Cisco Systems

IP COMMUNICATIONS

Troubleshooting Cisco IP Telephony

Reveals the methodology you need to resolve complex problems in an IP telephony network

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Addis Hallmark, CCNA, CIPT, is a senior technical marketing engineer with Cisco Systems. He has been installing, configuring, administering, and troubleshooting the Cisco IP Telephony solution since the 2.3 release of CallManager. He has contributed to numerous design guides, application notes, and white papers on a variety of IP Telephony subjects, including CallManager, IP Phones, and IP gateways.

Anne Smith is a technical writer in the CallManager engineering group at Cisco Systems. She has written technical documentation for the Cisco IP Telephony solution since CallManager release 2.0 and was part of the Selsius Systems acquisition in 1998. Anne writes internal and external documents for CallManager, IP phones, and other Cisco IP Telephony products. She is a co-author of *Cisco CallManager Fundamentals* (ISBN: 1-58705-008-0) and *Developing Cisco IP Phone Services* (ISBN: 1-58705-060-9), both from Cisco Press.

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Christina Hattingh is a member of the Technical Marketing organization at Cisco Systems. In this role she works closely with product management and engineering. Christina focuses on helping Cisco sales engineers, partners, and customers design and tune enterprise and service provider Voice over Packet network infrastructures with particular focus on QoS. Prior to this she was a software engineer and engineering manager of PBX Call Center products at Nortel Networks. Her earlier software development experience in X.25 and network management systems provide background for the issues involved today in migrating customers' traditional data and voice networks to packet-based technologies. Christina has a graduate degree in computer science and mathematical statistics.

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Chris Pearce is a technical leader in the Cisco CallManager software group at Cisco Systems, Inc. He has ten years of experience in telecommunications. His primary areas of expertise include call routing, call control, and telephone features. He was a member of the team that developed and implemented the Cisco CallManager software from its early stages, and he was directly involved in developing the system architecture and design.

Ana Rivas, CCIE No. 3877, is an escalation engineer in Cisco's EMEA region. She is one of the technical leaders for AVVID solutions in the Cisco TAC. She is responsible for technically leading the resolution of some of the most critical problems in voice and IP Telephony, spreading technical knowledge to other teams, and working with Cisco business units and the field to head IP Telephony solutions. She has been working as a network engineer for more than five years.

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Dedications

Paul Giralt

I dedicate this book to my parents, Vicia and Pedro, for being the best parents anyone could ask for and always providing the opportunity and encouragement to continue learning.

Addis Hallmark

I want to dedicate this book to my lovely wife, Stephanie. Her companionship is the most precious thing in the world to me. Her patience, understanding, and encouragement helped me write this book. I love and appreciate her dearly.

Anne Smith

For Herb for seeing me through the long nights and weekends without complaint. And, of course, for all those backrubs.

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Thank you to all the contributors to the VNT Voice University website as well as the AVVID TAC tips website on Cisco.com. Also thanks to all the other unnamed authors for the documentation scattered throughout various web pages.

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Addis Hallmark

First, I'd like to thank Paul Hahn and Richard Platt for bringing me on at Cisco. Paul in particular spent a lot of time with me, bringing me up to speed on these technologies, and for that, I am indebted to him.

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The TAC is on the front lines of troubleshooting, and much of the help I received was from the experiences that only solid TAC engineers could provide.

Also, the technical reviewers of this book were so helpful. Thank you so much to everyone for their hard work!

I really believe this is a great book, and one of the biggest reasons for that is Paul Giralt's invaluable contribution and hard work on this project. I couldn't have done this without him!

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Anne Smith

My many thanks go to Paul Giralt and Addis Hallmark for making this book a reality with their knowledge, experience, hard work, and sacrifice. In particular, I thank Paul for a highly enjoyable working experience. Paul's dedication to the quality, accuracy, and comprehensiveness of this book was unsurpassed; he spent countless hours reviewing every page of technical information and his experience with the many components in the Cisco AVVID IP Telephony solution made his extensive contribution invaluable. At every turn, Paul's dedication, commitment to quality, tireless drive for accuracy, and constant positive attitude made working with him a rewarding experience.

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Contents at a Glance

Contents

Foreword

In November of 1998, Cisco Systems acquired a small startup called Selsius Systems. For over a year this small company had been shipping the world's first IP phones and Windows NT-based call management software consisting of close to a million lines of C++ code with a small development staff of about 40 engineers. Since the acquisition, the code base has evolved into many millions of lines of C++, XML, and Java code, and the development staff now has over 500 engineers. The level of sophistication and capability has increased dramatically and is a key component of the Cisco Architecture for Voice, Video, and Integrated Data (AVVID). Current deployments range from extremely distributed enterprises with hundreds of remote offices to small 50-person offices. Geographically, systems are deployed across the world, including exotic locations such as Antarctica and the International Space Station!

AVVID's IP Telephony components (including the IP phones, gateways, and Cisco CallManager) comprise a telephony system that is both richer than and different from traditional TDM-based phone systems. For example, manageability and serviceability are achieved through either a browseable interface or an XML SOAP-based protocol for integration with existing IT systems. Geography disappears as a problem because telephony functions, manageability, and serviceability all traverse the IP network. Proprietary databases disappear in favor of standard SQL databases and LDAP directories. Nevertheless, this unification and standardization of telephony on IP networks also presents unique challenges. Voice quality can be impacted by poor IP network design. Capacity planning requires consideration of IP address numbering. Music on Hold as a multicast stream requires proper switch and router configuration. These are only a few examples of the unique considerations that must be given to IP Telephony deployments.

This book incorporates the authors' real-life experiences in planning and troubleshooting IP Telephony within the AVVID solution. The wisdom contained herein has been gained over the course of thousands of real customer experiences. Paul Giralt and Addis Hallmark are two of the very best troubleshooters in the industry, and Anne Smith has written about and worked with the system since the earliest releases. Paul has been with Cisco's customer support organization for several years. His depth and breadth of knowledge across all Cisco products are legendary, including his most recent focus on IP Telephony. I have seen him in action at some very large and sensitive customer installations, where he resolved extremely difficult problems and provided excellent guidance during upgrades and installations. We were fortunate to get him back, inasmuch as our customers were loathe to let him leave! Addis has been involved in the development and testing of many AVVID products. He has been personally engaged with many key customers during deployment and operation and has received numerous rave reviews from customers. Addis also has been instrumental in the security design aspects of Cisco CallManager. Anne is an author and the technical editor for this and several other AVVID books. She has been engaged with the technology since its inception at Selsius Systems.

I highly recommend this book to any individual or organization involved in installing, operating, or troubleshooting one of the most exciting advances in the long history of telephony. Written by three of its pioneers, this book serves as a guide for the rest of the pioneers who aren't afraid to help their organization communicate in its own way, the better way, the IP way.

Richard B. Platt
Vice President for Enterprise Voice, Video Business Unit
Cisco Systems, Inc.

Introduction

This book teaches you the troubleshooting skills you need to isolate and resolve IP telephony problems. IP telephony is a relatively new technology with many different components. The Cisco IP Telephony (CIPT) solution revolves around Cisco CallManager, the core call processing engine. CIPT includes many different endpoints, such as IP phones, various gateways, and various applications such as Cisco IP IVR, Cisco CallManager Attendant Console, Cisco IP SoftPhone, Cisco Conference Connection, extension mobility, and more. Additionally, the network infrastructure plays an important role in prioritizing voice packets to ensure quality of service (QoS).

With all these components involved in transmitting voice across packet networks, it is essential that you be able to identify and resolve issues in the entire solution. This requires knowledge of the functionality of these components and how they interact with each other, as well as what tools are available to help you find the root cause when problems arise. This book educates you about the techniques, tools, and methodologies involved in troubleshooting an IP telephony system.

Target CallManager Release

This book is written to CallManager release 3.3. Updates to this book may be provided after publication. You should periodically check the ciscopress.com web site for updates (go to ciscopress.com and search for "Troubleshooting Cisco IP Telephony").

Goals and Methods

This book intends to deliver a methodology you can follow when troubleshooting problems in an IP telephony network, particularly a Cisco IP Telephony solution. This book provides detailed troubleshooting information that applies to a variety of problems that can occur in any IP telephony deployment.

"Best Practices" sections in each chapter provide tips and design considerations to help you avoid common configuration problems.

Who Should Read This Book?

This book is designed to teach you how to isolate and correct problems in an IP telephony network. If you are a networking professional responsible for administering a Cisco IP Telephony (CIPT) system, this book is for you. Although this book's main focus is on CIPT, some concepts apply to IP telephony in general as well.

You will best be able to assimilate the information in this book if you already have a working knowledge of a CIPT network.

How This Book Is Organized

Although you could read this book cover-to-cover, it is designed to help you find solutions to specific problems. The chapters are organized by the various components of a Cisco IP Telephony solution. Four appendixes provide reference information.

- **Chapter 1, "Troubleshooting Methodology and Approach"**—You can troubleshoot even the most complex problems if you have a good methodology in place for finding the root cause. This chapter focuses on teaching that methodology: learning how to find clues and track down your "suspect" by breaking the problem into smaller pieces and tackling each piece individually.

- **Chapter 2, "IP Telephony Architecture Overview"**—Cisco AVVID includes many different components that come together to form a comprehensive architecture for voice, video, and integrated data. This chapter covers the basic components of the IP Telephony architecture in order to provide a big-picture view of the system.
- **Chapter 3, "Understanding the Troubleshooting Tools"**—To effectively troubleshoot problems in a Cisco IP Telephony network, you must be familiar with the many tools at your disposal. In addition, you need to know how to best use those tools to achieve maximum results. This chapter describes the various tools and their different uses.
- **Chapter 4, "Skinny Client Registration"**— IP phone registration is a common source of problems. This chapter describes how Skinny protocol-based device registration works, including discussions of inline power, network connectivity, and potential TFTP and CallManager issues.
- **Chapter 5, "IP Phones"**—IP phones can encounter various problems, from unexpected resets to directory and service problems, and more. This chapter explains proper IP phone behavior and examines problems that can occur after an IP phone successfully registers.
- **Chapter 6, "Voice Gateways"**—Voice gateways are the interface that bridges the Voice over IP (VoIP) world with the Public Switched Telephone Network (PSTN). Voice gateways can be Cisco IOS Software gateways or modules within voice-enabled LAN switches. They can be analog or digital, and they can use a wide variety of signaling protocols. This chapter teaches you how to identify and resolve gateway problems by breaking these components into logical groups and following a methodical troubleshooting approach.
- **Chapter 7, "Voice Quality"**—Voice quality is a broad term that covers the following conditions: delayed audio, choppy or garbled audio, static and noise, one-way or no-way audio, and echo. This chapter focuses on the information you need to investigate and resolve voice quality problems in an IP Telephony network.
- **Chapter 8, "Fax Machines and Modems"**—Fax machines and modems present unique challenges when carried over an IP Telephony network, primarily due to their unforgiving nature concerning any modification to the audio stream. This chapter discusses the effect of packet loss and jitter, fax passthrough, fax relay, and how to troubleshoot modems and faxes.
- **Chapter 9, "Call Routing"**—Possessing a strong understanding of call routing is arguably one of the most important aspects of a smooth-operating CIPT solution. This chapter discusses closest-match routing, calling search spaces and partitions, transformations, and translation patterns as well as troubleshooting hold, transfer, park, and call pickup.
- **Chapter 10, "Call Preservation"**—Call preservation is easier to predict when you understand the protocol interaction with CallManager. This chapter provides guidelines for determining call survivability based on endpoint type and protocol.

- **Chapter 11, "Conference Bridges, Transcoders, and Media Termination Points"**—Conference bridges, transcoders, and media termination points are media resources. This chapter discusses the role of media resource groups and media resource group lists, codec selection, and troubleshooting transcoder and conference bridge resources.
- **Chapter 12, "Music on Hold"**—The Music on Hold feature allows callers to hear streaming audio while on hold. This chapter describes this feature and provides steps to take if you encounter problems.
- **Chapter 13, "Call Admission Control"**—Call admission control is used in situations where a limited amount of bandwidth exists between telephony endpoints such as phones and gateways. This chapter discusses the two types of call admission control—locations-based and gatekeeper—and the mechanisms available to reroute calls through the PSTN in the event of WAN congestion.
- **Chapter 14, "Voice Mail"**—CallManager is compatible with a variety of voice mail systems that integrate with CallManager through various methods. This chapter focuses on troubleshooting the integration of CallManager and three types of voice mail systems: Cisco Unity, third-party voice mail systems integrated via Simple Message Desk Interface (SMDI), and Octel Voice Mail, integrated through Cisco DPA Voice Mail gateways.
- **Chapter 15, "Survivable Remote Site Telephony (SRST)"**—SRST allows a router at a remote branch to assume call processing responsibilities in the event that phones at a remote site are unable to contact the central CallManager. This chapter describes SRST and provides detailed information about the various problems that can occur.
- **Chapter 16, "Applications"**—Cisco AVVID allows for the creation of many different applications to interoperate within the converged network. This chapter discusses some of the primary applications in a Cisco AVVID IP Telephony solution, such as IP AA and IP IVR, extension mobility, Cisco IP SoftPhone, Personal Assistant, and Cisco CallManager Attendant Console.
- **Chapter 17, "SQL Database Replication"**—The SQL relational database stores the majority of CallManager configuration information. This chapter discusses the Publisher-Subscriber model for database replication, name resolution, Enterprise Manager, Replication Monitor, broken subscriptions, and CDR database replication.
- **Chapter 18, "LDAP Integration and Replication"**—User information is stored in a Lightweight Directory Access Protocol (LDAP) database. This chapter describes directory integration versus directory access, using the CallManager embedded directory, and integrating with Active Directory and Netscape iPlanet.
- **Appendix A, "Cisco IP Telephony Protocol and Codec Information and Reference"**—Cisco IP Telephony employs many different protocols and codecs. This appendix provides a list of applicable protocols and codecs with descriptions and the standards body corresponding to the protocol or the Request for Comments (RFC) number. Compression rates are given for each codec.

- **Appendix B, "NANP Call Routing Information"**—CallManager provides a built-in dial plan for the North American numbering plan (NANP). This appendix provides information from the NANP file located in the C:\Program Files\Cisco\Dial Plan directory. This file shows you how each part of an NANP number corresponds to a specific placeholder. It is particularly useful when you're learning how to apply route filters.
- **Appendix C, "Decimal to Hexadecimal and Binary Conversion Table"**—This appendix provides a cheat sheet that shows you how to quickly convert between decimal, hexadecimal, and binary values.
- **Appendix D, "Performance Objects and Counters"**—Microsoft Performance (PerfMon) and the Real-Time Monitoring Tool allow you to monitor your system through the use of performance counters. This appendix lists and describes the performance objects and counters in a Cisco IP Telephony network. Some pertinent Windows 2000 counters are also described.
- **Glossary**—The glossary defines terms and acronyms used in this book.

Best Practices

In a perfect world, there would be no need for this book, because systems would always run perfectly. Unfortunately, in the real world, problems do arise, and they usually don't go away on their own. However, an administrator/installer can proactively take steps to ensure reliability and high availability and minimize the number of problems that arise.

Best practices include not only design considerations but also monitoring and management. A properly monitored system can detect failures before they become service-affecting. Each chapter contains a section outlining best practices as they apply to the chapter topic.

In a properly designed network, you can achieve 99.999 percent reliability—a rating that is expected of a telephone system.

High Availability in an IP Telephony Environment

High availability for IP telephony is based on distribution and core layers in the network and servers (call processing, application servers, and so on). BellCore Specification GR-512 defines what criteria must be met to achieve "five 9s" (99.999 percent) reliability. A careful examination of this document is recommended if you are interested in understanding 99.999 percent reliability. Note that many "events" are not counted against five 9s reliability. Some of these events include the following:

- Outages of less than 64 devices
- Outages less than 30 seconds in duration
- Outages due to outside causes, such as power loss from utility or network circuit failures caused by the provider
- Outages due to planned maintenance

The Cisco AVVID IP Telephony solution can achieve 99.999 percent reliability per the BellCore FR-512 specification.

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:

- Vertical bars | separate alternative, mutually exclusive elements.
- Square brackets [ ] indicate an optional element.
- Braces { } indicate a required choice.
- Braces within brackets [{ }] indicate a required choice within an optional element.
- **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 the user inputs (such as a **show** command).
- *Italic* indicates arguments for which you supply actual values.

OSI Reference Model

Throughout the book, a few references are made to the OSI model. Table I-1 provides a brief primer on the OSI reference model layers and the functions of each. You can learn more about the OSI model in any of the Cisco Press books that target the CCNA certification.

Table I-1 *OSI Reference Model Overview*

OSI Layer Name	Functional Description	Examples
Physical (Layer 1)	Responsible for moving bits of data between devices. Also specifies characteristics such as voltage, cable types, and cable pinouts.	EIA/TIA-232, V.35
Data link (Layer 2)	Combines bytes of data into frames. Provides access to the physical media using a Media Access Control (MAC) address, which is typically hard-coded into a network adapter. Also performs error detection and recovery for the data contained in the frame.	802.3/802.2, HDLC
Network (Layer 3)	Uses logical addressing which routers use for path determination. Can fragment and reassemble data if the upper-layer protocol is sending data larger than the data link layer can accept.	IP, IPX
Transport (Layer 4)	Provides reliable or unreliable delivery of data packets. Allows for multiplexing of various conversations using a single network-layer address. Can also ensure data is presented to the upper layers in the same order it was transmitted. Can also provide flow control.	TCP, UDP
Session (Layer 5)	Sets up, coordinates, and terminates network connections between applications. Also deals with session and connection coordination between network endpoints.	Operating systems and application access scheduling

Table I-1 *OSI Reference Model Overview (Continued)*

OSI Layer Name	Functional Description	Examples
Presentation (Layer 6)	Defines how data is presented to the application layer. Can perform special processing, such as encryption, or can perform operations such as ensuring byte-ordering is correct.	JPEG, ASCII
Application (Layer 7)	Interface between network and application software.	Telnet, HTTP

Comments for the Authors

The authors are interested in your comments and suggestions about this book. Please send feedback to the following address:

troubleshootingcipt@external.cisco.com

Further Reading

The authors recommend the following sources for more information.

Cisco Documentation

This book provides comprehensive troubleshooting information and methodology. However, details about common procedures might not be provided. You should be familiar with and regularly use the documentation that is provided with the Cisco IP Telephony system to supplement the information in this book.

You can find Cisco IP Telephony documentation by searching for a specific product on Cisco.com or by starting at the following link:

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

You can examine the following books at a technical bookseller near you or online by entering the title in the search box at www.ciscopress.com.

Cisco CallManager Fundamentals: A Cisco AVVID Solution

You can find detailed information about CallManager's inner workings in the book *Cisco CallManager Fundamentals* (ISBN 1-58705-008-0).

Developing Cisco IP Phone Services: A Cisco AVVID Solution

You can find instructions and tools for creating custom phone services and directories for Cisco IP Phones in the book *Developing Cisco IP Phone Services* (ISBN 1-58705-060-9).

Cisco IP Telephony

You can find installation, configuration, and maintenance information for Cisco IP Telephony networks in the book *Cisco IP Telephony* (ISBN 1-58705-050-1).

Integrating Voice and Data Networks

You can find information on how to integrate and configure packetized voice networks in the book *Integrating Voice and Data Networks* (ISBN 1-57870-196-1).

Cisco Router Configuration, Second Edition

Cisco Router Configuration, Second Edition (ISBN 1-57870-241-0) provides example-oriented Cisco IOS Software configuration for the three most popular networking protocols used today—TCP/IP, AppleTalk, and Novell IPX.

Icons Used in This Book

Throughout this book, you will see a number of icons used to designate Cisco-specific and general networking devices, peripherals, and other items. The following icon legend explains what these icons represent.

Network Device Icons


CallManager


IP Phone


Stations


SRST Router


Used For:
Application Server
DHCP
DNS
MOH Server
MTP
SW Conference Bridge
Voice Mail Server


Router


Switch


Layer 3 Switch


PIX Firewall


Gateway or
3rd-party
H.323
Server


Modem


Access Server


ATM Switch


Used For:
Analog Gateway
Gatekeeper
Gateway
H.323 Gateway
Voice-enabled Router


PBX/PSTN Switch


PBX (Small)


Cisco
Directory
Server


Local Director


DAT Tape


PC


Laptop

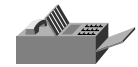

Server


PC w/Software


Used For:
HW Conference Bridge
Transcoder
Voice-enabled Switch


POTS Phone


Relational
Database


Fax Machine

Media/Building Icons


Network Cloud

Ethernet Connection

Serial Connection


Telecommuter


Building


Branch
Office

CHAPTER 3

Understanding the Troubleshooting Tools

To effectively troubleshoot problems in a Cisco IP Telephony (CIPT) network, you must be familiar with the many tools at your disposal. In addition, you need to know how best to use those tools to achieve maximum results. This chapter describes the various tools and their different uses.

Depending on the problem you encounter and your particular skill set, you might find certain tools more helpful than others. Nevertheless, knowing what tools are available and how they can help you solve the problem is essential to a successful resolution. Also, some tools might be more useful to you based on your past experience. If you are strong in IP, a sniffer trace might be your preferred tool when applicable. However, if you are stronger in call processing-related traces, the Cisco CallManager (CCM, also sometimes called SDI) traces might prove helpful once you understand how they work.

Later chapters demonstrate the use of these tools in different scenarios you might encounter.

This chapter covers the following topics:

- **Time synchronization**—Explains how to synchronize the clocks on all devices in an IP telephony network to ensure that timestamps on your trace files and debugs are synchronized with each other.
- **Reading CCM (or SDI) traces**—Describes one of the most important trace files you use to troubleshoot CallManager-related problems. CCM trace files provide information about call processing events and all messages exchanged between Skinny, MGCP, and H.323 endpoints.
- **Reading SDL traces**—Discusses the components of the less-used Signal Distribution Layer (SDL) trace files in CallManager. SDL traces describe the events occurring in the CallManager software at a code level. These traces are usually reserved for Cisco development engineering use; however, there are a few key pieces of information you can use to your advantage when troubleshooting CallManager problems.
- **Microsoft Performance (PerfMon)**—Details the capabilities of PerfMon, a built-in Windows 2000 utility that helps you troubleshoot CallManager.
- **CCEmail**—Details the third-party alerting tool that can be used in conjunction with PerfMon to configure alerts for the performance counters.

- **CallManager Serviceability**—Discusses various web-enabled tools provided with CallManager for reading alarms and XML-based tracing.
- **Real Time Monitoring Tool (RTMT)**—Describes the Cisco web-based monitoring application that allows you to view CallManager cluster details, monitor performance objects (much like PerfMon), and monitor devices and CTI applications.
- **Call detail records (CDR) and the CDR Analysis and Reporting (CAR) Tool**—Describes the CAR tool that helps you analyze the raw data that comprises the CDR database and create reports based on your search criteria.
- **CDR Time Converter**—Describes how to use this small utility that allows you to convert the UNIX Epoch-based date and time format stored in a CDR to standard date and time format.
- **Event Viewer**—Briefly explains the function of another built-in Windows 2000 tool that plays a key role in troubleshooting CallManager.
- **Q.931 Translator and Enhanced Q.931 Translator**—Describes two tools that read a CCM trace file and analyze all Q.931 and H.225 messages. The various information elements are decoded for ease of reading. The Enhanced Q.931 Translator also adds additional search and filtering options and decodes more information elements than the original Q.931 Translator.
- **Dick Tracy**—Describes an important tool used to troubleshoot the WS-X6608 and WS-X6624 voice gateways for the Catalyst 6000 series switches.
- **Sniffer traces**—Discusses when and why to use a network packet-capture tool.
- **Voice Codec Bandwidth Calculator**—Describes how to use the Voice Codec Bandwidth Calculator to determine the bandwidth used by different codecs with various voice protocols over different media.
- **Cisco Bug Toolkit (formerly Bug Navigator)**—Describes the web-based tool that allows you to find known bugs based on software version, feature set, and keywords. The resulting matrix shows when each bug was integrated or fixed if applicable.
- **Remote Access Tools**—Describes applications like Terminal Services and Virtual Network Computing, which allow you to access a server from a remote location.
- **Websites and Further Reading**—Provides URLs for websites that contain additional troubleshooting information. Also points you to a section in the "Introduction" with a recommended reading list.

Time Synchronization

Time synchronization is simply making sure that all the participating CallManager servers and network devices have the same exact time. Time synchronization is critical. A large CallManager cluster can have eight or more separate servers, not including any voice mail servers or application servers. This distributed architecture creates a highly available and

scalable system. It also makes the troubleshooting process more involved because you have to collect traces from all participating servers to see the full picture of what happened.

As endpoints such as IP phones and voice gateways call each other, signaling occurs between CallManager and the endpoint device. Signaling also occurs between the respective CallManagers of each endpoint device. If a problem occurs, you need to consolidate trace files from all involved servers. CallManager Serviceability, discussed later in this chapter, can collect this information into one file. However, if the timestamps of each file are mismatched, it can be impossible to tell what the real series of call processing events is.

All the CallManager servers can be time-synched using the Network Time Protocol (NTP). When you synchronize the time on all involved servers, all the trace files are timestamped the same. When trace files are collected for analysis, you can follow the true series of call processing events with accuracy.

In addition to CallManager servers, you should ensure all network devices such as switches, routers, and voice gateways are synchronized to the same time source as the CallManager servers. This ensures consistent timestamps regardless of which device you are looking at.

Configuring Automatic Time Synchronization on CallManager Servers

Use the following steps to configure the CallManager server to automatically synchronize—and stay synchronized—with a Time Server.

Step 1 Verify that the NetworkTimeProtocol service is configured to launch automatically upon startup. Right-click **My Computer** and select **Manage**.

Step 2 Expand the **Services and Applications** section, and select **Services**.

Step 3 Double-click the **NetworkTimeProtocol** service, and ensure that Startup Type is set to **Automatic**.

Step 4 Configure the **C:\WINNT\ntp.conf** file. This file contains the list of time servers that CallManager will synchronize with. You can configure CallManager to point to specific time servers (see Example 3-1), or you can configure it to receive NTP broadcasts (see Example 3-2) on the local LAN segment from the router (as long as the router is configured to do so).

Example 3-1 *Sample ntp.conf File Using Static Time Servers*

```
server 10.0.0.10
server 10.1.0.10
driftfile %windir%\ntp.drift
```

Example 3-2 *Sample ntp.conf File Using an NTP Broadcast Router*

```
broadcastclient
driftfile %windir%\ntp.drift
```

Step 5 Go to the Services Control Panel and stop/start the NetworkTimeProtocol service. Allow several minutes for the update to take place.

Synchronizing Time Manually on CallManager Servers

Use the following steps to manually configure time synchronization.

Step 1 Stop the NetworkTimeProtocol service in the Services Control Panel.

Step 2 Synchronize the clock by using one of the following commands from a command prompt.

To synchronize with a remote Time Server, use the following command where *x.x.x.x* is the IP address of the Time Server:

```
ntpdate x.x.x.x
```

To synchronize with a Broadcast Router, use the command where *x.x.x.x* is the IP address of the Ethernet port of the router:

```
ntpdate x.x.x.x
```

Step 3 Restart the NetworkTimeProtocol service in the Services Control Panel.

Synchronizing Time on Cisco IOS Devices

In addition to your CallManager servers, you should also ensure the time is synchronized on all your network devices, including IOS voice gateways, switches, and routers.

For Cisco IOS devices, you can use the Network Time Protocol (NTP) to synchronize the time. To enable NTP, you must configure the time zone the IOS device is in along with the IP address of the NTP server. You should configure the IOS device to take daylight savings time into account as well if you live in a time zone that observes daylight savings time.

First you must configure the time zone. Use the command **clock timezone** *name_of_time_zone offset_ from_GMT* to configure the time zone. For example, the following command configures the IOS device for Eastern Standard Time (EST):

```
clock timezone EST -5
```

If you are in a time zone that observes daylight savings time, use the command **clock summer-time** *name_of_time_zone* **recurring** to enable automatic daylight savings time. For example, the following configures Eastern Daylight Savings Time:

```
clock summer-time EDT recurring
```

You can determine time zones by performing a search at Google.com; search for a string such as "time zone GMT." Any one of the many hits should point you to a table showing the various time zones around the world, including daylight savings time.

Once your time zone is configured properly, you can enable NTP. If you do not have a device on the network running NTP, you can make an IOS device into an NTP master clock. You can do this by configuring the command **ntp master** on the IOS device. If you are not making the device an NTP master, you should configure the IP address of the NTP server using the command **ntp server** *NTP_server_IP_address*. For example, the following tells the IOS device to synchronize its clock with the NTP server at IP address 172.18.109.1:

```
ntp server 172.18.109.1
```

Once you have enabled NTP, all IOS devices should synchronize their clocks with the central NTP server. You should also synchronize the clocks on non-IOS network devices such as switches running CatOS software.

Synchronizing Time on CatOS Devices

You can use NTP to synchronize the clocks on switches running CatOS software. The configuration is similar to configuring NTP on a device running Cisco IOS Software.

First you must configure the time zone the switch is in using the command **set timezone** *name_of_time_zone offset_ from_GMT*. For example, the following sets the clock to Eastern Standard Time:

```
set timezone EST -5
```

You can also configure daylight savings time using the command **set summertime enable** *name_of_time_zone*. For example, the following enables Eastern Daylight Savings Time:

```
set summertime enable EDT
```

You can determine time zones by performing a search at Google.com; search for a string such as "time zone GMT." Any one of the many hits should point you to a table showing the various time zones around the world, including daylight savings time.

Once you have the time zone configured properly, you can set the NTP server IP address and enable the NTP client on the switch. First set the NTP server IP address using the command **set ntp server** *NTP_server_IP_address*. Then enable the NTP client using the command **set ntp client enable**. The following shows the configuration to use the NTP server with IP address 172.18.109.1:

```
set ntp server 172.18.109.1
set ntp client enable
```

Once you have enabled proper time synchronization in your network, you can move on to reading the variety of trace files available to you for troubleshooting problems in an IP Telephony network.

Reading CCM (or SDI) Traces

CCM (also known as System Diagnostic Interface or SDI) traces are the user-friendliest call processing trace files you have available to you. After you learn a few tricks, it is easy to follow the call flows and find the potential problem. Although you might see "SDI" and "CCM" used interchangeably to refer to this type of trace, in this book we primarily refer to this type of trace as a CCM trace. Some pages in CallManager Serviceability refer to CCM traces as SDI traces.

Because CallManager is at the heart of a CIPT network, CCM traces are usually the first place to look when troubleshooting most problems. You can analyze problems related to device registration, call flow, digit analysis, and related devices such as IP phones, gateways, gatekeepers, and more. By the end of this section, you should be able to follow some basic call flows in the CCM trace. Future chapters continue to show CCM trace examples as you learn how to troubleshoot more problems. Later in this chapter you also learn about the Q.931 Translator, which is useful for quickly troubleshooting a variety of gateway problems. It is not a substitute for learning how to read the CCM trace, but it helps you quickly examine a trace in a graphical format without having to wade through irrelevant debugs. See the later "Q.931 Translator and Enhanced Q.931 Translator" section for more information.

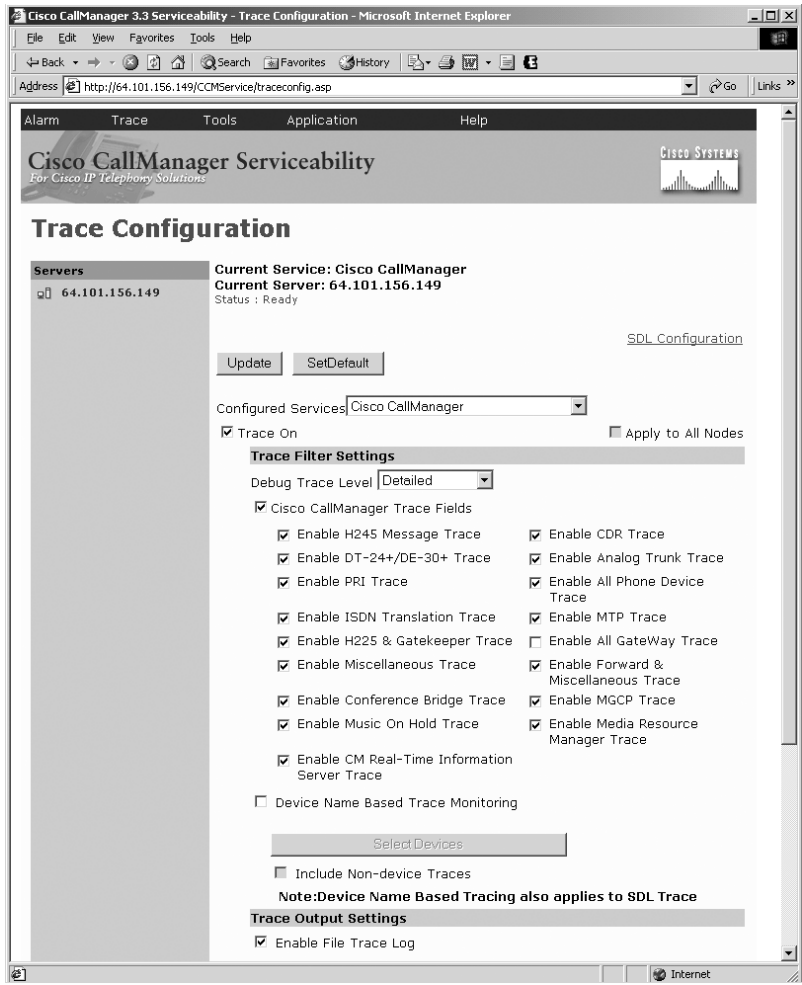
Setting the Appropriate Trace Level and Flags

CallManager allows you to select from a variety of different options that adjust which events are logged to the CCM trace files. If you know exactly what you are looking for, you can configure CallManager to log only specific events in the CCM trace file. If you configure the trace to collect too much information, the trace becomes more difficult to analyze. However, if you do not configure the trace to collect enough information, you might miss the problem you are trying to find. When you are beginning to learn how to read CCM trace files, it is best to enable more debugs than less. As you learn which trace settings are required for specific problems, you can enable just the settings you need.

Unfortunately you can never predict what problems will come up, so if you do not have a high level of tracing enabled, you may have to wait for a problem to happen a second time after enabling the appropriate trace settings before you can troubleshoot. For this reason, it is usually best to leave a majority of the trace flags enabled during normal operating conditions so you have trace data available if a problem occurs.

You configure a CCM trace in Cisco CallManager Serviceability (**Trace > Configuration**). Figure 3-1 shows the top half of the Trace Configuration page, and this section discusses the relevant fields and settings on that page.

Figure 3-1 *Trace Configuration (Part 1) in CallManager Serviceability*



First thing to note is the **Trace On** checkbox. This must be selected for any of the trace settings to be available.

The **Apply to All Nodes** checkbox allows you to apply the specified trace settings to all CallManager servers in the cluster. This is useful when you are troubleshooting a problem that is occurring on more than one CallManager server. You generally want to keep the same level of tracing enabled on all the servers in the cluster unless you are absolutely certain the problem is isolated to a single server in the cluster. Because of the distributed nature of CallManager, you may think a process only involves a single server when in reality part of the processing for a particular call is occuring on another server in the cluster.

The Trace Filter Settings area allows you to specify the exact parameters of your trace. First, you set the level of tracing you want to perform in the **Debug Trace Level** field.

CallManager Serviceability provides six different levels, but for nearly every kind of problem that you'll want traces for, the recommended levels are either Detailed or Arbitrary:

- **Detailed**—Provides detailed debug information and highly repetitive messages that are primarily used for debugging, including KeepAlives and responses. For CallManager releases 3.1 and earlier, be cautious about using this level during normal production hours on a heavily loaded system. It can cause performance degradation on CallManager. In release 3.2 and later, CallManager uses asynchronous tracing to reduce the impact that trace file generation has on call processing.
- **Arbitrary**—Provides low-level debug traces. This level is best suited for debugging difficult problems. This level includes nearly everything that is included in Detailed with the exception of KeepAlives. If you are not troubleshooting problems related to missed KeepAlives, this level is good for day-to-day troubleshooting.

Other trace levels are provided, but generally, they don't offer as much information as the Detailed and Arbitrary levels. The other levels are

- **Error**—Provides traces generated in abnormal conditions, such as coding errors or other errors that normally should not occur.
- **Special**—Provides traces for all informational, non-repetitive messages such as process startup messages, registration messages, and so on. All system and device initialization traces are at this level.
- **State Transition**—Provides traces for call processing events or normal events traced for the subsystem (signaling layers).
- **Significant**—Provides traces for media layer events.

Second, make sure the **Enable CallManager Trace Fields** checkbox is selected, which gives you the opportunity to select which specific traces you want to run and to choose your traces. Table 3-1 describes the trace fields to choose from. Most are reasonably self-explanatory. For example, if you're having problems with an H.323 device, you would enable the **H245 Message Trace** and **Enable H225 & Gatekeeper Trace** options. Likewise, for music on hold (MOH) issues, the **Enable Music on Hold** option would display trace relating to all MOH activity.

Table 3-1 *Cisco CallManager Trace Fields*

Field Name	Description
Enable H245 Message Trace	Debugs H.245 signaling for H.323 calls, including the media processing messages.
Enable DT-24+/DE-30+ Trace	Activates the logging of events related to the legacy gateways, Cisco Digital Access DT-24+/DE-30+.
Enable PRI Trace	Activates a trace of Primary Rate Interface (PRI) devices.
Enable ISDN Translation Trace	Activates a Layer 3 trace of Q.931 (ISDN messages).

Table 3-1 *Cisco CallManager Trace Fields (Continued)*

Field Name	Description
Enable H225 & Gatekeeper Trace	Activates a trace showing H.225 signaling messaging for H.323 calls.
Enable Miscellaneous Trace	Activates a trace of miscellaneous devices.
Enable Conference Bridge Trace	Activates a trace of the conference bridges. Use this level to trace conference bridge statuses such as • Registered with CallManager • Unregistered with CallManager • Resource allocation processed successfully • Resource allocation failed
Enable Music on Hold Trace	Activates a trace of MOH devices. Use this level to trace MOH device statuses such as • Registered with CallManager • Unregistered with CallManager • Resource allocation processed successfully • Resource allocation failed
Enable CM Real-Time Information Server Trace	Activates CallManager real-time information traces used by the real-time information server.
Enable CDR Trace	Enables tracing of call detail record (CDR) processing. However, this trace flag does not provide much information about CDRs because CDR processing is mostly handled by the Database Layer Monitor and CDR Insert services discussed in Chapter 17, "SQL Database Replication."
Enable Analog Trunk Trace	Activates a trace of all MGCP-based devices using an analog interface.
Enable All Phone Device Trace	Activates a trace of phone devices, including Cisco IP SoftPhones, and shows events such as on-hook, off-hook, key presses, and so on.
Enable MTP Trace	Activates a trace of media termination point devices and transcoders. Use this level to trace MTP device statuses such as • Registered with CallManager • Unregistered with CallManager • Resource allocation processed successfully • Resource allocation failed
Enable All Gateway Trace	Activates a trace of all analog and digital gateways.

continues

Table 3-1 *Cisco CallManager Trace Fields (Continued)*

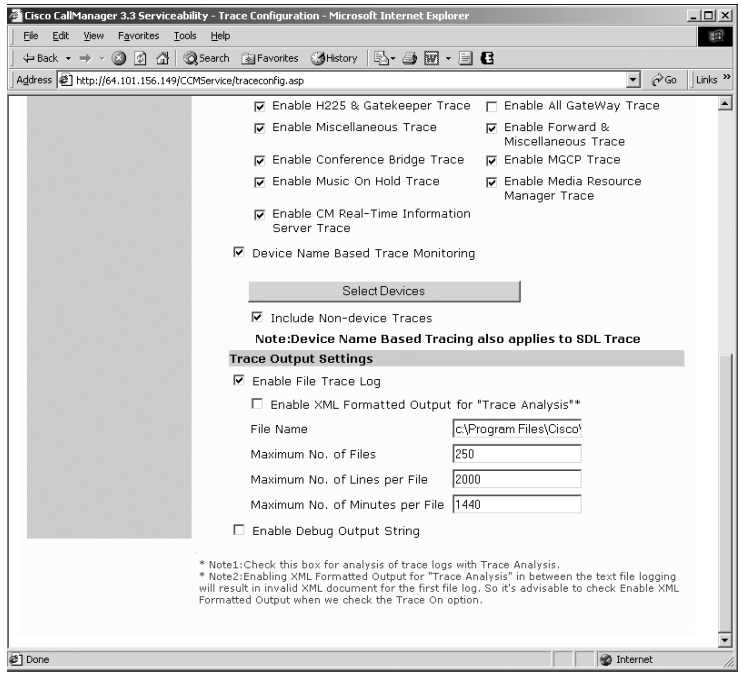
Field Name	Description
Enable Forward and Miscellaneous Trace	Activates a trace of call forwarding and all subsystems not covered by another checkbox.
Enable MGCP Trace	Activates a trace showing media gateway control protocol (MGCP) messages for MGCP-based devices.
Enable Media Resource Manager Trace	Activates a trace for media resource manager (MRM) activities.

Next, you can trace based on a specific device name by selecting the **Device Name Based Trace Monitoring** checkbox. When that checkbox is selected, you can click the **Select Devices** button to choose from a list of devices to trace. Tracing based on specific devices is very useful when you know which devices are involved in the problem, such as specific phones or gateways, and want to see trace output only for those devices.

The non-device option is a catch-all; if you're having issues that are not device-related, you can select the **Include Non-device Traces** checkbox to see traces not related to devices.

Figure 3-2 shows the bottom half of the Trace Configuration page.

Figure 3-2 *Trace Configuration (Part 2) in CallManager Serviceability*



For traces to be logged to a file, the **Enable File Trace Log** checkbox must be selected. The trace output is then sent to the path specified in the **File Name** field, which is C:\Program Files\Cisco\Trace\CCM\ccm.txt by default.

We're going to skip discussion of the **Enable XML Formatted Output for "Trace Analysis"** checkbox for a moment to finish the fields related to file settings.

The Trace Configuration page in CallManager Serviceability validates the filename and ensures that it has a .txt extension. Do not use a filename that exists on another computer. Use a filename that exists on the computer running the trace. It is a good practice to have each server use a filename format that has the server name in it. This could be **servera-ccm.txt**, for example. That way, if trace files are collected from several servers, each file is easily distinguished from those collected on a different server. One downside to this, though, is that you can't use the **Apply to all nodes** option. If you did, you would have to go back and change the names of the files on all nodes.

The **Maximum No. of Files** field defaults to 250. Normally, a maximum of 250 is never enough files because the files write in round-robin fashion—when the maximum limit is met, the next file starts at the beginning and overwrites the first file. Frequently, this means that the trace information you need to troubleshoot a problem that began occurring yesterday or the day before has already been overwritten by new files. For each server on which you're running trace, determine how much free disk space you have, deduct a safety net of 500 MB, and then use the rest of your disk space for trace space. Assuming you don't go above 10,000 lines—which we don't recommend—you can calculate the size of your trace files by figuring the average CCM trace at 10,000 lines consumes about 2.5 MB; the average SDL trace at 10,000 lines consumes about 3.5 MB. We don't recommend going above 10,000 lines because you want to keep your trace files at a manageable size. At these average sizes, you can easily zip the file and e-mail it if needed.

The default of 1440 for the **Maximum No. of Minutes per File** is adequate; you'll probably never reach it.

The **Enable XML Formatted Output for "Trace Analysis"** checkbox takes the trace output and formats it in XML, which is required if you want use the Trace Analysis feature in CallManager Serviceability. With Trace Analysis, you can view the trace files in a web page, and the XML tagging lets you filter the trace results. The downside, however, is that the number of lines per file is limited to 2000 instead of 10,000. In most cases, you'll probably want to stick with standard text-based tracing because the 2000 line limit severely restricts the amount of information in the trace file. Also, reading XML-formatted traces manually can be far more time-consuming than non-XML-formatted traces because you need to weed out all the extra XML tags that get added to each trace line. If you do not enable XML-formatted output, the log file compiles in text format.

The **Enable Debug Output String** checkbox sends debugging information to a Microsoft development tool useful only to Cisco development engineers. You should never need to or be asked to enable this checkbox.

Click **Update** to save your settings. The new trace settings take effect immediately when you click **Update**.

Once you've specified the trace settings, you can go to the Trace Collection page (**Trace > Collection**) to collect traces from one or more servers after an event has occurred. Set the service you want to retrieve the trace files for (such as Cisco CallManager), along with the date and time period you want to trace and click the **Submit Form** button. Depending on the time period you specified, tracing can impact performance, so be sure to trace short intervals that won't impact CallManager or during non-production hours.

It is usually quicker and easier to manually collect the traces yourself using either Terminal Services or VNC (discussed later in this chapter) to access the CallManager server and copy the files to another machine for analysis.

Once the trace has been gathered, you are given the option to view it in a new window or use Save As to save the output to a file. If you choose to view the text-based output in a new window, it looks similar to Figure 3-3.

Figure 3-3 *Viewing Text-based Trace Output in Web Browser*

If you choose the Save As option, the text-based output displays in the CallManager Serviceability window in the same format as Figure 3-3, and you can click **File > Save As** to save the file.

We generally recommend that you do not use the trace collection utility for anything other than very small traces on a system that is not busy. Manually copy the traces off the server(s) in question and analyze them offline.

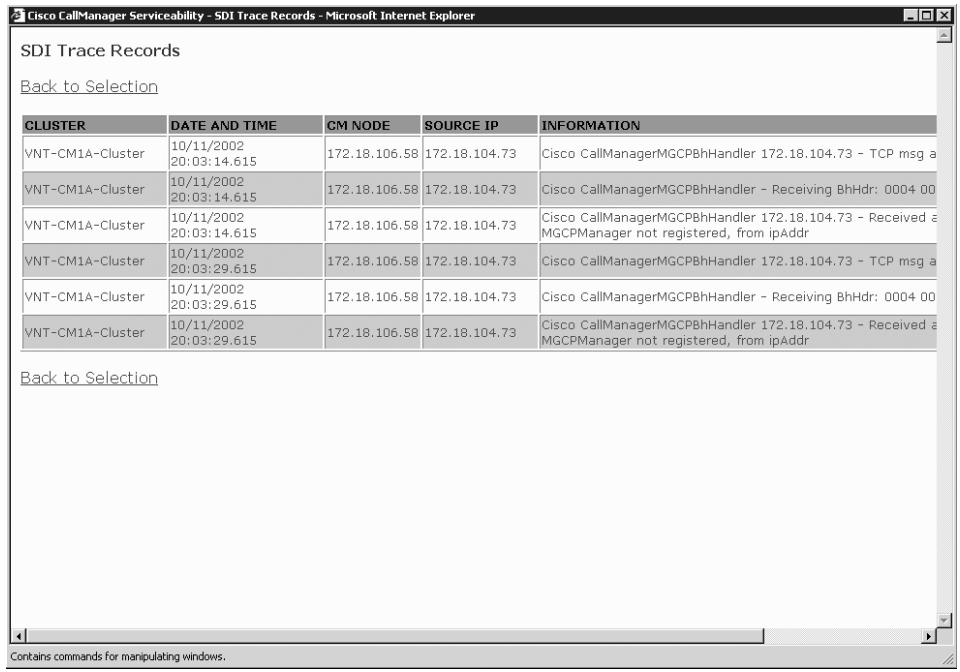
If you selected the **Collect XML Trace File(s)** checkbox and choose to view the output in a new window, the Trace Analysis dialog appears. In this dialog box, you can filter the trace results based on the options described in Table 3-2.

Table 3-2 *Trace Analysis Filtering*

Field Name	Description
CallManager Host	Choose ALL CallManagers or select just one CallManager.
Device Name	Choose ALL or specify the device's name. If you specify a device name, only trace information pertaining to that device name appears in the search results.
IP Address	Choose ALL or a specific server's IP address.
Trace Type	Choose ALL, Alarm, or Trace. Trace shows all events as specified in the trace settings. Alarm shows only specific messages that meet the criteria of being an information Alarm message.
Cluster	Select this checkbox if you want to include the cluster name in the trace output.
Date and Time	Select this checkbox if you want to include the date and time of each event listed in the trace output.
CM Node	Select this checkbox if you want to include the CallManager node (IP address or host name) in the trace output.
Trace Type	Select this checkbox if you want to include the trace type in the trace output.
IP Address	Select this checkbox if you want to include the device's source IP address in the trace output.
Correlation Tag	Select this checkbox if you want to include the number that correlates traces with each other in the trace output.
Application Name	Select this checkbox if you want to include the directory numbers (DNs) and other service-specific information in the trace output.
Information	Select this checkbox if you want to include a description of what the trace found in the trace output.
Device Name	Select this checkbox if you want to include the device name in the trace output.

Figure 3-4 shows a trace file formatted as XML output and filtered to display only one CallManager host, one IP address, and the fields Cluster, Date and Time, CM Node, Source IP, and Information.

Figure 3-4 *Filtered Trace Output in XML Format*



You can click the **Back to Selection** link to return to the Trace Analysis dialog and filter based on different criteria.

As you can see, reading through a large amount of trace files using the trace collection utility can be very cumbersome and time-consuming in relation to reading the trace files manually. You should learn how to read the CCM traces directly from the text files to help you troubleshoot problems more quickly and accurately. For that reason, all the examples of CCM traces that we show in this book are in plain text files. The following section describes how to read a text-based CCM trace.

Reading CCM Traces

This section shows you a few call flow examples and highlights key information in each example that helps you understand the CCM trace.

For brevity, the header and tail of the trace line have been omitted in many examples. For example, the complete trace would look like the following:

```
03/15/2001 05:34:41.956 CCM |StationInit: 1a3e8b54 OffHook.|
  <CLID::DLS2-CM152-SRV4-Cluster><NID::DLS2-CM152-SRV4><CT::1,100,96,1.69>
  <IP::172.28.238.62><DEV::SEP003094C2D11F>
```

But for many examples, the header and tail do not add value. So the same trace is shown in this book as

```
StationInit: 1a3e8b54 OffHook.
```

The header portion of the trace line just specifies the date and time when the trace event was generated and which trace file you are looking at. For a CCM trace file, every line starts with the date and time followed by the letters "CCM." Prior to CallManager 3.3 the trace files begin with the date and time followed by the words "Cisco CallManager."

You should understand a few things about CCM traces:

- Many places in the trace files use hexadecimal equivalents for the IP addresses—The IP address 172.28.232.164 is shown in trace files as a4e81cac, which is a hexadecimal representation of 172.28.232.164. You can determine the IP address by working backwards: take the last two digits, **ac**, which is hex for 172; then **1c**, which is hex for 28; then **e8**, which is hex for 232; and finally, **a4**, which is hex for 164. The IP address is 172.28.232.164. Appendix C, "Decimal to Hexadecimal and Binary Conversion Table," provides a quick cheat sheet to determine how to quickly convert between decimal, hexadecimal, and binary values.
- Trace files sometimes use ASCII for directory numbers—Consider the value 33 30 30 31, which is how the directory number 3001 is sometimes displayed in the trace file.
- Trace files may include messages for a variety of protocols—Details on each of these protocols is described in the appropriate chapters. For example, the Skinny protocol is discussed in Chapter 4, "Skinny Client Registration," and Chapter 5, "IP Phones." Other protocols such as H.323 and MGCP are discussed in Chapter 6, "Voice Gateways." Appendix A, "Cisco IP Telephony Protocol and Codec Information and References," lists the protocols in an IP Telephony environment and the standards body or specification governing the protocol.

When you first open a CCM trace file, you might feel intimidated by the large amount of information presented in the trace file. We recommend that you use the default trace settings for CCM traces except you set the trace level to either **Arbitrary** or **Detailed**. Click the **SetDefault** button on the Trace Configuration page for CCM traces in CallManager Serviceability (**Trace > Configuration**) and then change the **Debug Trace Level** setting to **Detailed**.

A Sample CCM Trace for a Call Between Two IP Phones

As a first example at looking at CCM traces, go through a simple call between two IP phones. IP phones use the Skinny protocol to communicate with CallManager. All messages to and from a Skinny device are preceded by either the words **StationInit** or **StationD**.

For example, assume you have two phones, Phone A and Phone B. Phone A calls Phone B, Phone A goes off-hook and you see the following line in the CCM trace:

```
StationInit: 1a3e8b54 OffHook.
```

StationInit means that an inbound Transmission Control Protocol (TCP) message from a Skinny station reached CallManager. A Skinny station is any endpoint that uses the Skinny protocol to communicate with CallManager. This includes the Cisco 79*xx* family of

IP phones. In other words, any message that starts with StationInit is a message *from* an IP phone.

1a3e8b54 is probably the most important piece of this trace example. It is called a *TCP handle* and it represents a unique value that identifies a specific IP phone registered to this CallManager server. With the TCP handle, you can follow every message to and from that IP phone and see the full series of messages exchanged between CallManager and the phone. When searching through a CCM trace file in Notepad, copy the TCP handle to the clipboard (**Ctrl+C**), then open the Find box in Notepad (**Ctrl+F**), and paste the TCP handle (**Ctrl+V**). Once you get into the habit of highlighting the TCP handle and then pressing Ctrl+C, Ctrl+F, and Ctrl+V to enter the TCP handle into the Find window, you will be able to search for Skinny messages related to a device very quickly.

If you want to find the TCP handle for a particular IP phone, obtain the MAC address of the phone from either CallManager Administration or from the phone itself and search for the MAC address in the CCM trace until you find a KeepAlive to that phone. For example,

```
StationInit - InboundStim - KeepAliveMessage –
  Send KeepAlive to Device Controller. DeviceName=SEP003094C2D11F,
  TCPHandle=1a3e8b54, IPAddr=10.80.1.147, Port=51763,
  Device Controller=[2,89,2992]
```

Notice the KeepAlive message contains both the TCP handle and the device name for the IP phone. Once you find the KeepAlive message, copy the TCPHandle field and use that to search through the CCM trace.

The **OffHook** message means that CallManager received a Skinny message indicating the phone went off-hook.

The next message is

```
StationD:    1a3e8b54 DisplayText text='                    3000        '
```

Notice that instead of **StationInit** you see **StationD**. This signifies that CallManager is sending a Skinny message to the phone. StationInit messages are sent from the IP phone to CallManager, while StationD messages are sent from CallManager to the IP phone. Skinny message transmission such as this between the IP phone and CallManager occurs for every action undertaken by the IP phone, including initialization, registration, on-hook, off-hook, dialing of digits, key presses on the phone, and so much more.

Again you see the same TCP handle, 1a3e8b54, listed for this message.

The number **3000** represents the directory number of the phone. If you know the phone number of the calling IP phone, you can often find the beginning of a call by simply searching for the calling phone's directory number. In CallManager 3.3 and later the DisplayText message actually shows

```
StationOutputDisplayText don't need to send, because mIsALegacyDevice = 0
```

The reason for this message is that the 79*xx* series Cisco IP Phones never paid attention to the DisplayText message in the first place, so in CallManager 3.3 and beyond, the message

is no longer sent. This means that if you search for the directory number of the IP phone, you might not find exactly the beginning of the sequence of events. It is best to search for the device name you are looking for and find a KeepAlive to get the TCP handle as discussed earlier.

Other messages sent to the IP phone include the following:

```
StationD:    1a3e8b54 StartTone tone=33(InsideDialTone), direction=0
StationD:    1a3e8b54 SetLamp stimulus=9(Line) stimulusInstance=1
  lampMode=2(LampOn).
StationD:    1a3e8b54 CallState callState=1 lineInstance=1 callReference=16777217
StationD:    1a3e8b54 DisplayPromptStatus timeOutValue=0
  promptStatus='Enter number' lineInstance=1 callReference=16777217.
StationD:    1a3e8b54 SelectSoftKeys instance=1 reference=16777217
  softKeySetIndex=4 validKeyMask=-1.
StationD:    1a3e8b54 ActivateCallPlane lineInstance=1.
```

Again you see that all the trace lines begin with **StationD** indicating that these are messages from CallManager to the IP phone and you see each line has the same TCP handle.

Do not concern yourself at this point about exactly what each of the pieces in the trace mean. These are all Skinny messages sent to the IP phone. At this point, you should just familiarize yourself with the basic call flow to understand how to read the trace files, not necessarily what each piece of the trace file means. Chapters 4 and 5 provide additional detail relating to the Skinny messaging you see in the preceding output. In particular, see Table 5-1, "Skinny Message Definitions," and the section, "Examining Skinny Protocol Messages in a CCM Trace" in Chapter 5 for detailed explanations.

You will notice, however, that many of the trace messages are relatively self-explanatory. For example, the **StartTone** message with **tone=33(InsideDialTone)** tells the IP phone to start playing dial tone.

Note the **callReference** ID. A callReference ID is created for each participant in a call and you can use this ID to track a particular call through a CCM trace. A new callReference ID is created for each participant in a call and when some features are invoked, such as transfer and conference. Each leg of a call gets its own callReference ID assigned, so in a call between two IP phones, each phone gets assigned a separate callReferenceID.

So far you have only seen Skinny protocol messages; however, this callReference ID can help you correlate the Skinny messages with other messages to devices involved in the same call.

Next, the user on the IP phone begins dialing digits. This time, notice the difference between **StationD** and **StationInit** messages, indicating communication back and forth between CallManager and the IP phone.

```
StationInit: 1a3e8b54 KeypadButton kpButton=3
StationD:    1a3e8b54 StopTone
StationD:    1a3e8b54 SelectSoftKeys instance=1 reference=16777217
  softKeySetIndex=6 validKeyMask=-1
StationInit: 1a3e8b54 KeypadButton kpButton=0
StationInit: 1a3e8b54 KeypadButton kpButton=0
StationInit: 1a3e8b54 KeypadButton kpButton=1
Digit analysis: match(fqcn="3000", cn="3000", pss="IPMA:PA:Line1", dd="3001")
```

Notice that a 3 is dialed and a tone is then stopped. The Skinny protocol does not provide a mechanism to specify which tone to stop, so it sends a generic **StopTone** message. This stops any tones the IP phone happened to be playing at the time. In this case, remember you saw CallManager instruct the IP phone to play inside dial tone in the previous trace section.

The **kpButton=** message is always followed by the dialed digit. As soon as the first digit is dialed, the phone is told to stop playing dial tone. That makes sense because when you pick up the phone, you hear dial tone, but as soon as you dial a digit, the tone stops. Notice that after the phone is told to stop the tone, the soft keys are updated on the display. By looking at all the **kpButton=** messages, you can see that 3001 is the number dialed.

Note that the digit ***** is shown in a trace as the letter **e** and a **#** is shown as **f**. So for example, the message **kpButton=e** means the user entered the ***** key.

CallManager is constantly analyzing the digits the user dials, and once it finds an exact match, digit analysis returns the results for the match. Now it is time to ring Phone B, which is configured with DN 3001. Chapter 9, "Call Routing" provides additional detail about how digit analysis works in CallManager. There is, however, one important concept you should understand about digit analysis: Whenever digit analysis makes a match for a call, it displays the digit analysis results in the CCM trace. For example,

```
Digit analysis: analysis results
|PretransformCallingPartyNumber=3000
|CallingPartyNumber=3000
|DialingPartition=Line1
|DialingPattern=3001
|DialingRoutePatternRegularExpression=(3001)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(2,34,3500)
|PretransformDigitString=3001
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=3001
|CollectedDigits=3001
|UnconsumedDigits=
|TagsList=SUBSCRIBER
|PositionalMatchList=3001
|VoiceMailbox=
|VoiceMailCallingSearchSpace=IPMA:PA:Line1
|VoiceMailPilotNumber=5678
|DisplayName=James
|RouteBlockFlag=RouteThisPattern
|InterceptPartition=
|InterceptPattern=
|InterceptWhere=
|InterceptSdlProcessId=(0,0,0)
|InterceptSsType=0
|InterceptSsKey=0
|WithTags=
|WithValues=
|CgpnPresentation=NotSelected
|CallManagerDeviceType=UserDevice
```

Do not be concerned about what each of the fields means at this point. Chapter 9 explains some of the concepts such as partitions and calling search spaces. The important concept to grasp here is that any time digit analysis makes a match, you see a digit analysis result

similar to this one in the CCM trace. These digit analysis results are easy to spot in a CCM trace because of the white space to the right of the digit analysis results. If you look at a CCM trace, you will see that the majority of trace lines are over 100 characters in length; however, the digit analysis results are usually no more than 20 to 30 characters long, making the digit analysis results easy to find while scrolling quickly through a CCM trace file.

Because CallManager has collected all the required digits, it is ready to notify the destination IP phone there is an incoming call. The next example shows CallManager sending Skinny messages to Phone B.

```
StationD:    1a3e8af0 DisplayText text='                    3001        '
StationD:    1a3e8af0 CallState callState=4 lineInstance=1 callReference=16777218
StationD:    1a3e8af0 CallInfo callingPartyName='James' callingParty=3000
                      cgpnVoiceMailbox=
                      calledPartyName='Mary' calledParty=3001 cdpnVoiceMailbox=
                      originalCalledPartyName='Mary' originalCalledParty=3001
                      originalCdpnVoiceMailbox= originalCdpnRedirectReason=0
                      lastRedirectingPartyName='Mary' lastRedirectingParty=3001
                      lastRedirectingVoiceMailbox=
                      lastRedirectingReason=0
                      callType=1(InBound) lineInstance=1
                      callReference=16777218
StationD:    1a3e8af0 SetLamp stimulus=9(Line) stimulusInstance=1
  lampMode=5(LampBlink)
StationD:    1a3e8af0 SetRinger ringMode=2(InsideRing)
StationD:    1a3e8af0 DisplayNotify timeOutValue=10 notify='From 3000'
StationD:    1a3e8af0 DisplayPromptStatus timeOutValue=0 promptStatus='From 3000'
  lineInstance=1 callReference=16777218
StationD:    1a3e8af0 SelectSoftKeys instance=1 reference=16777218
  softKeySetIndex=3 validKeyMask=-1
```

Notice first that a **StationD** message is generated. This means that a message is sent from CallManager to the IP phone. Also notice that the TCP handle is different than in the preceding trace output. Each IP phone has a unique TCP handle assigned to it at registration. You can use the unique TCP handle to differentiate between the Skinny messages sent to and from Phone A (TCP handle 1a3e8b54) and those sent to and from Phone B (TCP handle 1a3e8af0). A new TCP handle is assigned any time an IP phone such as Phone A unregisters and reregisters to CallManager, resets, or fails over or back from one CallManager to another.

Notice also that the **callReference** value (16777218) is different from the previous output. As we mentioned earlier, this is because each leg of a call is assigned a different call reference. In this case, this is the call reference for Phone B. This call reference persists for the duration of the call on Phone B.

Phone B rings and the call information shows that James called Mary. You see several messages that seem to suggest the call is being redirected; it's not. These are just standard messages sent by CallManager.

Once again, do not be concerned about exactly what each message means at this point. Future chapters go into detail about each message; however, you can see from reading the trace that Phone B is being told to ring **(SetRinger ringMode=2(InsideRing))** and display "From 3000" on the prompt line of the IP phone **(promptStatus='From 3000')**.

Now that a call is in progress, Phone A gets some updated information, including display information such as called and calling party names.

```
StationD:    1a3e8b54 SelectSoftKeys instance=1 reference=16777217
  softKeySetIndex=8 validKeyMask=-1.
StationD:    1a3e8b54 CallState callState=12 lineInstance=1 callReference=16777217
StationD:    1a3e8b54 CallInfo callingPartyName='James' callingParty=3000
             cgpnVoiceMailbox= calledPartyName='Mary' calledParty=3001
             cdpnVoiceMailbox= originalCalledPartyName='' originalCalledParty=
             originalCdpnVoiceMailbox= originalCdpnRedirectReason=0
             lastRedirectingPartyName='' lastRedirectingParty=
             lastRedirectingVoiceMailbox= lastRedirectingReason=0
StationD:    1a3e8b54 StartTone tone=36(AlertingTone).
StationD:    1a3e8b54 CallState callState=3 lineInstance=1 callReference=16777217
StationD:    1a3e8b54 SelectSoftKeys instance=1 reference=16777217
  softKeySetIndex=8 validKeyMask=-1.
StationD:    1a3e8b54 DisplayPromptStatus timeOutValue=0 promptStatus='Ring Out'
  lineInstance=1 callReference=16777217.
```

Basically, these messages perform two actions. First, the display on Phone A changes now that the call is in progress. Second, the phone is told to play an alerting tone. The alerting tone is the standard ringback tone you hear when placing a call. You also see the first **callReference** value, 16777217, indicating the original call placed by Phone A. Remember each call reference is only valid for one leg of the call.

```
StationInit: 1a3e8af0 OffHook
StationD:    1a3e8af0 ClearNotify
StationD:    1a3e8af0 SetRinger ringMode=1(RingOff)
StationD:    1a3e8af0 SetLamp stimulus=9(Line) stimulusInstance=1
  lampMode=2(LampOn)
StationD:    1a3e8af0 CallState callState=1 lineInstance=1 callReference=16777218
StationD:    1a3e8af0 ActivateCallPlane lineInstance=1
```

The **StationInit OffHook** message indicates that Phone B goes off-hook and answers the call. CallManager sends a **SetRinger ringMode=1(RingOff)** message, which tells Phone B to stop ringing, and the preparation is now complete for the actual media connection.

Next, we'll examine how the audio stream is set up. As with all VoIP protocols, Skinny uses Real-Time Transport Protocol (RTP) streams over User Datagram Protocol (UDP) packets to send and receive Voice over IP (VoIP) samples. Each RTP stream is called a logical channel. A logical channel is a unidirectional RTP stream, so to have a two-way conversation, you must have two logical channels opened—one from the calling device to the called device and one from the called device to the calling device.

In a call involving a Skinny device, CallManager asks the IP phone to open a connection to receive RTP streams. CallManager asks the IP phone for specific parameters for this connection, including the codec and packet size. You can see the following **OpenLogicalChannel** messages from CallManager to each IP phone requesting that they open a connection to receive RTP packets using G.711.

```
StationD:    1a3e8b54 OpenReceiveChannel conferenceID=0
  passThruPartyID=17 millisecondPacketSize=20
  compressionType=4(Media_Payload_G711Ulaw64k)
  qualifierIn=?.   myIP: 3eee1cac (172.28.238.62)
StationD:    1a3e8af0 OpenReceiveChannel conferenceID=0
  passThruPartyID=33 millisecondPacketSize=20
  compressionType=4(Media_Payload_G711Ulaw64k)
  qualifierIn=?.   myIP: 2fee1cac (172.28.238.47)
```

Upon receiving an **OpenReceiveChannel** message, the IP phone selects the UDP port number it wants to use to receive RTP packets and reports this information back to CallManager in an **OpenReceiveChannelAck** message. Phone A responds first:

```
StationInit: 1a3e8b54 OpenReceiveChannelAck
  Status=0, IpAddr=0x3eee1cac, Port=20096, PartyID=17
```

Once CallManager receives this information from Phone A, it can tell Phone B where to send its RTP stream. Until this point, CallManager could not tell Phone B which UDP port number to use because Phone A had not reported it to CallManager. Once CallManager receives the port number in the **OpenReceiveChannelAck** message from Phone A, it sends a **StartMediaTransmission** message to Phone B giving it the IP address and port number of Phone A along with information about which voice codec to use.

```
StationD:    1a3e8af0 StartMediaTransmission
  conferenceID=0 passThruPartyID=33
  remoteIpAddress=3eee1cac(172.28.238.62)
  remotePortNumber=20096 milliSecondPacketSize=20
  compressType=4(Media_Payload_G711Ulaw64k)
  qualifierOut=?. myIP: 2fee1cac (172.28.238.62)
```

Next, CallManager receives an **OpenReceiveChannelAck** from Phone B containing the UDP port number information and passes this on to Phone A in a **StartMediaTransmission** message.

```
StationInit: 1a3e8af0 OpenReceiveChannelAck Status=0,
  IpAddr=0x2fee1cac, Port=19648, PartyID=33
StationD:    1a3e8b54 StartMediaTransmission
  conferenceID=0 passThruPartyID=17
  remoteIpAddress=2fee1cac(172.28.238.47)
  remotePortNumber=19648 milliSecondPacketSize=20
  compressType=4(Media_Payload_G711Ulaw64k) qualifierOut=?.
  myIP: 3eee1cac (172.28.238.47)
```

So at this point, Phone A (TCP handle 1a3e8b54) is sending RTP packets to 172.28.238.47, which happens to be Phone B, and Phone B (TCP handle 1a3e8af0) is sending RTP packets to 172.28.238.62, which happens to be Phone A.

Notice that for the duration of this call, Phone A has never sent nor received any Skinny signaling to or from Phone B. This is because all the signaling goes through CallManager. The only time IP phones send packets to each other is for the actual voice stream. This is what allows CallManager to set up calls between devices that use different signaling protocols. For example, if Phone A called a phone number on the PSTN instead of another IP phone, the signaling between CallManager and Phone A remains the same. Phone A has no idea that it is sending RTP packets to a voice gateway and vice versa.

When reading CCM traces, you can usually separate each leg of the call and concentrate on one part at a time. For example, if you have a call that goes out through a voice gateway, once you have verified that the IP phone dialed the correct digits and CallManager is routing the call to a voice gateway, you can focus on the gateway debugs to determine how the call gets set up.

Also it is important to separate the signaling aspects of a call set up from the RTP media streams. All VoIP devices are blindly told to send RTP packets to an IP address and port number without knowing what type of device they are sending these packets to. As long as the terminating device provided the correct IP address and port number and CallManager relayed this information correctly, everything works properly. However, pay attention to the signaling aspects to ensure that the port number received from the terminating device is the same as the port number reported to the originating device.

Tracing a Call Through an MGCP T1 PRI Gateway

The next call to dissect is an IP phone (3000) making a call through a WS-X6608-T1 PRI MGCP gateway. MGCP is not described in detail in this example; however, you will see ISDN Q.931 messages because this is the protocol used over the PRI D-channel. A detailed discussion on troubleshooting ISDN PRI problems can be found in Chapter 6. Appendix A includes the standards body and specification for ISDN Q.931.

All Q.931 and H.323 (including intercluster trunk) calls look basically the same in a CCM trace. Also, some non-ISDN gateways such as the WS-X6624 and the WS-X6608 when using T1 CAS also use Q.931 messages to communicate with CallManager. This makes understanding the basic structure of this kind of trace message important.

The majority of information presented in the CCM trace file for a gateway call is in hexadecimal notation. The Q.931 specification states that phone numbers should be encoded in ASCII as well as character strings, such as display names. The CCM trace also uses hexadecimal equivalents for the IP addresses on some occasions. Do not be intimidated by the hexadecimal values. Once you understand how to decode them, they are actually easy to understand. Appendix C provides a cheat sheet for conversions.

Whenever an H.323 or Q.931 call is made, you will see a section of trace similar to the following outgoing ISDN setup message:

```
Out Message -- PriSetupMsg -- Protocol= PriNi2Protocol.
Ie - Ni2BearerCapabilityIe IEData= 04 03 80 90 A2
Ie - Q931ChannelIdIe IEData= 18 03 A9 83 97
Ie - Q931CallingPartyIe IEData= 6C 06 00 80 33 30 30 30
Ie - Q931CalledPartyIe IEData= 70 05 80 33 30 30 31
MMan_Id= 0. (iep=  0 dsl=  0 sapi=  0 ces=  0 IpAddr=a4e81cac IpPort=2427)
IsdnMsgData2= 08 02 00 09 05 04 03 80 90 A2 18 03 A9 83 97 6C 06 00 80 33 30 30
   30 70 05 80 33 30 30 31
```

The first line gives you information about the direction of the call (either **Out Message** or **In Message**) followed by the type of message (**PriSetupMsg**) and the protocol used for the message (**PriNi2Protocol**). The direction is from the perspective of CallManager, so this means a device registered to CallManager is placing a call out through a gateway.

As with the IP phone, a unique identifier is used to keep track of the call. Use the hexadecimal IP address **IpAddr=a4e81cac** in conjunction with call reference number **02 00 09** to track the call throughout the trace. When viewing an ISDN trace, look at the **IsdnMsgData2** line to find the call reference ID. In some cases you will see **IsdnMsgData**

or **IsdnMsgData1** instead of **IsDNMsgData2**. They are equivalent. Ignore the first two numbers (usually **08**). The next two numbers are the call reference length (**02**), and the next four are the call reference value (**00 09**).

You might be wondering how **a4e81cac** is converted to an IP address in dotted decimal notation. This is a hexadecimal representation of the IP address. You can figure out the IP address by working backwards. In this example, first, take the last two digits, **ac**, which is hex for 172. Next, consider the **1c**, which is hex for 28. Third, take **e8**, which is hex for 232. Finally, take **a4**, which is hex for 164. The IP address is 172.28.232.164.

NOTE Appendix C provides a quick cheat sheet to determine how to quickly convert between decimal, hexadecimal, and binary values.

So far you know you are looking at an outbound setup message on a PRI configured for the NI2 ISDN protocol, all of which you determined from the first line of the trace. You also know the call reference from decoding the first few bytes of the **IsdnMessageData2**. The lines in the middle that begin with **Ie** are ISDN Q.931 information elements. Information elements are covered in detail in Chapter 6; however, they are all formatted the same way in the CCM trace. For example, the following is the called party information element (IE):

```
Ie - Q931CalledPartyIe IEData= 70 05 80 33 30 30 31
```

Each information element line begins with the letters **Ie** followed by the name of the information element, in this case **Q931CalledPartyIe**. After the name follows the data contained in that information element. The format of the data that follows is dependent on the particular information element. The Q.931 and H.225 specifications describe the format of each information element. Having a copy of the ITU-T Q.931 specification can prove to be invaluable when troubleshooting ISDN problems because you can get down to the exact details of each bit in the IE data.

Fortunately for most day-to-day activities you do not need to reference the Q.931 specification because the information you need is easily identifiable. For example, in the called party number information element shown above, notice the IE data contains the sequence **33 30 30 31**, which is 3001 in ASCII. This represents the directory number that is being called. Now you can follow the call's events.

Just as you searched through the CCM trace for the TCP handle of the IP phone to find the next message associated with a particular phone, you can do the same for a Q.931 or H.323 call. The call reference for a call remains the same for the duration of a call. The first bit, however, of the call reference is flipped depending on the direction of the call. Chapter 6 goes into detail about how this works, so don't worry about it right now. All you need to know is to search for the call reference minus the first digit. In this case the call reference is **00 09**, so search for **0 09**.

If you search through the CCM trace, you come to the next message. Notice that the message that follows is an **In Message**. This means it is an inbound message sent to CallManager by the ISDN network:

```
In  Message -- PriCallProceedingMsg -- Protocol= PriNi2Protocol
Ie - Q931ChannelIdIe -- IEData= 18 03 A9 83 97
MMan_Id= 0. (iep=  0 dsl=  0 sapi=  0 ces= 0 IpAddr=a4e81cac IpPort=2427
IsdnMsgData1= 08 02 80 09 02 18 03 A9 83 97
```

Notice that the call reference is now **02 80 09**. The most-significant bit (MSB) is set on the call reference value. This bit determines if this message is the originating or terminating side; however, you do not need to know what this bit means because CallManager clearly tells you which direction the message is going when it says **In Message** or **Out Messsage**. When searching, you need to look for **02 00 09** or **02 80 09** to track all events relating to this call event, although searching for just **0 09** is usually good enough.

These are just some of the tricks that help you follow call flows through the CCM trace. Subsequent chapters provide additional trace examples as we investigate other troubleshooting scenarios. Half the battle in reading a CCM trace is knowing which pieces of the trace file to ignore so that you can focus on the important messages in the trace. As you read through the following chapters, you will get a better understanding of the different messages you might find in a CCM trace.

Reading SDL Traces

An SDL trace is a very detailed trace mainly used by Cisco development engineers for code-level analysis of call processing events. To the average CallManager administrator, SDL trace files are far too detailed for normal practical use.

If you're working with the Cisco Technical Assistance Center (TAC) on problem resolution, TAC might ask you to collect SDL traces so that it can forward them to CallManager developers for analysis. You need to know how to configure the trace files to capture the right information for TAC.

Although the SDL trace files are generally not used for troubleshooting purposes, you will find a few occasions in this book where you must look in the SDL trace file to get a full understanding of a particular problem. In cases like this, we will provide details on exactly what to look for.

SDL Overview

SDL traces provide a C programming language interface to alarms and trace information in CallManager. Alarms are used to inform a TAC engineer or CallManager developer of unexpected events, such as being unable to access a file, database, Winsock, or other operating system resources.

SDL traces can span multiple servers, allowing a process on one server to communicate with a process on another server transparently. This mechanism is supported by the use of

SDL links. An SDL link spans from one server supporting SDL to another server supporting SDL.

SDL maintains a circular queue of files to log information. Over time, a file will be overwritten. The number of files (determined by the CallManager service parameter **SdlTraceTotalNumFiles**) and the number of lines per file (determined by the CallManager service parameter **SdlTraceMaxLines**) governs how long it takes to overwrite old log files.

SDL generates two types of files:

- **Log files**—Contain the actual tracing information.
- **Index files**—Indicate which log file in the circular queue is currently being used for writing. An index file allows SDL logging to start where it left off each time an application is restarted.

Log filenames are composed of the following:

SDL*nnn_qqq_xxxxxx*.txt

where

nnn represents the node ID
qqq represents the application ID (CallManager = 100)
xxxxxx represents the unique file index

Index filenames are composed of the following:

SDL*nnn_qqq*.index

where

nnn represents the node ID
qqq represents the application ID (CallManager = 100)

The actual SDL log and index files are text-based. All columns in a log entry are delimited by a vertical bar (|) character (0x7c). A log entry is broken into two components, prefix and detail, which appear in the following format:

```
| Prefix Component | Detail Component |
```

Every log entry contains a prefix. The prefix always has the same format. The common prefix of a trace line is as follows:

```
uuuuuuuuu | yy/mm/dd-hh:mm:ss:vvv | nnn | xxxx |
```

where

uuuuuuuuu represents a unique line sequence timestamp
yy represents the year
mm represents the month
dd represents the day
hh represents the hour
mm represents the minutes
ss represents the seconds

vvv represents the milliseconds
nnn represents the node ID
xxxx represents the log entry type (which defines the type of entry being logged)

The SDL process running on each CallManager allows each node in the cluster to communicate with other nodes in the cluster to exchange information. SDL is what allows CallManager clustering to work by allowing processing for tasks to run on any node in the cluster. Generally, processing for an event occurs on the node where the device performing the given action is registered. For example, if an IP phone is registered to a particular CallManager in a cluster, that CallManager usually handles all call processing operations for that phone. But what if an IP phone registered to one CallManager in the cluster places a call out a gateway registered to a different CallManager in the cluster? In this case, some of the processing might occur on one node while the rest of the processing occurs on another node.

CallManager functions by sending *signals* from one process to another. These signals are just messages from one piece of software to another internal to CallManager. These signals can be sent to a piece of code on an entirely different CallManager node in the cluster. The SDL trace lets you see this happening. The best way to see this is by looking at the log entry type field at the beginning of an SDL trace file. For example,

```
011381785 | 02/09/16 14:51:37.874 | 002 | SdlSig-O
```

The log entry type is SdlSig-O. This means this CallManager server sent an SDL signal to another node (O means outgoing). Conversely, if you see SdlSig-I, this tells you the CallManager node you are looking at received an SDL signal from another node or another process. The CallManager and CTI Manager process communicate with each other using SDL links as well, so you see communication to and from CTI Manager in the SDL trace as SdlSig-O and SdlSig-I.

So how do you know which node the signal was coming from or destined to? After the SdlSig-O or SdlSig-I message you should see several columns of text separated by the vertical bar (|) character. For example,

```
SdlSig-O  | DbDeviceClose  | initialized    | Db(1,100,20,1)  | Db(2,100,20,1)   |
```

An actual SDL trace has many more columns and spaces between columns than what is shown here. We have condensed the trace to fit on the printed page. The two columns to examine are the third and fourth columns after SdlSig-O. These describe the process sending the signal and the process to which the signal is being sent. The third column shows the destination (**Db(1,100,20,1)**) and the fourth shows the source (**Db(2,100,20,1)**).

Now you need to understand what you are looking at when you see Db(2,100,20,1). Db is the process name. You do not need to understand what the various process names signify. The important part is the four comma-separated numbers that follow the process name. The four numbers represent (in this order)

- Node number
- Application number (100 = CallManager, 200 = CTI Manager)
- Process type
- Process instance

The only ones you should be concerned with are the node number and the application number. In this case, you can see the signal is from CallManager on node 2 to CallManager on node 1.

You might be wondering why you need to know all this. Because CallManager is a distributed architecture, you might find that when looking at a CCM trace the events occurring don't seem to make sense. For example, you might see an IP phone making a call but never see the call go out a gateway to its destination, or you might see an IP phone told to play reorder for no apparent reason. When you see an unexplained event in the CCM trace, look at the same timestamp in the SDL trace to see if there was a signal from another node at that time. If so, look in the CCM trace on the other node at the same time to see if there is any additional detail about the call on the other node. This is the reason why having the clocks synchronized on all servers is so vitally important. Without time synchronization, you would have a very difficult task trying to match the events shown in the CCM trace on one server with the events on another server.

Although you will never read through an SDL trace the way you read through a CCM trace, you might occasionally have to look at the SDL trace to see what triggered a particular event in the CCM trace. In some cases, you will take the timestamp for an event in the CCM trace and match up that same timestamp in the SDL trace.

Enabling SDL Trace and Setting the Appropriate SDL Trace Level

Now that you know what the SDL trace is for, you need to know how to turn on SDL tracing and set the appropriate bit mask for the data you need. A *bit mask* is a string of bits that each represent a particular trace setting. For example, 10010011 is a bit mask. Each 1 in the bit mask indicates that a particular trace should be enabled, while each 0 indicates that a particular trace should be disabled. Bit masks are usually represented as strings of hex digits. For example, 10010011 is 0x93.

Detailed SDL tracing consumes a lot of disk space and affects the processor on the CallManager server, which can result in performance degradation under very high call volumes. However, as of CallManager 3.3, SDL trace writing is performed asynchronously which means CallManager is allowed first access to the disk and CPU for call processing. If insufficient disk or CPU bandwidth exists to write the trace file, lines are skipped in the trace, but call processing is not affected.

SDL traces are enabled in the **Service > Service Parameter** area in CallManager Administration. Table 3-3 shows the parameters you can adjust.

Table 3-3 *SDL Service Parameters*

SDL Service Parameter	Description
SdlListeningPort	This is the TCP port with which SDL links can be established between nodes in a cluster. The port is 8002 by default. There is rarely any reason to change this value.
SdlMaxRouterLatencySecs	Indicates the maximum number of seconds of signal latency before forcing a restart of the CallManager service. The default is 20.
SdlMaxUnHandledExceptions	Specifies the maximum number of CallManager exceptions that can occur before CallManager stops running. The default is 5.
SdlTraceDataFlags	This is a bit mask used to enable the tracing of SDL non-application-specific components or to modify the behavior of SDL tracing. • The recommended value for normal system debugging is **0x110**. • The recommended value when tracking problems with SDL links is **0x13D**. See Table 3-4 for details.
SdlTraceFlushImmed	Determines whether SDL trace entries are to be flushed to disk immediately. If this parameter is set to False, SDL trace entries are flushed to disk when there is spare disk bandwidth not being used for call processing. Setting this parameter to True causes higher disk input/output but ensures that all entries are written to the disk in the unlikely event of a software error. You should set this to False during normal operating conditions. The default is True.
SdlXmlTraceFlag	Determines whether XML-formatted tracing is allowed for SDL traces. The default is False.
SdlTraceDataSize	For signal types, this constrains the number of bytes that can be dumped from the data portion of a signal. This information appears in the freeform information column at the end of each line in the SDL trace file. The default is 100 bytes.
SdlTraceFilePath	This is the directory path that SDL uses to generate the log files. If this path is not defined or is defined incorrectly, SDL uses the default root path: C:\ProgramFiles\Cisco\Trace\SDL\.
SdlTraceFlag	This is a Boolean flag that indicates if SDL tracing is enabled or disabled. Set this flag to True to turn tracing on or False to turn tracing off. The default is True.

Table 3-3 *SDL Service Parameters (Continued)*

SDL Service Parameter	Description
SdlTraceMaxLines	This value indicates the maximum number of lines written to a log file before a new log file is created. The default is 10,000 lines.
SdlTraceTotalNumFiles	This value indicates the maximum number of files that can be created for logging purposes. The default is 250 files. Normally, a maximum of 250 files is not enough because the files write in round-robin or circular fashion—when the maximum limit is met, the next file starts at the beginning and overwrites the first file. Determine how much free disk space you have, deduct a safety net of 500 MB, and then use the rest of your disk space for trace space. You can calculate the size of your trace files by figuring the average CCM trace at 10,000 lines consumes about 2.5 MB; the average SDL trace at 10,000 lines consumes about 3.5 MB. Be sure to monitor free disk space by using other tools mentioned in this chapter such as PerfMon or CCEmail.
SdlTraceTypeFlags	This field indicates the bit mask value for collecting the trace type flag of choice. • The recommended value for normal call debugging is **SdlTraceTypeFlags=0x00000B04**. • The recommended value for low-level debugging or the debugging of voice gateways is **0xA000EB15**. See Table 3-5 for more details.

The bit mask definitions shown in Table 3-4 correlate to the Trace Characteristics on the SDL Trace Configuration page in CallManager Serviceability (**Trace > Configuration >** *select a server* **> Cisco CallManager >** *click the link to* **SDL Configuration**).

Table 3-4 *Non-application-specific Bits*

Name	Trace Characteristic in CallManager Serviceability	Bit Mask	Description
traceSdlLinkState	Enable SDL Link States Trace	0x00000001	Enables the tracing of SDL link states.
traceSdlLowLevel	Enable Low-level SDL Trace	0x00000002	Enables low-level SDL tracing.
traceSdlLinkPoll	Enable SDL Link Poll Trace	0x00000004	Enables the tracing of SDL link poll.

continues

Table 3-4 *Non-application-specific Bits (Continued)*

Name	Trace Characteristic in CallManager Serviceability	Bit Mask	Description
traceSdlLinkMsg	Enable SDL Link Messages Trace	0x00000008	Enables the tracing of SDL Link messages.
traceRawData	Enable Signal Data Dump Trace	0x00000010	Enables signal data dump.
traceSdlTagMap	Enable Correlation Tag Mapping Trace	0x00000020	Enables the tracing of correlation tag mapping.
traceCreate	Enable SDL Process States Trace	0x00000100	Enables the tracing of SDL process states.
traceNoPretyPrint	Disable Pretty Print of SDL Trace	0x00000200	Enables no pretty print of the SDL trace. Pretty print adds tabs and spaces in a trace file without performing post processing.
traceSdlEvent	Enable SDL TCP Event Trace	0x80000000	Enables TCP event traces.

The bit mask definitions shown in Table 3-5 correlate to the Trace Filter Settings on the SDL Trace Configuration page in CallManager Serviceability (**Trace > Configuration >** *select a server* **> Cisco CallManager >** *click the link to* **SDL Configuration**).

Table 3-5 *Bit Mask Definitions*

Name	Trace Filter Setting in CallManager Serviceability	Bit Mask	Description
traceLayer1	Enable All Layer 1 Trace	0x00000001	Enables all Layer 1 traces.
traceDetailLayer1	Enable Detailed Layer 1 Trace	0x00000002	Enables detailed Layer 1 trace.
Not used	—	0x00000008	This bit is not used.
traceLayer2	Enable All Layer 2 Trace	0x00000010	Enables all Layer 2 traces.
traceLayer2Interface	Enable Layer 2 Interface Trace	0x00000020	Enables Layer 2 interface trace.
traceLayer2TCP	Enable All Layer 2 TCP Trace	0x00000040	Enables Layer 2 TCP trace.

Table 3-5 *Bit Mask Definitions (Continued)*

Name	Trace Filter Setting in CallManager Serviceability	Bit Mask	Description
traceDetailLayer2	Enable Detailed Dump Layer 2 Trace	0x00000080	Enables a detailed dump of Layer 2 frames.
traceLayer3	Enable All Layer 3 Trace	0x00000100	Enables all Layer 3 traces.
traceCc	Enable All Call Control Trace	0x00000200	Enables all Call Control traces.
traceMiscPolls	Enable Miscellaneous Polls Trace	0x00000400	Enables miscellaneous polls traces.
traceMisc	Enable Miscellaneous Trace (Database Signals)	0x00000800	Enables miscellaneous traces (database signals).
traceMsgtrans	Enable Message Translation Signals Trace	0x00001000	Enables message translation signals **TranslateIsdnToSdlReq**, **TranslateIsdnToSdlRes**, **TranslateSdlToIsdnReq**, **TranslateSdlToIsdnRes** traces.
traceUuie	Enable UUIE Output Trace	0x00002000	Enables UUIE output traces.
traceGateway	Enable Gateway Signals Trace	0x00004000	Enables gateway signals traces.
traceCti	Enable CTI Trace	0x00008000	Enables CTI signal traces.
traceNetworkSvc	Enable Network Service Data Trace	0x10000000	Enables network service data traces.
traceNetworkEvent	Enable Network Service Event Trace	0x20000000	Enables network service event traces.
traceIccpAdmin	Enable ICCP Admin Trace	0x40000000	Enables Intracluster Control Protocol (ICCP) administration traces.
traceDefault	Enable Default Trace	0x80000000	Enables default traces.

Microsoft Performance (PerfMon)

Microsoft Performance is an administrative tool provided by the Windows 2000 operating system. It is colloquially referred to as PerfMon. PerfMon can be used to monitor a variety of performance objects. A *performance object* is a set of counters reported by a process or application running on the system that can be monitored. An example of a standard performance object is the Processor object, which contains a variety of processor-related counters such as processor utilization. PerfMon allows you to look at real-time statistics. It contains logging facilities to take snapshots of any counter at user-defined intervals.

The StatisticsEnabled service parameter in CallManager Administration (**Service > Service Paramaters >** *select a server* **> Cisco CallManager**) must be set to True to generate data in the counters. If statistics are disabled, neither PerfMon nor the Real-Time Monitoring Tool (RTMT) can collect data. Statistics are enabled by default.

CallManager includes several performance objects that let you monitor various counters related to the operation of the CallManager services and associated devices. The RTMT, which is discussed later in this chapter as part of the "CallManager Serviceability" section, provides much the same functionality as PerfMon.

Comparing PerfMon and the Real-Time Monitoring Tool (RTMT)

Although PerfMon and the RTMT allow you to view the same performance objects, each tool has its strengths and weaknesses.

PerfMon Advantages

You can configure PerfMon to log specific counters to a comma-separated values (CSV) file. This CSV file can then be imported into a spreadsheet application for further analysis.

A third-party tool called CCEmail (discussed in the next section) allows you to configure alerts in PerfMon.

Another advantage PerfMon has over RTMT is that the RTMT web browser must always be running for the counters to be monitored and alerts to be sent. However, that will likely be fixed in releases of RTMT subsequent to release 3.3(3).

RTMT Advantages

You can run the RTMT from a web browser on any PC that has IP connectivity to CallManager. PerfMon can be run only from a Windows NT/2000/XP PC that has PerfMon installed.

You can configure specific counters and save the configurations in RTMT. Then, each time you run RTMT, the pre-defined configurations are available.

In addition to performance-monitoring capabilities, RTMT provides two tabs—Devices and CTI Apps. The tabs allow you to search for devices and check the state of TAPI and JTAPI applications that are connected to CallManager.

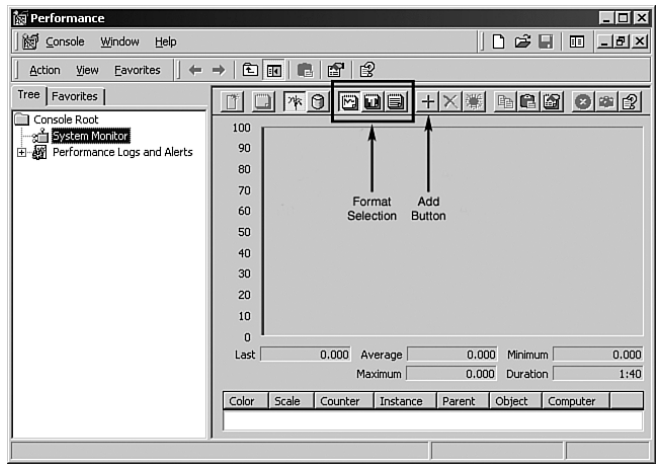
You'll read more about the additional capabilities of the RTMT later in this chapter.

Using PerfMon to View Real-time Statistics

PerfMon's most basic function is to view real-time statistics on a machine. For example, you might want to know how many calls are currently active on CallManager or the status of the channels on a PRI. This kind of information can easily be obtained through PerfMon.

An example of how to view some real-time statistics with PerfMon will better familiarize you with the tool. Launch PerfMon from a CallManager server by selecting **Start > Programs > Administrative Tools > Performance**. The application looks similar to Figure 3-5.

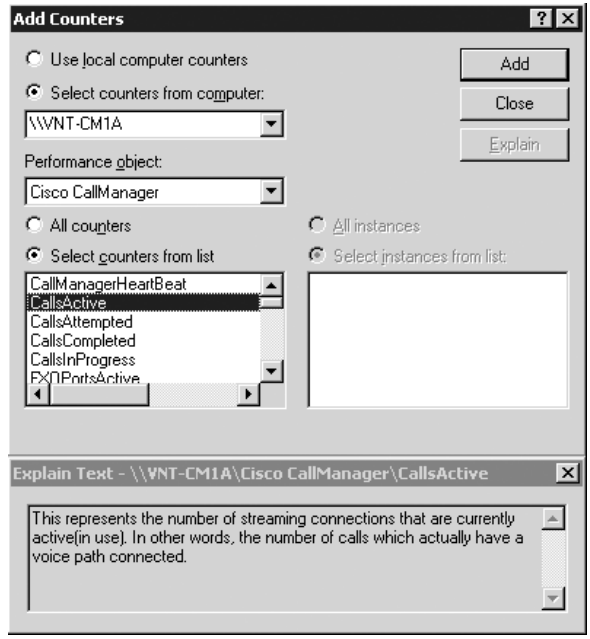
Figure 3-5 *PerfMon*



PerfMon has three formats to display data: chart, histogram, and report. Three buttons on the toolbar shown in Figure 3-5 are used to switch between the different formats. For viewing real-time statistics, you usually want to use the report format. Click the **View Report** button on the toolbar to gray out the area below.

Next, add the counters you want to monitor. Click the **Add** button on the toolbar, as shown in Figure 3-5. You see a dialog box similar to Figure 3-6.

Figure 3-6 *PerfMon Add Counters Dialog Box*



From this screen, you can add counters from either the machine on which you are running PerfMon or any other machine that has IP connectivity to the machine on which you are running PerfMon. For you to be able to view counters on remote machines, the account you are logged in with must have administrator privileges on the machine you want to monitor.

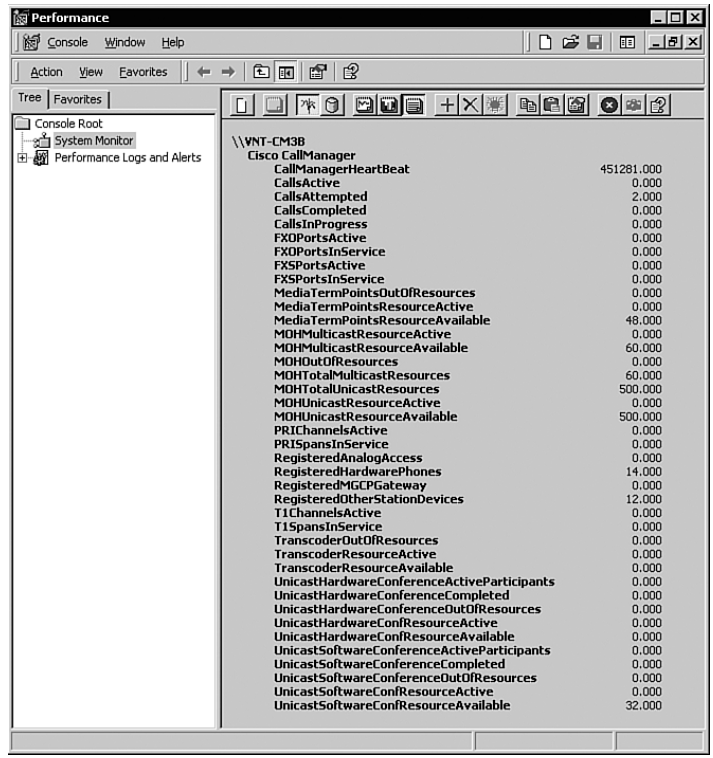
Next, select the object that contains the counter(s) you want to monitor. For example, if you want to monitor the number of active calls on a CallManager, select the Cisco CallManager object.

Below the object selection you can choose which counter to monitor. If you select the Cisco CallManager object, you see a counter labeled CallsActive. Select this counter and click **Add**. The dialog box remains open so that you can continue selecting and adding counters. To monitor all the counters for a particular object, click the **All counters** button. You can click the **Explain** button to get a short description of the counter, as shown in Figure 3-6. When you are done adding counters, click **Close**. Figure 3-7 shows all the counters in the Cisco CallManager object. You can see that the counters are updated every second or so.

Appendix D, "Performance Objects and Counters," describes the meanings of each of these counters, as well as the rest of the CallManager-related objects.

Objects and counters are only available for installed components. For example, if you do not have Cisco CallManager Attendant Console installed on the server you are trying to monitor, you will not see the Cisco CallManager Attendant Console object.

Figure 3-7 *PerfMon Displaying the CallManager Performance Object*



Some counters give you the option of selecting a specific instance of the counter to monitor. For example, the Cisco Lines object has only one counter, named **Active**. However, for this one counter, you can select one or more instances to monitor. In this case, each instance corresponds to a particular line on CallManager. You can select a range by holding down the Shift key and clicking the first and last instances in the range. Or you can select several individual instances by holding down the Ctrl key while clicking to select specific instances.

See Appendix D for detailed descriptions of all the Cisco CallManager-related performance objects and counters available in either PerfMon or the RTMT.

Using Counter Logs

One of PerfMon's most powerful features is the ability to periodically log performance information to a file. This can be useful for monitoring trends or determining exactly what time a problem occurred. Probably the most common use of counter logs is to monitor memory and CPU utilization for trends. If memory utilization continues to increase, you might be running into a memory leak. If the CPU spikes during specific times, you might be encountering one of many problems that cause high CPU utilization, such as a call routing loop.

Use the following steps to configure a counter log to monitor memory and CPU utilization on a per-process basis:

Step 1 On a CallManager server, open PerfMon by selecting **Start > Programs > Administrative Tools > Performance**.

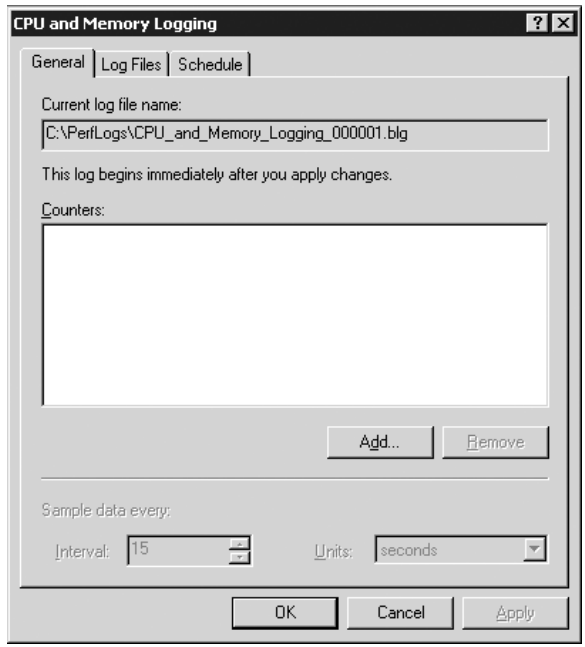
Step 2 In the left column, in the **Performance Logs and Alerts** section, select **Counter Logs**.

Step 3 Select **Action > New Log Settings**.

Step 4 In the New Log Settings dialog box, type a name for the log, such as **CPU and Memory Logging**.

Step 5 A dialog box similar to the one shown in Figure 3-8 appears. Click **Add**.

Figure 3-8 *Counter Log Configuration Dialog Box*



Step 6 The Select Counters dialog box appears (previously shown in Figure 3-6). The **Processor** object and **% Processor Time** counter are selected by default. Click **Add** to add this counter to the log.

Step 7 In the **Performance object** field, select the **Memory** object.

Step 8 In the list of counters, select the **Available MBytes** counter.

Step 9 Click **Add** and then **Close** to return to the counter log configuration dialog box.

Step 10 In the **Sample data every** area, adjust the interval depending on how often you want to monitor. For example, set it to 5 seconds to take a snapshot of the selected counters every 5 seconds. Microsoft recommends this counter not be set lower than 2 seconds when logging many different counters.

Step 11 Select the **Log Files** tab.

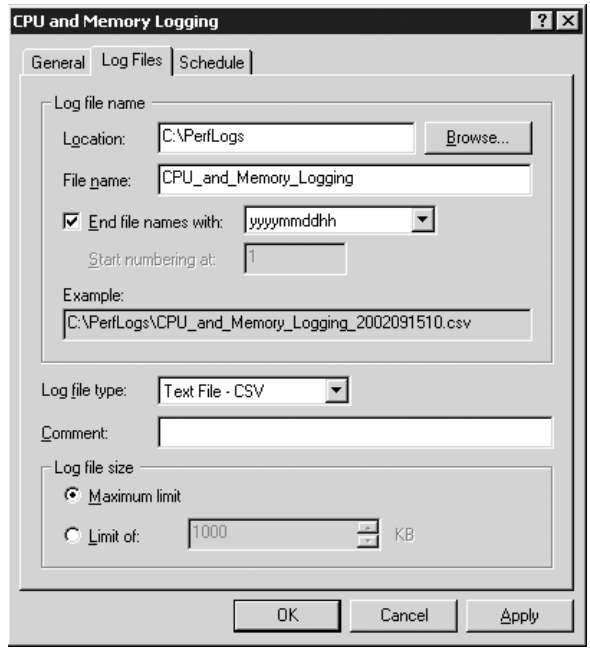
Step 12 In the **Location** field, type the path where you want to save the log files (the default is C:\PerfLogs).

Step 13 In the **File Name** field, type a filename for this log file, such as **CPU_and_Memory_Logging**.

Step 14 The **End file names with** checkbox and popup menu let you append a number to the end of the filename based on either the date and time the file is created or just an arbitrary number that increments each time a new log file is created. Select the **End file names with** checkbox, and choose a date format from the popup menu.

Step 15 Change the **Log file type** from **Binary File** to **Text File – CSV**. With the data in a CSV format. you can take the resulting data and import it into a variety of applications. Figure 3-9 shows the Log Files tab as described in the preceding steps.

Figure 3-9 *Log Files Tab for CPU and Memory Logging in PerfMon*



Step 16 Click the **Schedule** tab. If this is the first log on this system, you'll be asked if you want to create the directory, C:\PerfLogs. Click **Yes**.

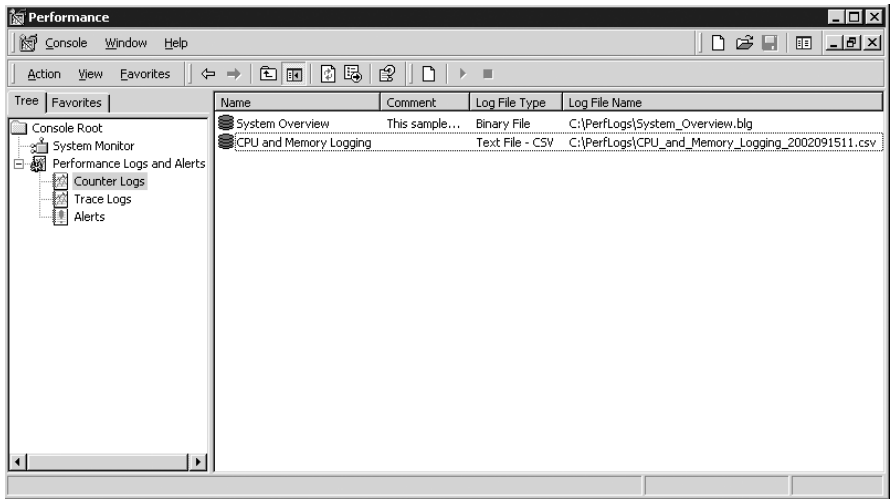
Step 17 Instead of creating one huge log file, you can have PerfMon automatically create a new file periodically. To do this, in the **Stop log** area, click **After** and select how often you want to create a new file. For example, to create a new file every day, enter **1** and set the **Units** to days.

Step 18 Check the **Start a new log file** checkbox so that PerfMon starts a new file for the frequency you specified.

Step 19 Click **OK**.

If you followed the preceding instructions, you should now see a log with a green icon to the left of it on the Counter Logs screen, as shown in Figure 3-10.

Figure 3-10 *CPU and Memory Logging Counter Log in PerfMon*



If the icon is red, right-click it and select **Start**. An error message should appear, telling you PerfMon is unable to start the log. The error will likely refer you to the Windows event log to get more details on the error. Correct the error, right-click the icon again, and select **Start**.

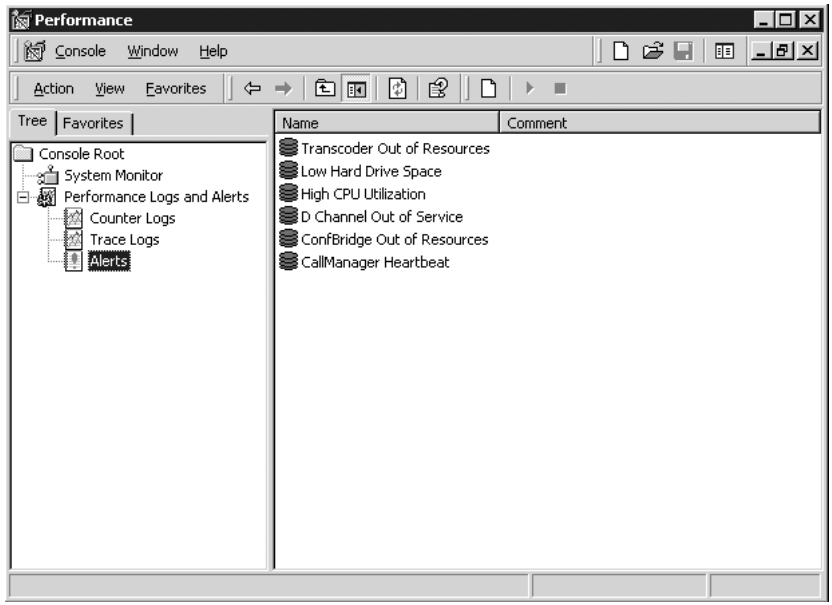
CAUTION When logging a small number of objects, as in the preceding example, setting the logging interval to something small such as 1 or 2 seconds is not a problem. However, if you are logging a large number of counters, be careful not to set the interval too low because this can lead to performance degradation.

CAUTION Unlike CCM trace files, PerfMon log files do not have a way to do circular wrapping (also known as round-robin). This means that if left unattended, PerfMon log files can use up all the available hard drive space on the server, leading to a multitude of other potential problems. If you are using PerfMon logging, ensure you periodically delete old files to free up hard drive space.

Using Alerts

PerfMon allows you to configure alerts based on selected counters. For example, you can monitor the OutOfResources counter in the Cisco HW Conference Bridge Device object and be alerted when the value in that counter crosses a threshold you specify. Figure 3-11 shows some common alerts you may want to configure.

Figure 3-11 *Alerts in PerfMon*



Other common things you might want to set an alert for are low disk space, low available memory, the D-channel going out of service on an MGCP gateway, and the number of registered phones falling below a certain value. You can place an alert on any counter in PerfMon, so the reasons and counters you may want to monitor vary based on your system.

The default action for an alert is to log an entry in Event Viewer. You can also configure the alert to send a network message to a computer you specify, run a counter log, or have a specified program run when the alert is triggered.

The Help in PerfMon (**Help > Help Topics**) provides detailed configuration steps for setting alerts.

The RTMT in CallManager Serviceability also allows you to configure alerts based on counters. However, in RTMT, you can have the alert sent as a network message to a computer you specify (just like in PerfMon) or to a pager or via e-mail to a specified recipient. You can also achieve this functionality using the CCEmail tool in combination with PerfMon.

CCEmail

You can use the Windows 2000-based tool CCEmail in conjunction with PerfMon to configure alerts for the performance counters. Alerts can be sent via pager or e-mail so long as the CallManager node for which the alert is configured has connectivity to an SMTP server. Also, the pager specified to receive alerts must be capable of receiving alphanumeric pages through an e-mail.

As part of the free tools provided with this book, you can download a copy of CCEmail from the Cisco Press web site. See the "Acquiring CCEmail" section for details.

Perform the following steps to install and configure CCEmail:

Step 1 Create a folder called **ccemail** on the C: drive of each CallManager server where monitoring is to take place.

Step 2 Download the CCEmail program from the Cisco Press website (as specified in the "Acquiring CCEmail" section) and save it to the C:\ccemail directory on each CallManager server.

Step 3 Open PerfMon (**Start > Programs > Administrative Tools > Performance**) and click on **Performance Logs and Alerts** and then **Alerts**.

Step 4 Right-click in the window and select **New Alert Settings**.

Step 5 In the **Name** field, type **minor**. Once you finish all the steps in this section, you are asked to repeat them again to create two alert settings, minor and major. The second time you perform this step, type **major** instead.

Step 6 Click **OK**.

Step 7 (Optional) Add a comment, such as **Alert settings for minor alarms** or **Alert settings for major alarms**.

Step 8 Click **Add**. The Select Counters dialog box appears.

Step 9 Make sure that **Select counters from computer** is selected and select the name of the CallManager server.

Step 10 In the **Performance object** list, choose **Cisco CallManager**.

Step 11 With the **Select counters from list** button selected, select counters you want to include in the alert and click **Add**. Add as many counters as you want to monitor with this alert. You can press and hold the Control key while clicking on counters from the list.

Some recommended counters to include for the minor alert are

- RegisteredAnalogAccess
- RegisteredDigitalAccess
- RegisteredPhones

Some recommended counters to include for the major alert are

- CallManagerHeartBeat
- RegisteredAnalogAccess
- RegisteredDigitalAccess
- RegisteredPhones

Step 12 In the **Performance object** list, choose **Process**. Select the **Private Bytes** counter and the instance **_Total**.

Step 13 In the **Performance object** list, choose **Processor**. Select the **% Processor Time** counter and the instance **_Total**.

Step 14 Click **Add** and then **Close**.

Step 15 Click on the first counter in the **Counters** list and set the threshold for when to be notified. Refer to Table 3-6 for suggestions for determining thresholds.

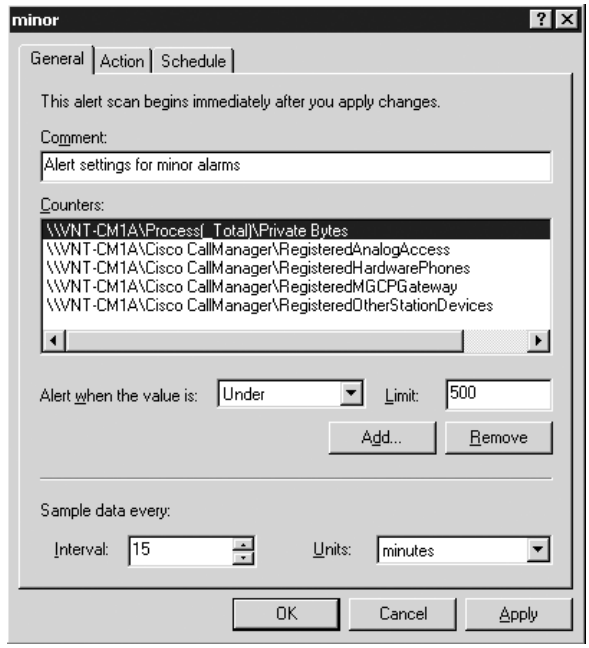
Table 3-6 *Threshold Recommendations*

Counter	Minor Alert	Major Alert
RegisteredAnalogAccess	Under 1 less than total	Under 50 percent less than total
RegisteredDigitalAccess	Under 1 less than total	Under 50 percent less than total
RegisteredPhones	Under 20 percent of total or 50, whichever is smaller	Under 40 percent of total or 150, whichever is smaller
Private Bytes (1 GB Total Memory)	900 MB	950 MB
Private Bytes (512 MB Total Memory)	450 MB	500 MB
% Processor Time	85 percent	95 percent

Step 16 Repeat the previous step until the thresholds have been set for all counters.

Step 17 Set the interval for how often the counters should be monitored. The recommended time is 15 minutes but you can choose any interval. Figure 3-12 shows the General tab for the minor alert.

Figure 3-12 *Minor Alert Settings, General Tab*



Step 18 Click the **Action** tab.

Step 19 Select the **Log an entry in the application event log** checkbox.

Step 20 Select the **Run this program** checkbox and click **Browse**.

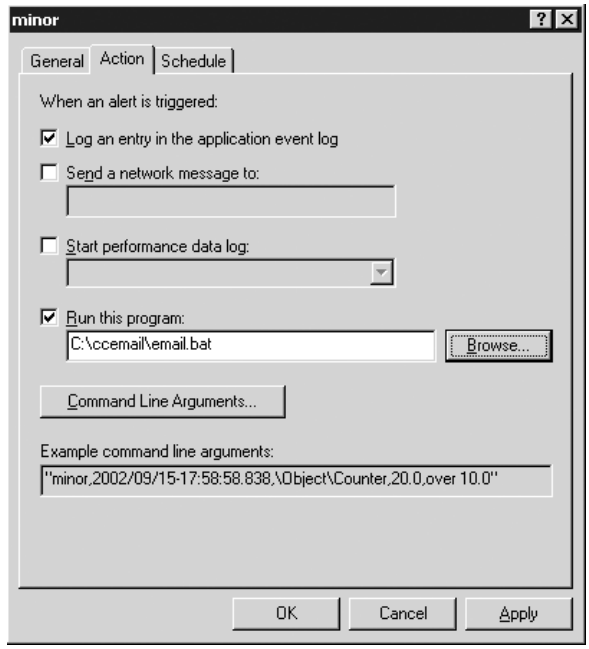
Step 21 Navigate to the C:\ccemail directory and choose the proper .bat file.

If you only want to use e-mail alerts (no paging), choose email.bat. If you only want to use paging alerts (no e-mail), choose page.bat. You can also use one method for minor alarms and the other method for major alarms. For example, major alerts always page someone and minor alerts always e-mail someone. In that case, choose page.bat for the major alerts and email.bat for the minor alerts.

Figure 3-13 shows the Action tab for the minor alert.

Step 22 Click the **Schedule** tab.

Figure 3-13 *Minor Alert Settings, Action Tab*



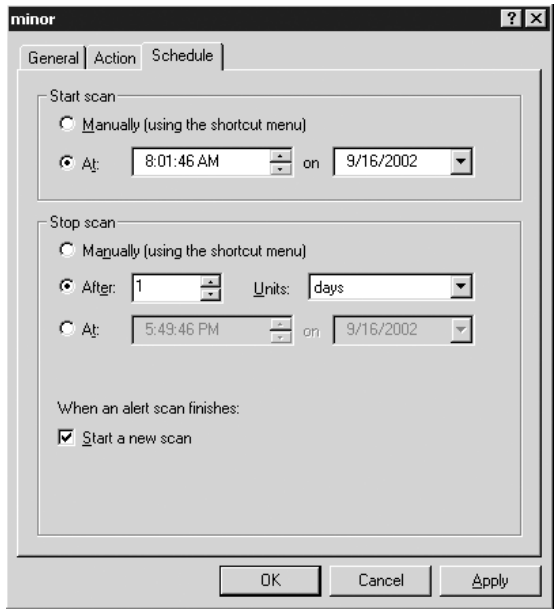
Step 23 In the **Start scan** area, make sure that **At** is selected.

Step 24 In the **Stop scan** area, make sure that **After** is selected and that the setting is **1 day**. Also, select the **Start a new scan** checkbox. Figure 3-14 shows the Schedule tab for the minor alert.

Step 25 Click **OK**.

Step 26 The alert settings for minor alarms are now complete. Repeat Steps 4 through 25 to create the alert settings for major alarms.

Figure 3-14 *Minor Alert Settings, Schedule Tab*



Perform the following steps to configure the .bat files for CCEmail:

Step 1 In the C:\ccemail directory, double-click **Setup.exe**.

Step 2 The CCEmail Setup Program displays. Click **Next**.

Step 3 In the Sender Information window, type the name of the SMTP server to be used and the sender's e-mail address.

Step 4 Click **Next**.

Step 5 Confirm the settings you just entered and click **Next** again.

Step 6 In the Email Addresses window, type the e-mail addresses, if any, to be e-mailed. For multiple addresses, separate entries with this string **-to:**.

For example: **sysadmin-alias@abc.com -to:john.smith@abc.com**

Step 7 Click **Next**.

Step 8 Confirm the settings you just entered and click **Next** again.

Step 9 In the Pager Addresses window, type the pager addresses, if any, to be paged. For multiple addresses, separate entries with this string, as described in Step 6 **-to:**.

Step 10 Click **Next**.

Step 11 Confirm the settings you just entered and click **Next** again.

Step 12 Click **Finish**.

Alerting Methods During Production and Non-production Hours

You can configure CCEmail to automatically switch between paging and e-mailing alerts depending on the time of day. Major alerts are typically important enough that a page should be received when the alert is triggered, regardless of the hour of day. Minor alerts are still important, but they're normally not considered critical enough to warrant a page during non-production hours, such as the middle of the night.

The following steps explain how to use .bat files and the Windows Task Scheduler so that minor alerts send pages during business hours and e-mails at night.

Step 1 Download the **ccemail_auto.exe** file from the Cisco Press website (as specified in the "Acquiring CCEmail" section) file and save it to the CallManager server where the changes are to be made.

Step 2 Double-click **ccemail_auto.exe** and make sure that the **Unzip to folder** field is set to **C:\ccemail**.

Step 3 Open the Windows Task Scheduler (**Start > Settings > Control Panel > Scheduled Tasks > Add Scheduled Task**).

Step 4 The Scheduled Task Wizard displays. Click **Next**.

Step 5 Click **Browse**, choose the **C:\ccemail\today.bat** file, and click **Open**.

Step 6 Type a name for the task (such as **today**), select **Daily** and click **Next**.

Step 7 For **Start time**, enter the beginning of business hours, such as 8:00 AM, select **Every Day**, and set the start date as today's date.

Step 8 Click **Next**.

Step 9 Enter the username and password of any user that has read/write permission for the C: drive and click **Next**.

Step 10 Click **Finish**.

Step 11 Repeat Steps 3 through 10 with the following changes:

- In Step 5, choose **tonight.bat** instead of today.bat.
- In Step 7, enter the end of business hours instead of the beginning, such as 6:00 PM.

Step 12 Open PerfMon (**Start > Programs > Administrative Tools > Performance**) and click on **Performance Logs and Alerts** and then click **Alerts**.

Step 13 Double-click on **minor** and click on the **Action** tab.

Step 14 In the **Run this program** field, click **Browse** and choose the **C:\ccemail\minor.bat** file.

Step 15 Click **OK** and close PerfMon.

Acquiring CCEmail

Check the Cisco Press website for a free downloadable file containing this tool (**www.ciscopress.com >** *type* **1587050757** *in the Search field* **>** *click the link to* **Troubleshooting Cisco IP Telephony**). Check the site regularly as there may also be updates to the tool or the book chapters.

CAUTION This is not an officially supported tool. If you download, install, or use this tool, you do so at your own risk. Cisco Systems, Inc., is not responsible for correcting problems that may arise as a result of using this unsupported tool.

CallManager Serviceability

CallManager Serviceability is a collection of tools that help you troubleshoot various aspects of your CIPT system. CallManager Serviceability provides end user documentation online (make a selection from the **Help** menu) and on Cisco.com at the following location:

> **www.cisco.com/univercd/cc/td/doc/product/voice/c_callmg/index.htm >** *select your CallManager release* **> Serviceability**

CallManager Serviceability provides the following basic services:

- Alarms
- Tracing and the web-based Q.931 Translator
- Service Activation
- Control Center
- Real-Time Monitoring Tool

Alarms

Options under the Alarm menu let you configure the destination for the alarms (**Alarm > Configuration**) and search for alarm message definitions (**Alarm > Definitions**). *Alarms* are messages that notify you of basic errors. The messages can be inserted into CCM (SDI) and SDL traces and the Windows Event Viewer.

CallManager Serviceability provides pre-defined alarms, set at pre-defined levels. Use the Alarm Configuration page to set up which level of alarm you want to receive and where you want those alarms sent. Table 3-7 describes the alarm event levels.

Table 3-7 *Alarm Event Levels*

Level	Description
Emergency	This level designates the system as unusable.
Alert	This level indicates that immediate action is needed.
Critical	This level indicates that CallManager detects a critical condition.
Error	This level signifies that an error condition exists.
Warning	This level indicates that a warning condition is detected.
Notice	This level designates a normal but significant condition.
Informational	This level designates information messages only.
Debug	This level designates detailed event information used for debugging.

Tracing

Options under the Trace menu let you configure trace levels and parameters for CallManager, Database Layer, CTI Manager, and other core services. CallManager Serviceability then allows you to collect this information from one node or all nodes in the cluster when necessary. You can select trace information at the device level to target one or more specific devices in the trace output. The log files generated by the trace can be .txt format or XML-enabled for detailed analysis. We have already discussed text-based and XML-based tracing in the previous sections "Reading CCM Traces" and "Reading SDL Traces." In this section, we focus on XML-formatted tracing.

Using XML-enabled Traces

Using XML-enabled traces can make reading traces easier. Earlier in this chapter, you learned how to read trace files in their raw, text-based format and saw comparisons of standard text-based tracing and XML-formatted tracing. In some cases, you might find it easier to use XML-enabled traces and let the system do the searching for you. With XML-based tracing, you are given certain trace filters that you can apply to the trace output. Also, you can have the system search and compile trace information on just the devices you need (this feature is also available for standard text-based tracing). We don't normally recommend XML-based tracing because of a 2000-line limit in the trace files and the fact that searching through a large number of trace files is very slow. If you are searching a large amount of data, for instance, a problem that occurred over the course of several hours or several days, depending on the size of your system, you will probably need to use text-based tracing instead of XML-based tracing. If you select trace collection criteria that causes the trace output to be larger than XML-based tracing can handle, a message displays in CallManager Serviceability advising you that trace analysis cannot be performed on files

greater than 2 MB in size. You are then given the option to save the result without filtering using **File > Save As** in your web browser.

The previous section, "Reading CCM Traces" detailed the steps used to configure XML-enabled tracing. Once trace files are collected, XML-based tracing allows you to filter the trace output using Trace Analysis.

Searching for Devices with XML Traces

Using the Trace Analysis feature, you can search for devices in the XML traces and narrow the scope of the trace search by choosing different search criteria and display fields. Instead of scanning thousands of lines of trace files looking for an event here and there, the XML-enabled trace can compile only the lines that relate to the devices or analysis criteria you have selected. In other words, all trace file lines you aren't looking for are filtered out so that you can concentrate on what is important to your search. Trace Analysis was discussed in detail in the previous section, "Reading CM Traces."

Web-based Q.931 Translator

The Trace menu in CallManager Serviceability provides a link to a web-based interface to the Q.931 Translator. Although convenient, the web-based interface may slow the performance of the Q.931 Translator. See the "Q.931 Translator and Enhanced Q.931 Translator" section for more information about this tool.

Service Activation

Service Activation in CallManager Serviceability (**Tools > Service Activation**) lets you activate and deactivate CallManager services. You can activate or deactivate CallManager-related services from Automatic mode by selecting or deselecting checkboxes next to specific services and then clicking the **Update** button. Then you can use the Control Center or the Windows Services Microsoft Management Console (*right-click* **My Computer** *and select* **Manage >** *double-click* **Services and Applications >** *click* **Services**) to start and stop services.

CAUTION If you need to deactivate a service, you should do so from the Service Activation page. If you deactivate services from the Service Control Manager, you get an error message saying that some of the services are not configured properly. This is because deactivating services from the Service Control Manager does not remove the entries from the CallManager database; therefore, the services are out of sync with the configured services in the CallManager database.

Control Center

The Control Center in CallManager Serviceability (**Tools > Control Center**) lets you start and stop CallManager services and view their activation status. You can stop and start services in the Windows Services Microsoft Management Console as well if you have local access to the server. If not, you can use the Control Center web page in CallManager Serviceability to do the same thing.

Real-Time Monitoring Tool (RTMT)

The RTMT is a web-based application that provides up-to-the-second information about the state of a CallManager cluster, including run-time information about CallManager and CallManager-related components such as IP phones and gateways. This includes dozens of counters on items such as call activity, trunk usage, and even memory and processor utilization.

Appendix D describes the meanings of each of the CallManager-related performance objects and counters, as well as some commonly used Windows counters.

The RTMT provides much the same functionality as PerfMon in an easy-to-use web-based tool. See the section "Comparing PerfMon and the Real-Time Monitoring Tool (RTMT)," for more information about the differences the two tools provide. You may find yourself using PerfMon for some tasks and RTMT for others. The StatisticsEnabled service parameter in CallManager Administration (**Service > Service Paramaters >** *select a server* **> Cisco CallManager**) must be set to True to generate data in the counters. If statistics are disabled, neither the RTMT nor PerfMon can collect data. Statistics are enabled by default.

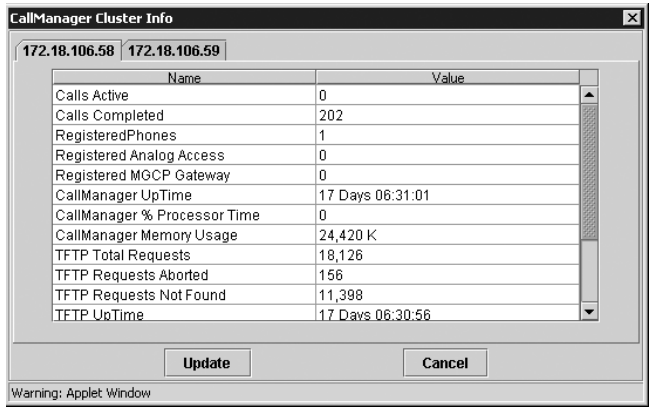
One very useful feature not provided by PerfMon (unless you use a third-party tool such as CCEmail described in this chapter) is the ability to configure alerts based on the objects. These alerts can be set to send a notification via pager, e-mail, or system message popups. These alerts can be configured for any performance object and can be set to trigger if a counter is greater or less than a specific threshold. The alerts can be configured to run only at specific times or all the time. Alerts are useful when you are troubleshooting problems because you can set the RTMT to alert you when a specific event you are trying to troubleshoot occurs. For example, if you are running out of channels on a gateway interface, you can set the RTMT to page or e-mail you when the counter for total calls to that gateway exceeds a certain value. Through release 3.3, the RTMT web browser must remain open and running for alerts to be sent. This requirement may change in future releases of CallManager Serviceability.

Objects and counters are only available for installed components. For example, if you do not have Cisco CallManager Attendant Console installed on the server you are trying to monitor, you will not see the Cisco CallManager Attendant Console object.

Performance Tab

The Performance tab in RTMT allows you to view CallManager cluster info as shown in Figure 3-15. To view this information, right-click on the cluster name and select **Properties**. The CallManager Cluster Info window displays basic statistic about the cluster, broken down by server.

Figure 3-15 *RTMT, Cluster Info*



This is also the tab where performance monitoring occurs, much the same functionality as with PerfMon. However, in RTMT, you can configure tabs for specific counter configurations. Each time you use RTMT, the pre-defined configurations are displayed. Figure 3-16 shows several counters used to monitor general activities on a system.

To add a new category, right-click on one of the existing tabs and select **New Category**. Building the category is as easy as selecting a counter and then dragging and dropping it onto the tab's frame. Multiple counters can be piled in a single frame, and six frames per tab are provided.

Devices Tab

The Devices tab in RTMT allows you to view device information that you configure for various device types—phones, gateways, H.323 devices, CTI applications, voice mail, and Cisco IP Voice Media Streaming Application devices such as MOH servers, MTP resources, and conference bridges. The Devices tab is shown in Figure 3-17.

Figure 3-16 *RTMT, Performance Tab*

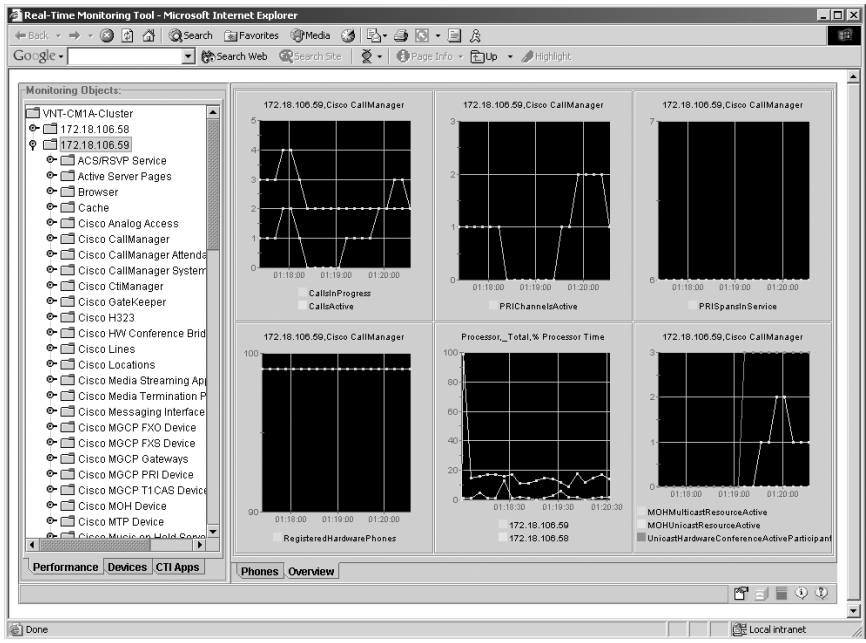
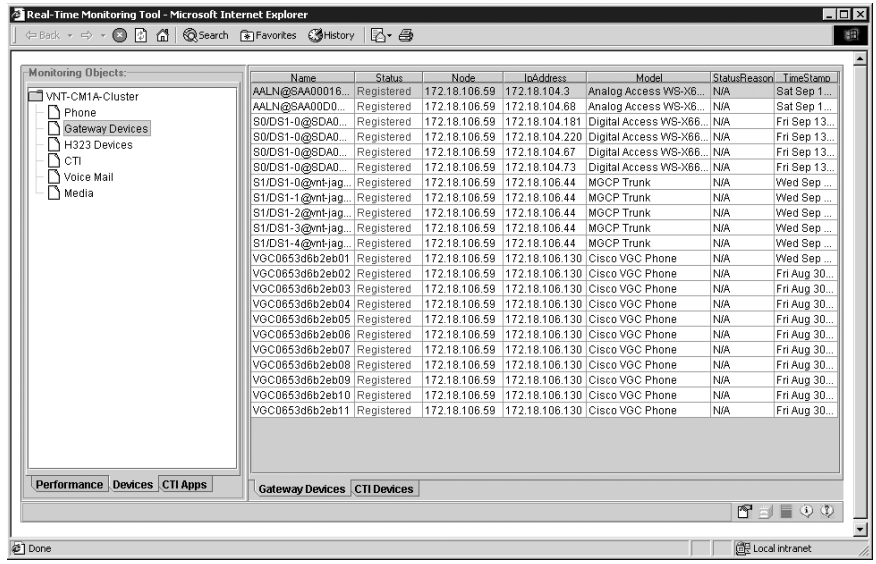


Figure 3-17 *RTMT, Devices Tab*



To add a new category, click on one of the existing tabs and select **New Category**. You build the category by first double-clicking on one of the device types and then making selections in a series of wizard-like screens that display, as shown in Figure 3-18 and 3-19.

Figure 3-18 *Device Search Criteria Screens, Part 1*

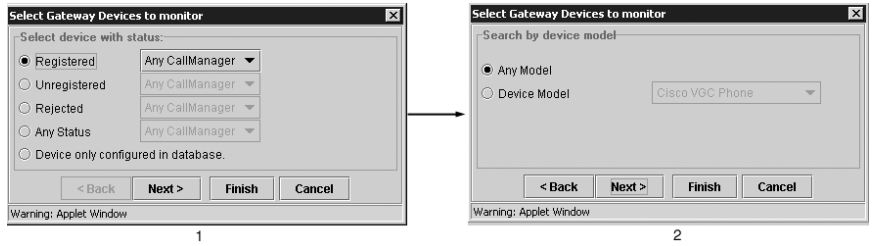
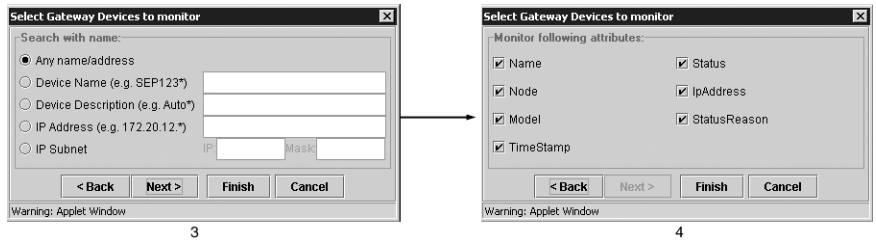


Figure 3-19 *Device Search Criteria Screens, Part 2*



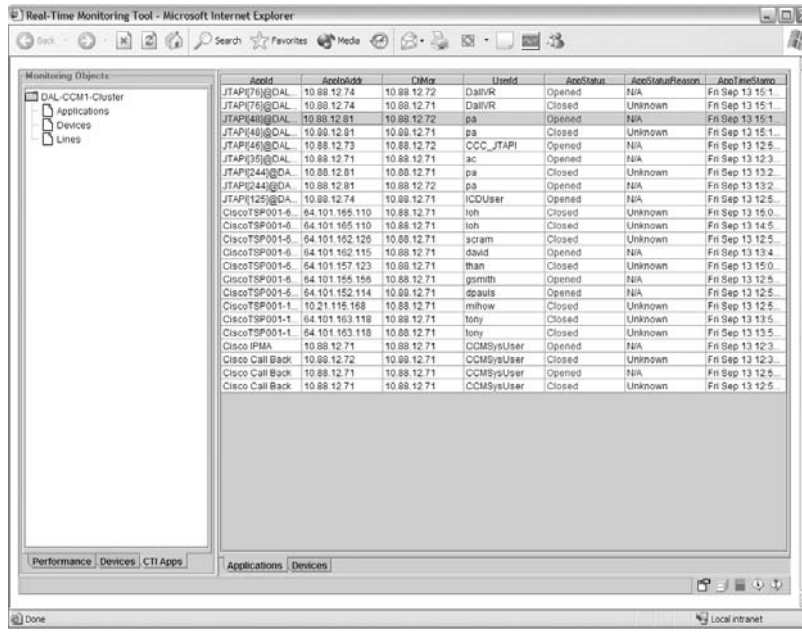
The screens shown in Figure 3-18 and 3-19 allow you to search for real-time information about devices in the cluster regardless of their registration status. You can search for devices based on device name, IP address, DN, IP subnet, and so on.

CTI Apps Tab

The CTI Apps tab in RTMT allows you to view application-, device-, and line-based information for CTI applications. You can check the registration status of TAPI and JTAPI applications such as Personal Assistant, Cisco IP Manager Assistant, Cisco IP SoftPhone, and so on. The CTI Apps tab shows whether the application has an open connection with CallManager and who's using it. The CTI Apps tab also shows you applications attempting to log in using an invalid username or password. Figure 3-20 shows the CTI Apps tab.

To add a new category, right-click on one of the existing tabs and select **New Category**. You build the category by first double-clicking on one of Applications, Devices, or Lines in the left frame and then making selections in a series of wizard-like screens that appear.

Figure 3-20 *RTMT, CTI Apps Tab*



Call Detail Records (CDR) and the CDR Analysis and Reporting (CAR) Tool

CallManager can be configured to store CDRs for all calls generated throughout a CallManager cluster. Typically CDRs are used for billing and accounting purposes, but they can also be useful when troubleshooting certain types of problems. Usually CDRs do not provide enough information to diagnose a problem, but they can help you narrow down a problem and provide information about the specific time when a problem occurred, leading you to other trace files or debugging tools that might give you more details on the problem's root cause.

CDRs are stored in their own SQL database on the Publisher server in the CallManager cluster. For CallManager to generate CDRs, you must set the **CdrEnabled** service parameter on each CallManager to True (select **Service > Service Parameters**). This service parameter is set to False by default. For additional details, you should also enable the **Call Diagnostics Enabled** service parameter. If the **Call Diagnostics Enabled** service parameter is set to True, CallManager also generates Call Management Records (CMRs, also known as Diagnostic CDRs). CMRs contain data such as packets sent and received, packets lost, and jitter for the duration of the call. One CDR might have multiple CMRs associated with it because each media stream creates a CMR. This means that if a call is placed on hold and then resumed, two CMRs are created—one for the media stream before

the call is on hold, and one for the media stream after the call is on hold. CMRs are especially useful for diagnosing voice quality problems because they allow you to see patterns. For example, you might notice that all the phones across a specific WAN link are experiencing high jitter or packet loss. This can indicate a possible quality of service (QoS) misconfiguration or line errors on the WAN link.

The CDR Analysis and Reporting (CAR) tool (**CallManager Serviceability > Tools > CDR Analysis and Reporting**) can help you analyze the raw data that comprises the CDR database and create reports based on your search criteria. For example, if you are receiving complaints of poor voice quality, one of the first things you should do is find out which phones or gateways are experiencing the poor voice quality. Although you can wait to collect data from additional user complaints, you can proactively use the data in the CDRs and CMRs to identify any trends in high jitter or packet loss to hone in on where the problem is.

For example, if a user tells you they had a problem calling someone, you can use CAR to search for the call in the CDRs. This gives you the time the call occurred, which helps you when you examine the CCM traces related to the problem.

You can search CDRs by user or specific extensions for the period that you specify. This helps you trace calls placed from specific extensions for diagnostic or informational purposes. All associated records, such as transfer and conference calls, appear together as a logical group.

CAR can also be used to send you alerts if the number of calls with poor QoS is exceeded or if the CDR database size exceeds a percentage of the maximum number of records.

For more information on the various features available in CAR, review the CAR section in the CallManager Serviceability documentation at

> **www.cisco.com/univercd/cc/td/doc/product/voice/c_callmg/index.htm >** *select your CallManager release* **> Serviceability > Serviceability System Guide > CDR Analysis and Reporting**

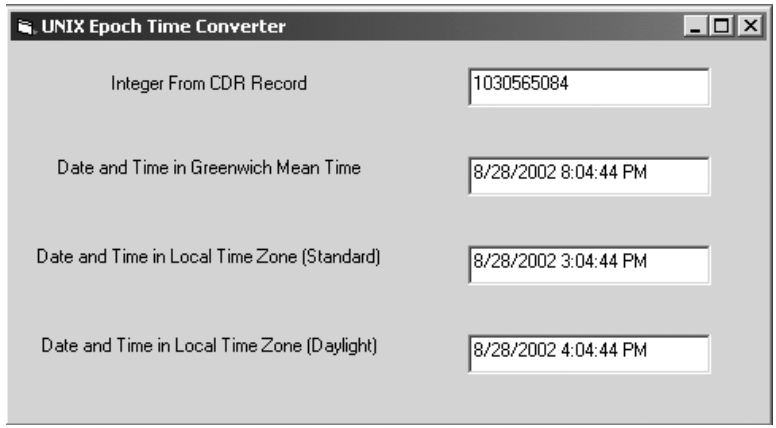
If you look at the raw timestamp information stored in the CallManager CDRs, you notice they are stored in a format that is not easily recognizable in standard date and time format. To quickly convert the date and time from the format in the CDRs to standard format, use the CDR Time Converter utility.

CDR Time Converter

Timestamps in the CallManager CDRs are stored in a format known as Epoch time. This is the number of milliseconds that have elapsed since midnight January 1, 1970 GMT. While this might be a convenient format for things like computers, humans usually prefer a more readable date and time format.

The CDR Time Converter utility allows you to enter the time as stored in a CDR—for example, 1030565084—and convert it to standard date and time format—such as 8/28/2002 3:04:44 PM. Figure 3-21 shows the output of the CDR Time Converter tool.

Figure 3-21 *CDR Time Converter Tool*



Notice the tool converts the number in Epoch time to Greenwich Mean Time (GMT) and the local time zone of the PC on which the tool is installed for both standard and daylight savings time.

Acquiring the CDR Time Converter

Check the Cisco Press website for a free downloadable file containing this tool (**www.ciscopress.com >** *type* **1587050757** *in the Search field* **>** *click the link to* **Troubleshooting Cisco IP Telephony**). Check the site regularly as there may also be updates to the tool or the book chapters.

CAUTION This is not an officially supported tool. If you download, install, or use this tool, you do so at your own risk. Cisco Systems, Inc., is not responsible for correcting problems that may arise as a result of using this unsupported tool.

Event Viewer

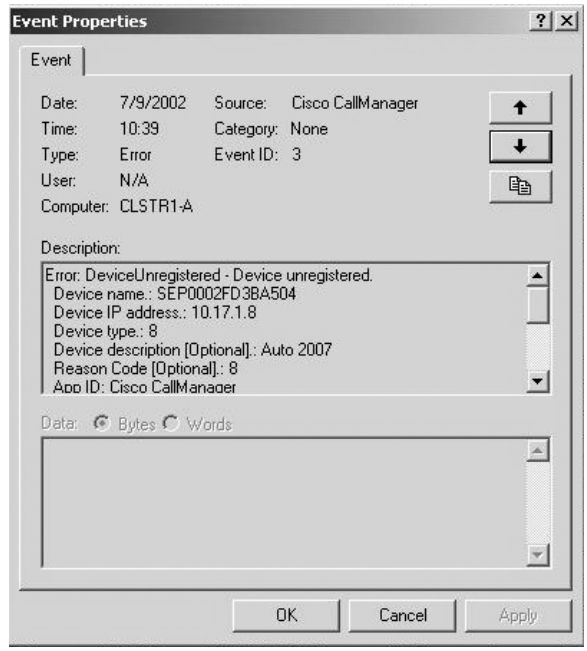
Microsoft Event Viewer is a Windows 2000 Server application that displays system, security, and application events (including CallManager) for the Windows 2000 Server. These events are alarm messages generated by CallManager. CallManager Serviceability is used to configure alarm messages to be sent to the Event Viewer (**Alarm > Configuration**).

Open Event Viewer on the server running CallManager by clicking **Start > Settings > Control Panel > Administrative Tools > Event Viewer**. CallManager errors are logged in the Application log. You can double-click an event in the log to learn more about it.

Alarm definitions can be found in CallManager Serviceability (**Alarm > Definitions**).

Figure 3-22 shows an example of an alarm message in Event Viewer.

Figure 3-22 *Event Viewer*



Notice the App ID and the Error message. This tells you that the IP phone specified in the Device Name details has unregistered from CallManager.

All alarms fall into seven catalogs, as shown in Table 3-8.

Table 3-8 *Alarm Definition Catalogs*

Catalog Name	Description
CallManager	All CallManager alarm definitions such as CallManagerFailure, DChannelOOS, DeviceUnregistered, and RouteListExhausted.
TFTPAlarmCatalog	All Cisco TFTP alarm definitions such as kServingFileWarning and kCTFTPConnectSendFileTimeoutOccurred.

Table 3-8 *Alarm Definition Catalogs (Continued)*

Catalog Name	Description
CMIAlarmCatalog	All Cisco Messaging Interface (CMI) alarm definitions such as kCCMConnectionError, kSMDIMessageError and kSerialPortOpenningError.
CtiManagerAlarmCatalog	All Cisco Computer Telephony Integration (CTI) alarm definitions such as kCtiProviderOpenRej, kCtiMaxConnectionReached, and kCtiProviderOpenFailure.
DBAlarmCatalog	All Cisco database alarm definitions such as kPrimaryDbIsLost, kUnableToConnectToDB, and kErrorBuildingCnfFile.
GenericAlarmCatalog	All generic alarm definitions shared by all applications such as OutOfMemory, ServiceStopped, and ServiceStartupFailed.
IpVmsAlarmCatalog	All Cisco IP Voice Media Streaming Application alarm definitions, including MOH, conference, media termination point, and transcoder alarms, such as kIPVMSDeviceDriverNotFound, CreateAudioSourcesFailed, and kDeviceDriverError.
JavaApplications	All Java applications that run on the CallManager server including extension mobility and Cisco IP Manager Assistant (IPMA), such as EMAppServiceError and IPMAApplicationError.

Based on the information given in the Event log entry in Figure 3-22 you can go to CallManager Serviceability to search for a better definition of the problem in question.

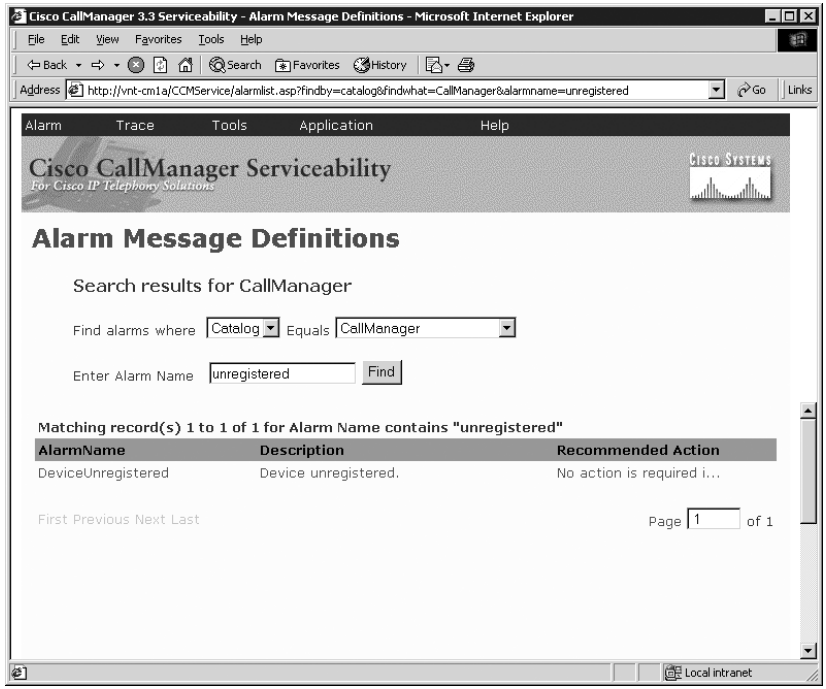
To search for alarm definitions, perform this procedure:

Step 1 In CallManager Administration, choose **Application > Cisco CallManager Serviceability**. The Cisco CallManager Serviceability window appears.

Step 2 Choose **Alarm > Definitions**.

Step 3 Choose the catalog of alarm definitions from the **Find alarms where** drop-down box, or click the **Enter Alarm Name** field to enter the alarm name. Figure 3-23 shows this interface.

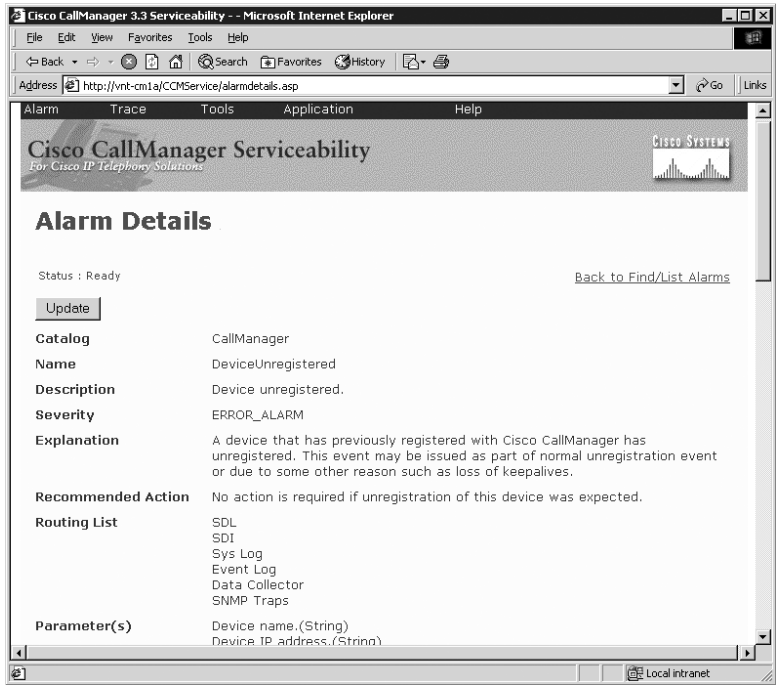
Figure 3-23 *CallManager Alarm Message Definitions*



Step 4 Click the **Find** button. The definitions list appears for the alarm catalog or search string you entered.

Step 5 In the list, click the alarm definition for which you want alarm details. Based on the Event Viewer entry shown in Figure 3-22, choose the CallManager catalog and the DeviceUnregistered error. You see Figure 3-24, showing the severity, an explanation, and the recommended action. All alarm messages found in the Event Viewer can be researched in this fashion.

Figure 3-24 *CallManager Alarm Details*



Q.931 Translator and Enhanced Q.931 Translator

Q.931 Translator is an application that takes a CCM trace and decodes the hex-formatted Q.931 messages into human-readable format. Depending on the version of CallManager you are using, there are two ways to access the application. Q.931 Translator is bundled with every CallManager installation. It can be found in the C:\Program Files\Cisco\bin directory. The file is called **Q931 translator.exe**. In CallManager version 3.2 and later, the application is also part of the CallManager Serviceability web page (select **Trace > Q931 Translator**).

Q.931 Translator is useful for quickly troubleshooting a variety of gateway problems. It is not a substitute for learning how to read the CCM trace, but it helps you resolve some gateway signaling problems without ever having to look in the CCM trace.

The name Q.931 Translator is a bit misleading because this application does more than just translate ISDN messages into a human-readable format. The Q.931 Translator also helps you troubleshoot problems with hardware that does not use the Q.931 protocol, including the WS-X6624 analog FXS card, T1 CAS on the WS-X6608 card, and calls to and from an

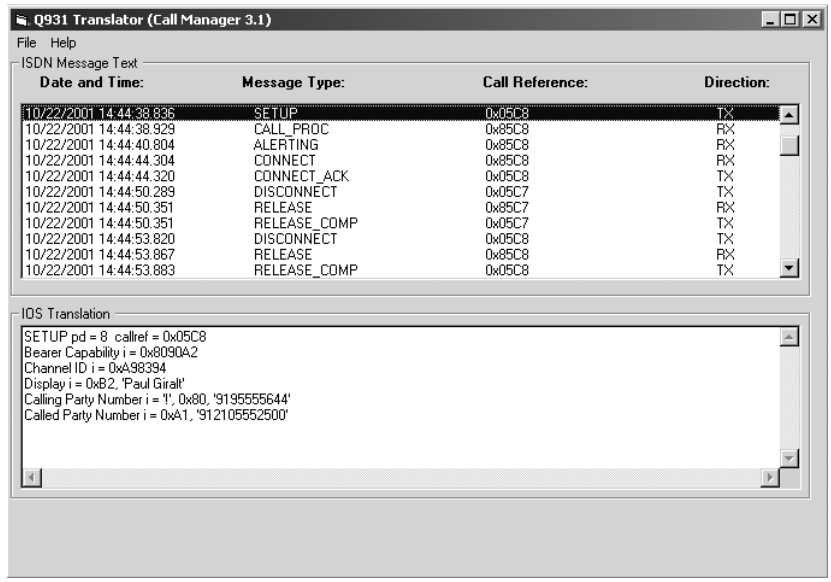
H.323 gateway. The Translator is helpful for troubleshooting problems with these hardware components because some of these gateways convert their native signaling to Q.931 messages. H.225 messages used in the H.323 protocol are based on the Q.931 specification; hence, the Q.931 Translator can decode H.225 messages.

The amount of information contained in a CCM trace can be intimidating. Q.931 Translator helps you quickly examine a trace in a graphical format without having to wade through irrelevant debugs. For example, the following sample is from a CCM trace of an outgoing call setup:

```
Out Message -- Pri250SetupMsg -- Protocol= Pri250Protocol
Ie - Ni2BearerCapabilityIe IEData= 04 03 80 90 A2
Ie - Q931ChannelIdIe IEData= 18 03 A9 83 94
Ie - Q931DisplayIe IEData= 28 0C B2 50 61 75 6C 20 47 69 72 61 6C 74
Ie - Q931CallingPartyIe IEData= 6C 0C 21 80 39 31 39 35 35 35 35 36 34 34
Ie - Q931CalledPartyIe IEData= 70 0D A1 39 31 32 31 30 35 35 35 32 35 30 30
MMan_Id= 0. (iep=  0 dsl=  0 sapi=  0 ces=  0 IpAddr=346812ac IpPort=2427)
IsdnMsgData2= 08 02 05 C8 05 04 03 80 90 A2 18 03 A9 83 94 28 0C B2 50 61 75
  6C 20 47 69 72 61 6C 74 6C 0C 21 80 39 31 39 35 35 35 35 36 34 34 70 0D A1
  39 31 32 31 30 35 35 35 32 35 30 30
```

Without a copy of the ITU Q.931 specification, which explains each bit in the trace information elements, this sample looks like a bunch of numbers and letters. However, the Q.931 Translator decodes the output into a readable format, as shown in Figure 3-25.

Figure 3-25 *Q.931 Translator Application*



One thing you should know right away is not to trust the Direction column. Information in the Direction column can occasionally be inaccurate because of how the tool decodes hex messages. To be sure of the direction, look at the first line in the debug, which states **Out**

Message. This indicates that this is an outbound (TX) setup message. Similarly, an inbound (RX) message would show **In Message** in the CCM trace.

Calls in the trace can be distinguished by the call reference value. The most significant bit (MSB) of the call reference toggles between 0 and 1, depending on whether the message was inbound to CallManager or outbound from CallManager to the gateway. Table 3-9 shows the difference between the first bit being 1 versus being 0. This table also shows the binary representation of the hexadecimal digits to clearly show how the last three binary digits in the left and right columns are identical. The only difference between the left and right columns is that the MSB for all the digits in the left column is 0, and the MSB for all the digits in the right column is 1.

Table 3-9 *Binary Representation of Hexadecimal Digits*

MSB = 0	MSB = 1
0 (0000)	8 (1000)
1 (0001)	9 (1001)
2 (0010)	A (1010)
3 (0011)	B (1011)
4 (0100)	C (1100)
5 (0101)	D (1101)
6 (0110)	E (1110)
7 (0111)	F (1111)

The output in Figure 3-25 shows that the call reference is **0x05C8** in the transmit (TX) direction and **0x85C8** in the receive (RX) direction. Any row that has a 0x05C8 or 0x85C8 in the Call Reference column is a message for the same call. As Table 3-9 shows, 0 and 8 are equivalent, with the exception of the MSB, which is different. Call reference values are eventually reused, but it is impossible for the same call reference to appear in a single CCM trace file for one gateway because several hundred or thousand calls must occur before the value is reused.

Q.931 Translator shows the calling and called party numbers converted to easily readable text, along with the bearer channel identifier, bearer capability, and display information element. The channel identifier and bearer capability are not fully decoded. You need a copy of the Q.931 specification to understand what the various bits in those fields signify. Don't worry if you don't have this specification. Chapter 6 explains how to decode some of these. Some of these are decoded for you in the Enhanced Q.931 Translator explained in the next section.

Q.931 Translator is most useful for decoding cause codes that are sent as an information element (IE) in various Q.931 messages. These cause codes are always sent as part of a DISCONNECT message and may be included as part of other messages as well. Decoding the cause code IE gives you some insight into why the call was disconnected.

The following is a sample of a CCM trace showing a Q.931 DISCONNECT message:

```
Out Message -- PriDisconnectMsg -- Protocol= Pri250Protocol
Ie - Q931CauseIe IEData= 08 02 80 81
MMan_Id= 0. (iep=  0 dsl=  0 sapi=  0 ces=  0 IpAddr=a86a12ac IpPort=2427)
IsdnMsgData2= 08 01 AA 45 08 02 80 81
```

The first thing you should notice is that this is a DISCONNECT message being sent by CallManager to the PSTN. You can see that the hex representation of the data in the Q931CauseIe is **08 02 80 81**. These cause codes are defined in the ITU Q.850 specification; however, Q.931 Translator decodes these for you. Opening the trace file containing the DISCONNECT message in Q.931 Translator reveals the following:

```
DISCONNECT pd = 8  callref = 0xAA
Cause i = 0x8081 - Unallocated/unassigned number
```

You can see that 0x8081 is decoded to **Unallocated/unassigned number**, meaning that CallManager does not have a phone or route pattern that matches the digits that were sent to CallManager by the ISDN network.

NOTE See Chapter 6 for additional information about what various cause codes mean and how to continue troubleshooting problems like these.

Enhanced Q.931 Translator

Although the Q.931 Translator allows you to quickly observe Q.931 and H.225 events, its functionality is somewhat limited. Two Cisco TAC engineers took the original source code for Q.931 Translator and enhanced it to provide additional functionality not available from the official Q.931 Translator product bundled with CallManager.

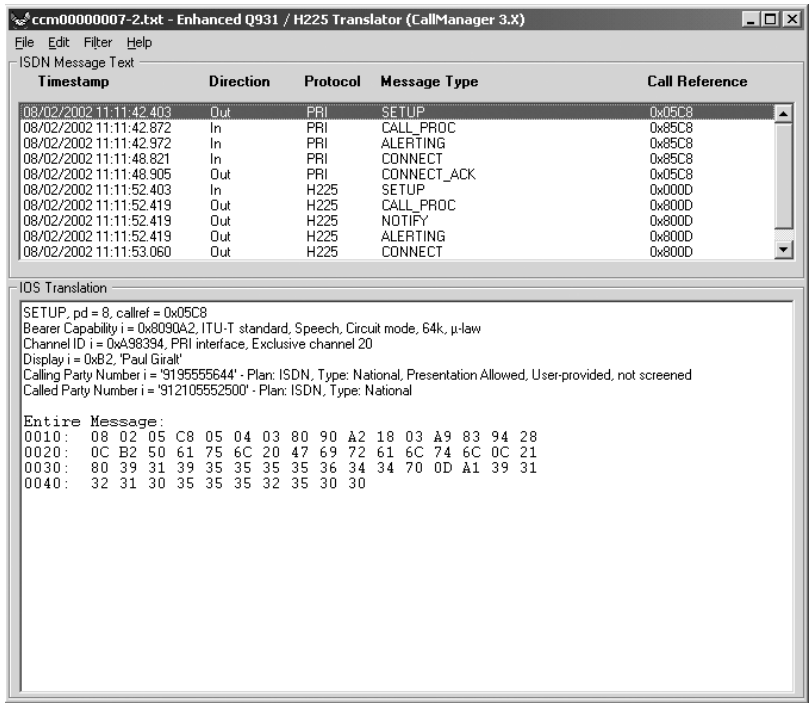
The Enhanced Q.931 Translator offers the following advantages over the standard Q.931 Translator:

- **Direction column is correct**—As mentioned earlier, the method by which the standard Q.931 Translator decodes the direction is flawed, and therefore, the direction column is sometimes incorrect. The Enhanced Q.931 Translator properly decodes the messages as **In** or **Out**.
- **Protocol column**—Tells you whether the message is a Q.931 message or an H.225 message.
- **Expanded IE decoding**—Decodes far more Q.931 information elements than the original Q.931 Translator, including bearer capability, channel ID, numbering plan and type in the calling and called party number IEs, call state, and many more.
- **Find in messages**—Allows you to search for any text that appears in the decoded message data. This means you can search for a calling or called party phone number, disconnect cause code, or any other value and quickly find the message you are looking for.

- **Filter messages**—Allows you to filter messages based on a specific call reference or protocol type. When filtering on call reference, Enhanced Q.931 Translator automatically includes all messages for that call reference regardless of the setting of the MSB of the call reference.
- **Raw ISDN message data**—After the decoded information, the raw hex bytes for the ISDN message are presented.
- **Resizable window**—Allows you to resize the window to view more messages.
- **File name display**—The currently-open filename is displayed in the title bar in case you forget where you left off when searching through several traces in a folder.

Figure 3-26 shows the same trace shown in Figure 3-25, this time as it looks in the Enhanced Q.931 Translator.

Figure 3-26 *Enhanced Q.931 Translator Application*



Notice the amount of additional data presented in the bottom pane of the tool. The bearer capability (ITU-T standard, Speech, Circuit mode, 64k, µ-law), channel ID (PRI interface, Exclusive channel 20), calling party numbering plan and type (Plan: ISDN, Type: National, Presentation Allowed, User-provided, not screened), and called party numbering plan and type (Plan: ISDN, Type: National) are all decoded for you automatically.

Future versions of CallManager may include the Enhanced Q.931 Translator, but until then, you must download the tool from the Cisco Press website (see "Acquiring Enhanced Q.931 Translator" later in this chapter).

One thing to remember is that the Q.931 Translator and Enhanced Q.931 Translator tools are not a replacement for the CCM trace. They are just another tool you can use to make decoding the CCM trace easier and finding the location of your problem quicker. Often you will have to refer back to the CCM trace after finding the call in Q.931 Translator to get additional detail regarding the events that surround a message. For example, if you find a message in Q.931 Translator indicating CallManager disconnected a call with a cause code of 0xAF, "Resources Unavailable," Q.931 Translator tells you the cause code and the timestamp in the CCM trace. You must then go to the CCM trace at that timestamp and determine what resource was unavailable that caused CallManager to disconnect the call.

As mentioned before, Q.931 Translator decodes more than just Q.931 messages. The H.225 protocol used to communicate between H.323 gateways and other CallManager clusters uses messages similar to ISDN Q.931. Q.931 Translator translates H.225 messages that appear in CCM traces.

Some gateways use Q.931 to communicate with CallManager even though the gateway's interface to the PSTN is not actually using Q.931. For example, the WS-X6624 24-port FXS gateway uses analog FXS signaling to communicate with analog phones, fax machines, and modems; however, this analog signaling is converted to Q.931 messages between the gateway and CallManager. These Q.931 messages appear in the CCM trace and are translated by the Q.931 Translator. The same is true of the WS-X6608-T1 gateway when communicating with the PSTN using channel associated signaling (CAS). The gateway converts CAS to Q.931 messages.

So if the gateways are converting these various protocols to Q.931, you might wonder how you can troubleshoot the signaling before it is converted to Q.931. This is one of the various uses of the Dick Tracy tool discussed in the section "Dick Tracy."

Acquiring Enhanced Q.931 Translator

Check the Cisco Press website for a free downloadable file containing this tool (**www.ciscopress.com >** *type* **1587050757** *in the Search field* **>** *click the link to* **Troubleshooting Cisco IP Telephony**). Check the site regularly as there may also be updates to the tool or the book chapters.

CAUTION This is not an officially supported tool. If you download, install, or use this tool, you do so at your own risk. Cisco Systems, Inc., is not responsible for correcting problems that may arise as a result of using this unsupported tool.

Dick Tracy

The Dick Tracy tool is a complex and powerful tool used to troubleshoot problems on various gateways based on the Skinny or MGCP protocols. Specifically, the Dick Tracy tool is used on the following voice gateways:

- DT-24+
- DE-30+
- WS-X6608-T1
- WS-X6608-E1
- WS-X6624

There is little to no documentation on the Dick Tracy tool because it was created as an internal development tool. However, it has been released under the condition that the tool itself is unsupported. This means that no formal documentation or release mechanism exists for the tool. This also means that the tool's behavior might change from one release to another without warning, but as long as you understand the basics of how the tool works, you should be able to pick up any changes without too much difficulty. The "Acquiring Dick Tracy" section explains how you can download a copy of the Dick Tracy tool.

There are two versions of Dick Tracy:

- A standalone Windows 95/98/NT/2000/XP application that connects to the gateway you want to monitor over TCP/IP. This version is commonly called Dick Tracy.
- A version is embedded in the Catalyst 6000 operating system (CatOS). It can be invoked from the Catalyst 6000 command-line interface (CLI) to diagnose problems on WS-X6608 and WS-X6624 gateways in the chassis. This version is commonly called CLI Tracy or embedded Tracy.

The CLI Tracy tool can be used to connect only to gateways that are in the same Catalyst 6000 chassis to which you are connected. Using the Windows-based Dick Tracy tool is recommended here because it is more flexible and easier to use than CLI Tracy.

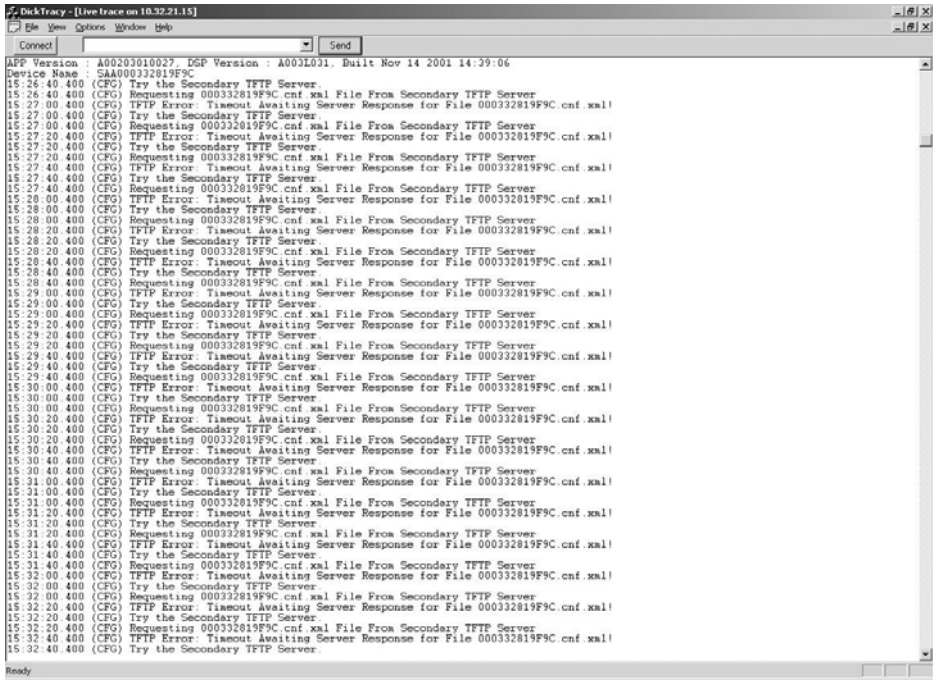
Why do you need the Dick Tracy tool? In case you've never seen a Cisco DT-24+ gateway, it is a PCI card that plugs into a PCI slot in any PC chassis. The gateway only uses the PCI slot to draw power. The PCI bus is not used for any kind of data transfer to or from the DT-24+. The DT-24+ also provides an Ethernet port and a T1 port. There is no console port or other method of out-of-band communication to the gateway; therefore, you need a tool to access the gateway so that you can determine what is going on inside the gateway. The WS-X6608 and WS-X6624 are similar to the DT-24+ because they use the Catalyst 6000 chassis they reside in for power and IP connectivity. Other than that, the Catalyst 6000 Supervisor (the management interface on the Catalyst 6000) has no out-of-band management interface to the gateways.

CAUTION Before we discuss the tool itself, you should know that the Dick Tracy tool is not the most user-friendly piece of software. It also can damage your gateway if it's not used properly. A good working rule is that if you don't know what something does, you probably shouldn't mess with it—particularly most of the **set** commands that are available. This book covers a few useful **set** commands, but other than those, you should not need to use Dick Tracy to set commands on the gateways.

Using the Dick Tracy Tool

Figure 3-27 shows the Windows version of the Dick Tracy tool. The tool itself is a very simple program. Clicking the **Connect** button on the toolbar reveals a box indicating **Connect to remote target device**. Enter the IP address of the gateway you want to troubleshoot in the **Target IP Address** box. The port number should always be 2005.

Figure 3-27 *Dick Tracy Tool for Windows*



After you connect to a gateway, you see a text box labeled **Live trace on** *ip address,* where *ip address* is your gateway's IP address. You can open additional connections to other gateways using the same procedure. You will likely see several lines of tracing when you connect. This is because the gateway buffers the last few lines of tracing and shows them to you when you connect.

After you connect to a gateway, it is a good idea to enable logging of the live trace. Click **Options > Start Logging**. A dialog box appears, asking where you want to save the logged traces. Choose a convenient location on your hard drive that has enough space available. The traces are just a text log of everything you see on your screen while connected to the gateway. They usually do not get very big unless you are running a debug for an extended period of time. Logging does not affect the performance of Dick Tracy or the gateway. Enabling logging keeps a record of all the traces you capture until you close the trace window or stop logging using the menu.

At this point, you've started a trace, and the tool is up and running. On the menu bar you see a one-line text box and a **Send** button. Think of this field as the *command box* to communicate with the gateway. To communicate with the gateway, you must understand the concept of task IDs. Each gateway has various tasks that are responsible for various components of the software. For example, one of the tasks is responsible for sending and receiving messages to and from the digital signal processors (DSPs) on the gateway.

To view the list of available tasks on a gateway, enter the command **0 show tl** (this command uses the number 0, not the letter O, and the letter L, not the number 1) in the command box and click **Send** or press **Enter**. You should see something similar to the following sample:

```
03:15:15.450 (GEN) Lennon Tasks
         0 : GEN
         1 : AUD
         2 : TRC
         3 : SNMP
         4 : SPAN
         5 : NMP
         6 : DSP
         7 : LINE
         8 : CFG
         9 : GMSG
        10 : SOCK
        11 : TMR
        12 : Q921
        13 : XA
```

The 0 (zero) is the number of the task ID you want to issue a command to. In this case, you are issuing a command to the GEN task (ID 0). Each task has various commands that can be issued to it. Some tasks do not respond to commands at all. The **show tl** portion is the command being issued to task ID 0.

Dick Tracy offers context-sensitive help for the tasks that accept commands. For example, to see which commands are available for task 0, enter **0 ?** in the command box and press **Enter**. You see something similar to the following sample:

```
03:22:21.320 GEN  --> Help -> Available Commands:
03:22:21.320 reset <hard/soft>
03:22:21.320 read func[index] <count>
03:22:21.320 write func[index] <value>
03:22:21.320 show func
03:22:21.320 clear func
03:22:21.320 set func
```

You can see that **show** is one of the available commands for task ID 0. You can get additional context-sensitive help by entering **0 show ?**. You see the following:

```
03:24:31.500 GEN   Help: show modifiers ->
03:24:31.500 Show Task List : show tl
03:24:31.500 Show Ethernet Stat's : show ether
03:24:31.500 Show Version's : show ver
03:24:31.500 Show Time : show time
03:24:31.500 Show Reset : show reset
```

You can see that the command **show tl** performs the command **Show Task List** according to the context-sensitive help. It is difficult to provide a list of the available task IDs and Tracy commands because they vary from gateway to gateway and can change from one version of CallManager to the next. Chapter 6 covers some specific task IDs when discussing troubleshooting gateways.

You should always precede each command with a task ID. If you do not precede a command with a task ID, the gateway applies the command to the last task ID you sent a command to.

In addition to the various **show** commands, a few other commands are useful. One in particular is **set mask**. You use the **set mask** command to turn on various debugs. To view what debugs you can turn on for a particular task, issue the **show mask** command for the task in question. For example, issuing the command **6 show mask** details the various trace bits for the DSP task:

```
(DSP) Mask<0x0>
   Where Bit0 = Debug Msg's
         Bit1 = Call Progression Msg's
         Bit2 = Boot Msg's
         Bit3 = Stat Msg's
         Bit4 = Cmd Msg's
         Bit5 = RTP Msg's
         Bit6 = SID Frames
         Bit7 = Status Msg's
```

As you can see from this output, the mask is currently set to **0x0**, and eight different trace bits can be enabled. The mask is set as a hexadecimal digit. In this case because you have eight bits, 256 possible combinations of tracing can be turned on, depending on which bits are enabled. For example, to enable Debug Msg's (Bit0) and Call Progression Msg's (Bit1), the mask must be set to 00000011 in binary. This translates to 0x03 in hex. Therefore, the command is **6 set mask 0x03**.

CAUTION Be sure to set the masks back to **0x0** after enabling any trace masks. The traces continue to run until the gateway is reset, even if you close your Dick Tracy session. Failure to turn off the debug masks could create a performance impact on the gateway.

For the time being, don't worry about what each of these trace masks or task IDs does. Chapter 6 covers the masks and IDs in detail.

Using the CLI Tracy/Embedded Tracy Tool

The less-used version of Dick Tracy is the CLI Tracy or embedded Tracy tool, available on the Catalyst 6000 CLI. To enable tracing for a particular port, enter the command **tracy_start** *module port,* where *module* and *port* are the module and port numbers of the port you want to debug. For example, for the gateway on port 6/3, you would enter

```
tracy_start 6 3
```

Notice that the syntax differs from the traditional Catalyst Operating System (CatOS) notation, in which ports are specified using *module/port*.

After you start CLI Tracy for a port, all the debug information for that port appears on your console or Telnet session. You can send Tracy commands to the port using the command **tracy_send_cmd** *module port taskid command*. When you need to send commands to a port, you should really use the Windows-based tool because sending commands via CLI Tracy can lead to switch instability. When you are done using the CLI Tracy tool, enter the command **tracy_close** *module port* to end the session. You can have only one CLI Tracy session open at any given time on the whole chassis. So if someone has a Tracy session open on the console port, you cannot start one from a Telnet session. Be sure to close the CLI Tracy session when you are done.

CLI Tracy does have one advantage over the Windows-based Dick Tracy tool: It can monitor a port before it obtains an IP address. For that reason, you can use CLI Tracy to troubleshoot problems where the port cannot obtain an IP address. The regular Dick Tracy tool cannot accomplish this because you need IP connectivity to the port in question before you can gather information from the port.

Acquiring Dick Tracy

Check the Cisco Press website for a free downloadable file containing this tool (**www.ciscopress.com >** *type* **1587050757** *in the Search field* **>** *click the link to* **Troubleshooting Cisco IP Telephony**). Check the site regularly as there may also be updates to the tool or the book chapters.

CAUTION This is not an officially supported tool. If you download, install, or use this tool, you do so at your own risk. Cisco Systems, Inc., is not responsible for correcting problems that may arise as a result of using this unsupported tool.

Sniffer Traces

One of the most powerful tools for troubleshooting a large number of CallManager problems is a packet capture/analyzer tool such as Network Associates' Sniffer Pro (www.nai.com), Finisar Surveyor (www.finisar.com/product/product.php?product_id=104), or Ethereal (www.ethereal.com). Because of Sniffer Pro's widespread appeal, a trace file generated by any packet-capture tool is commonly called a *sniffer trace*. This term is used here to refer to any packet-capture software, not just Sniffer Pro.

A sniffer trace lets you see exactly what is happening on the network at any given time. Examining a sniffer trace requires a good understanding of the various layers of the OSI model, which were briefly described in the "Introduction" to this book.

To get the most benefit from a packet-capture tool, you should use a tool that can decode the various protocols you might encounter in an IP telephony network. This includes, but is not limited to, H.323 (H.225 and H.245), MGCP, RTP, SQL, and LDAP. Also extremely important is the capability to decode Skinny packets. Most newer versions of commercially available network capture application can decode Skinny. Finisar Surveyor can decode Skinny as part of the standard package while Network Associates Sniffer Pro requires you purchase the Sniffer Voice add-on to get Skinny decodes. Also, the free protocol analyzer Ethereal supports decoding Skinny as of version 0.8.20.

This book does not teach you how to use the network analysis software, but instead focuses on the kind of information you can obtain using network analysis software. Sniffer traces are most important when you're troubleshooting problems that can't be examined using standard trace files or debugs because the problem is either network-related or the appropriate diagnostic tool is not built into the product. For example, device registration problems are much easier to troubleshoot with a sniffer trace, as are most voice quality problems.

Voice Codec Bandwidth Calculator

Use the Voice Codec Bandwidth Calculator to determine the bandwidth used by different codecs with various voice protocols over different media.

To get a detailed analysis of all the headers for your particular medium, use the automated Voice Codec Bandwidth Calculator available on Cisco.com. You must be a registered user on Cisco.com to access the tool.

http://tools.cisco.com/Support/VBC/jsp/Codec_Calc1.jsp

Bug Toolkit (Formerly Bug Navigator)

Cisco provides a bug search feature that allows you to find known bugs based on software version, feature set, and keywords. The resulting matrix shows when each bug was integrated or fixed, if applicable. It also allows you to save the results of a search in Bug Groups and also create persistent Alert Agents that can feed those Groups with new defect alerts.

Bug Toolkit is only available to registered users on Cisco.com. Access the bug toolkit by searching for "bug toolkit" on Cisco.com or at the following link:

www.cisco.com/cgi-bin/Support/Bugtool/launch_bugtool.pl

Remote Access Tools

There are several tools you can use to remotely access your system. In this chapter, we briefly discuss Terminal Services and Virtual Network Computing (VNC).

Terminal Services

Windows Terminal Services is a feature that comes standard on all servers running Windows 2000 Server software. Terminal Services allows you to remotely access the Windows interface on a CallManager server.

Terminal Services is extremely useful for remotely troubleshooting problems on CallManager when you do not have immediate access to the console. To use Terminal Services, you must install the Terminal Services Client on any PC running most Microsoft Windows operating systems.

The installer for the Terminal Services Client is included on all CallManager servers in the C:\WINNT\system32\clients\tsclient folder. There are several folders for the 32-bit and 16-bit versions. Use the 32-bit client on any PC running Windows 95 or later.

Once installed, launch the Terminal Services Client and enter the IP address of CallManager and the screen resolution you want to connect with. You receive a login prompt for the server. Enter the administrator username and password to authenticate. Once authenticated, you have access to the CallManager desktop almost as if you were on the console.

If you need to open a hole through your firewall to access CallManager via Terminal Services, open TCP port 3389. This is the only port needed to access a Windows 2000 Server via Terminal Services.

CAUTION Cisco does not support installing any CallManager applications, CallManager software patches, or operating system patches via Terminal Services. Terminal Services is designed for remote access to the CallManager server for troubleshooting purposes; however, some portions of the CallManager installer are not compatible with Terminal Services. You can use VNC to access the console of the server for performing upgrades and patches.

Virtual Network Computing (VNC)

VNC is basically a remote display system that allows you to view a remote desktop environment. VNC allows you to use one computer to drive actions on a target computer but differs from Terminal Services because, with VNC, any actions performed by you that occur on the target computer can be seen equally by the local user. All computers must have a local copy of VNC installed. You can use VNC to install, upgrade, or apply patches to CallManager.

You can access the VNC application and documentation files on the OS version 2000 2.2 and later CD or download. If you're running an older version of the OS, run the OS upgrade for version 2000 2.2 or later to gain access to the VNC files. OS upgrades are available on CCO at the following link:

www.cisco.com/cgi-bin/tablebuild.pl/cmva-3des

CAUTION Using VNC can expose you to a security risk. Please review the "Security Best Practices" section in the Cisco-produced document for installing VNC, which is available on the OS 2000 version 2.2 and later CD or at the download link previously shown.

Websites and Further Reading

There's a wealth of information available on the Internet. When you're looking for more information, use the following resources:

- **Cisco Technical Assistance website**—Provides the latest information and technical documentation from Cisco's Technical Assistance Center (TAC). Use the TAC website to search for tech tips and documentation, download software updates, or open a TAC case to obtain additional information. Access the TAC website by searching for "TAC" on Cisco.com or at

  www.cisco.com/tac

- **Technical tips for IP Telephony applications, servers and associated technologies**—This site contains a variety of technical documents written by TAC engineers that are useful for solving commonly encountered problems. Access the IP Telephony Applications website by searching for "IP Telephony Applications" on Cisco.com or at

  www.cisco.com/pcgi-bin/Support/browse/index.pl?i=Technologies&f=1533

- **Networking Professionals Connection**—This site is the gathering place for networking professionals to share questions, suggestions, and information about networking solutions, products, and technologies. Access the forum by searching for "networking professionals" on Cisco.com or at

  www.cisco.com/go/netpro

- **AnswerMonkey**—You'll find detailed information about Cisco Unity at the home page of one Unity's creators, Jeff Lindborg. Access the site by searching for "answermonkey" on Google.com or at

  www.answermonkey.net

- **Updates to this book**—Check the Cisco Press website regularly for updated information pertaining to the chapters in this book (**www.ciscopress.com >** *type* **1587050757** *in the Search field* **>** *click the link to* **Troubleshooting Cisco IP Telephony**).

Check out the section "Further Reading" in the "Introduction" to this book for additional books about IP Telephony and VoIP.

Best Practices

- Become familiar with the various troubleshooting tools at your disposal. Be sure to try each of them before you encounter a problem to understand how they work so you do not have to waste time learning the tool when under pressure to resolve a network outage.
- Keep a copy of all the tools in a centralized location and check frequently for updates that may add additional functionality.
- The only way to become efficient at reading CCM traces is by practicing. The more you look through CCM traces to troubleshoot problems, the better you will understand the intricacies of the CCM trace.
- Ensure remote access to your CallManager cluster is available before a problem occurs. If you need to be able to access the CallManager cluster from outside the office, make provisions for VPN or dialup access and use Terminal Services or VNC to access your CallManagers. Terminal Services works amazingly well even over slow dialup connections.

VNC Best Practices

- If you're using VNC and no longer plan to use Terminal Services for remote management, disable Terminal Services.
- Set the VNC service to Manual startup and start it only during remote management. This adds another layer of protection by requiring that users access the environment via Windows username/password authentication to start the VNC service.
- Use a complex alphanumeric password for VNC. VNC does not have a username/password structure; it only uses a single password, so make sure the password you choose is difficult to crack. VNC limits the password to eight characters. A good password includes numbers, upper- and lowercase letters, and special characters and does not use any known word. For example, 123eye67 is not as good a password choice as 4hW9Lv#g.

Summary

This chapter covered the various troubleshooting tools and resources you have at your disposal to troubleshoot a CIPT network. Which tool you use depends largely on the problem at hand. However, some problems can be resolved using more than one tool. As you become more familiar with each of the tools in your tool belt, you will begin to favor some tools over others for specific tasks. No matter how much you read about these tools, you will not learn about them until you use them to troubleshoot a real problem on your own. As you advance through the rest of this book, you will see frequent references to the tools presented in this chapter because they play an integral part in CIPT troubleshooting.

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Symbols

Numerics

A

B

C

D

E

F

G

H

I

J

K-L

M

N

O

P

Q

R

S

T

U

V

W-X-Y-Z