servers in a 1:1 redundancy design. If any one server fails, only half of the phones must failover to the backup server (which provides a faster failover process).

2:1 Redundancy Design

In a 2:1 Cisco CallManager redundancy deployment design, you have a backup server shared between every two primary servers, as shown in Figure 2-2. Although this design offers some redundancy, there is the risk of overwhelming the backup server if multiple primary servers fail. In addition, upgrading the Cisco CallManager servers can cause a temporary loss of service because you must reboot the Cisco CallManager servers after the upgrade is complete.

NOTE Figure 2-2 assumes that each Cisco CallManager server can support a maximum of 2500 IP Phones.

Figure 2-2 *1:2 Server Redundancy Design*



Network administrators use this 2:1 redundancy model in most IP telephony deployments because of the reduced server costs. If you are using a Cisco MCS 7835 (shown in the figure), that server is equipped with redundant, hot-swappable power supplies and hard drives. When you properly connect and configure these servers, it is unlikely that multiple primary servers will fail at the same time, which makes the 2:1 redundancy model a viable option for most businesses.

Because of tight budgets, some network administrators are forced into 3:1 and even 4:1 redundancy designs (three or four primary servers supported by a single backup server). Cisco does not support these designs because of the high risk involved.

TIP The Cisco CallManager architecture limits a cluster to nine servers (one publisher and eight subscribers) that have access to the SQL database. Only these servers can participate in call-processing functions. The redundancy model and server platform you choose has a huge impact on the total size of your Cisco CallManager cluster. Although it might be possible to create a cluster capable of supporting a tremendous amount of IP Phones by minimizing redundancy, Cisco TAC only supports a maximum cluster size of 30,000 IP Phones. If you reach this maximum cluster size, you will need to divide your network into multiple clusters.

Call-Processing Deployment Models

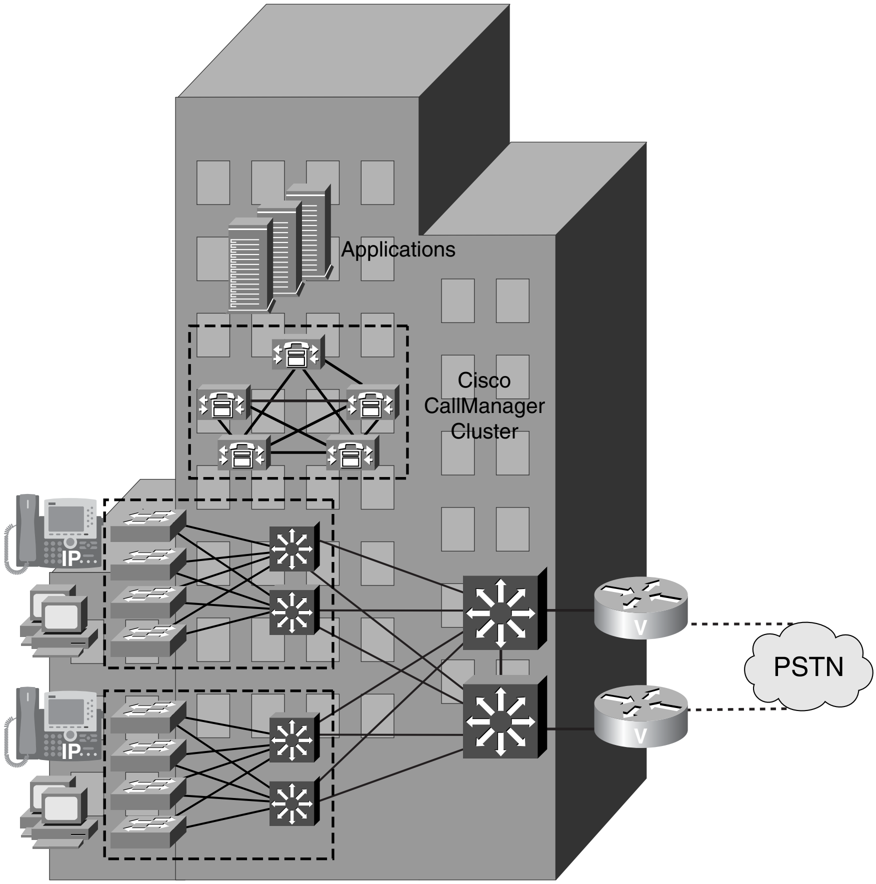
After you understand the concepts and design of a Cisco CallManager cluster, you can move on to the design of the organization IP telephony network. Cisco proposes four primary design models:

- Single-site deployment
- Multisite deployment with centralized call processing
- Multisite deployment with distributed call processing
- Clustering over the IP WAN

Single-Site Deployment

In a single-site deployment model, all Cisco CallManager servers, applications, and IP Phones are in the same physical location. A single Cisco CallManager cluster can support a maximum of 30,000 IP Phones. If you need to support more IP Phones at the site, you can implement multiple clusters within the single location and interconnect them through intercluster trunks (intercluster trunks are covered in Chapter 10, CallManager Route Plan Basics). Gateways that connect directly to the Public Switched Telephone Network (PSTN) handle external calls. If an IP WAN exists between sites, it is used to carry data traffic only; no telephony services are provided over the WAN. Figure 2-3 shows a typical network diagram of a single-site deployment.

Figure 2-3 *Single-Site Cisco CallManager Design*



Use the following guidelines for single-site deployments:

- You must understand the current calling patterns within the enterprise. How and where are users making calls? How many calls are intersite or interbranch versus intrasite? If calling patterns dictate that most calls are intrasite, use the single-site model to deploy IP telephony and make use of the relatively inexpensive PSTN. This design also simplifies the dial plans and avoids provisioning dedicated bandwidth for voice in the IP WAN.

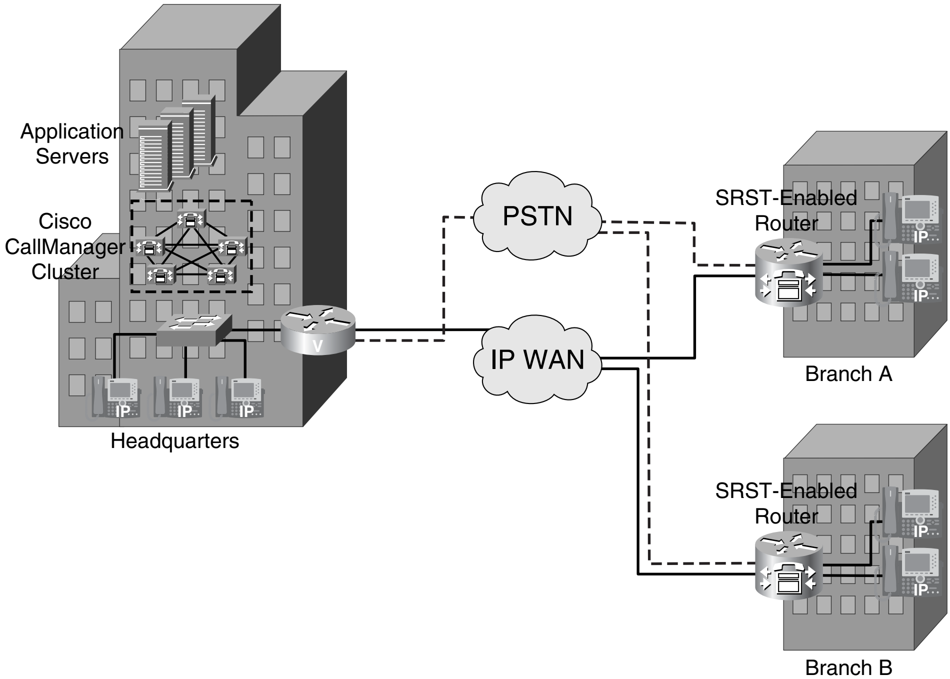
- The IP Phones should use the G.711 coder-decoder (codec). Because the call will stay in the LAN, G.711 is a simple and high-quality codec for deployment. It does not require dedicated Digital Signal Processor (DSP) resources for transcoding (which means converting between codec types, such as between G.711 and G.729), and older voice-mail systems might support only G.711. You can allocate these DSP resources to other functions, such as conferencing and Media Termination Point (MTP). Although the 64 kbps per-call bandwidth that G.711 consumes is higher than that of other commonly used codecs, it is not a concern in this design because the call is not traversing the WAN, where bandwidth is generally limited.

- All off-net calls will be diverted to the PSTN or sent to the legacy PBX for call routing if the PSTN resources are being shared during migratory deployments.
- Use Media Gateway Control Protocol (MGCP) gateways for the PSTN if the network does not require H.323 functionality. Centralize the gateway dial plans by using H.323 gatekeepers when deploying multiple clusters, rather than using MGCP gateways. In addition, PSTN gateway redundancy should also be a consideration because a single gateway failure could affect the inbound and outbound PSTN calls for an entire corporation.
- Deploy the recommended network infrastructure for high-availability connectivity options for telephones (inline power), quality of service (QoS) mechanisms, and other services.
- Do not oversubscribe Cisco CallManager to scale larger installations. In earlier software releases, Cisco used various schemes to allow the capacity of a system to be calculated using device weights, busy hour call attempt (BHCA) multipliers, and dial plan weights. With Cisco CallManager Release 4.0, this scheme has been replaced by a capacity tool to allow for more accurate planning of the system. The capacity planning tool is currently available only to Cisco employees. If your system does not meet the guidelines found in the Cisco IP Telephony Solution Network Reference Design, or if you consider the system to be more complex and would like to verify the capacity, please contact your Cisco Systems engineer. The Cisco IP Telephony Solution Reference Network Design (SRND) is found at: http://www.cisco.com/univercd/cc/td/doc/solution/esm/ (or can be browsed to through http://www.cisco.com/go/srnd).

Multisite Deployment with Centralized Call Processing

Figure 2-4 illustrates the multisite centralized call-processing deployment model with a Cisco CallManager cluster at a central site and a connection to several remote sites through a QoS-enabled IP WAN. The remote sites rely on the centralized Cisco CallManager cluster to handle call processing. Applications such as voice mail and interactive voice response (IVR) systems usually reside at the central site, thus reducing the overall cost of ownership and centralizing administration and maintenance. However, centralizing these application servers is not a requirement.

Figure 2-4 *Multisite Deployment with Centralized Call Processing*



Routers that reside at WAN edges require QoS mechanisms, such as low latency queuing (LLQ) and traffic shaping, to protect voice traffic from data traffic across the WAN (where bandwidth is typically scarce).

To avoid oversubscribing the WAN links with voice traffic (thus causing deterioration of the quality of established calls), the network might need a call admission control scheme. With the introduction of Cisco CallManager Release 3.3, centralized call-processing models can take advantage of automated alternate routing (AAR) features. AAR allows Cisco CallManager to dynamically reroute a call over the PSTN if the call exceeds the WAN bandwidth.

NOTE AAR is a feature specific to centralized call-processing environments and is discussed in depth in Chapter 14, Implementing Multiple-Site Deployments.

You can provide PSTN access for the voice network through a variety of Cisco gateways. When the IP WAN is down, the users at remote branches can place their calls through the PSTN using the Cisco Survivable Remote Site Telephony (SRST) feature that is available for Cisco IOS gateways that can provide call processing during the outage.

Follow these best-practices guidelines when deploying a centralized call-processing model:

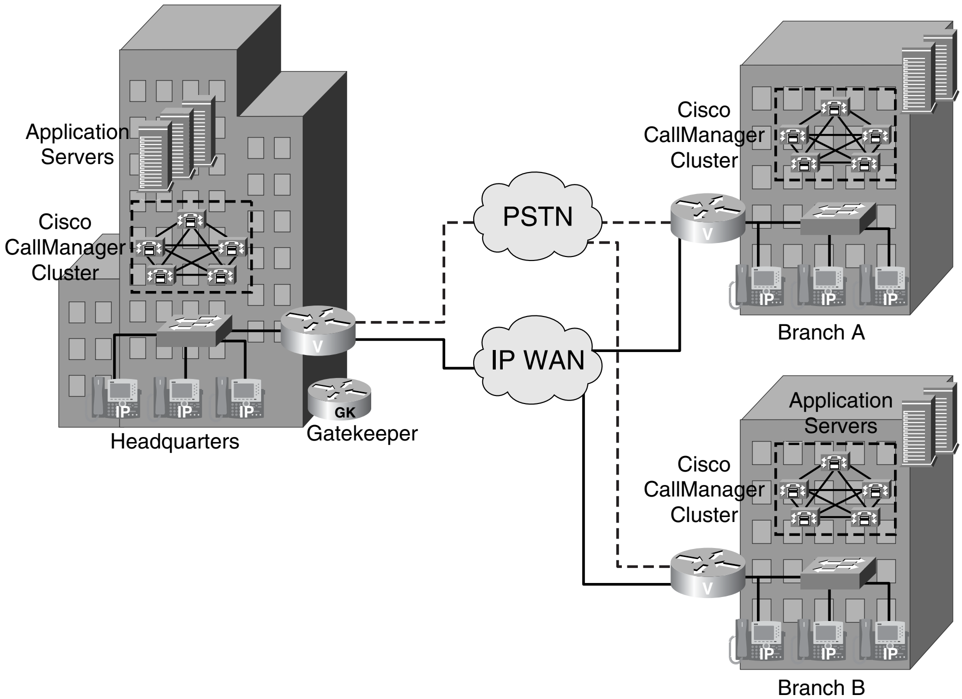
- Installations adopting the centralized call-processing deployment model are limited to hub-and-spoke topologies because the locations-based call admission control mechanism used in centralized call-processing deployments records only the available bandwidth in and out of each location.
- There is no limit to the number of IP Phones at each individual remote branch. However, the capability that is provided by the SRST feature in the branch router limits remote branches to 720 Cisco IP Phones using a 3845 router during failover. Smaller platforms have lower limits. SRST is covered in its entirety in Chapter 14, "Implementing Multiple-Site Deployments."
- Controlling WAN bandwidth utilization in a centralized call-processing deployment model is quite simple. Through the use of the Cisco CallManager Locations feature, the central Cisco CallManager cluster can decide if the remote-site call should use the WAN or PSTN for its connection.
- When communicating over the IP WAN, the G.729 compressed codec should be used to save considerable WAN bandwidth.

Multisite Deployment with Distributed Call Processing

A multisite distributed call-processing deployment has one or more call-processing agents at each site, and each site has its own Cisco CallManager cluster consisting of two or more servers. By separating these sites into different clusters, the Cisco CallManager servers do not share the SQL database and intracluster run-time data between sites. This requires some additional configuration because it will be necessary to trunk these sites together through an IP WAN. Figure 2-5 illustrates a typical multisite design using distributed call processing.

When deciding between a centralized or distributed call-processing model, you will usually base your decision on two factors: how many phones must you support at each site and what features does the company require to be available during failover. If you are using a smaller number of IP Phones and only require basic call-processing functionality, the centralized call-processing model might be most appropriate. If you are using a larger number of IP Phones or require more advanced features (such as call center applications), a distributed model might be more appropriate.

Figure 2-5 *Multisite Deployment with Distributed Call Processing*



Depending on your network design, an individual site in the distributed call-processing design might consist of the following:

- A single site with its own call-processing agent, which can be a Cisco CallManager or a third-party call agent
- A centralized call-processing site (and all of its remote sites) that the network views as a single site for distributed call processing
- A legacy PBX with a VoIP gateway or a legacy PBX that is attached using a time-division multiplexing (TDM) interface to a VoIP gateway

The distributed call-processing model requires all sites to be connected through an IP WAN. Cisco considers a site connected only through the PSTN to be a standalone site.

Multisite distributed call processing allows each site to be completely self-contained. In the event of an IP WAN failure or insufficient bandwidth, the site does not lose call-processing service or functionality. Cisco CallManager simply sends all calls between the sites across the PSTN.

The main benefits of this deployment model are as follows:

- Cost savings when you are using the IP WAN for intersite calls. The IP Phones and voice gateways can compress calls using the G.729 codec, reducing the bandwidth required for the audio and increasing efficiency over the standard PSTN connections.
- Toll-bypass savings when you are using remote gateways to drop off into the PSTN (known as "tail end hop off," or TEHO). For example, if one of your sites is located in Arizona, all the other sites can forward calls across the IP WAN to obtain toll-bypass when calling a number in Arizona.
- No loss of functionality during an IP WAN failure.
- Scalability to hundreds of clusters.

The multisite WAN with distributed call-processing deployment model is a superset of the single-site and multisite WAN with centralized call-processing models. You should follow the best-practices guidelines mentioned previously for single-site and multisite deployments in addition to those listed here, which are specific to this deployment model.

The H.323 gatekeeper or Session Initiation Protocol (SIP) proxy servers are among the key elements in the multisite WAN with distributed call processing. Both provide centralized dial plan resolution, with the gatekeeper also providing call admission control.

A gatekeeper is an H.323 device that provides call admission control and centralized dial plan resolution. Otherwise, the network administrator at each of the sites must configure a full dial plan to reach all devices on the network. Additional gatekeeper guidelines include the following:

- Cisco recommends that you use backup gatekeeper support to provide a gatekeeper solution with high availability. It is also recommended that you use multiple gatekeepers to provide spatial redundancy within the network.
- Cisco recommends that you use a single WAN codec. This design makes capacity planning easy and does not require you to overprovision the IP WAN to allow for worst-case scenarios.

SIP proxy servers provide resolution of dialed numbers (also called E.164 numbers) as well as SIP uniform resource identifiers (URIs) to enable endpoints to place calls to each other. Cisco CallManager supports the use of E.164 numbers only.

The following best practices apply to the use of SIP proxies:

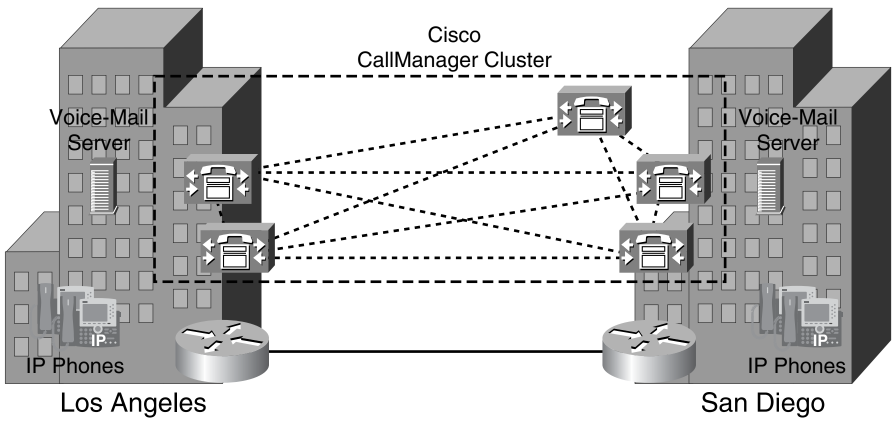
- Provide adequate redundancy for the SIP proxies.
- Ensure that the SIP proxies have the capacity for the call rate and number of calls required in the network.

For more detail on bandwidth capacity planning and call admission control for each deployment model, refer to the Cisco IP Telephony Solution Reference Network Design (SRND) for Cisco CallManager 4.0 at: http://www.cisco.com/univercd/cc/td/doc/solution/esm/iptele/index.htm.

Clustering over the IP WAN

Cisco supports Cisco CallManager clusters over a WAN, shown in Figure 2-6.

Figure 2-6 *Clustering over the IP WAN*



Although there are stringent requirements, this design offers the following advantages:

- Single point of administration for users for all sites within the cluster
- Feature transparency; features such as call transfers and conference calls work seamlessly using an internal dialing scheme
- Shared line appearances over a WAN connection, which allows multiple phones to share a line instance
- Extension mobility within the cluster, allowing users to log in to remote phones and have their phone profile follow them

This design is useful for customers who require more functionality at remote sites than the limited feature set that is offered by SRST. This network design also allows remote offices to support more Cisco IP Phones than SRST in the event that the connection to the primary Cisco CallManager is lost.

Although the distributed single-cluster call-processing model offers some significant advantages, it must adhere to these strict design guidelines:

- Two Cisco CallManager servers in a cluster must have a maximum round-trip delay of 40 ms between them. In comparison, high-quality voice guidelines dictate that one-way end-to-end delay should not exceed 150 ms. Because of this strict guideline, you can use this design only between closely connected, high-speed locations (Metropolitan Ethernet would be a good example of this connectivity type).
- For every 10,000 Busy Hour Call Attempts (BHCAs) within the cluster, you must support an additional 900 kbps of WAN bandwidth for intracluster run-time communication. The BHCA represents the number of call attempts made during the busiest hour of the day.
- Up to eight small sites are supported using the remote failover deployment model. Remote failover allows you to deploy the backup servers over the WAN. Using this deployment model, you can have up to eight sites with Cisco CallManager subscribers being backed up by Cisco CallManager subscribers at another site.
- SRST can function in this model but is not necessary to protect against Cisco CallManager failure. The telephones can failover across the WAN to other Cisco CallManager servers. This design may require significant additional bandwidth, depending on the number of telephones at each location.

NOTE If the WAN connection between sites fails, users will receive a fast busy signal for any intersite call. This might be confusing to users who are used to ringing followed by voice mail.

Summary

This chapter covered the key factors you must consider when you are initially designing a Cisco CallManager–based IP telephony deployment. The key to understanding your deployment options is first understanding the Cisco CallManager cluster relationship as defined by the Microsoft SQL replication and intracluster run-time data. The Cisco CallManager cluster provides redundancy and failover in an IP telephony environment. This redundancy can be implemented using the 1:1 server redundancy design, which offers the most redundancy available, or a 2:1 redundancy design, which balances the cost factors with overall redundancy. The chapter ended with a discussion of the four call-processing deployment models:

- Single-site deployment
- Multisite deployment with centralized call processing
- Multisite deployment with distributed call processing
- Clustering over the IP WAN

Review Questions

You can find the solutions to these questions in Appendix A, "Answers to Review Questions."

1. Which four of the following are IP telephony deployment models that are supported by Cisco? (Choose four.)
   a. a single site with one call-processing agent
   b. multiple sites with centralized call processing
   c. multiple sites each with its own call-processing agent
   d. a single cluster with distributed call processing
   e. multiple clusters with no call-processing agent
2. A 1:1 redundancy design offers _________.
   a. increased redundancy; however, the increased server cost is often prohibitive
   b. some redundancy; however, a server reboot is required after an upgrade
   c. maximum uptime; however, no more than a 20-ms round-trip delay can exist between servers
   d. high availability; however, you might overwhelm the backup servers
3. A single cluster that spans multiple sites can have which two benefits compared to a branch office in a multisite WAN with a centralized call-processing deployment? (Choose two.)
   a. completely self-contained individual sites
   b. WAN bandwidth cost savings
   c. a common dial plan across all sites
   d. more IP Phone features during failover
   e. scalability to hundreds of sites
4. Which two of the following enable the cluster to achieve redundancy? (Choose two.)
   a. Windows clustering
   b. database replication
   c. at least two servers
   d. directory access
5. When configuring the SQL database relationship between Cisco CallManager servers, how many publisher servers should you configure for the cluster?
   a. It depends on the server platform; you should have one publisher for each group of phones followed by a backup.
   b. You should have only one publisher server for the entire cluster.

c. You should have at least two publisher servers to provide database redundancy.
d. You must configure a publisher for each subscriber you install.

6. What is the maximum number of SQL subscribers that can exist in a Cisco CallManager cluster?
   a. 3
   b. 5
   c. 8
   d. 9
   e. 12
7. To support a centralized, multisite Cisco CallManager design, what feature should be employed at the remote offices?
   a. QoS
   b. SRST
   c. Classification
   d. SQL database replication
8. When using a multisite, single-cluster Cisco CallManager design, it is not necessary to employ SRST features to support the IP Phones in the case of Cisco CallManager failure. (True/False)
9. When designing a distributed, multicluster Cisco CallManager design, what pieces of equipment might be necessary to ensure the IP telephony network maintains a consistent and centralized dial plan? (Choose two.)
   a. H.323 gatekeeper
   b. H.323 gateways
   c. SIP proxy server
   d. Cisco CallManager
10. What is the maximum Cisco CallManager cluster size Cisco will support using Cisco CallManager 4.x?
    a. 7500 IP Phones
    b. 10,000 IP Phones
    c. 12,000 IP Phones
    d. 15,000 IP Phones
    e. 30,000 IP Phones

This chapter covers the following topics:

- The requirements to perform Cisco Unified CallManager server installation
- Cisco Unified CallManager installation process
- Cisco Unified CallManager postinstallation procedures
- Cisco Unified CallManager upgrade process

CHAPTER 3

Cisco Unified CallManager Installation and Upgrades

When Cisco first announced that the Cisco CallManager platform would use Microsoft Windows 2000 as a foundation operating system and Microsoft SQL 2000 as a database store, concern arose that the Cisco administrator would require a thorough understanding of these applications to successfully operate a Cisco IP telephony network. Although an understanding of these components can be useful for monitoring and troubleshooting purposes, it is not necessary for Cisco CallManager installation and setup. Cisco has done a fantastic job of "wizard-izing" and scripting the complete installation of both Windows 2000 and SQL 2000, hiding any complexity behind a friendly **Next** button.

The upgrade process of Cisco CallManager is not quite as friendly as a clean install; however, if you keep the potential pitfalls in mind, a Cisco CallManager upgrade can go quite smoothly. This chapter discusses both the clean install and upgrade process of Cisco CallManager.

Cisco Unified CallManager 4.x Clean Installation Process

A clean installation of Cisco CallManager has always been an extremely simple process. As you perform the Cisco CallManager installation, the automated setup process prompts you for the information that is necessary to build Windows 2000, Microsoft SQL Server 2000, and Cisco CallManager with a base configuration. The entire operating system installation process, excluding preinstallation tasks, takes approximately 25 to 45 minutes per server, depending on your server type. Installing Cisco CallManager, excluding pre- and postinstallation tasks, takes 45 to 90 minutes per server, depending on your server type.

Installation Disks

All Cisco MCSs and customer-provided servers that meet approved Cisco configuration standards ship with a blank hard drive. When you purchase a Cisco IP telephony application, you use the appropriate disks to install or upgrade the operating system and application:

- **Disk 1: Cisco IP Telephony Server Operating System Hardware Detection Disk—** Checks the server and displays an error message if it detects an unsupported server. After you boot the server using the Hardware Detection CD-ROM, the automated installation process prompts for the correct CD-ROMs to use based on the type of hardware platform detected.

- **Disk 2: Cisco IP Telephony Server Operating System Installation and Recovery Disk**—Installs the operating system. Use only one of the server-specific Cisco IP Telephony Server Operating System Installation and Recovery disks that come in your software kit. Depending on your platform, the Operating System disc could be CD-ROM or DVD-based. After the operating system installation, a prompt instructs you to insert the appropriate Cisco CallManager software disk into the drive.
- **Disk 3: Cisco CallManager 4.1 Software Disk**— This disk installs the Cisco CallManager application on the server.

You might also receive a Cisco IP Telephony Server Operating System Upgrade Disk. Use this disk to upgrade the operating system on existing (not new) servers in the cluster. You do not need to use this disk if you are performing a new operating system installation.

Installation Configuration Data

As mentioned previously, the installation process for Cisco CallManager is automated by a step-by-step wizard. You will initially boot off the Hardware Detection CD-ROM, which will walk you through a wizard prompting you for the basic configuration data to get the server running. The process erases all data on the server hard disk. During the installation, you are prompted for the following items:

- **New installation or server replacement**—Choose this option if you are installing the Cisco IP telephony application for the first time, overwriting an existing installation, or replacing a server. To replace the server, you must store the data to a network directory or tape device before the operating system installation. Choosing this setting erases all existing drives.
- **Cisco product key**—Cisco supplies a product key when you purchase a Cisco IP telephony product. The product key is based on a file encryption system that allows you to install only the components that you have purchased. It also prevents you from installing other supplied software for general use. The product key consists of alphabetical characters only.
- **Username and organization name**—The system will prompt you for a username and an organization name to register the software product that you are installing. Do not leave the field blank. You can enter letters, numbers, hyphens (-), and underscores (_).
- **Computer name**—The system will prompt you to assign a unique computer name, using 15 characters or fewer, to each Cisco CallManager server. The computer name can contain alphabetic and numeric characters, hyphens, and underscores, but it must begin with a letter of the alphabet. Follow your local naming conventions, if possible. If you want to change the computer name after the application installation, you must completely reinstall the operating system and the application.

- **Workgroup—**The system will also prompt you for a workgroup name. A workgroup consists of a collection of computers that share the same workgroup name. Computers in the same workgroup can more easily communicate with each other across the network. Ensure that this entry, which must also be 15 characters or fewer, follows the same naming conventions as the computer name.
- **Domain suffix—**When prompted, you must enter the Domain Name System (DNS) suffix in the format "mydomain.com" or "mycompany.mydomain.com." If you are not using DNS, use a fictitious domain suffix, such as fictitioussite.com.
- **TCP/IP properties—**You must assign an IP address, subnet mask, and default gateway when installing a Cisco CallManager server. Changing the Cisco CallManager IP address after you install the software can be a tedious process, so be sure to plan accordingly.

CAUTION It is strongly recommended that you choose static IP information, which ensures that the Cisco CallManager server obtains a fixed IP address. With this selection, Cisco IP Phones can register with Cisco CallManager when the telephones are plugged into the network. Using Dynamic Host Configuration Protocol (DHCP) can cause problems, including failure of the telephony system.

- **DNS—**You can identify a primary DNS server for this optional field. By default, the telephones will attempt to connect to Cisco CallManager using DNS. Therefore, you must verify that the DNS server contains a mapping of the IP address and the fully qualified domain name (FQDN) of the Cisco CallManager server. If you do not use DNS, use the server IP address, instead of a server name, to register the telephones with Cisco CallManager.

NOTE Before you begin installing multiple servers in a cluster, you must have a name resolution method in place, such as DNS, Windows Internet Naming Service (WINS), or local name resolution using a configured LMHOSTS file. If you use DNS, you must verify that the DNS server contains a mapping of the IP address and the hostname of the server that you are installing. This verification must take place before you begin the installation. If you use local name resolution, ensure that the LMHOSTS file is updated on the existing servers in the cluster before you begin the installation on the new subscriber server. You must add the same information to the LMHOSTS file on the new server during installation.

TIP Although it might seem tedious, Cisco considers the creation of LMHOST file IP address to hostname mappings on each Cisco CallManager server a better practice. Using DNS services introduces another point of failure for the voice network.

- **SNMP community string—**The Windows 2000 Simple Network Management Protocol (SNMP) agent provides security through the use of community names and authentication traps. Cisco sets the community rights to none for security reasons. If you want to use SNMP with this server, you must configure it.
- **Database server—**You must determine whether you will configure this server as a publisher database server or as a subscriber database server through a radio button selection during the Cisco CallManager installation. This selection is permanent. You must reinstall the Cisco CallManager server if you want to reassign the database server type at a later date.

NOTE You must install a Cisco CallManager publisher server before you can install any subscriber servers. When you are configuring a subscriber database server, ensure that the server that you are installing can connect to the publisher database server during the installation. This connection facilitates the copying of the publisher database to the local drive on the subscriber server. You must supply the name of the publisher database server and a username and password with administrator access rights on that server. The installation will be discontinued if, for any reason, the publisher server cannot be authenticated.

- **New password for the system administrator—**Cisco CallManager Releases 3.0 and later support password protection. A prompt at the end of the installation procedure will ask you to supply a new password for the system administrator.

NOTE For Cisco CallManager database replication, you must enter the same Administrator account password for the publisher and all of the subscribers in the cluster. The installation wizard will request this password.

Sample Configuration Data Worksheet

Table 3-1 shows the configuration information that you need to install the Cisco CallManager software on your server. You should complete all of the fields in the table, unless otherwise noted. You must gather this information for each Cisco CallManager server that you are installing in the cluster. Make copies of this table, and record your entries for each server in a separate table. Table 3-1 summarizes the data you should have available when you begin the installation.

Table 3-1 *Configuration Data for Cisco MCS*

Configuration	Data
Cisco product key	
Username	
Name of your organization	

Table 3-1 *Configuration Data for Cisco MCS (Continued)*

Configuration	**Data**
Computer name	
Workgroup	
Microsoft NT domain (optional)	
DNS domain suffix	
Current time zone, date, and time	
DHCP parameters	It is recommended that you program a fixed IP address in TCP/IP properties for the server instead of using DHCP.
TCP/IP properties (required if DHCP is not used): • IP address • Subnet mask • Default gateway	
DNS servers (optional): • Primary • Secondary WINS servers (optional): • Primary • Secondary • LMHOSTS file (optional)	
Database server (choose one): • Publisher • Subscriber If you are configuring a subscriber server, supply the username and password of the publishing database server: • Publisher username • *Publisher password*	
Backup (choose one or both): • Server • Target	
New Windows 2000 administrator password	

Postinstallation Procedures

After you complete the Cisco CallManager software installation, the installation wizard will prompt you to change all passwords used in the Cisco CallManager cluster. These passwords should be the same on all servers you install into the cluster. In addition, many supporting services are running on your server that you might be able to stop. The fewer services you have running on your server, the more server resources you will have available to support the IP telephony network. In addition, running more services on the Cisco CallManager server introduces more security vulnerabilities for the underlying Windows operating system. You should stop all of the following services on both the Publisher and Subscriber servers in your cluster and set them to manual-start status unless they are otherwise needed on the system:

- DHCP client
- Fax service
- FTP Publishing Service
- Smart Card (unless using security tokens)
- Smart Card Helper
- Computer browser
- Distributed File System
- License Logging Service

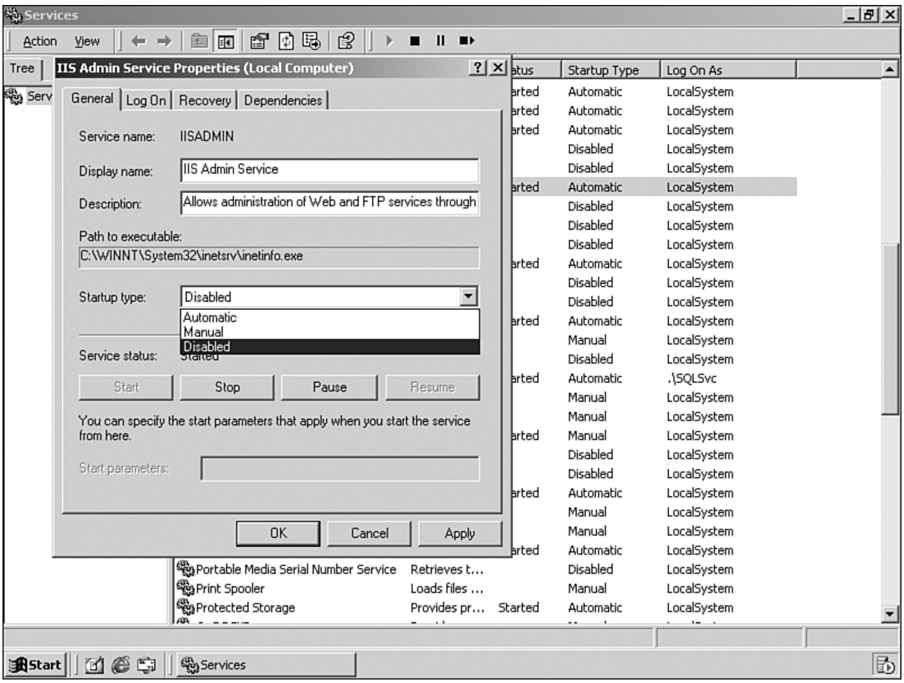
By default, the installation wizard configures all Subscribers with Internet Information Server (IIS) Services running. This allows you to make changes to the cluster by accessing the web interface on your subscriber servers. Even though you are accessing the web interface on the Subscriber server, the changes are actually being made on the Publisher server (because it has the only writable copy of the database). In addition, allowing the web services to run on all Subscriber servers introduces more security risk as there are now multiple points of access for the Cisco CallManager administration interface. Because of this, it is usually best to save the Subscriber resources by stopping the web services on all servers except the Publisher. You can accomplish this by stopping the following services:

- Microsoft Internet Information Server (IIS) Admin Service
- World Wide Web Publishing Service

You can stop all of these services through the Windows 2000 Services console. To open this console, click **Start > Programs > Administrative Tools > Services**. When the console opens, Windows lists all services in alphabetic order. Right-click on the service you want to disable and choose **Properties**. In the Properties window shown in Figure 3-1, use the drop-down box to select either a **Manual** startup or to **Disable** the service. These will take effect the next time you reboot the Cisco CallManager server. You can also choose to stop the service without restarting the server from this page.

> **TIP** Because Cisco CallManager requires these services to be active when upgrading to new Cisco CallManager versions, setting them to a state of Manual is suggested.

Figure 3-1 *Windows 2000 Service Properties*



Activating Cisco Unified CallManager Services

If you are installing Cisco CallManager for the first time, all services that are required to run Cisco CallManager automatically install on the system; however, none of the services are activated at the completion of the installation (except for the Cisco Database Layer Monitor service). Cisco CallManager Serviceability provides a web-based Service Activation tool that is used to activate or deactivate multiple services and to select default services to activate.

It is recommended that you activate only the required components for each server in the cluster. Each component that you activate adds to the server load.

If you are upgrading Cisco CallManager, the services that you have already started on your system will start after the upgrade.

Each service performs specific functions for the IP telephony network. Some services might need to run on a single Cisco CallManager server in a cluster; other services might need to run on all of the Cisco CallManager servers in the cluster.

CAUTION Be sure to activate at least the Cisco CallManager service before you apply any configuration to your Cisco CallManager server. Failure to do so can lead to unpredictable results, potentially leading to a server reinstall.

The following information briefly describes each available Cisco CallManager service:

- **Cisco CallManager Service**—Allows the server to participate in telephone registration, call processing, and other Cisco CallManager functions. Cisco CallManager Service is the core service of the Cisco CallManager platform.
- **Cisco TFTP**—Activates a TFTP server on Cisco CallManager. The TFTP service delivers Cisco IP Phone loads and configuration files to IP Phones, along with streamed media files, such as music on hold (MOH) and ring files.
- **Cisco Messaging Interface**—Allows Cisco CallManager to interface with a Simplified Message Desk Interface (SMDI)-compliant, external voice-mail system.
- **Cisco IP Voice Media Streaming Application**—Allows Cisco CallManager to act as a Media Termination Point (MTP), a conference bridge, a music on hold (MOH) server, and an annunciator. The voice network uses these media resources for feature functionality.
- **Cisco CTIManager**—Allows Cisco CallManager to support computer telephony integration (CTI) services and provides Telephony Application Programming Interface (TAPI) or Java Telephony Application Programming Interface (JTAPI) client support. Cisco CTIManager allows you to use applications such as Cisco IP SoftPhone.
- **Cisco Telephony Call Dispatcher**—Distributes calls to multiple telephone numbers (hunt groups). Cisco WebAttendant, Attendant Console, and Auto Attendant require Cisco Telephony Call Dispatcher (TCD).
- **Cisco MOH Audio Translator**—Allows Cisco CallManager to convert MP3 or WAV audio files into voice codec format used for MOH.
- **Cisco Real-Time Information Server (RIS) Data Collector**—Allows Cisco CallManager to write trace and alarm file information to a database, Microsoft Event Viewer, or alert an SNMP server.

- **Cisco Database Layer Monitor**—Monitors aspects of the Microsoft SQL 2000 database, as well as call detail records (CDRs).
- **Cisco CDR Insert**—Allows Cisco CallManager to write CDRs to the local database and replicates CDR files to the Microsoft SQL publisher at a configured interval.
- **Cisco CTL Provider**—Works with the Cisco Certificate Trust List (CTL) client to change the security mode for the cluster from nonsecure to secure (called mixed mode).
- **Cisco Extended Functions**—Provides support for some Cisco CallManager features, including Cisco Call Back and Quality Report Tool (QRT).
- **Cisco Serviceability Reporter**—Generates the following daily reports: Device Statistics, Server Statistics, Service Statistics, Call Activities, and Alert.
- **Cisco WebDialer**—Provides click-to-dial functionality by using a web page or a desktop application.
- **Cisco IP Manager Assistant**—Allows Cisco CallManager to support the Cisco IP Manager Assistant (IPMA), an application designed to allow a receptionist and manager to support special functionality between their phones.
- **Cisco CallManager Extension Mobility**—Allows Cisco CallManager to support extension mobility functions for roaming users. By using extension mobility features, you can assign users roaming phone profiles that activate when they log in to a phone. Works similarly to the roaming profiles in a Windows-based domain environment.
- **Cisco Certificate Authority Proxy Function**—Working in conjunction with the Cisco Certificate Authority Proxy Function (CAPF) application, the Cisco CAPF service can perform the following tasks, depending on your configuration:
    - Issues locally significant certificates to supported Cisco IP Phone models
    - Requests certificates from third-party certificate authorities on behalf of supported Cisco IP Phone models
    - Upgrades existing certificates on the phones
    - Retrieves phone certificates for troubleshooting
    - Deletes locally significant certificates on the phone

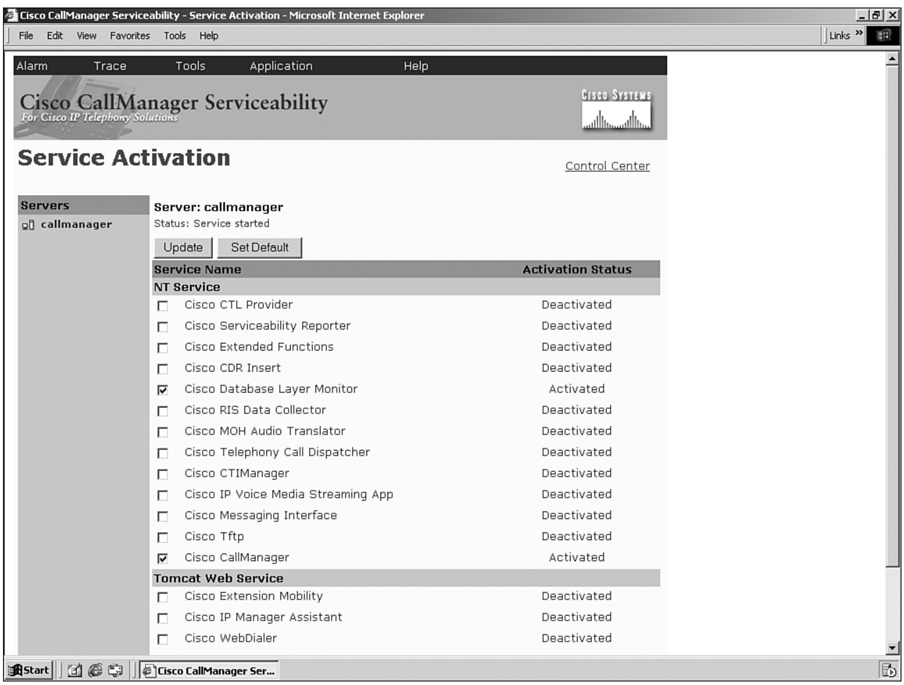
You must activate the Cisco CallManager services from the Service Activation web interface rather than the Windows 2000 Services control panel. To access this interface, perform the following steps:

Step 1 Open Internet Explorer, and go to https://<CallManager_IP_Address>/ccmadmin. The <CallManager_IP_Address> is the IP address of the Cisco CallManager server that is running IIS web services. Enter the administrative username and password information.

Step 2 From the Application menu, choose **Cisco CallManager Serviceability**. The Cisco CallManager Serviceability interface appears.

Step 3 From the Tools menu, choose **Service Activation**. A window similar to the window shown in Figure 3-2 appears.

Figure 3-2 *Cisco CallManager Service Activation Interface*



Step 4 Click the server that you want to configure from the Servers column. Next click the services that you want to activate, and click the **Update** button. (You will experience a slight delay.) The Service Activation window will refresh when the process is complete.

TIP The method shown is just one way to access the Cisco CallManager Serviceability pages. If you are working on the Cisco CallManager itself, you can get there quicker by using the Windows 2000 Start menu (**Start > Programs > Cisco CallManager > Cisco CallManager Serviceability**) or by accessing https://<CallManager-IP-Address>/CCMService.

CAUTION Remember to activate the Cisco CallManager services from the Service Activation web interface. Activating the services through the Windows 2000 Services console will produce unpredictable and unstable results.

When you click the **Set Default** button in the web interface, the Service Activation tool chooses the services required to run Cisco CallManager based on a single-server configuration. This is the bare minimum to have a working Cisco CallManager–based IP telephony network. Because Cisco highly advises against single-server installations, you will most likely use the **Set Default** button in a lab environment.

Upgrading Prior Cisco Unified CallManager Versions

Cisco supports the upgrade to Cisco CallManager 4.1 from Cisco CallManager Release 3.3(4), 3.3(5), 4.0(1), and 4.0(2a). You must first upgrades earlier versions to one of these releases before you can upgrade to version 4.1.

NOTE Cisco CallManager 4.1(3) is the most recent version available at the time of this writing. If you are upgrading to a more recent version, refer to the specific upgrade instructions provided in the documentation for the more recent Cisco CallManager version.

If your server runs a version of Cisco CallManager Release 3.2 or earlier, you must first upgrade every server in the cluster to the latest version of Cisco CallManager Release 3.3 before you can upgrade to a version of Cisco CallManager Release 4.1. This is because the database structure and storage system changes from versions prior to release 3.3. (Earlier versions use SQL 7.0 rather than SQL 2000.)

Before you perform any upgrade procedures, it is strongly recommended that you install the latest operating system upgrade and service release, SQL service releases and hotfixes, and Cisco CallManager service release for the versions that currently run in the cluster. Cisco provides the service release and corresponding “readme” documentation on Cisco.com. To obtain these documents, go to http://www.cisco.com/kobayashi/sw-center/sw-voice.shtml (CCO login is required).

NOTE Cisco releases all service packs, hotfixes, and Windows operating system upgrades through the CCO website. Updates released by Microsoft should *never* be applied to the Cisco CallManager server as they can potentially cause the Cisco CallManager software to become unstable or to crash. Cisco guarantees that they will release any critical operating system patch within 24 hours of the Microsoft announcement. Noncritical patches are rolled up into a monthly update.

Cisco requires that you install Cisco IP Telephony Server Operating System Version 2000.2.6 (available from Cisco CCO) before you upgrade to Cisco CallManager Release 4.1.

CAUTION Because the foundation operating system components might change, the upgrade procedure will many times require you to back up the database on the Publisher server, reimage Cisco CallManager completely (using the clean installation method discussed earlier in the chapter), and then restore the SQL data from backup. If your version of Cisco CallManager requires this, be absolutely sure to install the newer backup software on the Cisco CallManager you are upgrading before you initially back up the database. Many times, the backup software changes between the 3.x and 4.x versions of Cisco CallManager. Because of this, you will usually find the latest version of the backup software in the /backup folder on the first Cisco CallManager Installation CD-ROM. If you do not install the new backup software before performing the Cisco CallManager upgrade, your backup file (and SQL data) might be unreadable in the new Cisco CallManager version.

Because many of the Cisco CallManager servers are equipped with RAID 1 configurations (mirrored hard disks), a common upgrade strategy is as follows:

Step 1 Remove the mirrored hard disk from the production Cisco CallManager server.

Step 2 Boot a lab server using the mirrored hard disk.

Step 3 Perform the Cisco CallManager software upgrade on the mirrored hard disk.

Step 4 During a scheduled window of downtime, reboot the production server using the upgraded, mirrored drive.

Step 5 The hard drive running the older Cisco CallManager version will become the new, backup mirrored drive.

By using this upgrade process, you can reduce the amount of voice network downtime to the amount of time it takes to reboot the Cisco CallManager server on the upgraded drive.

CAUTION The upgrade process described will cause a loss of any database changes performed from the time you have removed the mirrored drive until the time you reboot your Cisco CallManager server using the new version. Plan your upgrade windows accordingly.

Summary

This chapter covered the Cisco CallManager installation and upgrade process. When you purchase a Cisco MCS and Cisco CallManager software, you will receive a number of CDs (and even DVDs). Of those, you will only need three discs for the installation: the hardware detection CD, the operating system CD/DVD, and the Cisco CallManager installation CD/DVD. The installation itself is very automated and disk image-based. It requires very little knowledge of the foundation Windows 2000 operating system and SQL 2000 database store. After the installation completes, you should stop any unnecessary services running on your Cisco CallManager and start the Cisco CallManager services required for your network.

The upgrade process for Cisco CallManager varies wildly depending on the version of Cisco CallManager you are using. The process can be as simple as inserting the Cisco CallManager installation disk and following a step-by-step wizard or as complex as a complete reimage of all servers in the cluster. The key is to make sure that you have backed up all data from the SQL server with the newest version of the backup software available from the Cisco CallManager installation CD-ROM or the Cisco website.

Review Questions

You can find the solutions to these questions in Appendix A, “Answers to Review Questions.”

1. Cisco CallManager uses which of these operating systems?
   a. Linux
   b. Windows NT
   c. Windows 2000
   d. Windows 2003

2. Why is it recommended that you stop IIS on the subscriber servers? (Choose two.)
   a. enhances server call processing and redundancy
   b. maximizes the number of devices in a cluster
   c. disables the use of remote terminal services
   d. helps prevent unauthorized access to the server
   e. makes more resources available for critical voice services
3. When you first install Cisco CallManager software, which CD-ROM should you use to boot the server to determine the correct CD-ROM to insert next?
   a. Cisco CallManager 4.0 Software Disk
   b. Cisco IP Telephony Server Operating System Hardware Detection Disk
   c. Cisco CallManager Installation, Upgrade, and Recovery Disk
   d. Cisco IP Telephony Server Backup and Restore Disk
   e. Cisco Extended Services and Locales Disk
4. If you are not using DNS, what must you configure to resolve server names for the Cisco CallManager installation process?
   a. DHCP
   b. backup server
   c. LMHOSTS file
   d. DNS reverse lookup
5. Which of the following represent Windows services that you are able to stop on both Publisher and Subscriber servers? (Choose three.)
   a. computer browser
   b. DHCP client
   c. database layer monitor
   d. CTL provider
   e. FTP Publishing Service
6. If you want to set up a Cisco CallManager as a single-server configuration in a lab environment, what button can you click on the Service Activation web page to start the necessary services?
   a. Single Server
   b. Lab
   c. Set Default
   d. Minimal Service

7. You have just completed the installation of a Cisco CallManager server, entered the new passwords for all accounts, and restarted. What is the next step you should take?

a. Configure the Cisco CallManager server settings to send IP address information to the IP Phones rather than hostname.

b. Activate the necessary services through the Service Activation web page.

c. Upgrade the backup software to the latest version.

d. Apply the latest operating system service pack from the Cisco website.

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Part II: IPT Devices and Users

This chapter covers the following topics:

- The basic features of Cisco IP Phones
- The entry-level Cisco IP Phones and their features
- The midrange Cisco IP Phones and their features
- The upper-end Cisco IP Phones and their features
- The features and functions of additional Cisco IP telephony endpoints, including video endpoints, conference stations, expansion modules for Cisco IP Phones, PC-based Cisco IP Phones, and analog adapters
- The six steps of the Cisco IP Phone startup process in the correct order
- The two audio codecs that are supported by Cisco IP Phones

Cisco IP Phones and Other User Devices

Thus far, you have focused on the Cisco Unified CallManager server as the management system of the Cisco IP telephony network. It is now time to focus on the devices that Cisco CallManager controls. An important task of implementing and supporting an IP telephony deployment is managing the end-user devices, or endpoints. You should be able to distinguish among the various Cisco IP telephony end-user devices that you might encounter during the course of deploying and administering a Cisco IP telephony network.

This chapter explains the various models of Cisco IP Phones and how they work within a Cisco IP telephony solution. You will learn the basic features of Cisco IP Phones, analog adapters, and conference stations; the IP Phone power-up and registration process; and the audio coders-decoders (codecs) that are supported by Cisco IP Phones.

Cisco IP Phones

To the user, the telephone is the most visible component of the voice communications network. Cisco IP Phones are next-generation, intelligent communication devices that deliver essential business communications. Fully programmable, the growing family of Cisco IP Phones provides the most frequently used business features.

The majority of Cisco IP Phones provide the following features:

- Display-based user interface
- Straightforward user customization
- Inline Power over Ethernet (PoE)
- Support for the G.711 and G.729 audio codecs

Each Cisco IP Phone provides toll-quality audio. Because it is an IP-based phone, you can install it in any location on a corporate local or wide-area IP network. Some corporations have even made the Cisco CallManager publicly available on the Internet (with appropriate firewall platforms in place), allowing Cisco IP Phones to register from any location that has an Internet connection.

Entry-Level Cisco IP Phones

Cisco has produced a number of low-cost, entry-level IP Phones for a variety of business functions. Depending on user requirements, these IP Phones may function well for employees or for use only in public areas, such as lobbies or break rooms. Figure 4-1 shows the four entry-level Cisco IP Phones available at the time of this writing.

Figure 4-1 *Entry-Level Cisco IP Phones*

Entry-level Cisco phones provide the following common features:

- Display-based user interface (except Cisco IP Phone 7902G)
- G.711 and G.729 codec
- Single line (directory number [DN])
- Cisco inline power, powered patch panel, or local power option support via a power cube (the same power supply as the Cisco IP Phone 7910, 7940, or 7960)
- Visual message waiting indicator (MWI)
- One-way speakerphone (no built-in microphone) and no headset port

The following list briefly describes the major features of each entry-level Cisco IP Phone:

- **Cisco IP Phone 7902G**—The Cisco IP Phone 7902G is a single-line, entry-level, no-display business phone with fixed feature keys that provide one-touch access to the redial, transfer, conference, and voice-mail access features. The following briefly describes the major features of the Cisco IP Phone 7902G:
  - — Fixed features: redial, transfer, conference, messages
  - — Hard Hold key
  - — Single 10-Mbps RJ-45 connection (no internal Ethernet switch)

NOTE Even though it might appear from the figure that the Cisco IP Phone 7902G has an LCD display, Cisco has equipped this low-end IP Phone with a piece of paper for a user to label key buttons and phone numbers.

- **Cisco IP Phone 7905G and Cisco IP Phone 7912G**—The Cisco IP Phone 7905G provides single-line access and four interactive softkeys that guide a user through call features and functions via the pixel-based liquid crystal display (LCD). Use this IP Phone for employees who do not need a full-featured phone or for a common area such as a hallway, manufacturing floor, break room, reception space, or office cubicle. The Cisco IP Phone 7912G includes an integrated Ethernet switch that provides LAN connectivity to a collocated PC (allowing only a single cable drop to each location). The following briefly describes the major features of the Cisco IP Phone 7905G and Cisco 7912G:
  - Pixel-based display (approximately five lines plus softkeys (on-screen feature buttons) and date, time, and menu title
  - Hard Hold key
  - Access to all standard IP Phone features through four physical buttons able to access multiple on-screen softkeys
  - Support for limited Extensible Markup Language (XML) script processing
  - Support for Cisco Skinny Client Control Protocol (SCCP), H.323 version 2 (Cisco 7905G only), and Session Initiation Protocol (SIP; compliant with RFC 2543)
- **Cisco IP Phone 7911G**—The Cisco IP Phone 7911G provides identical features as the Cisco 7912G with the following differences:
  - Support for IEEE 802.3af and Cisco proprietary inline power standards
  - Enhanced memory and XML application support similar to the Cisco IP Phone 7970
  - Support for enhanced security features
- **Cisco IP Phone 7910G+SW**—The Cisco IP Phone 7910G+SW is for common-use areas that require only basic features, such as dialing out, accessing 911, and intercom calls. Locations that might benefit from these limited features include lobbies, break rooms, and hallways. The Cisco IP Phone 7910G+SW includes a two-port switch for use in applications where you require basic IP Phone functionality and a collocated PC. The following briefly describes the major features of the Cisco 7910G+SW:
  - Ability to handle low-to-medium telephone usage
  - Single line with call waiting
  - Display area of 2 x 24 inches (5.08 x 60.96 cm)

— Cisco 10BASE-T/100BASE-T, two-port FastEthernet switch

— Basic features: line, hold, transfer, settings, messages, conference, forward, speed dial, redial

— Adjustable foot stand (flat to 60 degrees)

— Basic and optional wall mounting

> **NOTE** Cisco has positioned the Cisco IP Phone 7912G to replace the Cisco IP Phone 7910G+SW. The Cisco 7910+SW should reach End-of-Life (EOL) in January 2006.

Midrange and Upper-End Cisco IP Phones

Cisco designed the IP Phones 7940G, 7941G and 7941G-GE, 7960, 7961G and 7961G-GE, 7970G and 7971G-GE, and 7985G to meet the demand for a corporate-level, full-featured IP Phone for medium-to-high telephone use. Figure 4-2 shows the midrange and upper-end Cisco IP Phones available at the time of this writing.

Figure 4-2 *Midrange and Upper-End Cisco IP Phones*

A description of features that are common to all midrange and upper-end IP Phones follows:

- Multiline capability
- Large pixel-based displays, which allow for the inclusion of XML and future features
- Integrated two-port FastEthernet or Gigabit Ethernet switch
- Built-in headset connection and quality full-duplex speakerphone (does not come with a headset)
- Information key for "online" help with features
- A minimum of 24 user-adjustable ring tones
- Adjustable foot stand (flat to 60 degrees) for desktop use or appropriate kit included for wall mounting
- SCCP and SIP support
- XML service support
- An EIA/TIA-232 port for options, such as line expansion and security access

NOTE The 79XXG-GE IP Phone models offer features identical to the 79XXG models with the exception of the integrated two-port switch. The 79XXG-GE models have integrated, two-port 10/100/1000 switchports, whereas the 79XXG models have integrated, two-port 10/100 switchports.

Additional details of these phone models follow:

- **Cisco IP Phone 7941G and 7941G-GE**—The Cisco IP Phone 7941G and 7941G-GE is for medium traffic and has these features:
  - Ideal for transaction workers who use cubicle phones
  - Two lines and programmable feature buttons, and four interactive softkeys
  - PoE compatible (Cisco prestandard PoE and IEEE 802.3af industry standard)
  - Increased memory for advanced applications
  - Higher resolution display
  - Backlit screen and line keys
- **Cisco IP Phone 7961G and 7961G-GE**—The Cisco IP Phone 7961G and 7961G-GE is for high or busy telephone traffic and has these features:
  - Ideal for professionals or managers

- — Six lines and programmable feature buttons, and four interactive softkeys
- — PoE compatible (Cisco prestandard PoE and IEEE 802.3af industry standard)
- — Increased memory for advanced applications
- — Higher resolution display
- — Backlit screen and line keys

- **Cisco IP Phone 7970G**—The Cisco IP Phone 7970G is one of Cisco's current high-end devices featuring a high resolution, full-color display that can make a passerby look twice. It has the following features:
  - — Ideal for executives, decision makers, and high-visibility areas of your organization
  - — Eight lines and programmable feature buttons, and five interactive softkeys
  - — Larger, color display with touch-screen capabilities
  - — 3.5-mm stereo jack sockets for connection to PC-style speakers or headphones, and microphone
  - — PoE compatible (Cisco prestandard PoE and IEEE 802.3af industry standard)
  - — Advanced XML development platform for more dynamic applications
- **Cisco IP Phone 7971G-GE**—The Cisco IP Phone 7971G looks identical to the 7970G with the following differences:
  - — Delivers Gigabit Ethernet bandwidth to the desktop with an integrated 10/100/1000 Mbps switch
  - — "Feature identical" to the Cisco IP Phone 7970G
  - — PoE compatible (IEEE 802.3af PoE only; does not support Cisco prestandard PoE)
- **Cisco IP Phone 7980G**—The Cisco 7980G harnesses next-generation video technology offering the following features:
  - — Ideal for executives, decision makers, and high-visibility areas of your organization (such as conference rooms)
  - — Provides a desktop video phone making instant, face-to-face communication possible using the H.264 video codec
  - — Offers an integrated, two-port 10/100 Ethernet switch
  - — Enables digital clarity video