Cisco IP Telephony Troubleshooting Guide for Cisco CallManager Release 3.0(1)

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Purpose

This troubleshooting guide provides descriptions of the tools and utilities used to configure, monitor, and troubleshoot Cisco CallManager Release 3.0(1), Cisco IOS Gateways and Gatekeeper. Appendices provide detailed examples of three different call flows. In the first case study, a Cisco IP Phone calls another Cisco IP Phone within a cluster, which is called an intra-cluster call. In the second case study, a Cisco IP Phone calls through a Cisco IOS Gateway to a phone hanging off of a local PBX or somewhere on the PSTN. In the third case study, a Cisco IP Phone calls another Cisco IP Phone in a different cluster, which is called an inter-cluster call. Once you understand the call flow and debug traces, it will be easier to isolate a problem and determine which component is causing the problem. This document helps you understand the tools available to troubleshoot potential problems and to understand the call flows and series of events through the call traces and debug outputs.

In the event that you must contact the Cisco Technical Assistance Center (TAC), many of the tools explained here are instrumental in gathering the data required by TAC. Having this information before calling TAC assists with faster problem resolution.

Version

All discussions in this document are written for Cisco CallManager Release 3.0(1), unless otherwise stated.

Topology

It is very important to have an accurate topology of the network that contains the ports to which various components are connected, such as VLANs, routers, switches, gateways, and so on. Having a well-documented topology will assist you in troubleshooting problems with the system. You need to ensure that you have an accurate topology, access to all the network devices, and terminal services for management of the Cisco CallManager.

Adding IP telephony to a new or existing network requires significant planning to ensure success. Since real-time traffic has different requirements than data traffic, the network must be designed with low latency and quality-of-service (QoS) in mind. As with any network that carries mission-critical traffic, it is imperative that the network administrator maintains accurate, detailed diagrams of the network topology. In a crisis situation it is important to know not just the broad overview of the network, but also which ports are connected to network components, such as routers, switches, Cisco CallManager servers, gateways, and other critical devices. It is important to plan the network with redundancy and scalability in mind.

Caution: Cisco does not support using hubs for shared connectivity to the switches as they can interfere with correct operation of the IP telephony system.

When working with switched networks, knowing the state of the spanning-tree for redundancy is critical. The state of the network should be documented before any failure occurs.

Documentation Checklist

Use the following checklist to be sure you have the proper documentation on your network topology.

- Topology that shows all network devices and critical components with port/interface numbers to which they are attached, and what VLAN (if applicable) to which they belong. Special designations should be used for ports that are in trunking or channeling mode.
- The root of the spanning-tree should be configured and all normally blocking ports should be identified.
- Any WAN circuits should be identified with the amount of bandwidth (CIR in the case of frame-relay).

Note: The Cisco IP Phone 7960 has a 10/100-switched network port and a 10/100 PC port. Cisco does not support “cascading” phones off of the PC port. We do not recommend attaching both the network and PC ports to a switch (thereby creating a physical loop in the network).

Any WAN interface will require special consideration, since this is a potential source of congestion. Cisco IP Phones and gateways set the RTP stream IP precedence field to five, however this only tags the RTP packet. It is up to the network administrator to ensure that the network is configured for prioritization and call admission control so that the Voice over IP (VoIP) traffic can be serviced with minimal delay and contention for resources. For additional information on this topic, see:
http://www.cisco.com/warp/public/793/voip/

Glossary of Terms

Following are some common terms and acronyms that may be used in this document.

Glossary	
Acronym/Term	**Definition**
.cnf	Configuration file used by devices.
µ-law ("mu-law")	Companding technique commonly used in North America. µ-law is standardized as a 64-kbps codec in ITU-T G.711.
A-law	ITU-T companding standard used in the conversion between analog and digital signals in PCM systems. A-law is used primarily in European telephone networks and is similar to the North American µ-law standard.
ACF	Admission Confirm.
ANI	The calling number
ARQ	Admission Request.
B-Channel	Bearer channel. In ISDN, a full-duplex, 64-kbps channel used to send user data.
Calling Search Space	The Calling Search Space defines what directory numbers and route patterns a given device can call. It is a grouping of partitions to look through when making a call. For example, assume there are several Partitions in a Calling Search Space named "Executive." If a Cisco IP Phone number is in the "Executive" Calling Search Space, then when initiating a call, it looks for the example "NYInternationalCall," "NYLongDistance," "NYLocalCall," and "NY911" Partitions available to search through. A Cisco IP Phone number that has a "Guest" Calling Search Space, for example, might only be allowed to search through "NYLocalCall" and "NY911" Partitions, so that if the user tries to dial an international number, it won't find a match and the call can't be routed.
CCAPi	Call Control API. Used by Cisco IOS to handle VoIP call processing.
CCO	Cisco Connection Online (http://www.cisco.com). Provides the latest information on Cisco products, technical support information, and technical documentation.
CDR	Call Detail Record. Information about call origination, destination, and duration, used to create billing records.
Cisco IOS	Cisco system software that provides common functionality, scalability, and security for all products under the CiscoFusion architecture. Cisco IOS allows centralized, integrated, and automated installation and management of internetworks, while ensuring support for a wide variety of protocols, media, services, and platforms.
Cluster	Cisco CallManager cluster. A logical grouping of several Cisco CallManager servers.
CMR	Call Management Records, also known as Diagnostic CDRs. Records that contain the count of bytes sent, packets sent, jitter, latency, dropped packets, and so on.
codec	Coder-Decoder. A DSP software algorithm used to compress/decompress speech or audio signals.
D-Channel	Data channel. Full-duplex, 16-kbps (BRI) or 64-kbps (PRI) ISDN channel. Used for signaling and control.
DCF	Disengage Confirm.
DHCP	Dynamic Host Configuration Protocol. Provides a mechanism for allocating IP addresses dynamically so that addresses can be reused when hosts no longer need them.
DN	Directory Number. This is the phone number of an end device. It can be a number assigned to a Cisco IP Phone, a Cisco IP SoftPhone, fax

Glossary	
	machine, or analog phone attached to a gateway. Examples include 1000, 24231, and so on.
DNIS	Dialed Number Identification Service.
DNS	Domain Name System. System used in the Internet for translating names of network nodes into addresses.
DRQ	Disengage Request.
DTMF	Dual tone multifrequency. Use of two simultaneous voice-band tones for dialing (such as touch tone).
Flow	Stream of data traveling between two endpoints across a network (for example, from one LAN station to another). Multiple flows can be transmitted on a single circuit.
Full duplex	Capability for simultaneous data transmission from both a sending station and a receiving station.
G.711	Describes the 64-kbps PCM voice coding technique. In G.711, encoded voice is already in the correct format for digital voice delivery in the PSTN or through PBXs. Described in the ITU-T standard in its G-series recommendations.
G.729	Describes CELP compression where voice is coded into 8-kbps streams. There are two variations of this standard (G.729 and G.729 Annex A) that differ mainly in computational complexity; both provide speech quality similar to 32-kbps ADPCM. Described in the ITU-T standard in its G-series recommendations.
H.225	An ITU standard that governs H.225 session establishment and packetization. H.225 actually describes several different protocols: RAS, use of Q.931, and use of RTP.
H.245	An ITU standard that governs H.245 endpoint control.
H.323	Extension of ITU-T standard H.320 that enables videoconferencing over LANs and other packet-switched networks, as well as video over the Internet.
Half Duplex	Capability for data transmission in only one direction at a time between a sending station and a receiving station. BSC is an example of a half-duplex protocol.
Hookflash	Short on-hook period usually generated by a telephone-like device during a call to indicate that the telephone is attempting to perform a dial-tone recall from a PBX. Hookflash is often used to perform call transfer.
ICCP	Intra-Cluster Control Protocol
ISDN	Integrated Services Digital Network. Communication protocol, offered by telephone companies, that permits telephone networks to carry data, voice, and other source traffic.
Jitter	The variation in the arrival times of voice packets.
µ-law (“mu-law”)	Companding technique commonly used in North America. µ-law is standardized as a 64-kbps codec in ITU-T G.711.
MGCP	Media Gateway Control Protocol. A protocol for Cisco CallManager to control VoIP gateways (MGCP endpoints).
MTP	Media Termination Point.
Partition	A Partition is a logical grouping of Directory Numbers and Route Patterns with similar reachability characteristics. For simplicity, these are usually named for their characteristic, such as "NYLongDistance", "NY911", etc. When a DN or Route Pattern is placed into a certain partition, this creates a rule for who can call that device or Route List.
PBX	Private Branch Exchange. Digital or analog telephone switchboard located on the subscriber premises and used to connect private and public telephone networks.

Glossary	
PRI	PRI is Primary Rate Interface. Primary rate access consists of a single 64-Kbps D channel plus 23 (T1) or 30 (E1) B channels for voice or data.
PSTN	Public Switched Telephone Network. General term referring to the variety of telephone networks and services in place worldwide.
Q.931	ITU standard that describes ISDN signaling. The H.225.0 standard uses a variant of Q.931 to establish and disconnect H.323 sessions.
RAS	Registration, Admission, and Status protocol. Protocol used in the H.323 protocol suite for discovering and interacting with a gatekeeper.
Route Filter	A route filter can be used not only to restrict dialing, but also to identify a subset of a wildcard pattern (when using the @ wildcard in the North American Dialing Plan). For example, it could be used to block the dialing of 900 area codes. In can also be used in conjunction with Partitions and Calling Search Spaces to set up complex rules. For example, assume you have three user groups established, Executive, Staff, and Guest. A Route Filter can allow the Executive user group to dial international numbers; while the Staff user group can only dial local numbers or long distance calls; and the Guest user group can only dial local numbers, 911, and 800 numbers.
Route Group	A Route Group is a list of one or more gateways or ports on gateways that are seen as equal access. It is analogous to a trunk group in traditional PBX terminology. For instance, you may have two PRI circuits to the same carrier that can be used arbitrarily. A gateway (or a particular port on a gateway) can only be added to one Route Group.
Route List	Formerly called Route Point, the Route List allows Cisco CallManager to hunt through a list of Route Groups in a configured order of preference. Multiple Route Lists can point to the same Route Groups.
Route Pattern	A specific number or, more commonly, a range of dialed numbers that will be used to route calls to a device (such as a Cisco Access DT-24+ Gateway or a voice-capable router) or indirectly via a Route List. For example, 1XXX signifies 1000 through 1999. The 'X' in 1XXX signifies a single digit, a wildcard. There are other such wildcards (such as @, .,!, etc). A Route Pattern does not have to be unique within a partition as long as the Route Filter is different.
RRJ	Registration Reject.
RTP	Real-Time Transport Protocol. One of the IPv6 protocols. RTP is designed to provide end-to-end network transport functions for applications transmitting real-time data, such as audio, video, or simulation data, over Multicast or Unicast network services. RTP provides services such as payload type identification, sequence numbering, time stamping, and delivery monitoring to real-time applications.
SEP	Selsius Ethernet Phone. Acronym that precedes MAC Addresses on Cisco IP Phones, and represents a unique device identifier.
Silence Suppression (Voice Activation Detection)	Silence Suppression allows a Cisco IP Phone to detect the absence of audio and does not transmit packets over the network. The sound quality may be slightly degraded but the connection may also use less bandwidth. Silence Suppression is disabled by default.
SNMP	Simple Network Management Protocol. Network management protocol used almost exclusively in TCP/IP networks. SNMP provides a means to monitor and control network devices, and to manage configurations, statistics collection, performance, and security.
SQL	Structured Query Language. International standard language for defining and accessing relational databases.
T1/CAS	T1 is a digital WAN carrier facility, transmitting DS-1-formatted data at

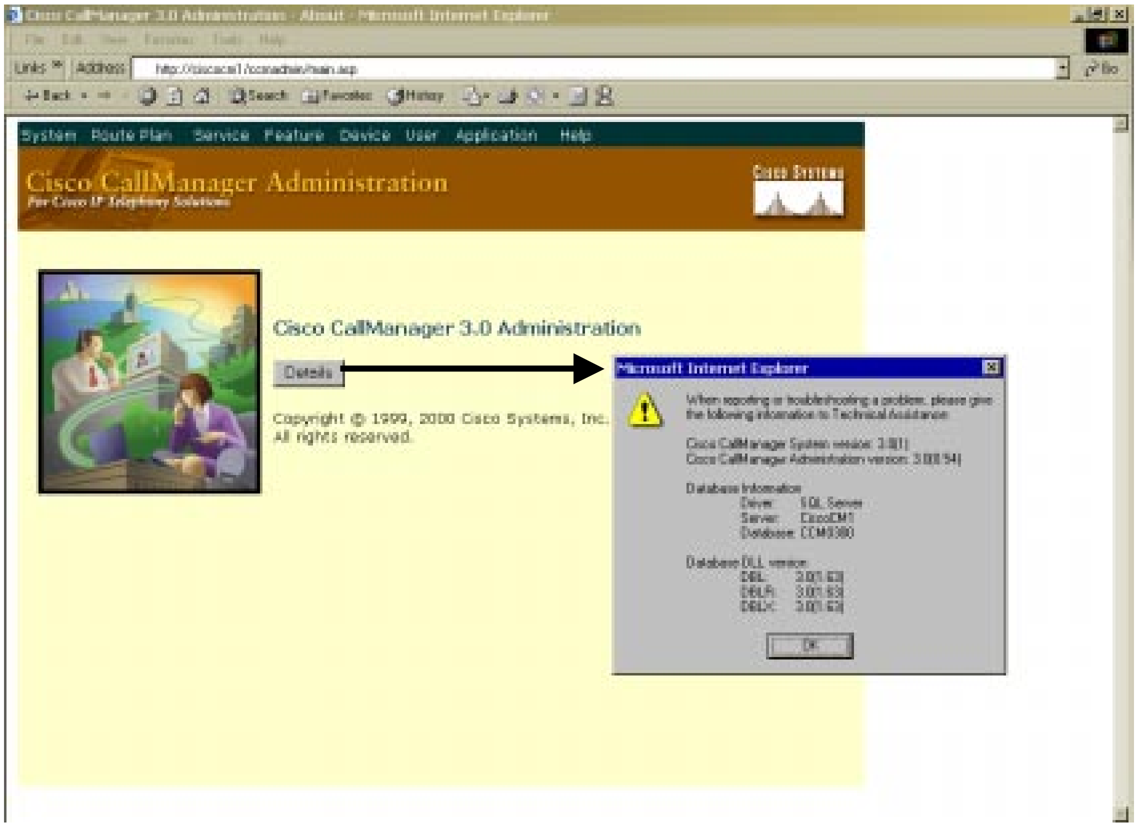
Glossary	
	1.544 Mbps through the telephone-switching network, using AMI or B8ZS coding. CAS is a Channel Associated Signaling interface.
T1/PRI	T1 is a digital WAN carrier facility, transmitting DS-1-formatted data at 1.544 Mbps through the telephone-switching network, using AMI or B8ZS coding. PRI is Primary Rate Interface. Primary rate access consists of a single 64-Kbps D channel plus 23 (T1) or 30 (E1) B channels for voice or data.
TCP	Transmission Control Protocol. Connection-oriented transport layer protocol that provides reliable full-duplex data transmission. TCP is part of the TCP/IP protocol stack.
TFTP	Trivial File Transfer Protocol. Simplified version of FTP that allows files to be transferred from one computer to another over a network.
Translation Pattern	Used to translate called (DNIS) and calling (ANI) numbers before routing the call. For example, a call may come in to a set of numbers 919 392-3XXX that need to be translated to a set of Cisco IP Phones that are in the range of 2XXX. Cisco CallManager has a Translation Pattern set up for 919 392-3XXX. This pattern translates the leading 919 392-3 simply to 2 while leaving the remaining digits intact. Then the call is routed to the appropriate Cisco IP Phone. Translation Patterns are used only for true translations and should not be used for simple digit stripping and prefixing.
UDP	User Datagram Protocol. Connectionless transport layer protocol in the TCP/IP protocol stack. UDP is a simple protocol that exchanges datagrams without acknowledgments or guaranteed delivery, requiring that error processing and retransmission be handled by other protocols. UDP is defined in RFC 768.
Voice Activation Detection (Silence Suppression)	Voice Activation Detection allows a Cisco IP Phone to detect the absence of audio and does not transmit packets over the network. The sound quality may be slightly degraded but the connection may also use less bandwidth. VAD/Silence Suppression is disabled by default.
VoIP	Voice over IP.
VLAN	virtual LAN. Group of devices on one or more LANs that are configured (using management software) so that they can communicate as if they were attached to the same wire, when in fact they are located on a number of different LAN segments. Because VLANs are based on logical instead of physical connections, they are extremely flexible.

Tools and Utilities to Monitor and Troubleshoot Cisco CallManager

This section addresses the tools and utilities to configure, monitor and troubleshoot Cisco CallManager.

Cisco CallManager Administration Details

Cisco CallManager Administration provides version information for the system, database and other components. On the opening page, press the Details button and write down the versions in use.


Cisco CallManager 3.0 Administration - About - Microsoft Internet Explorer
http://ciscocm1/ccmadmin/main.asp
System Route Plan Service Feature Device User Application Help
Cisco CallManager Administration
For Cisco IP Telephony Solutions
Cisco Systems
Cisco CallManager 3.0 Administration
Details
Copyright © 1999, 2000 Cisco Systems, Inc.
All rights reserved.
Microsoft Internet Explorer
OK

```
When reporting or troubleshooting a problem, please give
the following information to Technical Assistance:

Cisco CallManager System version: 3.0(1)
Cisco CallManager Administration version: 3.0(0.54)

Database Information
        Driver:     SQL Server
        Server:     CiscoCM1
        Database:   CCM0300

Database DLL version
        DBL:     3.0(1.63)
        DBLR:    3.0(1.63)
        DBLX:    3.0(1.63)
```

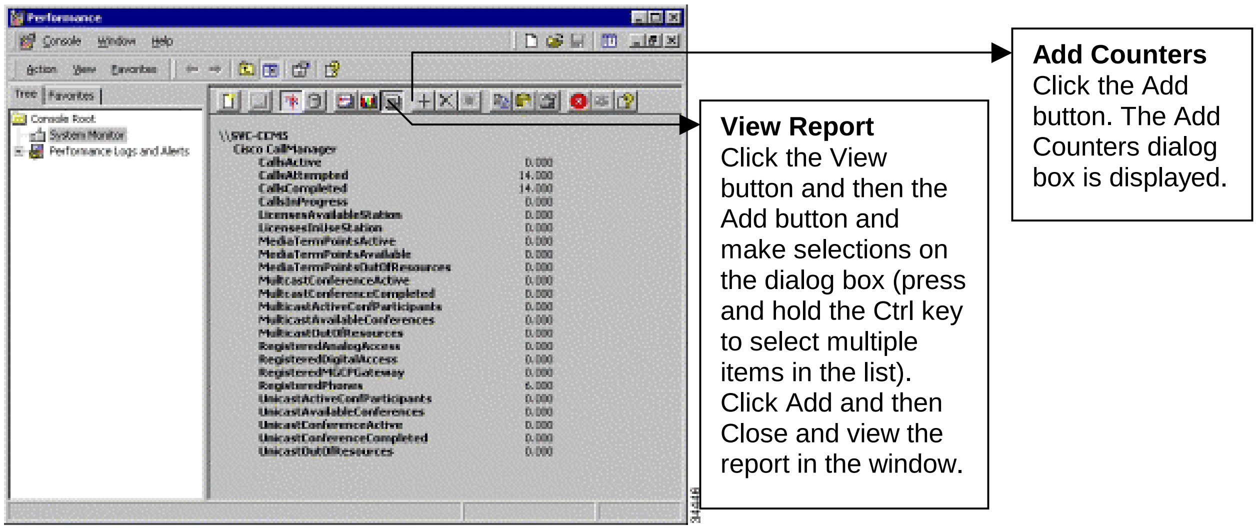
A more detailed explanation of Cisco CallManager Administration is available at the following location:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_callmg/3_0/index.htm

Microsoft Performance

Performance (Monitor) is a Windows 2000 server application that can display the activities and status of your Cisco CallManager system. It reports both general and specific information in real time. You can use Windows 2000 Performance to collect and display system and device statistics for any Cisco CallManager installation. This administrative tool allows you to gain a full understanding of a system without studying the operation of each of its components.

You can use Performance to monitor a variety of system variables in real time. After adding the Cisco CallManager parameters, you can define the terms under which Cisco CallManager will display statistics generated by the system. For example, you can monitor the number of calls in progress at any time, or the number of calls currently passing through a specific gateway. Performance shows both general and Cisco CallManager-specific status information in real-time.



```
\\SWC-CCM5
  Cisco CallManager
      CallsActive                            0.000
      CallsAttempted                        14.000
      CallsCompleted                        14.000
      CallsInProgress                        0.000
      LicensesAvailableStation               0.000
      LicensesInUseStation                   0.000
      MediaTermPointsActive                  0.000
      MediaTermPointsAvailable               0.000
      MediaTermPointsOutOfResources          0.000
      MulticastConferenceActive              0.000
      MulticastConferenceCompleted           0.000
      MulticastActiveConfParticipants        0.000
      MulticastAvailableConferences          0.000
      MulticastOutOfResources                0.000
      RegisteredAnalogAccess                 0.000
      RegisteredDigitalAccess                0.000
      RegisteredMGCPGateway                  0.000
      RegisteredPhones                       6.000
      UnicastActiveConfParticipants          0.000
      UnicastAvailableConferences            0.000
      UnicastConferenceActive                0.000
      UnicastConferenceCompleted             0.000
      UnicastOutOfResources                  0.000
```

View Report
Click the View button and then the Add button and make selections on the dialog box (press and hold the Ctrl key to select multiple items in the list). Click Add and then Close and view the report in the window.

Add Counters
Click the Add button. The Add Counters dialog box is displayed.

Opening Microsoft Performance

To open Performance on the server PC running Cisco CallManager, click **Start** > **Settings** > **Control Panel** > **Administration Tools** > **Performance**.

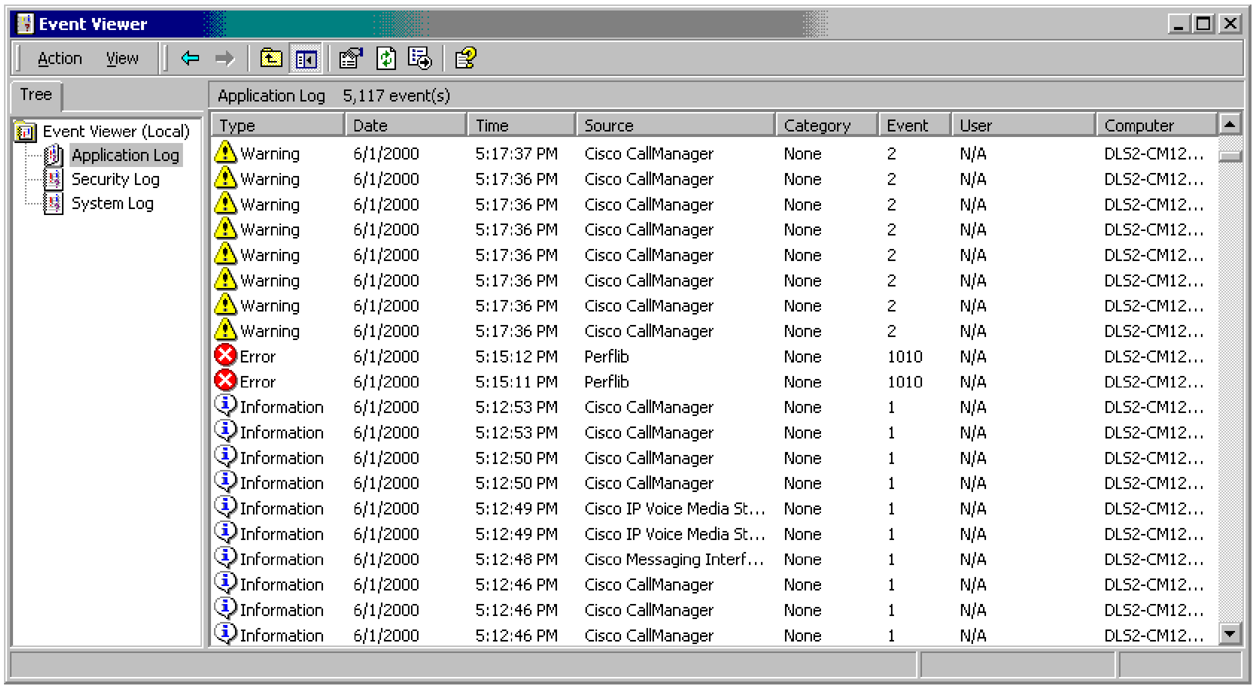
Customizing Performance

The Performance monitor must be customized to view the Cisco CallManager-related parameters that you wish to monitor. Choose the object, counter, and instance you want to include. Please refer to the Remote Serviceability documentation for instructions on how to use objects and counters to customize Microsoft Performance for Cisco CallManager operations.

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_callmg/3_0/service/index.htm

Microsoft Event Viewer

Microsoft Event Viewer is a Windows NT Server application that displays system, security, and application events (including Cisco CallManager) for the Windows NT Server. If a service (including TFTP) cannot read the database (where it gets the trace configuration), it will add errors to the Event Viewer. The Event Viewer is the only place where these types of errors will appear. The following illustration shows the application logs running on a Windows NT Server.



Application Log 5,117 event(s)

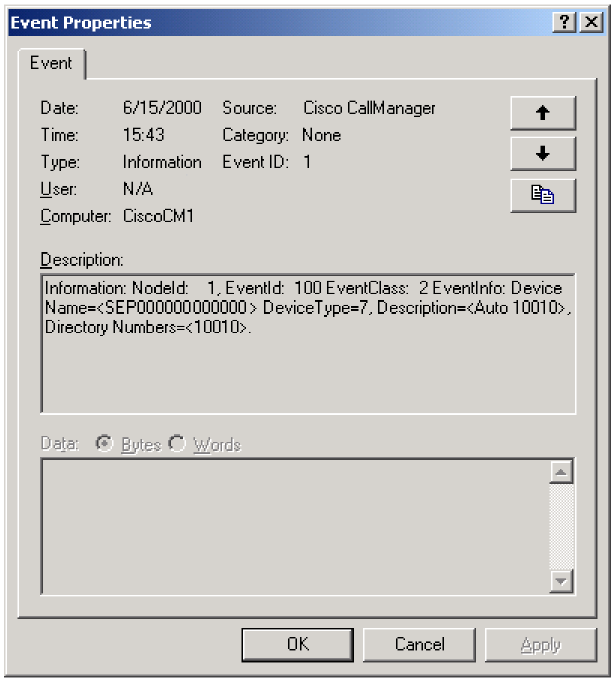
Type	Date	Time	Source	Category	Event	User	Computer
Warning	6/1/2000	5:17:37 PM	Cisco CallManager	None	2	N/A	DLS2-CM12...
Warning	6/1/2000	5:17:36 PM	Cisco CallManager	None	2	N/A	DLS2-CM12...
Warning	6/1/2000	5:17:36 PM	Cisco CallManager	None	2	N/A	DLS2-CM12...
Warning	6/1/2000	5:17:36 PM	Cisco CallManager	None	2	N/A	DLS2-CM12...
Warning	6/1/2000	5:17:36 PM	Cisco CallManager	None	2	N/A	DLS2-CM12...
Warning	6/1/2000	5:17:36 PM	Cisco CallManager	None	2	N/A	DLS2-CM12...
Warning	6/1/2000	5:17:36 PM	Cisco CallManager	None	2	N/A	DLS2-CM12...
Warning	6/1/2000	5:17:36 PM	Cisco CallManager	None	2	N/A	DLS2-CM12...
Error	6/1/2000	5:15:12 PM	Perflib	None	1010	N/A	DLS2-CM12...
Error	6/1/2000	5:15:11 PM	Perflib	None	1010	N/A	DLS2-CM12...
Information	6/1/2000	5:12:53 PM	Cisco CallManager	None	1	N/A	DLS2-CM12...
Information	6/1/2000	5:12:53 PM	Cisco CallManager	None	1	N/A	DLS2-CM12...
Information	6/1/2000	5:12:50 PM	Cisco CallManager	None	1	N/A	DLS2-CM12...
Information	6/1/2000	5:12:50 PM	Cisco CallManager	None	1	N/A	DLS2-CM12...
Information	6/1/2000	5:12:49 PM	Cisco IP Voice Media St...	None	1	N/A	DLS2-CM12...
Information	6/1/2000	5:12:49 PM	Cisco IP Voice Media St...	None	1	N/A	DLS2-CM12...
Information	6/1/2000	5:12:48 PM	Cisco Messaging Interf...	None	1	N/A	DLS2-CM12...
Information	6/1/2000	5:12:46 PM	Cisco CallManager	None	1	N/A	DLS2-CM12...
Information	6/1/2000	5:12:46 PM	Cisco CallManager	None	1	N/A	DLS2-CM12...
Information	6/1/2000	5:12:46 PM	Cisco CallManager	None	1	N/A	DLS2-CM12...

Opening Event Viewer

To open the Event Log on the server PC running Cisco CallManager, click **Start > Settings > Control Panel > Administrative Tools** > **Event Viewer**. The Event Viewer provides error logs for System, Security, and Applications. Cisco CallManager errors are logged under the Application log.

Detailed Information about Events

You can double-click an event in the log to learn more information about the event.



Date: 6/15/2000 Source: Cisco CallManager
Time: 15:43 Category: None
Type: Information Event ID: 1
User: N/A
Computer: CiscoCM1

Description:
Information: NodeId: 1, EventId: 100 EventClass: 2 EventInfo: Device Name=<SEP000000000000> DeviceType=7, Description=<Auto 10010>, Directory Numbers=<10010>.

Data: (● Bytes ○ Words)

SDI Trace

SDI traces are local log files. The IP address, TCP handle, device name or the time stamp can be used when reviewing the SDI trace to monitor the occurrence or the disposition of a request. This device name could be tracked back to the building of the file, which shows the device pool and model. The device pool and model can be tracked back to the building of the configuration file prototype, which will list the network address of the Cisco CallManager(s) and the TCP connection port.

When observing SDI traces, notice that C++ class and routine names are included with most trace lines. Most routines associated with the serving of a particular request include the thread ID in a standard format.

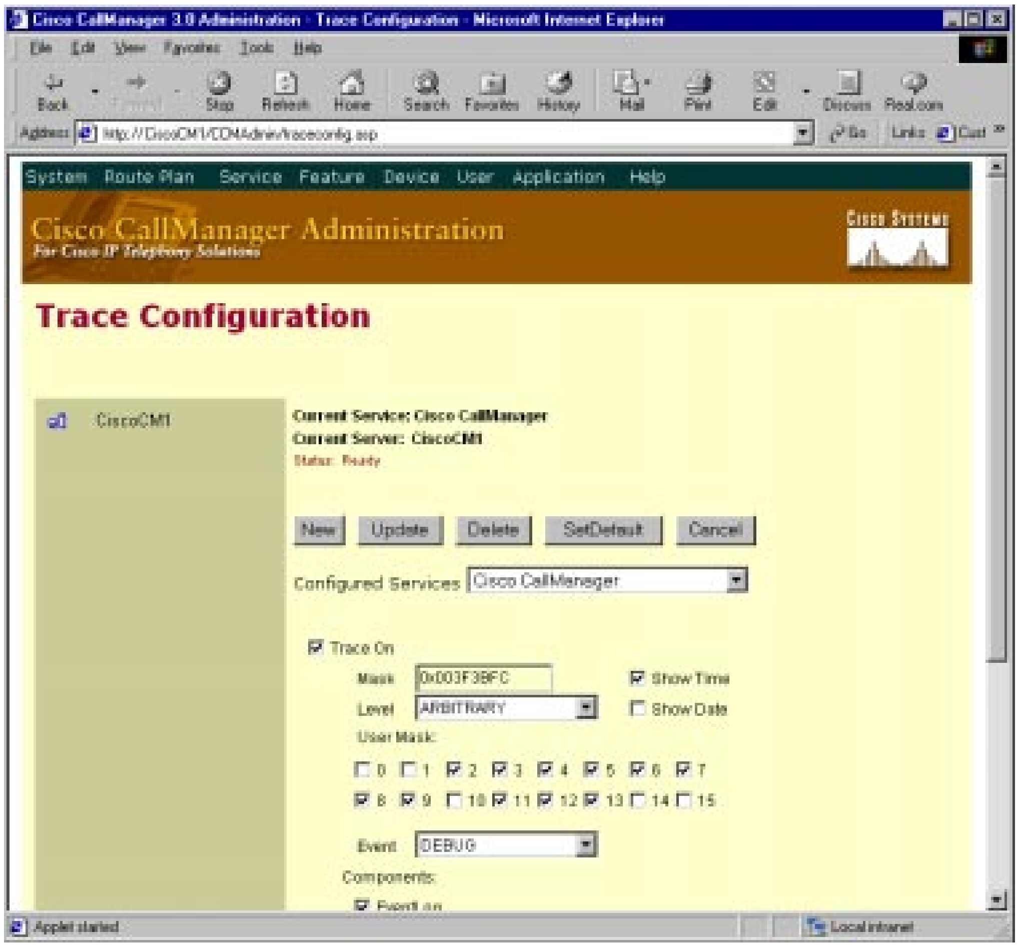
SDI traces will be explained in detail in the case studies in the appendices.

SDI Trace Output

SDI traces generate files (for example, CCM000000000) that store traces of Cisco CallManager activities. These traces provide information about the Cisco CallManager initialization process, registration process, KeepAlive process, call flow, digit analysis, and related devices such as Cisco IP Phones, Gateways, Gatekeepers, and more. This information can help you isolate problems when troubleshooting Cisco CallManager. To properly track the information you need—and only the information you need—it's important to understand how to set the options on the trace configuration interface.

The trace files are stored in the following default location: C:\Program Files\cisco\bin. A new trace file is started each time Cisco CallManager restarts, or when the designated number of lines has been reached.

Following is an illustration of the Cisco CallManager Administration trace configuration interface. You must enable the trace, choose the level on information needed, and check the user mask to obtain the desired level of information.


Cisco CallManager 3.0 Administration - Trace Configuration - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Forward Stop Refresh Home Search Favorites History Mail Print Edit Discuss Real.com
Address http://CiscoCM1/CCMAdmin/traceconfig.asp
System Route Plan Service Feature Device User Application Help
Cisco CallManager Administration
For Cisco IP Telephony Solutions
Cisco Systems
New Update Delete SetDefault Cancel
Applet started
Local intranet

Trace Configuration

CiscoCM1

Current Service: Cisco CallManager
Current Server: CiscoCM1
Status: Ready

Configured Services: Cisco CallManager

☑ Trace On
Mask: 0x003F3BFC ☑ Show Time
Level: ARBITRARY ☐ Show Date
User Mask:
☐ 0 ☐ 1 ☑ 2 ☑ 3 ☑ 4 ☑ 5 ☑ 6 ☑ 7
☑ 8 ☑ 9 ☐ 10 ☑ 11 ☑ 12 ☑ 13 ☐ 14 ☐ 15
Event: DEBUG
Components:
☑ Event on

If the trace is not configured properly, it will generate a large amount of information making it very difficult to isolate problems. The following section explains how to properly configure a useful trace.

Configuring Traces

Traces are composed of user mask flags (also known as bits) and trace levels. Open Cisco CallManager Administration. To turn on tracing, set your trace parameters (including configured service, bits, and so on) in the Service>Trace screen. Refer to the Cisco CallManager documentation for complete information about turning tracing on and off, and for descriptions of the User Masks and Levels for each configured service, and more.

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_callmg/3_0/admin_gd/admin_gd/index.htm

Following are two examples of trace mask bits that would be enabled based on the particular problem.

- For normal message debugging, turn on subsystem bits 5, 6, 7, 8, 11, and 12
- For debugging gateways, turn on subsystem bits 3,4,5,6,7,8,9,11,12,13

Following are two examples of desired trace levels based on the particular problem

- For normal debugging, the trace level should be set to SDI_LEVEL_ARBITRARY
- For normal running system, the trace level should be set to SDI_LEVEL_ERROR

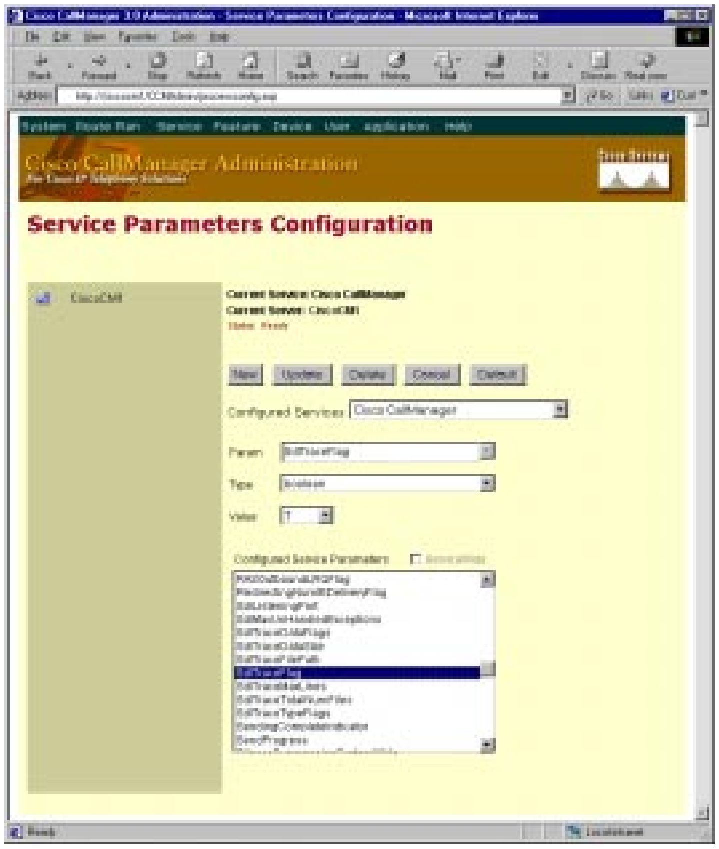
SDL Trace

Cisco engineers use SDL traces to find the cause of an error. You are not expected to understand the information contained in an SDL trace. However, while working with TAC, you may be asked to enable the SDL trace and provide it to the TAC. SDL trace files can be saved to local directories, the Windows NT Event Viewer, and CiscoWorks 2000. To avoid any performance degradation on the server, be sure that after the trace has been captured, you turn off SDL tracing.

SDL trace provides a C interface to trace and alarms. Alarms are used to inform the administrator of unexpected events, such as being unable to access a file, database, Winsock, or being unable to allocate other operating system resources.

Enabling SDL Trace

SDL traces are enabled in the **Service** > **Service Parameter** area in Cisco CallManager Administration. Remember that these traces should be turned on only when requested by a TAC engineer. Note the values chosen to turn on the SDL trace in the following illustration.



Service Parameters Configuration

Current Service: Cisco CallManager
Current Server: CiscoCM1
Status: Ready

Configured Services: Cisco CallManager
Param: SdlTraceFlag
Type: Boolean
Value: T

Configured Service Parameters:
[illegible]
[illegible]
SdlListeningPort
SdlMaxUnHandledExceptions
SdlTraceDataFlags
SdlTraceDataSize
SdlTraceFilePath
SdlTraceFlag
SdlTraceMaxLines
SdlTraceTotalNumFiles
SdlTraceTypeFlags
SendingCompleteIndicator
SendProgress

Once SDL traces are enabled, collect the traces. If the traces are being sent to the local drive, then you can retrieve them in the Cisco\Trace subdirectory. Alternatively, the trace files can be sent to an event log or to CiscoWorks 2000.

SDL flag bits described in the following table are set in the **Service** > **Service Parameters** area in Cisco CallManager Administration. Following are two examples of desired values based on the particular problem.

- The recommended value for normal call debugging is SdlTraceTypeFlags=0x00000b04
- The recommended value for low level debugging or debugging gateways is SdlTraceTypeFlags=0x00004b05

SdlTraceTypeFlags Definitions		
SDLTraceTypeFlag	**Value**	**Definition**
traceLayer1	= 0x00000001	All Layer 1 trace on
TraceDetailLayer1	= 0x00000002	Detail Layer 1 trace on
TraceSdlLinkAdmin	= 0x00000004	Trace inter-Cisco CallManager links within a cluster
traceUnused	= 0x00000008	Not used
traceLayer2	= 0x00000010	All Layer 2 trace on
traceLayer2Interface	= 0x00000020	Layer 2 interface trace on
traceLayer2TCP	= 0x00000040	Layer 2 TCP trace on
TraceDetailLayer2	= 0x00000080	More detail dump of Layer 2 Frames.
traceLayer3	= 0x00000100	All Layer 3 trace on
traceCc	= 0x00000200	All call control trace on
traceMiscPolls	= 0x00000400	Trace miscellaneous polls
traceMisc	= 0x00000800	Miscellaneous trace on (Database signals)
traceMsgtrans	= 0x00001000	Message Translation signals (TranslateIsdnToSdlReq, TranslateIsdnToSdlRes TranslateSdlToIsdnReq, TranslateSdlToIsdnRes)
traceUuie	= 0x00002000	UUIE output trace on
traceGateway	= 0x00004000	Gateway signals

Data bits described in the following table are set in the **Service** > **Service Parameters** area in Cisco CallManager Administration. Following are two examples of desired values based on the particular problem.

- The recommended value for normal system debugging is SdlTraceDataFlags=0x110
- The recommended value when tracking problems with SDL links is 0x13D (non-compacted trace; if a compact trace is desired, bit 0x200 must be set. It can be set in combination with any other bits)

SDLTraceDataFlags Definitions		
SDLTraceDataFlag	**Value**	**Definition**
TraceSdlLinkState	= 0x001	Enable trace of SDL Link Initialization
TraceSdlLowLevel	= 0x002	Enable tracing of low-level SDL events, for example, fileOpen, socket events, and so on
TraceSdlLinkPoll	= 0x004	Enable tracing of SDL Link Poll message
TraceSdlLinkMsg	= 0x008	Enable tracing of SDL Link Message
traceRawData	= 0x010	Enable raw signal data trace on all signals
TraceSdlTagMap	= 0x020	Enable tag mapping
traceCreate	= 0x100	Enable process create and stop traces
TraceNoPrettyPrint	= 0x200	Disable pretty printing of trace files

Disk Space Warning

IMPORTANT: Be advised that information obtained from this interface could be very detailed, and therefore consume a large amount of disk space. For this reason, we advise you to turn on the trace file for a specific amount of time, then review the information and turn off the trace.

Sniffer Trace

A Sniffer is a software application that monitors IP traffic on a network and provides information in the form of a trace. Sniffer traces provide information about the quantity and type of network traffic on your network. TCP/IP or UDP packets are protocols utilized by Cisco CallManager and endpoint devices such as phones and gateways. Sniffer traces can also help you identify high levels of broadcast traffic that could result in voice audio problems or dropped calls. Common Sniffer applications include Network Associates SnifferPro, Hewlett Packard Internet Advisor, and W&G Domino. Domino offers sniffing hardware and software solutions and a network analyzer. If you want to use Domino, we recommend using the analysis software to evaluate a captured sniffer file (such as from the SnifferPro application).

Sniffer Trace Applications

Use the following links to learn more about some available sniffer trace applications. Any sniffer application will work with Cisco CallManager.

- Network Associates SnifferPro: http://www.sniffer.com/
- Hewlett Packard Network Analyzer: http://www.hp.com/rnd/products/netman/netmgt.htm
- W&G Domino Analyzer: http://www.wwgsolutions.com/products/domino/domino.html

Call Detail Records (CDR) and Call Management Records (CMR)

Call Detail Record (CDR) is a reporting option that logs every call made (or attempted) from any Cisco IP Phone. There are two kinds of CDRs—basic CDRs and Diagnostic CDRs, or CMRs. Once enabled, you can open CDRs or Diagnostic CDRs (CMRs) in the SQL Server Enterprise Manager. CDR files are saved in a SQL database that can be exported to nearly any application, including Microsoft Access or Excel.

CDR records contain information needed to generate billing records. In a distributed environment, all CDR data is collected in a central location, or a set of locations. The failure of a Cisco CallManager node does not make the CDR data associated with that node unavailable, since the data is no longer stored on the Cisco CallManager disk as a flat file, but is instead stored in a central database in tables.

If the Cisco CallManager fails before any records are written, then no record of the call will exist. This means that no record will be written for calls that are active on a given Cisco CallManager when it fails before the calls terminate.

Refer to the Appendix in the back of this book for detailed information about CDRs and CMRs. The information provided includes:

- Reading and Writing Records
- Known Issues
- List of record types generated

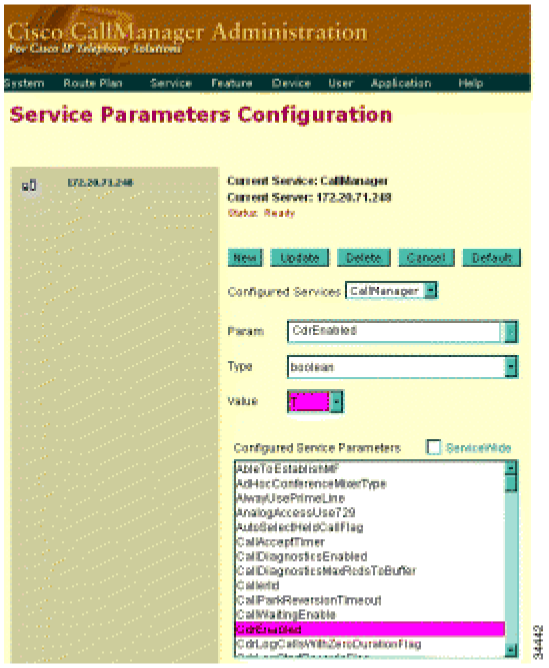
- List of fields contained in each record and a description of what that field represents
- Description of the types of calls logged, and the fields logged with each of them
- List of cause codes that may appear in the CDR records

Enabling or Disabling CDRs

CDR record creation is disabled by default when the system is installed. If you wish to have CDR data, you must enable CDRs in the **Service > Service Parameters** area of Cisco CallManager Administration. CDR processing can be enabled and disabled at any time while the system is in operation. You do not need to restart Cisco CallManager for the enabling or disabling of CDRs to take effect. The system will respond to all changes within a few seconds. CMR or diagnostic data is enabled separately from CDR data. CMR data will not be generated unless both CDRs and Call Diagnostics are enabled, but CDR data may be generated and logged without CMR data.

Use the following steps to enable CDRs.

1. Open Cisco CallManager Administration.
2. Select **Service** > **Service Parameters**.
3. Select the IP address of your Cisco CallManager installation.
4. From the list of Parameters, select **CDREnabled**.
5. Define type as **boolean**.
6. Select T for True.



Service Parameters Configuration

172.20.71.248

Current Service: CallManager
Current Server: 172.20.71.248
Status: Ready

Configured Services: CallManager

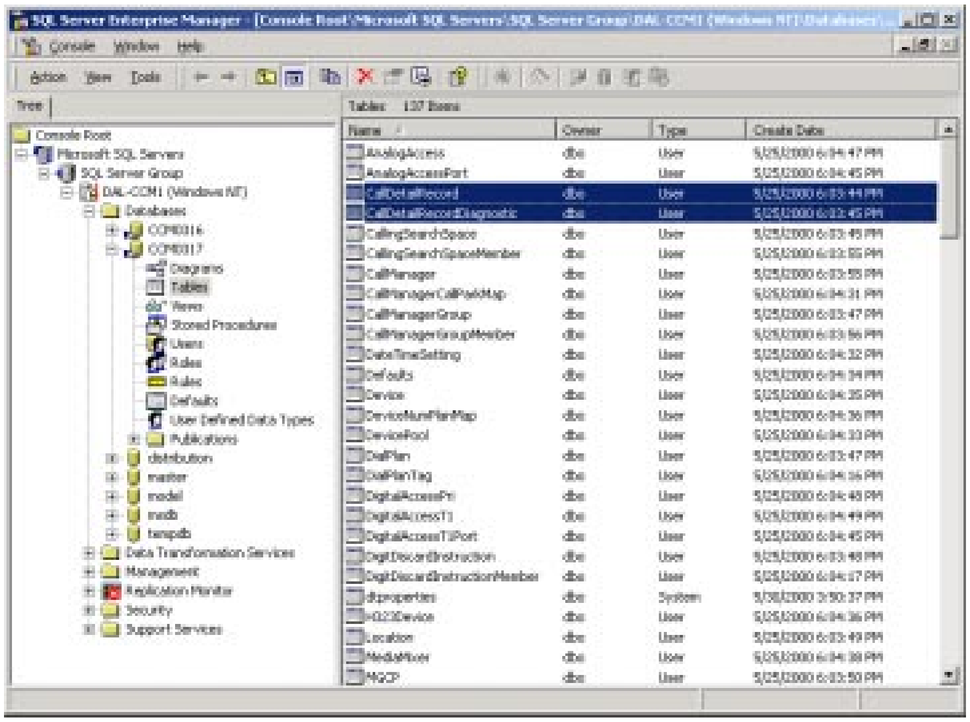
Param: CdrEnabled
Type: boolean
Value: T

Configured Service Parameters ☐ ServiceWide

AbleToEstablishMF
AdHocConferenceMixerType
AlwaysUsePrimeLine
AnalogAccessUse729
AutoSelectHeldCallFlag
CallAcceptTimer
CallDiagnosticsEnabled
CallDiagnosticsMaxRcdsToBuffer
CallerId
CallParkReversionTimeout
CallWaitingEnable
CdrEnabled
CdrLogCallsWithZeroDurationFlag

7. Update.
   Result: Call Detail Records will start logging immediately.

Caution: Tracing voice connectivity requires that CDR logging be enabled on every Cisco CallManager installation in a cluster.



Name	Owner	Type	Create Date
AnalogAccess	dbo	User	5/25/2000 6:04:47 PM
AnalogAccessPort	dbo	User	5/25/2000 6:04:45 PM
CallDetailRecord	dbo	User	5/25/2000 6:03:44 PM
CallDetailRecordDiagnostic	dbo	User	5/25/2000 6:03:45 PM
CallingSearchSpace	dbo	User	5/25/2000 6:03:45 PM
CallingSearchSpaceMember	dbo	User	5/25/2000 6:03:55 PM
CallManager	dbo	User	5/25/2000 6:03:55 PM
CallManagerCallParkMap	dbo	User	5/25/2000 6:04:31 PM
CallManagerGroup	dbo	User	5/25/2000 6:03:47 PM
CallManagerGroupMember	dbo	User	5/25/2000 6:03:56 PM
DateTimeSetting	dbo	User	5/25/2000 6:04:32 PM
Defaults	dbo	User	5/25/2000 6:04:34 PM
Device	dbo	User	5/25/2000 6:04:35 PM
DeviceNumPlanMap	dbo	User	5/25/2000 6:04:36 PM
DevicePool	dbo	User	5/25/2000 6:04:33 PM
DialPlan	dbo	User	5/25/2000 6:03:47 PM
DialPlanTag	dbo	User	5/25/2000 6:04:16 PM
DigitalAccessPri	dbo	User	5/25/2000 6:04:48 PM
DigitalAccessT1	dbo	User	5/25/2000 6:04:49 PM
DigitalAccessT1Port	dbo	User	5/25/2000 6:04:45 PM
DigitDiscardInstruction	dbo	User	5/25/2000 6:03:48 PM
DigitDiscardInstructionMember	dbo	User	5/25/2000 6:04:17 PM
dtproperties	dbo	System	5/30/2000 3:50:37 PM
H323Device	dbo	User	5/25/2000 6:04:36 PM
Location	dbo	User	5/25/2000 6:03:49 PM
MediaMixer	dbo	User	5/25/2000 6:04:38 PM
MGCP	dbo	User	5/25/2000 6:03:50 PM

CDRs

CDRs provide basic information that can help you understand the more detailed information contained in SDI traces. Basic CDRs provide information such as the calling number, called number, originating IP address, destination IP address, call duration, and so on. CDRs can help you troubleshoot phone problems. For example, if a user reports a problem with a call occurring at a specific time, you can consult the CDRs that occurred around the time indicated to learn additional information about that call and others. CDRs are commonly used for billing.

Diagnostic CDRs (Also Known As CMRs)

Diagnostic CDRs provide detailed call information, such as the number of packets sent, received, and lost, the amount of jitter and latency, and so on. This level of detail can provide explanations for some problems, such as one-way audio. For example, a one-way audio problem is indicated if a packet size of 10,000 is sent, but the received size is only 10.

Problem Categories

This section addresses some common problem categories that may occur with Cisco CallManager and related devices. Each problem category provides suggestions for the troubleshooting tools you should use to help isolate the problem. This document provides general categories of potential problems and suggestions about how to troubleshoot those problems. It does not provide an exhaustive list of problems and resolutions. If you encounter a problem that cannot be resolved using the tools and utilities described in this document, consult the Cisco Technical Assistance Center (TAC) for assistance. Be sure to have available the Cisco CallManager Administration Details, plus the diagnostic information (traces, etc.) you have gathered up to the point of calling the TAC.

Voice Quality

Voice quality issues include lost or distorted audio during phone calls. Common problems can be breaks in the sound which cause the audio to be intermittent (like broken words), or the presence of odd noises that distort the audio, such as echo, or effects that cause spoken words to sound watery or robotic. One-way audio, that is, a conversation between two people where only one person can hear anything, is not actually a voice quality issue, but will be discussed later in this section.

One or more of the following components could cause audio problems:

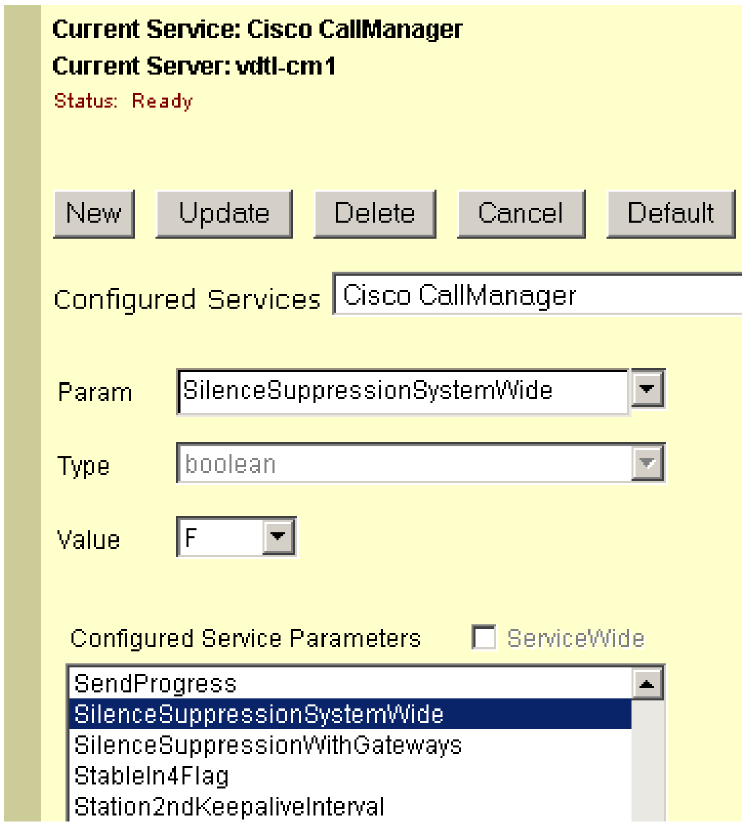
- Gateway
- Phone
- Network

To properly troubleshoot voice quality issues, you must consider the infrastructure and all the devices for drops and delays.

Lost or Distorted Audio

One of the most common problems encountered is a breaking up of audio (often described as garbled speech, or a loss of syllables within a word or sentence). There are two common causes for this: packet loss and/or jitter. Packet loss means that audio packets do not arrive at their destination because they were dropped or arrived too late to be useful. Jitter is the variation in the arrival times of packets. In the ideal situation, all VoIP packets from one phone to another would arrive exactly at a rate of 1 every 20 ms. Notice that this does not mention how *long* it takes for a packet to get from point A to point B, simply the *variation* in the arrival times. There are many sources of variable delay in a real network. Some of these cannot be controlled, and some can. Variable delay cannot be eliminated entirely in a packetized voice network. Digital Signal Processors (DSPs) on phones and other voice-capable devices are designed to buffer some of the audio, in anticipation of variable delay. This “dejittering” is done only when the audio packet has reached its destination and is now ready to be put into a conventional audio stream (to be played out into the user’s ear to be sent to the PSTN via a digital PCM stream). The Cisco IP Phone 7960 can buffer as much as one second of voice samples. The jitter buffer is adaptive, meaning if a burst of packets is received, the Cisco IP Phone 7960 can play them out in an attempt to control the jitter. The network administrator needs to minimize the variation between packet arrival times by applying quality-of-service (QoS) and other measures in advance (especially if calls cross a wide-area network).

When faced with a lost or distorted audio problem, the first thing to do is to try to isolate the path of the audio. Try to identify each network device (switches and routers) in the path of the call's audio stream. Keep in mind that the audio may be between two phones, or between a phone and a gateway, or it could have multiple legs (from a phone to a transcoding device and from there to another phone). Try to identify if the problem occurs only between two sites, only through a certain gateway, on a certain subnet, and so on. This will help narrow down which devices you need to look at more carefully. Next, if is often best to disable silence suppression (also known as Voice Activation Detection or VAD) if this hasn't been done already. This mechanism does save bandwidth by not transmitting any audio when there is silence, but may cause noticeable clipping at the beginning of words that may be unacceptable. You can disable this in Cisco CallManager Administration, under **Service** > **Service Parameters**. From there, select the server and the Cisco CallManager service. Then set SilenceSuppressionSystemWide to "F" (alternatively you can set SilenceSuppressionWithGateways to "F", but this does not apply to H.323 gateways or MGCP gateways). When in doubt, turn both off by selecting the Value F for each.


Current Service: Cisco CallManager
Current Server: vdtl-cm1
Status: Ready

Configured Services: Cisco CallManager
Param: SilenceSuppressionSystemWide
Type: boolean
Value: F

Configured Service Parameters ☐ ServiceWide
SendProgress
SilenceSuppressionSystemWide
SilenceSuppressionWithGateways
StableIn4Flag
Station2ndKeepaliveInterval

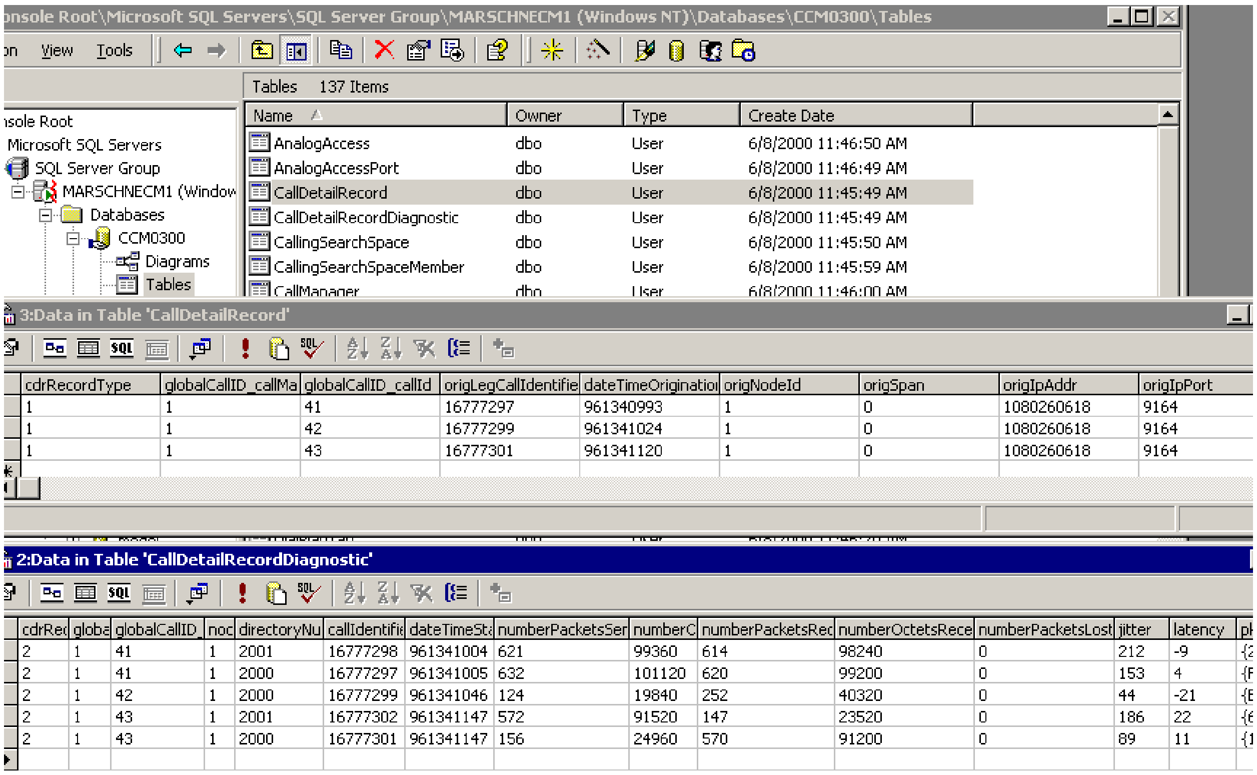
If a network analyzer is available, then a monitored call between two phones should have 50 packets per second (or 1 packet every 20 ms) when silence suppression is disabled. With proper filtering, it should be possible to identify if packets are being lost or delayed excessively.

Remember that delay by itself won't cause clipping, only variable delay will. Notice in the table below, which represents a perfect trace, the arrival times between the audio packets (which will have an RTP header), will be 20 ms. In a poor quality call (such as a call with a lot of jitter), the arrival times would vary greatly.

A Perfect Trace		
Packet Number	**Time – absolute (ms)**	**Time – delta (ms)**
1	0	
2	0.02	20
3	0.04	20
4	0.06	20
5	0.08	20

Placing the packet analyzer into various points in the network will help narrow down where the delay is coming from. If no analyzer is available, other methods will be required. It is important to examine interface statistics of each device in the path of the audio. Another tool for tracking calls with poor voice quality is the Diagnostic Call Detail Records (CDRs). See the CDR section in the Problem Categories section above, or Appendix D for more information about CDRs.

Then values for jitter and latency can be retrieved for all calls (but only **after** the call has terminated). Following is a sample Diagnostic CDR (CallDetailRecordDiagnostic is the actual table name). The number of packets sent, receive, lost, jitter, and latency are all recorded. The globalCallID value can be used to find the call in the regular CDR table so that the disconnect cause and other information can be obtained. The diagram below shows both tables open. Notice that in the Diagnostic CDR, every device that can possibly report this information is included. So if the problem is between two Cisco IP Phones, we see two table entries per call. If we have a call through a Cisco IOS Gateway, for example, we only see the diagnostic information from the Cisco IP Phone, not the gateway because there is no mechanism for it to notify the SQL database with this information.


onsole Root\Microsoft SQL Servers\SQL Server Group\MARSCHNECM1 (Windows NT)\Databases\CCM0300\Tables

on View Tools

Tables 137 Items

nsole Root
Microsoft SQL Servers
SQL Server Group
MARSCHNECM1 (Window
Databases
CCM0300
Diagrams
Tables

Name	Owner	Type	Create Date
AnalogAccess	dbo	User	6/8/2000 11:46:50 AM
AnalogAccessPort	dbo	User	6/8/2000 11:46:49 AM
CallDetailRecord	dbo	User	6/8/2000 11:45:49 AM
CallDetailRecordDiagnostic	dbo	User	6/8/2000 11:45:49 AM
CallingSearchSpace	dbo	User	6/8/2000 11:45:50 AM
CallingSearchSpaceMember	dbo	User	6/8/2000 11:45:59 AM
CallManager	dbo	User	6/8/2000 11:46:00 AM

3:Data in Table 'CallDetailRecord'

cdrRecordType	globalCallID_callMa	globalCallID_callId	origLegCallIdentifie	dateTimeOriginatio	origNodeId	origSpan	origIpAddr	origIpPort
1	1	41	16777297	961340993	1	0	1080260618	9164
1	1	42	16777299	961341024	1	0	1080260618	9164
1	1	43	16777301	961341120	1	0	1080260618	9164

2:Data in Table 'CallDetailRecordDiagnostic'

cdrRe	globa	globalCallID_	noc	directoryNu	callIdentifi	dateTimeSt	numberPacketsSen	numberC	numberPacketsRec	numberOctetsRece	numberPacketsLost	jitter	latency	pk
2	1	41	1	2001	16777298	961341004	621	99360	614	98240	0	212	-9	{2
2	1	41	1	2000	16777297	961341005	632	101120	620	99200	0	153	4	{F
2	1	42	1	2000	16777299	961341046	124	19840	252	40320	0	44	-21	{E
2	1	43	1	2001	16777302	961341147	572	91520	147	23520	0	186	22	{6
2	1	43	1	2000	16777301	961341147	156	24960	570	91200	0	89	11	{1

i Button Help

The Cisco IP Phone 7960 provides another tool for diagnosing possible audio problems. On an active call, you can press the ⓘ button twice rapidly and the phone will display an information screen that contains packet receive and transmit statistics, as well as average and maximum jitter counters. Note that on this screen, jitter is the average of the last 5 packets that arrived; the maximum jitter is the high-water mark for the average jitter.

The most common sources for delay and packet loss are devices where a higher speed interface feeds into a lower speed interface. For example: a router may have a 100 Mb fast Ethernet interface connected to the LAN and a slow frame-relay, for example, connected to the WAN. If the poor quality occurs only when communicating to the remote site (only the remote site may be reporting the poor voice quality while in the other direction everything appears to be fine), then the most likely causes of the problem include:

- The router has not been properly configured to give the voice traffic priority over the data traffic
- There are too many calls active for the WAN to support (that is, there is no call admission control to restrict the number of calls that can be placed)
- There are physical port errors
- There is congestion in the WAN itself

On the LAN, the most common problems are physical-level errors (such as CRC errors) caused by faulty cables, interfaces, or by incorrectly configured devices (such as a port speed or duplex mismatch). Make sure that the traffic is not crossing any shared-media device, such as a hub. There could also be situations where the traffic is taking a slower path through the network than expected. If QoS has been configured correctly, then the possibility exists that there is no call admission control. Depending on your topology, this can be accomplished through the use of Locations in Cisco CallManager Administration configuration, or by using a Cisco IOS router as a gatekeeper. In any case, you should always know how many calls could be supported across your WAN. If possible, test this by disabling silence suppression as described earlier, then place calls between the two sites. Do not place the calls on hold or on mute, since this will stop packets from being transmitted. With the maximum number of calls across the WAN, the calls should all have acceptable quality. Test to make sure that a fast busy is returned when trying to make one more call.

Crackling

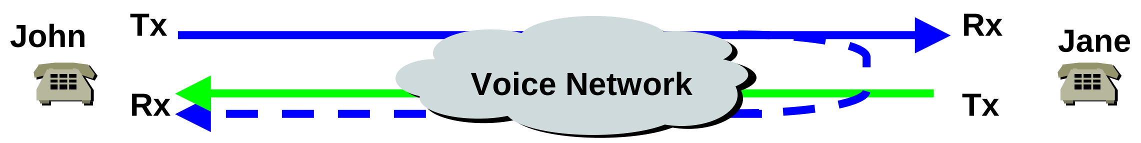
Another “poor quality” symptom may be a crackling, which is sometimes caused by a defective power supply or some kind of strong electrical interference close to the phone. Try swapping the power supply and moving the phone around.

Check Your Loads

You should also always check the phones and gateways to ensure the latest software loads are in use. When in doubt, check CCO (Cisco Connection Online at www.cisco.com) for the latest software loads, new patches, or release notes relating to the problem.

Echo

Echo (also known as “talker echo”) occurs when a talker’s speech energy, transmitted down the primary signal path, is coupled into the receive path from the far end. The talker then hears his or her own voice, delayed by the total echo path delay time.



In the diagram above, John’s voice (in blue) is being reflected back. This can happen but go unnoticed in a traditional voice network because the delay is so low. To the user, it sounds more like a side-tone than an echo. In a VoIP network, it will always be noticeable, since packetization and compression always contribute enough delay. The important thing to remember is that the cause of the echo is always with analog components and wiring. For instance, IP packets cannot simply turn around and go back to the source at a lower audio level. The same is impossible on digital T1/E1 circuits. So on a call from one Cisco IP Phone to another, there should never be any problem. The only exception may be if one party is using a speakerphone that has the volume set too high or some other situation where an audio loop is created.

When troubleshooting echo problems, make sure that the phones that are being tested or examined are not using the speakerphone, and that they have the headset volume to reasonable levels (start with 50% of the maximum audio level). Most of the time, the problems will occur when attaching to the PSTN by way of a digital or analog gateway. Cisco IP Phone users may complain that they hear their own voice being reflected back to them. Now, although the true source of the problem is almost always at the far end, it is nearly always impossible to change anything in the PSTN. So the first step is to determine which gateway is being used. If a digital gateway is in use, then it may be possible to add additional padding in the transmit direction (towards the PSTN), in the hopes that the lower signal strength will yield less reflected energy. Additionally, you can adjust the receive level so that any reflected audio is reduced even further. It is very important to remember to make small adjustments at a time. Too much attenuation of the signal will make the audio impossible to hear on both sides. Alternatively, you can contact the carrier and request to have the lines checked. On a typical T1/PRI circuit in North America, the input signal should be –15 dB. If the signal level is much higher (-5 dB, for example), then echo will be the likely result.

A log should be kept of all calls that experience echo. The time of the problem, the source phone number, and the number called should all be recorded. Gateways have a fixed time of 16 ms of echo cancellation. If the delay in the reflected audio is longer than this, the echo chancellor will be unable to work properly. This should not be an issue for local calls, and long distance calls should have external echo chancellors built into the network at the Central Office. This is one of the reasons why it is important to note the external phone number of a call that experiences echo.

Check Your Loads

Gateway and phone loads should be verified. Check CCO (Cisco Connection Online at www.cisco.com) for the latest software loads, new patches, or release notes relating to the problem.

One-Way Audio or No Audio

One-way audio occurs when one person cannot hear another person during a call. This can be caused by an improperly configured Cisco IOS Gateway, a firewall, or a routing or default gateway problem, among other things.

There are a number of causes for one-way audio or no audio during a call. The most common cause is an improperly configured device. For instance, Cisco CallManager handles the call setup for a Cisco IP Phone. The actual audio stream occurs between the two Cisco IP Phones (or between the Cisco IP Phone and a gateway). So it is entirely possible that the Cisco CallManager is able to signal to a destination phone (making it ring) when the phone originating the call does not have an IP route to the destination phone. A common cause for this is when the default gateway in the phone is improperly configured manually or on the DHCP server.

If a call consistently has one-way audio, take a PC that is on the same subnet as the phone and has the same default gateway and try to ping the destination Cisco IP Phone. Then take a PC that is on the same subnet as the destination phone (with the same default gateway as the destination phone) and ping the source phone. Both of those tests should work. Other things can affect the audio traffic include a firewall or packet filter (such as access lists on a router) that may be blocking the audio in one or both directions. If the one-way audio occurs only through a voice-enabled Cisco IOS Gateway, then check the configuration carefully. IP routing must be enabled (look at the configuration to make sure that “no ip routing” is not found near the beginning of the configuration). Also, make sure that if you’re using RTP header compression to save bandwidth across the WAN, that it is enabled on each router carrying voice traffic that attaches to the WAN circuit. There should not be a situation where the RTP header is compressed on one end but cannot be de-compressed on the other side of the WAN. A sniffer is a very useful tool when troubleshooting one-way audio problems, since you can then verify that the phone or gateway is actually sending or receiving packets. Diagnostic Call Detail Records (CDRs) are useful for determining if a call is experiencing one-way audio since they log transmitted and received packets (see “Lost or Distorted Audio” section). You can also press the ⓘ button twice quickly on a Cisco IP Phone 7960 during an active call to view details about transmitted and received packets.

Note: When a call is muted, no packets will be transmitted from the phone that has pressed the mute button. The Hold button stops the audio stream, so no packets are sent in either direction. When the Hold button is released, all the packet counters are reset. Remember that Silence Suppression must be disabled on both devices for the TX and RX counters to stay equal. Disabling Silence Suppression system-wide will not affect Cisco IOS Gateways.

MTP and One-Way Audio

If you are using Media Termination Point (MTP) in a call (to support supplementary services such as hold and transfer with H.323 devices that do not support H.323 version 2), check to see if the MTP allocated is working correctly. Cisco IOS routers support H.323 version 2 beginning in release 11.3(9)NA and 12.0(3)T. Starting with Cisco IOS release 12.0(7)T, the optional H.323 Open/Close LogicalChannel is supported, so that software-based MTP is no longer required for supplementary services.

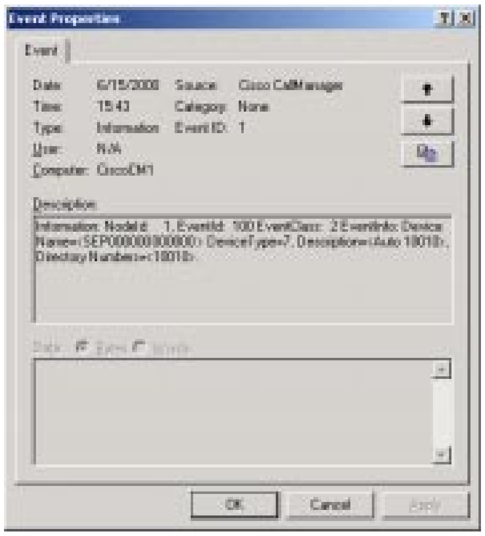
The MTP device, as well as Conference Bridge and Transcoder, will bridge two or more audio streams. If the MTP, Conference Bridge, or Transcoder is not working properly, one-way audio or audio loss might be experienced. Shut down MTP to find out if MTP is causing the problem.

Phone Resets

Phones will power cycle or reset for two reasons: 1) TCP failure connecting to Cisco CallManager, or 2) failure to receive an acknowledgement to the phone’s KeepAlive messages.

Steps for troubleshooting phone resets:

1. Check the phones and gateways to ensure that you are using the latest software loads.
2. Check CCO (Cisco Connection Online at www.cisco.com) for the latest software loads, new patches, or release notes relating to the problem.
3. Check the Event Viewer for instances of phone(s) resetting. Phone resets are considered Information events, as shown in the following illustration.



Date: 6/15/2000 Source: Cisco CallManager
Time: 15:43 Category: None
Type: Information Event ID: 1
User: N/A
Computer: CiscoCM1

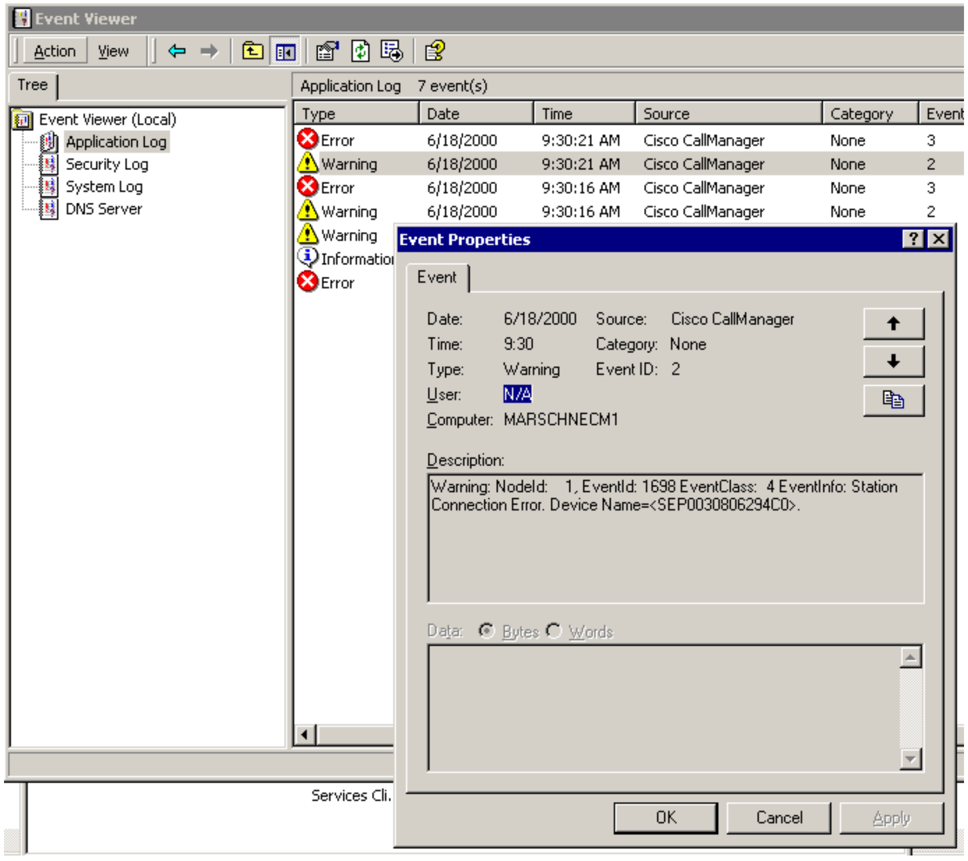
Description:
Information: NodeId: 1, EventId: 100 EventClass: 2 EventInfo: Device Name=<SEP000000000000> DeviceType=7, Description=<Auto 18010>, Directory Numbers=<18010>

4. Look for these and any errors that may have occurred around the time that the phone(s) reset.
5. Start an SDI trace and try to isolate the problem by identifying any common characteristics in the phones that are resetting. For example, check whether they are all located on the same subnet, same VLAN, and so on. Look at the trace and determine:
   - If the resets occur during a call or happen intermittently
   - If there any similarities of phone model – Cisco IP Phone 7960, Cisco IP Phone 30VIP, etc.
6. Start a Sniffer trace on a phone that frequently resets. After it has reset, look at the trace to determine if there are any TCP retries occurring. If so, this indicates a network problem. The trace may show some consistencies in the resets, such as the phone resetting every seven days. This might indicate DHCP lease expiration every seven days (this value is user-configurable; could be every two minutes, etc.).

Dropped Calls

Dropped calls occur when a call is prematurely terminated. You can use CDRs to determine the possible cause of dropped calls, particularly if the problem is intermittent. Dropped calls can be the result of a phone or gateway resetting (see above section) or a circuit problem, such as incorrect PRI configuration or error.

The first step is to determine if this problem is isolated to one phone or a group of phones. Perhaps the affected phones are all on a particular subnet or location. The next step is to check the Event Viewer for phone or gateway resets.



Application Log 7 event(s)

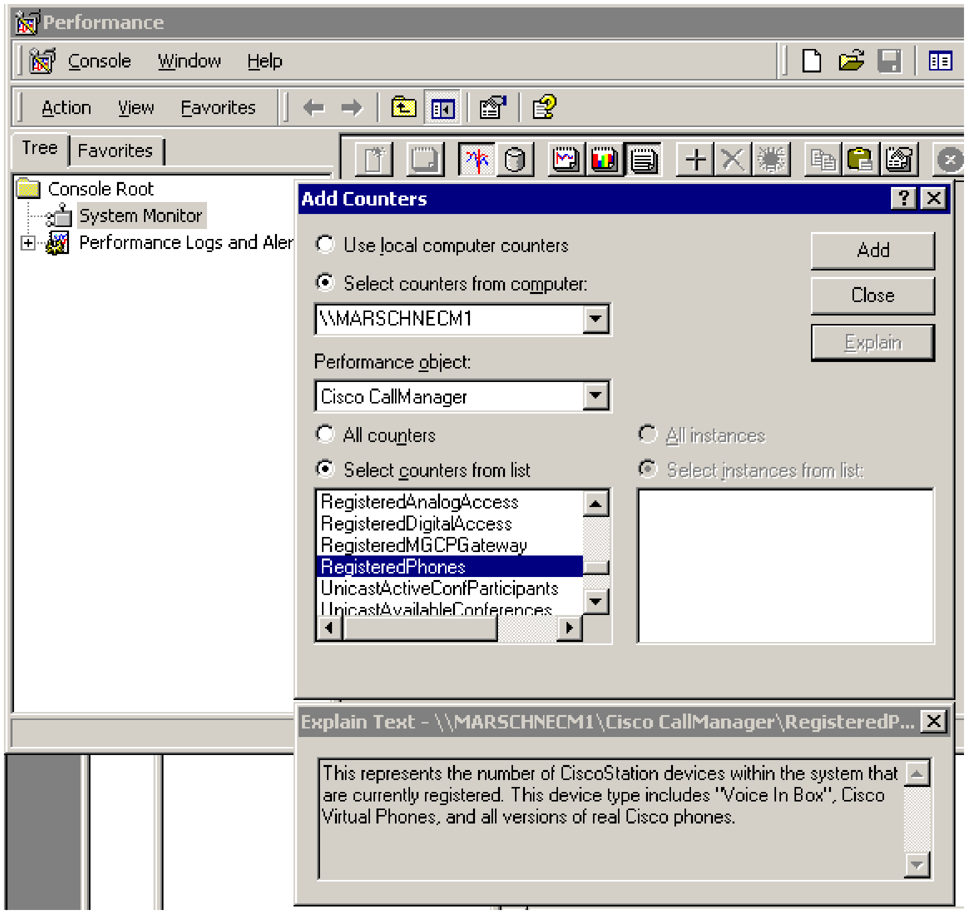
Type	Date	Time	Source	Category	Event
Error	6/18/2000	9:30:21 AM	Cisco CallManager	None	3
Warning	6/18/2000	9:30:21 AM	Cisco CallManager	None	2
Error	6/18/2000	9:30:16 AM	Cisco CallManager	None	3
Warning	6/18/2000	9:30:16 AM	Cisco CallManager	None	2
Warning					
Informatio					
Error					

Date: 6/18/2000 Source: Cisco CallManager
Time: 9:30 Category: None
Type: Warning Event ID: 2
User: N/A
Computer: MARSCHNECM1

Description:

Warning: NodeId: 1, EventId: 1698 EventClass: 4 EventInfo: Station Connection Error. Device Name=<SEP0030806294C0>.

There should be one Warning and one Error message for each phone that resets. In this case, the problem is often that the phone cannot keep its TCP connection to the Cisco CallManager alive, so the Cisco CallManager resets the connection. This may be because a phone was turned off or there may be a problem in the network. If this is an intermittent problem, it may be useful to use Microsoft Performance to record phone registrations.

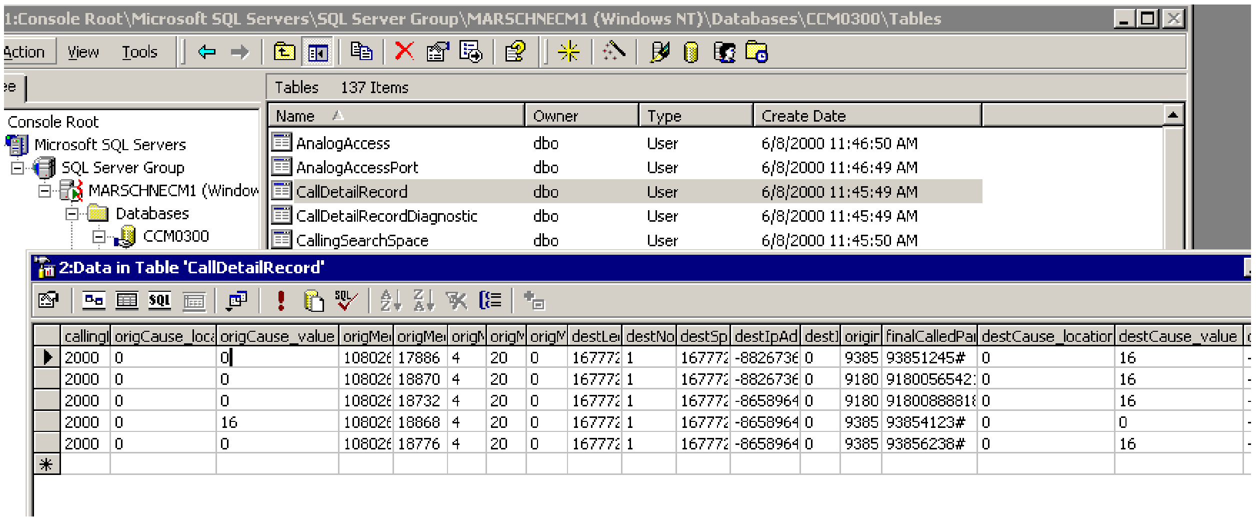


Use local computer counters
(●) Select counters from computer:
\\MARSCHNECM1
Performance object:
Cisco CallManager
All counters
(●) Select counters from list
RegisteredAnalogAccess
RegisteredDigitalAccess
RegisteredMGCPGateway
RegisteredPhones
UnicastActiveConfParticipants
UnicastAvailableConferences

Explain Text - \\MARSCHNECM1\Cisco CallManager\RegisteredP...

This represents the number of CiscoStation devices within the system that are currently registered. This device type includes "Voice In Box", Cisco Virtual Phones, and all versions of real Cisco phones.

If the problem seems to be occurring only through a certain gateway, such as a Cisco Access DT-24+, then the best course of action is to enable tracing and/or view the Call Detail Records (CDR). The CDR files will give a Cause Of Termination (COT) that may help determine the cause of the problem. See the CDR section in the Problem Categories section above, or Appendix D for more information about CDRs.


1:Console Root\Microsoft SQL Servers\SQL Server Group\MARSCHNECM1 (Windows NT)\Databases\CCM0300\Tables
Action View Tools
Console Root
Microsoft SQL Servers
SQL Server Group
MARSCHNECM1 (Window
Databases
CCM0300
Tables 137 Items
2:Data in Table 'CallDetailRecord'

Name	Owner	Type	Create Date
AnalogAccess	dbo	User	6/8/2000 11:46:50 AM
AnalogAccessPort	dbo	User	6/8/2000 11:46:49 AM
CallDetailRecord	dbo	User	6/8/2000 11:45:49 AM
CallDetailRecordDiagnostic	dbo	User	6/8/2000 11:45:49 AM
CallingSearchSpace	dbo	User	6/8/2000 11:45:50 AM

callingI	origCause_loc	origCause_value	origMe	origMe	origN	origN	origM	destLe	destNo	destSp	destIpAd	destI	origir	finalCalledPa	destCause_locatior	destCause_value	c
2000	0	0	108026	17886	4	20	0	167772	1	167772	-8826736	0	9385	93851245#	0	16	-
2000	0	0	108026	18870	4	20	0	167772	1	167772	-8826736	0	9180	918005654	0	16	-
2000	0	0	108026	18732	4	20	0	167772	1	167772	-8658964	0	9180	918008888	0	16	-
2000	0	16	108026	18868	4	20	0	167772	1	167772	-8658964	0	9385	93854123#	0	0	-
2000	0	0	108026	18776	4	20	0	167772	1	167772	-8658964	0	9385	93856238#	0	16	-

The disconnect cause values (origCause_value and destCause_value — depending on which side hung up the call), map to Q.931 disconnect cause codes (in decimal) that can be found at http://www.cisco.com/univercd/cc/td/doc/product/software/ios113ed/dbook/disdn.htm. In the example above, cause 16 refers to a normal call clearing. If the call is going out a gateway to the PSTN, then the CDR can be used to determine which side is hanging up the call. Much of the same information can be obtained by enabling tracing on the Cisco CallManager. Use the trace tool only as a last resort or if the network is not yet in production.

Check Your Loads

As with any problem, check the phone and gateway loads and CCO (Cisco Connection Online at www.cisco.com) for the latest software loads, new patches, or release notes relating to the problem.

Cisco CallManager Feature Issues

Problems may occur with features that are used in conjunction with Cisco CallManager, such as Conference Bridge or Media Termination Point. Some feature problems can be caused by configuration errors or lack of resources. For example, users may not be able to conference calls if the specified number of Ad Hoc conference resources has been exceeded. The result would be a dropped call when the user attempted to initiate the conference feature. This could appear to be a Cisco CallManager feature issue, when in fact it is a problem with the number of available conference resources. The number of times a conference resource was required but not available is one of the counters logged in Microsoft Performance. The same behavior would occur if there are conference resources available, but the conferencing service had stopped.

Codec/Regions: Codec Mismatch

If a user gets a reorder tone when going off-hook, it could be the result of codec disagreement between regions. Verify that both call ends support at least one common codec (for example, G.711). If not, you will need to use transcoders.

A region specifies the range of supported codecs that can be used with each of the other regions. Every device belongs to a region.

Note: Codec negotiation with a Cisco IOS router is not supported.

For example, Region1<->Region2 = G.711, means that a call between a device in Region1 and a device in Region2 can use G.711 or any other supported codec that requires the same or less bandwidth as G.711 (any supported codecs within G.711, G.729, G.723, and so on).

Note: The following codecs are supported for each device:
Cisco IP Phone 7960 — G.711A-law/µ-law, G.729AnnexB
Cisco IP Phone SP12 series and VIP 30 — G.711A-law/µ-law, G.723.1
Cisco Access Gateway DE30 and DT-24+ — G.711A-law/µ-law, G.723.1

Locations

If a user gets a reorder tone after dialing a number, it could be because the Cisco CallManager bandwidth allocation for the location of one of the call end devices has been exceeded (less than 24k). Cisco CallManager checks for 24k available bandwidth for each device before making a call. If less than 24k bandwidth is available, Cisco CallManager will not setup the call and the user will hear a reorder tone.

```
12:42:09.017 Cisco CallManager|Locations:       Orig=1 BW=12  Dest=0 BW=-1  (-1 implies infinite bw
available)
12:42:09.017 Cisco CallManager|StationD - stationOutputCallState tcpHandle=0x4f1ad98
12:42:09.017 Cisco CallManager|StationD - stationOutputCallInfo CallingPartyName=,
CallingParty=5003, CalledPartyName=, CalledParty=5005, tcpHandle=0x4f1ad98
12:42:09.017 Cisco CallManager|StationD - stationOutputStartTone: 37=ReorderTone
tcpHandle=0x4f1ad98
```

Once the call is established, the Cisco CallManager will subtract bandwidth from the locations depending on the codec used in that call. If the call is using G.711, Cisco CallManager will subtract 80k; if the call is using G.723, Cisco CallManager will subtract 24k; if the call is using G729, Cisco CallManager will subtract 24k.

Conference Bridge

Use the following information to help troubleshooting a “No Conference Bridge available” problem. This could be either software or a hardware problem.

First, check to see if you have any available Conference Bridge resources registered with Cisco CallManager (either software or hardware). To do so, you can use Microsoft Performance to check the number of “Unicast AvailableConferences.”

The Cisco IP Voice Media Streaming application performs the conference bridge function. One software installation of Cisco IP Voice Media Streaming will support 16 Unicast Available Conferences (3 people/conference), as shown in the following trace.

```
10:59:29.951 Cisco CallManager|UnicastBridgeControl - wait_capabilities_StationCapRes - Device=
CFB_kirribilli - Registered - ConfBridges= 16, Streams= 48, tcpHandle=4f12738
10:59:29.951 Cisco CallManager|UnicastBridgeManager - UnicastBridgeRegistrationReq - Device
Registration Complete for Name= Xoð´ ô%ð´  - DeviceType= 50, ResourcesAvailable= 16,
deviceTblIndex= 0
```

One E1 port (WS-X6608-E1 card contains 8x E1 ports) provides five Unicast Available Conferences (max conference size = 6), as shown in the following trace.

```
11:14:05.390 Cisco CallManager|UnicastBridgeControl - wait_capabilities_StationCapRes - Device=
CFB00107B000FB0 - Registered - ConfBridges= 5, Streams= 16, tcpHandle=4f19d64
11:14:05.480 Cisco CallManager|UnicastBridgeManager - UnicastBridgeRegistrationReq - Device
Registration Complete for Name= Xoð´ ô%ð´  - DeviceType= 51, ResourcesAvailable= 5,
deviceTblIndex= 0
```

The following hardware trace on the Cisco Catalyst 6000 8 Port Voice T1/E1 and Services Module, indicates that the E1 port 4/1 in the card has registered as a Conference Bridge with Cisco CallManager.

```
greece-sup (enable) sh port 4/1
Port  Name               Status     Vlan       Duplex Speed Type
----- ------------------ ---------- ---------- ------ ----- ------------
 4/1                     enabled    1          full     - Conf Bridge

Port     DHCP    MAC-Address       IP-Address      Subnet-Mask
-------- ------- ----------------- --------------- ---------------
 4/1     disable 00-10-7b-00-0f-b0 10.200.72.31    255.255.255.0

Port     Call-Manager(s)   DHCP-Server     TFTP-Server     Gateway
-------- ----------------- --------------- --------------- ---------------
 4/1     10.200.72.25      -               10.200.72.25    -

Port     DNS-Server(s)     Domain
-------- ----------------- -------------------------------------------------
 4/1     -                 0.0.0.0

Port     CallManagerState DSP-Type
-------- ---------------- --------
 4/1     registered       C549

Port  NoiseRegen NonLinearProcessing
----- ---------- -------------------
 4/1  disabled   disabled
```

Second, check the maximum number of users configured in the conference (Ad Hoc or Meet-Me) to determine if the problem occurred because this number was exceeded.

Transcoding Problems

If you have installed a hardware transcoder in the Cisco Catalyst 6000 8 Port Voice T1/E1 and Services Module, and it doesn't work as expected (meaning you cannot make calls between two users with no common codec), check to see if you have any available Transcoder resources registered with Cisco CallManager (must be hardware). Use Microsoft Performance to check the number of "MediaTermPointsAvailable" available.

One E1 port (WS-X6608-E1 card contains 8x E1 ports) provides Transcoder/MTP resources for 16 calls, as shown in the following trace.

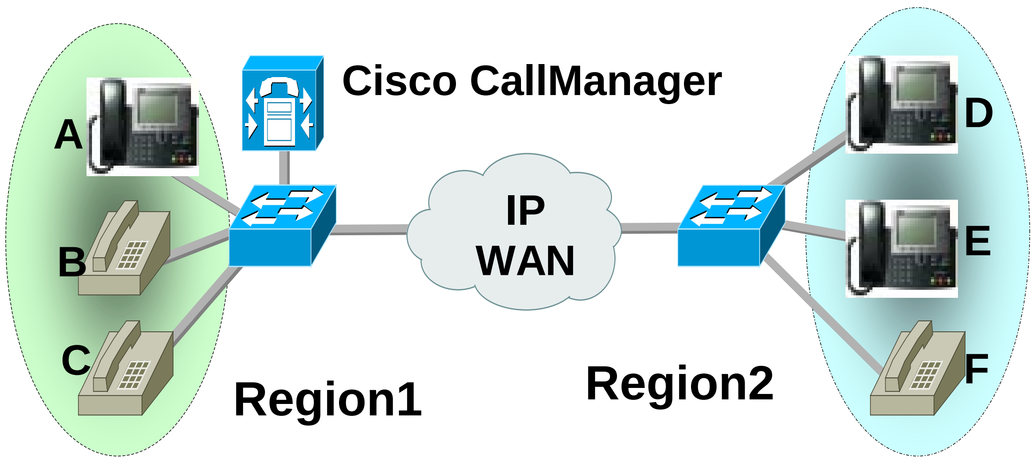
```
11:51:09.939 Cisco CallManager|MediaTerminationPointControl - Capabilities Received - Device=
MTP00107B000FB1 - Registered - Supports 16 calls
```

The following hardware trace on the Cisco Catalyst 6000 8 Port Voice T1/E1 and Services Module, indicates that the E1 port 4/2 in the card has registered as an MTP/transcoder with Cisco CallManager.

```
greece-sup (enable) sh port 4/2
Port  Name               Status     Vlan       Duplex Speed Type
----- ------------------ ---------- ---------- ------ ----- ------------
 4/2                     enabled    1          full      - MTP

Port     DHCP    MAC-Address       IP-Address      Subnet-Mask
-------- ------- ----------------- --------------- ---------------
 4/2     disable 00-10-7b-00-0f-b1 10.200.72.32    255.255.255.0

Port     Call-Manager(s)   DHCP-Server     TFTP-Server     Gateway
-------- ----------------- --------------- --------------- ---------------
 4/2     10.200.72.25      -               10.200.72.25    -

Port     DNS-Server(s)     Domain
-------- ----------------- -------------------------------------------------
 4/2     -                 0.0.0.0

Port     CallManagerState DSP-Type
-------- ---------------- --------
 4/2     registered       C549

Port  NoiseRegen NonLinearProcessing
----- ---------- -------------------
 4/2  disabled   disabled
```

Note: The same E1 port cannot be configured for both Conference Bridge and Transcoder/MTP

In order to make a call between two devices using a low bit rate code (such as G.729 and G.723) that do not support the same codec, a transcoder resource is required. Consider the following illustration:



Assume Cisco CallManager has been configured such that between Region1 and Region2, the codec is G.729. The following scenarios are possible:

- If caller on Phone A initiates a call, Cisco CallManager realizes it is a Cisco IP Phone 7960, which happens to support G.729. After the digits have been collected, the Cisco CallManager determines that the call is destined for User D who is in Region2. Since the destination device also supports G.729, the call is set up and the audio flows directly between Phone A and Phone D.
- If a caller on Phone B, who has a Cisco IP Phone 12SP+, were to initiate a call to Phone D, then this time the Cisco CallManager would realize that the originating phone only supports G.723 or G.711. Cisco CallManager would need to allocate a transcoding resource so that audio would flow as G.711 between Phone B and the transcoder, but as G.729 between the transcoder and Phone D. If no transcoder were available, Phone D's phone would ring, but as soon as the call was answered, the call would disconnect.
- If a user on Phone B were to call Phone F (Cisco IP Phone 12SP+), the two phones would actually use G.723, even though G.729 is configured as the codec to use between the regions. G.723 is used because both endpoints support it and it uses less bandwidth than G.729.
- If a Cisco uOne voicemail system is added (which only supports G.711) or a Cisco IOS router configured for G.711 to Region1, then a transcoding device must be used if calling from Region2. If none is available, then the call will fail.

MTP Resource Problems

An MTP resource problem could be the culprit if a call is established, but supplementary services are not available on an H.323 device that does not support H323v2. First, determine whether you have any available MTP resources (either software or hardware) registered with Cisco CallManager. You can do so by using Microsoft Performance to check the number of "MediaTermPointsAvailable."

One MTP software application supports 24 calls (using MTP to support supplementary services with H.323 devices that not support H.323v2), as shown in the following trace.

```
10:12:19.161 Cisco CallManager|MediaTerminationPointControl - Capabilities Received - Device=
MTP_kirribilli. - Registered - Supports 24 calls
```

One E1 port (WS-X6608-E1 card contains 8x E1 ports) provides MTP resources for 16 calls, as shown in the following trace.

```
11:51:09.939 Cisco CallManager|MediaTerminationPointControl - Capabilities Received - Device=
MTP00107B000FB1 - Registered - Supports 16 calls
```

The following hardware trace from the Cisco Catalyst 6000 8 Port Voice T1/E1 and Services Module, indicates that the E1 port 4/2 in the card has registered as an MTP/transcoder with Cisco CallManager.

```
greece-sup (enable) sh port 4/2
Port  Name               Status     Vlan       Duplex Speed Type
----- ------------------ ---------- ---------- ------ ----- ------------
 4/2                     enabled    1           full     - MTP

Port     DHCP    MAC-Address       IP-Address      Subnet-Mask
-------- ------- ----------------- --------------- ---------------
 4/2     disable 00-10-7b-00-0f-b1 10.200.72.32    255.255.255.0

Port     Call-Manager(s)   DHCP-Server     TFTP-Server     Gateway
-------- ----------------- --------------- --------------- ---------------
 4/2     10.200.72.25      -               10.200.72.25    -

Port     DNS-Server(s)     Domain
-------- ----------------- -------------------------------------------------
 4/2     -                 0.0.0.0

Port     CallManagerState DSP-Type
-------- ---------------- --------
 4/2     registered       C549

Port  NoiseRegen NonLinearProcessing
----- ---------- -------------------
 4/2  disabled   disabled
```

Second, check to see if the **Media Termination Point Required** box is checked in the Gateway Configuration screen of Cisco CallManager Administration.

Third, verify that Cisco CallManager has allocated the required number of MTP devices.

From the SDI file:

```
15:22:23.848 Cisco CallManager|MediaManager(40) started
15:22:23.848 Cisco CallManager|MediaManager - wait_AuConnectRequest
15:22:23.848 Cisco CallManager|MediaManager - wait_AuConnectRequest - Transcoder Enabled
15:22:23.848 Cisco CallManager|MediaManager - wait_AuConnectRequest - party1(16777357),
party2(16777358), proxies=1, connections=2, current proxies=0
15:22:23.848 Cisco CallManager|MediaManager - wait_AuConnectRequest - proxy connections
15:22:23.848 Cisco CallManager|MediaManager - wait_AuConnectRequest - allocating
MTP(ci=16777359)
15:22:23.848 Cisco CallManager|MediaManager - wait_AllocateMtpResourceRes
15:22:23.848 Cisco CallManager|MediaManager - wait_AllocateMtpResourceRes - start 2 connections
15:22:23.848 Cisco CallManager|MediaManager - wait_AllocateMtpResourceRes - creating connection
between party1(16777357) and party2(16777359)
15:22:23.848 Cisco CallManager|MediaManager - wait_AllocateMtpResourceRes - creating connection
between party1(16777358) and party2(16777359)
```

```
15:22:23.848 Cisco CallManager|MediaCoordinator - wait_MediaCoordinatorAddResource -
CI=16777359 count=1
15:22:23.848 Cisco CallManager|MediaCoordinator - wait_MediaCoordinatorAddResource -
CI=16777359 count=2
```

Dial Plans

A Dial Plan is a list of numbers and groups of numbers that tell the Cisco CallManager what devices (phones, gateways, and so on) to send calls to when a certain string of digits are collected. It is analogous to a static routing table in a router. Please be certain your dial plan concepts, basic call routing, and planning have been carefully considered and properly configured before trying to troubleshoot a potential dial plan issue. Very often, the problem lies with planning and configuration when a problem occurs.

Use the following tips to help troubleshoot dial plans problems:

- What is the Directory Number (DN) that is originating the call?
- What is the Calling Search Space of this DN?
- If applicable, what is the Calling Search Space of the device (such as a Cisco IP Phone) with which the DN is associated? Make sure that you identify the correct device; because multiple line appearances are supported, it’s possible to have a DN on multiple devices. Note the device’s Calling Search Space. If this is a Cisco IP Phone originating the call, remember that a particular line (DN) and the device that line is associated with have Calling Search Spaces. They will be combined when making a call. For example, if line instance 1000 has a Calling Search Space of AccessLevelX and the Cisco IP Phone that has extension 1000 configured on it has AccessLevelY as its Calling Search Space, then when making a call from that line appearance, Cisco CallManager will search through partitions contained in Calling Search Space AccessLevelX and AccessLevelY.
- What Partitions are associated with the Calling Search Space(s)?
- What is the Partition of the device to which the call should (or should not) go?
- What is the number that is being dialed? Note if and when they are getting secondary dial tone at any stage. Also what do they hear after all the digits have been entered (re-order, fast-busy)? Do they get the progress tones before they expect to hear anything? Make sure callers wait at least 10 seconds after typing the last digit, since they may have to wait for the inter-digit timer to expire.
- Generate a Route Plan Report in Cisco CallManager Administration, and use it to examine all the route patterns for the partitions that are in the Calling Search Space for the call.
- If necessary, add or modify the Route Patterns or Route Filters.
- If you can find the Route Pattern to which the call is being sent, note the Route List or Gateway to which the pattern points.
- If it's a Route List, check which Route Groups are part of the list and which Gateway(s) is part of the Route Groups.
- Verify that the applicable devices are registered with Cisco CallManager.
- If there's no access to Cisco CallManager, get the "show tech" to capture this information and verify.

- Watch out for the @ sign. This is a macro that can expand to include many different things. It is often used in combination with filtering options.
- If a device isn't part of a partition, it is said to be part of the Null or default partition. **Every** user should be able to call that device. The Null partition is always searched **last**.
- If you dial an outside number that is matching a 9.@ pattern and it takes 10 seconds before the call goes through, check the filtering options. By default, with a 9.@ pattern, when dialing a 7-digit number, it **will** wait 10 seconds. You need to apply a Route Filter to the pattern that says LOCAL-AREA-CODE DOES-NOT- EXIST and END-OF-DIALING DOES-NOT-EXIST.

Partitions

Route partitions inherit the error handling capabilities of the Cisco CallManager software. That is, a console and SDI file trace is provided for logging information and error messages. These messages will be part of the digit analysis component of the traces. Even with the traces below, knowing how the Partitions and Calling Search Spaces are configured and what devices are in each partition and its associated calling search space, is vital in determining the problem.

The trace below is an example of a number dialed that is in the Calling Search Space of the device. For more detailed explanations about SDI traces, please review the case studies in this document.

```
08:38:54.968 Cisco CallManager|StationInit - InboundStim - OffHookMessageID tcpHandle=0x6b88028
08:38:54.968 Cisco CallManager|StationD - stationOutputDisplayText tcpHandle=0x6b88028, Display=
5000
08:38:54.968 Cisco CallManager|StationD - stationOutputSetLamp stim: 9=Line instance=1
lampMode=LampOn tcpHandle=0x6b88028
08:38:54.968 Cisco CallManager|StationD - stationOutputCallState tcpHandle=0x6b88028
08:38:54.968 Cisco CallManager|StationD - stationOutputDisplayPromptStatus tcpHandle=0x6b88028
08:38:54.968 Cisco CallManager|StationD - stationOutputSelectSoftKeys tcpHandle=0x6b88028
08:38:54.968 Cisco CallManager|StationD - stationOutputActivateCallPlane tcpHandle=0x6b88028
08:38:54.968 Cisco CallManager|Digit analysis: match(fqcn="5000", cn="5000",
pss="RTP_NC_Hardwood:RTP_NC_Woodland:Local RTP", dd="")
```

Above you can see in the Digit Analysis component of the trace the "pss" (Partition Search Space, also known as Calling Search Space) is listed for the device placing the call. Below, you can see that "RTP_NC_Hardwood;RTP_NC_Woodland;Local_RTP" are the partitions this device is allowed to call.

```
08:38:54.968 Cisco CallManager|Digit analysis: potentialMatches=PotentialMatchesExist
08:38:54.968 Cisco CallManager|StationD - stationOutputStartTone: 33=InsideDialTone
tcpHandle=0x6b88028
08:38:55.671 Cisco CallManager|StationInit - InboundStim - KeypadButtonMessageID kpButton: 5
tcpHandle=0x6b88028
08:38:55.671 Cisco CallManager|StationD - stationOutputStopTone tcpHandle=0x6b88028
08:38:55.671 Cisco CallManager|StationD - stationOutputSelectSoftKeys tcpHandle=0x6b88028
08:38:55.671 Cisco CallManager|Digit analysis: match(fqcn="5000", cn="5000",
pss="RTP_NC_Hardwood:RTP_NC_Woodland:Local RTP", dd="5")
08:38:55.671 Cisco CallManager|Digit analysis: potentialMatches=PotentialMatchesExist
```

```
08:38:56.015 Cisco CallManager|StationInit - InboundStim - KeypadButtonMessageID kpButton: 0
tcpHandle=0x6b88028
08:38:56.015 Cisco CallManager|Digit analysis: match(fqcn="5000", cn="5000",
pss="RTP_NC_Hardwood:RTP_NC_Woodland:Local RTP", dd="50")
08:38:56.015 Cisco CallManager|Digit analysis: potentialMatches=PotentialMatchesExist
08:38:56.187 Cisco CallManager|StationInit - InboundStim - KeypadButtonMessageID kpButton: 0
tcpHandle=0x6b88028
08:38:56.187 Cisco CallManager|Digit analysis: match(fqcn="5000", cn="5000",
pss="RTP_NC_Hardwood:RTP_NC_Woodland:Local RTP", dd="500")
08:38:56.187 Cisco CallManager|Digit analysis: potentialMatches=PotentialMatchesExist
08:38:56.515 Cisco CallManager|StationInit - InboundStim - KeypadButtonMessageID kpButton: 3
tcpHandle=0x6b88028
08:38:56.515 Cisco CallManager|Digit analysis: match(fqcn="5000", cn="5000",
pss="RTP_NC_Hardwood:RTP_NC_Woodland:Local RTP", dd="5003")
08:38:56.515 Cisco CallManager|Digit analysis: analysis results
08:38:56.515 Cisco CallManager||PretransformCallingPartyNumber=5000
```

The key thing to note above is that "PotentialMatchesExist" is the result of Digit Analysis of the numbers that were dialed until the exact match is found and the call is routed accordingly.

Below is a trace when the number attempted to be dialed "1001" is not in the Calling Search Space of the device. Again, the key thing to note is the digit analysis routine had potential matches until only the first digit was dialed. The route pattern associated with the digit "1" is in a partition that is not in the device's calling search space, "RTP_NC_Hardwood;RTP_NC_Woodland;Local_RTP". Therefore the phone was sent Reorder Tone.

```
08:38:58.734 Cisco CallManager|StationInit - InboundStim - OffHookMessageID tcpHandle=0x6b88028
08:38:58.734 Cisco CallManager|StationD - stationOutputDisplayText tcpHandle=0x6b88028, Display=
5000
08:38:58.734 Cisco CallManager|StationD - stationOutputSetLamp stim: 9=Line instance=1
lampMode=LampOn tcpHandle=0x6b88028
08:38:58.734 Cisco CallManager|StationD - stationOutputCallState tcpHandle=0x6b88028
08:38:58.734 Cisco CallManager|StationD - stationOutputDisplayPromptStatus tcpHandle=0x6b88028
08:38:58.734 Cisco CallManager|StationD - stationOutputSelectSoftKeys tcpHandle=0x6b88028
08:38:58.734 Cisco CallManager|StationD - stationOutputActivateCallPlane tcpHandle=0x6b88028
08:38:58.734 Cisco CallManager|Digit analysis: match(fqcn="5000", cn="5000",
pss="RTP_NC_Hardwood:RTP_NC_Woodland:Local RTP", dd="")
08:38:58.734 Cisco CallManager|Digit analysis: potentialMatches=PotentialMatchesExist
08:38:58.734 Cisco CallManager|StationD - stationOutputStartTone: 33=InsideDialTone
tcpHandle=0x6b88028
08:38:59.703 Cisco CallManager|StationInit - InboundStim - KeypadButtonMessageID kpButton: 1
tcpHandle=0x6b88028
08:38:59.703 Cisco CallManager|StationD - stationOutputStopTone tcpHandle=0x6b88028
08:38:59.703 Cisco CallManager|StationD - stationOutputSelectSoftKeys tcpHandle=0x6b88028
08:38:59.703 Cisco CallManager|Digit analysis: match(fqcn="5000", cn="5000",
pss="RTP_NC_Hardwood:RTP_NC_Woodland:Local RTP", dd="1")
08:38:59.703 Cisco CallManager|Digit analysis: potentialMatches=NoPotentialMatchesExist
08:38:59.703 Cisco CallManager|StationD - stationOutputStartTone: 37=ReorderTone
tcpHandle=0x6b88028
```

Route partitions work by associating a partition name with every directory number in the system. The directory number can be called only if the calling device contains the partition within a list of partitions to which it is permitted to place calls — its partition search space. This provides for extremely powerful control over routing.

When a call is being placed, Digit Analysis attempts to resolve the dialed address only in those partitions that the partition search space specifies. Each partition name comprises a discrete subset of the global dial-able address space. From each listed partition, Digit Analysis retrieves the pattern that best matches the sequence of dialed digits. Then, from among the matching patterns, Digit Analysis chooses the best match. If two patterns equally match the sequence of dialed digits, Digit Analysis breaks the tie by selecting the pattern associated with the partition listed first in the partition search space (for more information, review the documentation about Closest-Match Routing).

Security

Cisco CallManager can be configured to create a secure dialing plan for users. This can be done through the use of partitions and calling search spaces, in addition to more common filtering based on sections of the "@" macro (which stands for the North American Numbering Plan) in a route pattern, such as the Area Code. Partitions and Calling Search Spaces are an integral part of security and are especially useful for multi-tenant environments and creating an individual user level. Filtering is a subset of the Calling Search Space/Partition concept that can add additional granularity to the security plan.

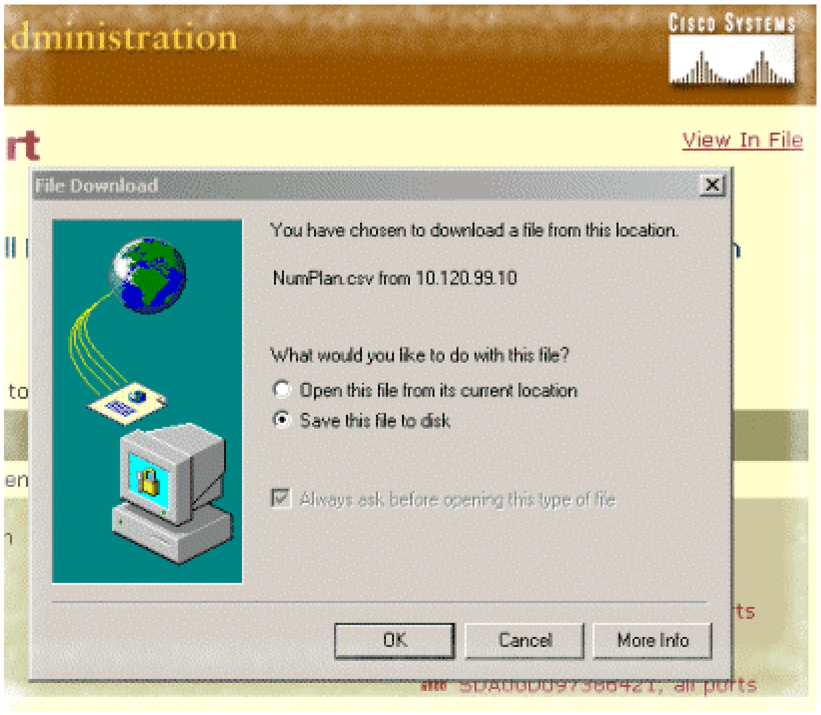
This is an extension to the Dial Plan section above. Be advised, usually the last thing you want to do when trying to fix a filtering problem is to run an SDI trace. There is simply not enough information and the potential for causing more harm is too great.

Run a "show tech" on Cisco CallManager. The following information appears in the Route Filter section.

Show-tech

Name	dialPlanWizardG	Clause
CiscoDallasInte	1	(INTERNATIONAL-
CiscoRTPTollByP	1	(AREA-CODE == 9
CiscoRTPLongDis	1	(AREA-CODE EXIS
CiscoDallasToll	1	(AREA-CODE == 9
CiscoDallas911R	1	(SERVICE == 911
CiscoRTPLocal7D	1	(AREA-CODE DOES
CiscoDallasLong	1	(AREA-CODE EXIS
CiscoRTP911RF	1	(SERVICE == 911
CiscoRTPInterna	1	(INTERNATIONAL-
CiscoDallasLoca	1	(LOCAL-AREA-COD

Unfortunately the display is cut off. It does at least give a listing of all the Route Filters in the system. There is no way from the "show" command to see what filters are associated with which Route Pattern. Another way to get a handle on the dial plan is to go to the Route Plan Report page. Following is an option on the far right-hand side to "View In File".



You have chosen to download a file from this location.

NumPlan.csv from 10.120.99.10

What would you like to do with this file?

( ) Open this file from its current location
[✓ Save this file to disk]

[✓] Always ask before opening this type of file

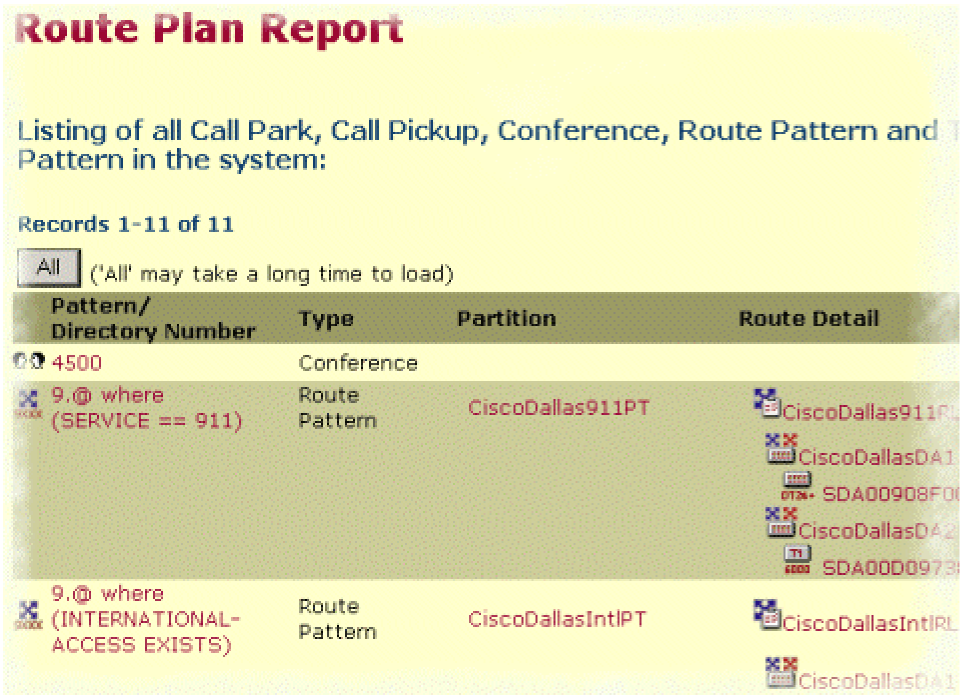
The output will be a comma-separated file that can be viewed in Microsoft Excel or a similar application:

Show-tech File Output

Pattern/ DN	Partition	Pattern Usage	Device Name	Device Description
1000		Device	SEP003094C2635E	Telecaster
1010		Device	SEP003094C2635E	Telecaster
1111		Device	SEP00308062CDF1	SEP00308062CDF1
1211		Device	SEP00308062CDF1	SEP00308062CDF1
2999		Device	SAA0010EB007FFE	SAA0010EB007FFE
4444		Device	SEP003094C26302	Guest
4500		Conference		
9.@	CiscoRTPLocalPT	Route	CiscoRTPLocalRL	
9.@	CiscoDallasLocalPT	Route	CiscoDallasLocalRL	
9.@	CiscoRTPIntlPT	Route	CiscoRTPIntlRL	
9.@	CiscoDallasLongDistPT	Route	CiscoDallasLongDistRL	
9.@	CiscoRTP911PT	Route	CiscoRTP911RL	
9.@	CiscoRTPLongDistPT	Route	CiscoRTPLongDistRL	
9.@	CiscoTollByPassToDallasPT	Route	CiscoTollByPassToDallasRL	
9.@	CiscoDallasIntlPT	Route	CiscoDallasIntlRL	
9.@	CiscoDallas911PT	Route	CiscoDallas911RL	
9.@	CiscoTollByPassToRTPPT	Route	CiscoTollByPassToRTPRL	

This shows the Route Patterns and their corresponding partitions. It does not show the Route Filters or the Calling Search Spaces of the directory numbers. More information is available on the actual Route Plan Report. If you must contact the Cisco TAC, send this page via email if the Cisco CallManager is inaccessible.

Following is a nice layout of the Route Patterns, the Partitions, and the Route List/Route Group/Gateway.



Route Plan Report

Listing of all Call Park, Call Pickup, Conference, Route Pattern and
Pattern in the system:

Records 1-11 of 11

All ('All' may take a long time to load)

Pattern/ Directory Number	Type	Partition	Route Detail
4500	Conference		
9.@ where (SERVICE == 911)	Route Pattern	CiscoDallas911PT	CiscoDallas911R CiscoDallasDA1 SDA00908F0 CiscoDallasDA2 SDA00D0973
9.@ where (INTERNATIONAL-ACCESS EXISTS)	Route Pattern	CiscoDallasIntlPT	CiscoDallasIntlRL CiscoDallasDA1

Slow Server Response

Slow response from the server could result if the duplex of the switch does not match the duplex of the Cisco CallManager server. For optimal performance set both switch and server to '100/Full." We do not recommend using "Auto" on either the switch or the server. You must restart the Cisco CallManager server for the change to take effect.

Reorder Tone Through Gateways

Users placing a call through the gateway might get a reorder tone if they are attempting to make a restricted call or call a number that has been blocked. A reorder tone may occur if the dialed number is out of service or if the PSTN has an equipment or service problem. Check to be sure the device giving the reorder tone has registered. Also, check your dial plan configuration to ensure that the call can be successfully routed.

Steps for troubleshooting reorder tones through gateways:

1. Check the gateways to ensure that you are using the latest software loads.
2. Check CCO (Cisco Connection Online at www.cisco.com) for the latest software loads, new patches, or release notes relating to the problem.
3. Start an SDI trace and recreate the problem. Reorder tones could be the result of a configuration issue with location-based admission control or gatekeeper-based admission control where the Cisco CallManager might limit the number of allowable calls. In the SDI

trace, locate the call to determine if it was blocked intentionally by a route pattern or the calling search space, or by any other configuration setting.

4. Reorder tones can also occur when calling through the PSTN. Check the SDI trace for Q.931 messages, in particular, for disconnect messages. If a Q.931 disconnect message is present, it means the other party caused the disconnect and we cannot correct for that.

Gateway Registration Problems

One of the most common issues encountered with gateways on a Cisco CallManager is a registration problem. Registration can fail for a variety of reasons.

This section deals with two similar but different categories of gateways. The Analog Access AS-X, AT-X and Digital Access DT-24+ and DE-30+ belong to one category. These gateways are stand-alone units that are not directly connected to a Network Management Processor (NMP). The second category includes the Analog Access WS-X6624 and Digital Access WS-X6608. These gateways are blades installed in a Catalyst 6000 chassis with direct connectivity to the NMP for control and statusing.

In the examples below, we have bolded lines of text to make it easier for you to see the messages being explained. In the actual display output, text is not bolded. The examples are from an WS-X6624.

The first thing to check is that the gateway is up and running. All of the gateways have a "heartbeat" LED that blinks 1-second on, 1-second off when the gateway software is running normally. If this LED is not blinking at all, or blinking very rapidly, then the gateway software is not running. Normally this will result in an automatic reset of the gateway. Also, it is normal for the gateway to reset itself if it cannot complete the registration process after about 2 to 3 minutes. So you may happen to look at the heartbeat LED while the device is resetting, but if the normal blinking pattern does not appear in 10 to 15 seconds, then the gateway has suffered a serious failure. On the AS-X or AT-X gateway, the heartbeat LED is the far right green LED showing on the front panel. On the DT-24+ or DE-30+ gateway, it is the far left red LED on the top edge of the card. On the Analog Access WS-X6624, it is a green LED inside the blade (not visible from the front panel) on the far right card edge near the front. Finally, on the Digital Access WS-X6608 there is a separate heartbeat LED for each of the 8 spans on the blade. There are 8 red LEDs across the card (not visible from the front panel) about 2/3 of the way towards the back.

The second thing to check is that the gateway has received its IP address. A standalone gateway **must** receive its IP address via DHCP or BOOTP. A Catalyst gateway may receive its IP address by DHCP, BOOTP, or by manual configuration through the NMP. If you have access to the DHCP server, the best way to check a standalone gateway is to verify that the device has an outstanding lease on an IP address. If the gateway shows up on your server, this is a good indication, but not definitive. Delete the lease at the DHCP server, and then reset the gateway. If the gateway reappears on the server with a lease within a couple of minutes, then everything is working fine in this area. If not, then either the gateway cannot contact the DHCP server (Is a router improperly configured and not forwarding DHCP broadcasts? Is the server running?), or cannot get a positive response (Is the IP address pool depleted?). If checking these suggestions does not yield the answer, use a sniffer trace to determine the specific problem.

For a Catalyst 6000 gateway, you should check to make sure the NMP can communicate with the gateway. You can check this by trying to ping its internal IP address from the NMP. The IP address is in the format:

127.1.*module.port*

So, in our example, we would do:

```
Console (enable) ping 127.1.7.1
127.1.7.1 is alive
```

If pinging works, then the 'sh-port' command will show IP address information. Make sure the IP address information and the TFTP IP address is correct as well. If the gateway is failing to obtain valid DHCP information, the tracy utility (which can be supplied by Cisco TAC) can be used to determine the problem. Issue the command from the Cat6000 CLI:

tracy_start *mod port*

In this example, the WS-X6624 is module 7 and it only has a single 860 processor, so it is port 1. The command we would issue is '**tracy_start 7 1**'

The following output is actually from the 860-console port on the gateway board itself, however, the output of the tracy command is nothing more than a remote copy of the 860-console port.

```
      |            |
      |            |
    |||          |||
   |||||       |||||
||||||||:..:||||||||:..
Cisco  Systems
CAT6K Analog Gateway (ELVIS)
APP Version : A0020300, DSP Version : A0030300, Built Jun  1 2000 16:33:01

ELVIS>> 00:00:00.020 (XA) MAC Addr : 00-10-7B-00-13-DE
00:00:00.050 NMPTask:got message from XA Task
00:00:00.050 (NMP) Open TCP Connection ip:7f010101
00:00:00.050 NMPTask:Send Module Slot Info
00:00:00.060 NMPTask:get DIAGCMD
00:00:00.160 (DSP) Test Begin -> Mask<0x00FFFFFF>
00:00:01.260 (DSP) Test Complete -> Results<0x00FFFFFF/0x00FFFFFF>
00:00:01.260 NMPTask:get VLANCONFIG
00:00:02.870 (CFG) Starting DHCP
00:00:02.870 (CFG) Booting DHCP for dynamic configuration.
00:00:06.570 (CFG) DHCP Request or Discovery Sent, DHCPState = INIT_REBOOT
00:00:06.570 (CFG) DHCP Server Response Processed, DHCPState = INIT_REBOOT
00:00:06.780 (CFG) IP Configuration Change!  Restarting now...
00:00:10.480 (CFG) DHCP Request or Discovery Sent, DHCPState = INIT
00:00:14:480 (CFG) DHCP Timeout Waiting on Server, DHCPState = INIT
```

```
00:00:22:480 (CFG) DHCP Timeout Waiting on Server, DHCPState = INIT
00:00:38:480 (CFG) DHCP Timeout Waiting on Server, DHCPState = INIT
```

If the above timeout message continues to scroll by, then there is a problem contacting the DHCP server. First thing to check is that the Catalyst 6000 gateway port is in the correct VLAN. This information is in the '**sh port**' command from before. If the DHCP server is not on the same VLAN as the Catalyst 6000 gateway, then make sure the appropriate IP Helper addresses have been configured to forward the DHCP requests to the DHCP server. It is possible for the gateway to get stuck in the INIT state after a VLAN number change until the gateway resets. When in this state, it would not hurt to try resetting the gateway. Every time the 860 gets reset, your tracy session will be lost, so you must close your existing session and re-establish a new one by issuing the following commands:

tracy_close *mod port*
tracy_start *mod port*

If all this checks out and you're still seeing the **DHCPState = INIT** messages, then check to see if the DHCP server is functioning correctly. If so, start a sniffer trace to see if the requests are being sent and if the server is responding or not.

Once DHCP is working correctly, the gateway will have an IP address that will allow the use of the tracy debugging utility. This utility is a built-in feature of the NMP command set for the Catalyst gateways, and available as a helper application that runs on Windows 98/NT/2000 for the standalone gateways. To use the helper application tracy utility, you need to "Connect" to the gateway by using the IP address to which it is assigned. This tracy application works on all the gateways, provides a separate trace window for each gateway (up to eight may be traced at once), and allows traces to be logged directly to a file you specify.

The next step is to verify that the TFTP server IP address was correctly provided to the gateway. This is normally provided by DHCP in either Option 66 (by name or IP address), Option 150 (IP address only), or si_addr (IP address only). If your server has multiple Options configured, si_addr will take precedence over Option 150, which will take precedence over Option 66. If Option 66 provides the DNS_NAME of the TFTP server, then the DNS server(s) IP address(es) must have been specified by DHCP, **and** the name entered in Option 66 must resolve to the correct TFTP server IP address. A Catalyst gateway could be configured by the NMP to disable DHCP, and the NMP operator must then enter all configuration parameters by hand at the console, including the TFTP server address.

Additionally, the gateways will always attempt to resolve the name "CiscoCM1" via DNS. If successful, the CiscoCM1 IP address will take precedence over anything the DHCP server or NMP tells it for the TFTP server address, even if the NMP has DHCP disabled.

You can check the current TFTP server IP address in a gateway by using the tracy utility. Enter the following command to get the configuration task number:

```
TaskID: 0
Cmd:    show tl
```

Look for a line with “config” or “CFG” and use the corresponding number as the taskID for the next line. For example, for the Digital Access WS-X6624 gateway, the command to dump the DHCP information is:

```
TaskID: 6
Cmd:    show dhcp
```

The TFTP server IP address is then clearly shown. If it is not correct, verify that your DHCP options and other information it provides are correct.

Once the TFTP address is correct, the next step would be to ensure that the gateway is getting its configuration file from the TFTP server. If you see the following in the tracy output, your TFTP service may not be working correctly or the gateway might not be configured on the Cisco CallManager:

```
00:09:05.620 (CFG) Requesting SAA00107B0013DE.cnf File From TFTP Server
00:09:18.620 (CFG) TFTP Error: Timeout Awaiting Server Response for .cnf File!
```

The gateway will attempt to connect to the same IP address as the TFTP server if it does not get a configuration file. This is fine unless you are in a clustered environment in which the gateway needs to receive its list of redundant Cisco CallManagers. If the card is not getting its TFTP information correctly, check the TFTP service on the Cisco CallManager and make sure it is running. Also, check the TFTP trace on the Cisco CallManager as well.

Another common problem is that the gateway is not configured correctly on the Cisco CallManager. A typical error is entering and incorrect MAC address for the gateway. If this is the case, for a Catalyst 6000 gateway, you will probably get the following messages on the NMP console every two minutes:

```
2000 Apr 14 19:24:08 %SYS-4-MODHPRESET:Host process (860) 7/1 got reset asynchronously
2000 Apr 14 19:26:05 %SYS-4-MODHPRESET:Host process (860) 7/1 got reset asynchronously
2000 Apr 14 19:28:02 %SYS-4-MODHPRESET:Host process (860) 7/1 got reset asynchronously
```

This is what the tracy output would look like if the gateway is not in the Cisco CallManager database:

```
00:00:01.670 (CFG) Booting DHCP for dynamic configuration.
00:00:05.370 (CFG) DHCP Request or Discovery Sent, DHCPState = INIT_REBOOT
00:00:05.370 (CFG) DHCP Server Response Processed, DHCPState = BOUND
00:00:05.370 (CFG) Requesting DNS Resolution of CiscoCM1
00:00:05.370 (CFG) DNS Error on Resolving TFTP Server Name.
00:00:05.370 (CFG) TFTP Server IP Set by DHCP Option 150 = 10.123.9.2
00:00:05.370 (CFG) Requesting SAA00107B0013DE.cnf File From TFTP Server
00:00:05.370 (CFG) TFTP Error: .cnf File Not Found!
00:00:05.370 (CFG) Requesting SAADefault.cnf File From TFTP Server
00:00:05.380 (CFG) .cnf File Received and Parsed Successfully.
00:00:05.380 (CFG) Updating Configuration ROM...
00:00:05.610 GMSG: GWEvent = CFG_DONE --> GWState = SrchActive
00:00:05.610 GMSG: CCM#0 CPEvent = CONNECT_REQ --> CPState = AttemptingSocket
```

```
00:00:05.610 GMSG: Attempting TCP socket with CCM 10.123.9.2
00:00:05.610 GMSG: CCM#0 CPEvent = SOCKET_ACK --> CPState = BackupCCM
00:00:05.610 GMSG: GWEvent = SOCKET_ACK --> GWState = RegActive
00:00:05.610 GMSG: CCM#0 CPEvent = REGISTER_REQ --> CPState = SentRegister
00:00:05.680 GMSG: CCM#0 CPEvent = CLOSED --> CPState = NoTCPSocket
00:00:05.680 GMSG: GWEvent = DISCONNECT --> GWState = Rollover
00:00:20.600 GMSG: GWEvent = TIMEOUT --> GWState = SrchActive
00:00:20.600 GMSG: CCM#0 CPEvent = CONNECT_REQ --> CPState = AttemptingSocket
00:00:20.600 GMSG: Attempting TCP socket with CCM 10.123.9.2
00:00:20.600 GMSG: CCM#0 CPEvent = SOCKET_ACK --> CPState = BackupCCM
```

Another possible registration problem could be if the load information is incorrect or the load file is corrupt. The problem could also occur if the TFTP server is not working. In this case, tracy clearly shows that the TFTP server reported the file is not found:

```
00:00:07.390 GMSG: CCM#0 CPEvent = REGISTER_REQ --> CPState = SentRegister
00:00:08.010 GMSG: TFTP Request for application load A0021300
00:00:08.010 GMSG: CCM#0 CPEvent = LOADID --> CPState = AppLoadRequest
00:00:08.010 GMSG: ***TFTP Error: File Not Found***
00:00:08.010 GMSG: CCM#0 CPEvent = LOAD_UPDATE --> CPState = LoadResponse
```

In this case, you can see that the gateway is requesting application load A0021300, although the correct load name would be A0020300. For a Catalyst 6000 gateway, the same problem can occur when a new application load needs to get its corresponding DSP load as well. If the new DSP load is not found, a similar message will appear.

The following shows the output when an Analog Access WS-X6224 has been configured to retrieve an incorrect application load. The output looks similar to that of a gateway that has not been configured on the Cisco CallManager:

```
      |           |
      |           |
    |||         |||
   |||||       |||||
|||||||:..:|||||||:..
C i s c o   S y s t e m s
CAT6K Analog Gateway (ELVIS)
APP Version : A0020300, DSP Version : A0030300, Built Jun  1 2000 16:33:01

ELVIS>> 00:00:00.020 (XA) MAC Addr : 00-10-7B-00-13-DE
00:00:00.050 NMPTask:got message from XA Task
00:00:00.050 (NMP) Open TCP Connection ip:7f010101
00:00:00.050 NMPTask:Send Module Slot Info
00:00:00.060 NMPTask:get DIAGCMD
00:00:00.160 (DSP) Test Begin -> Mask<0x00FFFFFF>
00:00:01.260 (DSP) Test Complete -> Results<0x00FFFFFF/0x00FFFFFF>
00:00:01.260 NMPTask:get VLANCONFIG
00:00:02.030 (CFG) Starting DHCP
00:00:02.030 (CFG) Booting DHCP for dynamic configuration.
00:00:05.730 (CFG) DHCP Request or Discovery Sent, DHCPState = INIT_REBOOT
00:00:05.730 (CFG) DHCP Server Response Processed, DHCPState = BOUND
00:00:05.730 (CFG) Requesting DNS Resolution of CiscoCM1
00:00:05.730 (CFG) DNS Error on Resolving TFTP Server Name.
```

```
00:00:05.730 (CFG) TFTP Server IP Set by DHCP Option 150 = 10.123.9.2
00:00:05.730 (CFG) Requesting SAA00107B0013DE.cnf File From TFTP Server
00:00:05.730 (CFG) .cnf File Received and Parsed Successfully.
00:00:05.730 GMSG: GWEvent = CFG_DONE --> GWState = SrchActive
00:00:05.730 GMSG: CCM#0 CPEvent = CONNECT_REQ --> CPState = AttemptingSocket
00:00:05.730 GMSG: Attempting TCP socket with CCM 10.123.9.2
00:00:05.730 GMSG: CCM#0 CPEvent = SOCKET_ACK --> CPState = BackupCCM
00:00:05.730 GMSG: GWEvent = SOCKET_ACK --> GWState = RegActive
00:00:05.730 GMSG: CCM#0 CPEvent = REGISTER_REQ --> CPState = SentRegister
00:00:06.320 GMSG: CCM#0 CPEvent = LOADID --> CPState = LoadResponse
00:01:36.300 GMSG: CCM#0 CPEvent = TIMEOUT --> CPState = BadCCM
00:01:36.300 GMSG: GWEvent = DISCONNECT --> GWState = Rollover
00:01:46.870 GMSG: CCM#0 CPEvent = CLOSED --> CPState = NoTCPSocket
00:01:51.300 GMSG: GWEvent = TIMEOUT --> GWState = SrchActive
00:01:51.300 GMSG: CCM#0 CPEvent = CONNECT_REQ --> CPState = AttemptingSocket
00:01:51.300 GMSG: Attempting TCP socket with CCM 10.123.9.2
00:01:51.300 GMSG: CCM#0 CPEvent = SOCKET_ACK --> CPState = BackupCCM
00:01:51.300 GMSG: GWEvent = SOCKET_ACK --> GWState = RegActive
00:01:51.300 GMSG: CCM#0 CPEvent = REGISTER_REQ --> CPState = SentRegister
00:01:51.890 GMSG: CCM#0 CPEvent = LOADID --> CPState = LoadResponse
```

The difference here is that the gateway gets stuck in the '**LoadResponse**' stage and eventually times out. This problem can be resolved by correcting the load file name in the Device Defaults area of Cisco CallManager Administration.

Gatekeeper Problems

Before starting any gateway-to-gatekeeper troubleshooting, verify that there is IP connectivity within the network. Assuming that there is IP connectivity, use the information in this section to troubleshoot your gateway.

Inter-Cluster Trunks Only

Note that gatekeeper control for Cisco CallManager Release 3.0(1) is only available for inter-cluster trunks. Gatekeeper control is configurable for other devices, but the configuration is not supported.

Admission Rejects (ARJ)

ARJs are issued when Cisco CallManager has registered with the Gatekeeper, but can't send a phone call. Configuration issues on the gatekeeper should be the primary focus when the gatekeeper is issuing a ARJ. However, here are the general guidelines for troubleshooting:

1. Verify IP connectivity from the gateway to the gatekeeper.
2. Show gatekeeper status – verify the gatekeeper state is up.
3. Is there a zone subnet defined on the gatekeeper? If so, verify that the subnet of the gateway is in the allowed subnets.

Registration Rejects (RRJ)

RRJs are issued when Cisco CallManager cannot register with the Gatekeeper. Configuration issues on the gatekeeper should be the primary focus when the gatekeeper is issuing a RRJ. However, here are the general guidelines for troubleshooting:

1. Verify IP connectivity from the gateway to the gatekeeper.
2. Show gatekeeper status – verify the gatekeeper state is up.
3. Is there a zone subnet defined on the gatekeeper? If so, verify that the subnet of the gateway is in the allowed subnets.

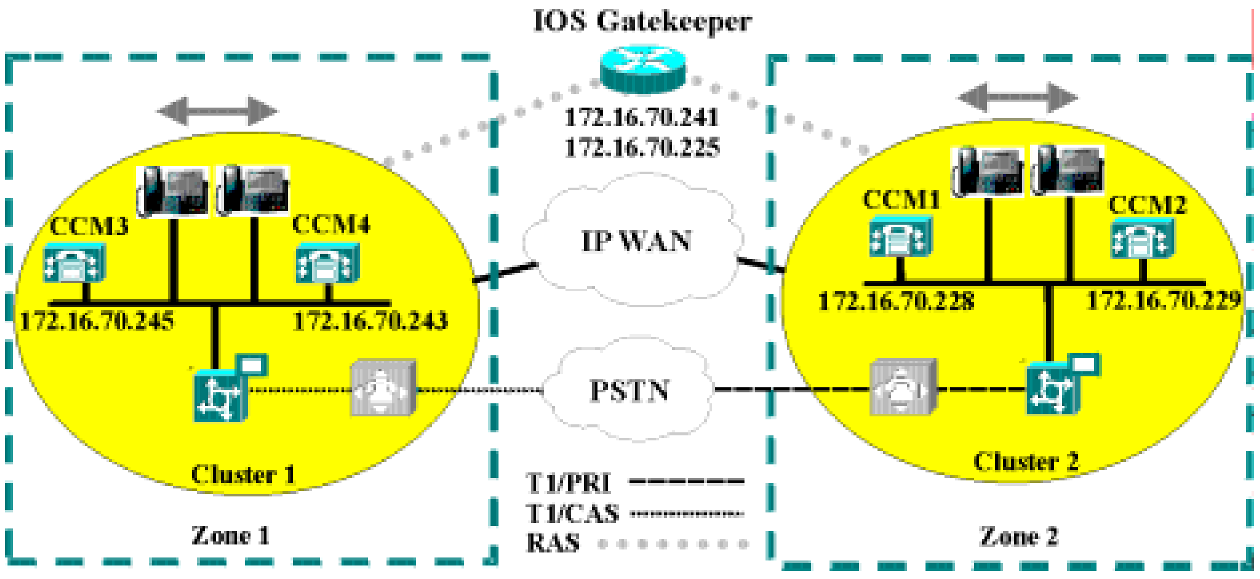
Appendix A – Troubleshooting Case Study 1

Intra-Cluster Cisco IP Phone-to-Cisco IP Phone Calls

This case study discusses in detail the call flow between two Cisco IP Phones within a cluster, called an intra-cluster call. This case study also focuses on Cisco CallManager and Cisco IP Phone initialization, registration, and KeepAlive processes. That is followed by a detailed explanation of an intra-cluster call flow. These processes are explained using trace utilities and tools discussed in a previous section.

Sample Topology

The following diagram depicts the sample topology for this case study. In the diagram there are two clusters named Cluster 1 and Cluster 2. The two Cisco CallManagers in Cluster 1 are called CCM3 and CCM4, while the two Cisco CallManagers in Cluster 1 are named CCM1 and CCM2. The traces collected for this case study are from CCM1, which is located in Cluster 2. The call flow is based on the two Cisco IP Phones in Cluster 2. The IP addresses of these two Cisco IP Phones are 172.16.70.230 (directory number 1000), and 172.16.70.231 (directory number 1001), respectively.



Cisco IP Phone Initialization Process

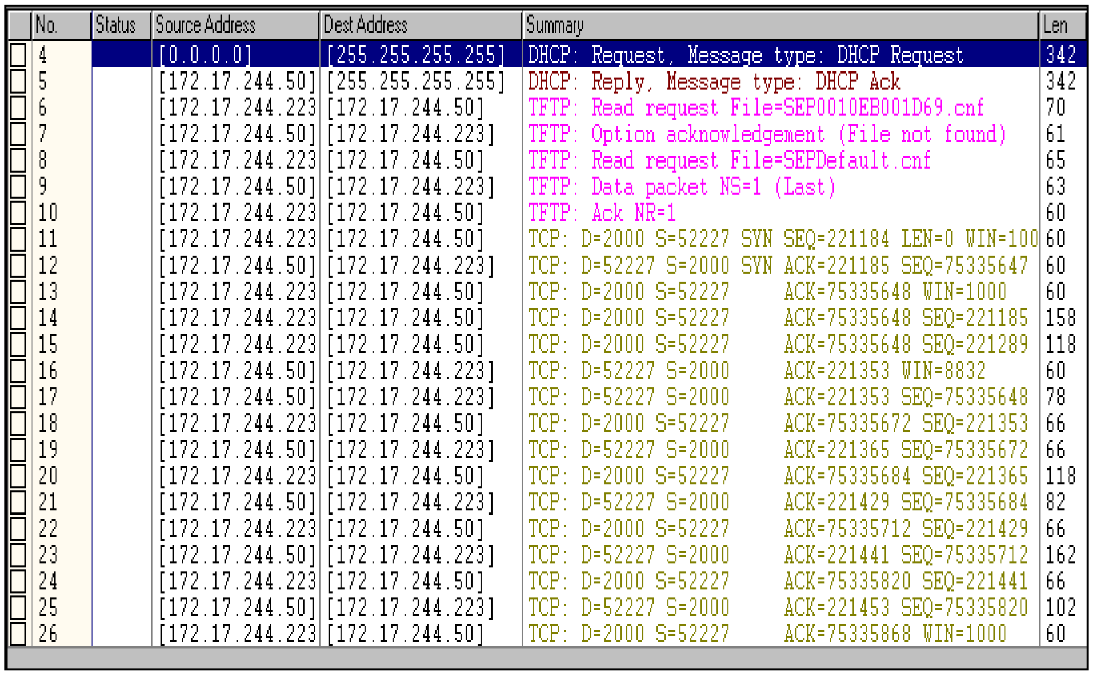
The Cisco IP Phone initialization (or "boot up") process is explained in detail below.

1. At initialization, the Cisco IP Phone sends a request to the DHCP server to get an IP address, DNS server address, and TFTP server name or address, if appropriate options are set in DHCP server (Option 066, Option 150, etc). It also gets a default gateway address if set in DHCP server (Option 003).
2. If a DNS name of the TFTP sever is sent by DHCP, then a DNS sever IP address is required to map the name to an IP address. This step is bypassed if the DHCP server sends the IP

address of the TFTP server. In this case study, the DHCP server sent the IP address of TFTP because DNS was not configured.

3. If a TFTP server name is not included in the DHCP reply, then the Cisco IP Phone uses the default server name.
4. The configuration file (.cnf) file is retrieved from the TFTP server. All .cnf files have the name SEP<mac_address>.cnf, where "SEP" is an acronym for Selsius Ethernet Phone. If this is the first time the phone is registering with the Cisco CallManager, then a default file, SEPdefault.cnf, is downloaded to the Cisco IP Phone. In this case study, the first Cisco IP Phone uses the IP address 172.16.70.230 (it's MAC address is SEP0010EB001720), and the second Cisco IP Phone uses the IP address 172.16.70.231 (it's MAC address is SEP003094C26105).
5. All .cnf files include IP address(es) of the primary and secondary Cisco CallManager(s). The Cisco IP Phone uses the IP address to contact the primary Cisco CallManager and register.
6. Once the Cisco IP Phone has connected and registered with Cisco CallManager, the Cisco CallManager tells the Cisco IP Phone which executable version (called a load ID) to run. If the specified version does not match the executing version on the Cisco IP Phone, the Cisco IP Phone will request the new executable from the TFTP server and reset automatically.

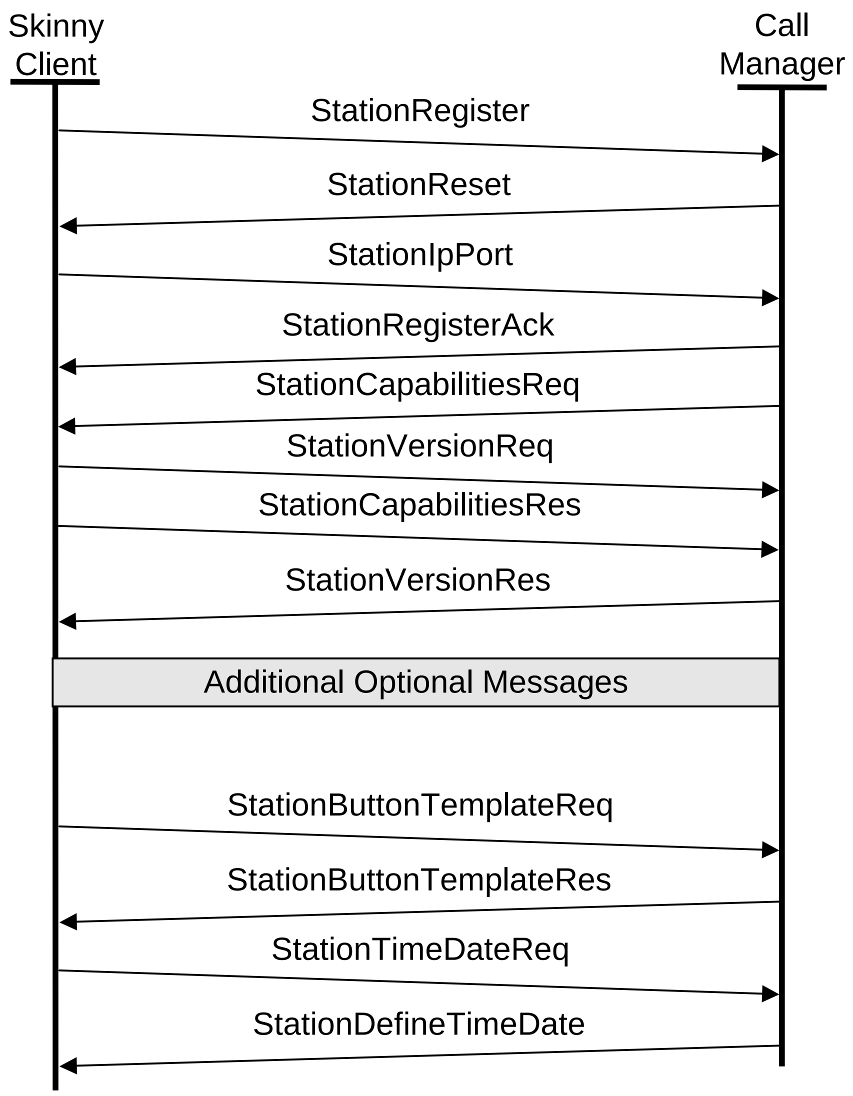
The following Sniffer trace example summarizes the phone initialization process. This trace example is not taken for this case study's sample topology, but it does provide an example of the series of events that occur during the Cisco IP Phone initialization process.



No.	Status	Source Address	Dest Address	Summary	Len
4		[0.0.0.0]	[255.255.255.255]	DHCP: Request, Message type: DHCP Request	342
5		[172.17.244.50]	[255.255.255.255]	DHCP: Reply, Message type: DHCP Ack	342
6		[172.17.244.223	[172.17.244.50]	TFTP: Read request File=SEP0010EB001D69.cnf	70
7		[172.17.244.50]	[172.17.244.223]	TFTP: Option acknowledgement (File not found)	61
8		[172.17.244.223	[172.17.244.50]	TFTP: Read request File=SEPDefault.cnf	65
9		[172.17.244.50]	[172.17.244.223]	TFTP: Data packet NS=1 (Last)	63
10		[172.17.244.223	[172.17.244.50]	TFTP: Ack NR=1	60
11		[172.17.244.223	[172.17.244.50]	TCP: D=2000 S=52227 SYN SEQ=221184 LEN=0 WIN=100	60
12		[172.17.244.50]	[172.17.244.223]	TCP: D=52227 S=2000 SYN ACK=221185 SEQ=75335647	60
13		[172.17.244.223	[172.17.244.50]	TCP: D=2000 S=52227 ACK=75335648 WIN=1000	60
14		[172.17.244.223	[172.17.244.50]	TCP: D=2000 S=52227 ACK=75335648 SEQ=221185	158
15		[172.17.244.223	[172.17.244.50]	TCP: D=2000 S=52227 ACK=75335648 SEQ=221289	118
16		[172.17.244.50]	[172.17.244.223]	TCP: D=52227 S=2000 ACK=221353 WIN=8832	60
17		[172.17.244.50]	[172.17.244.223]	TCP: D=52227 S=2000 ACK=221353 SEQ=75335648	78
18		[172.17.244.223	[172.17.244.50]	TCP: D=2000 S=52227 ACK=75335672 SEQ=221353	66
19		[172.17.244.50]	[172.17.244.223]	TCP: D=52227 S=2000 ACK=221365 SEQ=75335672	66
20		[172.17.244.223	[172.17.244.50]	TCP: D=2000 S=52227 ACK=75335684 SEQ=221365	118
21		[172.17.244.50]	[172.17.244.223]	TCP: D=52227 S=2000 ACK=221429 SEQ=75335684	82
22		[172.17.244.223	[172.17.244.50]	TCP: D=2000 S=52227 ACK=75335712 SEQ=221429	66
23		[172.17.244.50]	[172.17.244.223]	TCP: D=52227 S=2000 ACK=221441 SEQ=75335712	162
24		[172.17.244.223	[172.17.244.50]	TCP: D=2000 S=52227 ACK=75335820 SEQ=221441	66
25		[172.17.244.50]	[172.17.244.223]	TCP: D=52227 S=2000 ACK=221453 SEQ=75335820	102
26		[172.17.244.223	[172.17.244.50]	TCP: D=2000 S=52227 ACK=75335868 WIN=1000	60

Skinny Station Registration Process

Cisco IP Phones communicate with Cisco CallManager using the Cisco Skinny Station Protocol. The registration process allows a Skinny Station, such as a Cisco IP Phone, to inform Cisco CallManager of its existence and to make calling possible. The following figure shows the different messages that are exchanged between the Cisco IP Phone (the “station”) and the Cisco CallManager.



The primary messages in the Skinny Station registration process are described in the following table.

Skinny Station Registration Process Descriptions

Message	**Description**
Station Register	The station sends this message to announce its existence to the controlling Cisco CallManager.
Station Reset	Cisco CallManager sends this message to command the station to reset its processes.
Station IP Port	The station sends this message to provide Cisco CallManager with the UDP port to be used with the RTP stream.
Station Register Acknowledge	Cisco CallManager sends this message to acknowledge the registration of a station.

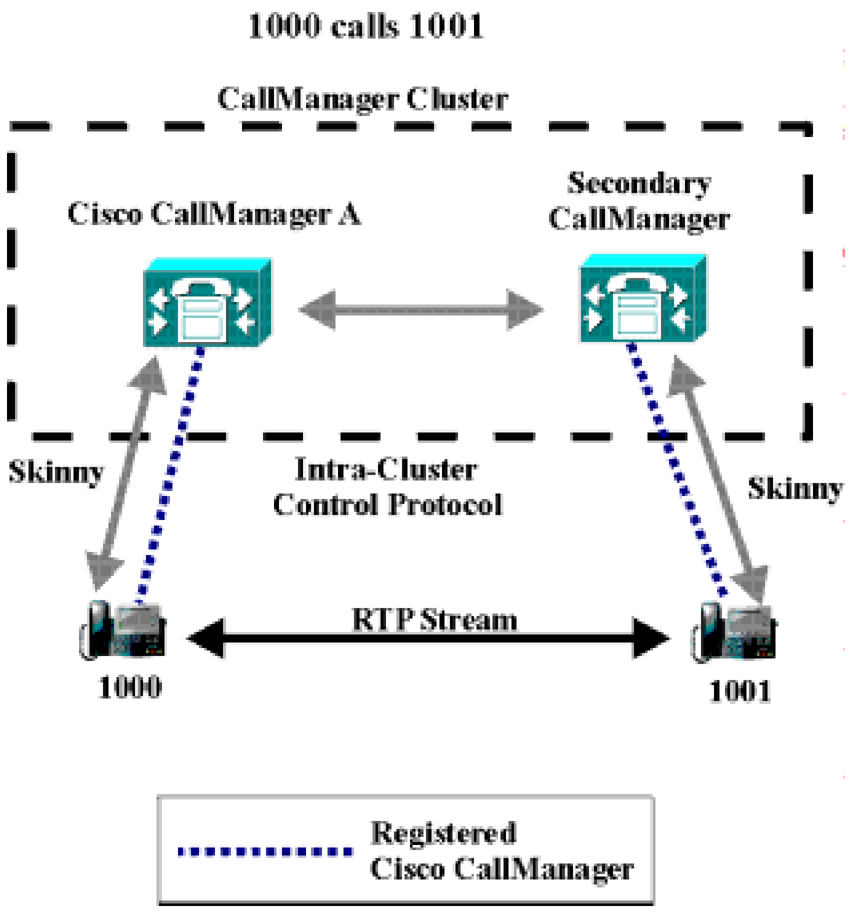
Station Register Reject	Cisco CallManager sends this message to reject a registration attempt from the indicated phone.
	`char text[StationMaxDisplayTextSize];`
	`    };`
	Where:
	`text` is a character string, maximum length of 33 bytes, containing a textual description of the reason that registration is rejected.

Station Capabilities Request	Cisco CallManager sends this message to request the current capabilities of the station. Station capabilities may include compression standard and other H.323 capabilities.
Station Version Request	The station sends this message to request the version number of the software load for the station.
Station Version Response	Cisco CallManager sends this message to inform the station of the appropriate software version number.
Station Capabilities Response	The station sends this message to Cisco CallManager in response to a Station Capabilities Request. The station's capabilities are cached in the Cisco CallManager and used to negotiate terminal capabilities with an H.323 compliant Terminal.
Station Button Template Request	The station sends this message to request the button template definition for that specific terminal or Cisco IP Phone.
Station Button Template Response	Cisco CallManager sends this message to update the button template information contained in the station.
Station Time Date Request	The station sends this message to request the current date and time for internal usage and for displaying as a text string.
Station Define Time and Date	Cisco CallManager uses this message to provide the date and time information to the station. It provides time synchronization for the stations.

Cisco IP Phone-to-Cisco IP Phone Call Flow within a Cluster

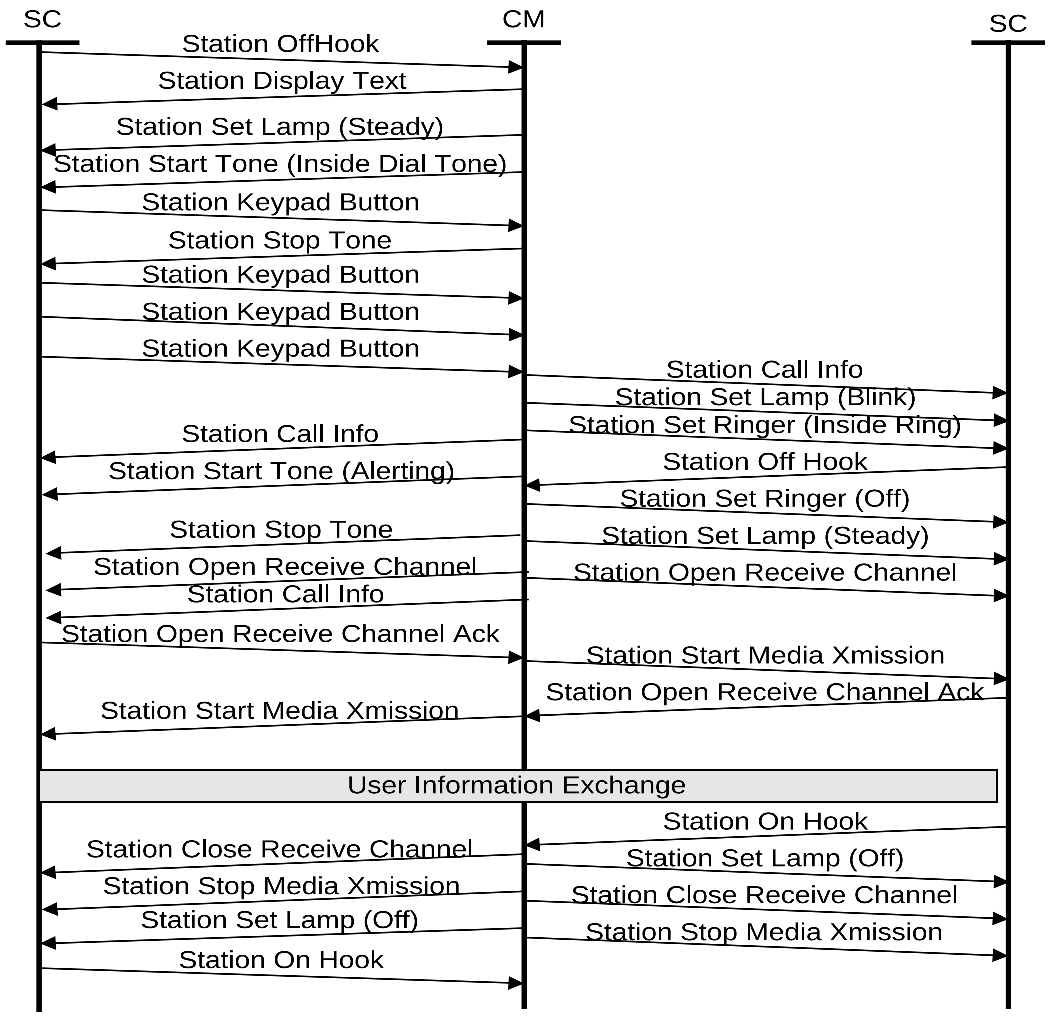
This section describes a Cisco IP Phone (directory number 1000) calling another Cisco IP Phone (directory number 1001) within the same cluster. The cluster is a group of Cisco CallManagers having one common Publisher SQL database and many Subscriber SQL databases.

In our sample topology, CCM1 is the publisher and CCM2 is a subscriber. The two Cisco IP Phones (1000 and 1001) are registered to CCM1 and CCM2, respectively. The call flow is shown in the diagram below. The two Cisco CallManagers within a cluster communicate with each other using Intra-Cluster Control Protocol(ICCP). When a Cisco IP Phone goes off-hook , it opens a control Skinny Station session (with TCP as the underlying protocol) with the Cisco CallManager. After call control signaling is established between the two Cisco IP Phones and their respective Cisco CallManagers, the RTP stream starts flowing directly between the two phones, as shown in the diagram below. The Skinny Station call flow messages for this intra-cluster call are explained in next section.



Cisco IP Phone-to-Cisco IP Phone Exchange of Skinny Station Messages during Call Flow

The following figure shows a sample exchange of messages between two Skinny Stations. The Skinny Station, or Cisco IP Phone, initiates a connection to the Cisco CallManager, and then Cisco CallManager performs digit analysis before opening a control session with the destination Skinny Station. As the following diagram indicates, the Skinny Station messages are written using simple English so they can be readily understood by end-users. Because of this, these messages are not explained in this section. However, these call flow Skinny Station messages are explained in more detail in later sections when traces are being examined.


SC
CM
SC
Station OffHook
Station Display Text
Station Set Lamp (Steady)
Station Start Tone (Inside Dial Tone)
Station Keypad Button
Station Stop Tone
Station Keypad Button
Station Keypad Button
Station Keypad Button
Station Call Info
Station Set Lamp (Blink)
Station Set Ringer (Inside Ring)
Station Call Info
Station Start Tone (Alerting)
Station Off Hook
Station Set Ringer (Off)
Station Stop Tone
Station Set Lamp (Steady)
Station Open Receive Channel
Station Open Receive Channel
Station Call Info
Station Open Receive Channel Ack
Station Start Media Xmission
Station Open Receive Channel Ack
Station Start Media Xmission
User Information Exchange
Station On Hook
Station Close Receive Channel
Station Set Lamp (Off)
Station Stop Media Xmission
Station Close Receive Channel
Station Set Lamp (Off)
Station Stop Media Xmission
Station On Hook

Cisco CallManager Initialization Process

In this section the initialization process of Cisco CallManager will be explained with the help of traces that are captured from CCM1 (identified by the IP address 172.16.70.228). As described previously, SDI traces are a very effective troubleshooting tool because they detail every packet sent between endpoints. This section will describe the events that occur when Cisco CallManager is initialized. Understanding how to read the trace helps you to properly troubleshoot the various Cisco CallManager processes, and the effect of those processes on services such as conferencing, call forwarding, and so on.

The following messages from the Cisco CallManager SDI trace utility show the initialization process on one of the Cisco CallManagers, in this case, CCM1. Review the descriptions of each message below.

- The first message indicates Cisco CallManager started its initialization process.

- The second message indicates Cisco CallManager read the default database values, which, for this case, would be the primary or publisher database.
- The third message indicates Cisco CallManager listened to the various messages on TCP port 8002.
- The fourth message shows that, after listening to these messages, Cisco CallManager added a second Cisco CallManager to its list: CCM2 (172.16.70.229).
- The fifth message indicates that Cisco CallManager has started and is running Cisco CallManager version 3.0.20.

```
16:02:47.765 CCM|CMProcMon - CallManagerState Changed - Initialization Started.
16:02:47.796 CCM|NodeId:  0, EventId: 107 EventClass: 3 EventInfo: Cisco CM Database Defaults Read
16:02:49.937 CCM| SDL Info - NodeId: [1], Listen IP/Hostname: [172.16.70.228], Listen Port: [8002]
16:02:49.984 CCM|dBProcs - Adding SdlLink to NodeId: [2], IP/Hostname: [172.16.70.229]
16:02:51.031 CCM|NodeId:  1, EventId:  1 EventClass:  3 EventInfo: Cisco CallManager
Version=<3.0(0.20)> started
```

Self-Starting Processes

Once Cisco CallManager is up and running, it starts several other processes within itself. Some of these processes are shown below, including MulticastPoint Manager, UnicastBridge Manager, digit analysis, and route list. The messages described during these processes can be very useful when troubleshooting a problem related to the features in Cisco CallManager.

For example, assume that the route lists are not functioning and are unusable. To troubleshoot this problem, you would monitor these traces to determine whether the Cisco CallManager has started RoutePlanManager and if it is trying to load the RouteLists. In the sample configuration below, RouteListName="ipwan" and RouteGroupName="ipwan" are loading and starting.

```
16:02:51.031 CCM|MulicastPointManager - Started
16:02:51.031 CCM|UnicastBridgeManager - Started
16:02:51.031 CCM|MediaTerminationPointManager - Started
16:02:51.125 CCM|MediaCoordinator(1) - started
16:02:51.125 CCM|NodeId:    1, EventId: 1543 EventClass:  2 EventInfo: Database manager started
16:02:51.234 CCM|NodeId:    1, EventId: 1542 EventClass:  2 EventInfo: Link manager started
16:02:51.390 CCM|NodeId:    1, EventId: 1541 EventClass:  2 EventInfo: Digit analysis started
16:02:51.406 CCM|RoutePlanManager - Started, loading RouteLists
16:02:51.562 CCM|RoutePlanManager - finished loading RouteLists
16:02:51.671 CCM|RoutePlanManager - finished loading RouteGroups
16:02:51.671 CCM|RoutePlanManager - Displaying Resulting RoutePlan
16:02:51.671 CCM|RoutePlanServer - RouteList Info, by RouteList and RouteGroup Selection Order
16:02:51.671 CCM|RouteList - RouteListName=”ipwan”
16:02:51.671 CCM|RouteList - RouteGroupName=”ipwan”
16:02:51.671 CCM|RoutePlanServer - RouteGroup Info, by RouteGroup and Device Selection Order
16:02:51.671 CCM|RouteGroup - RouteGroupName=”ipwan”
```

The following trace shows the RouteGroup adding the device 172.16.70.245, which is CCM3 located in Cluster 1 and considered an H.323 device. In this case, the RouteGroup is created to route calls to CCM3 in Cluster 1 with Cisco IOS Gatekeeper permission. If there is a problem routing the call to a Cisco IP Phone located in Cluster 1, then the following messages would help you find the cause of the problem.

```
16:02:51.671 CCM|RouteGroup - DeviceName=”172.16.70.245”
16:02:51.671 CCM|RouteGroup -AllPorts
```

Part of the initialization process shows that Cisco CallManager is adding “Dns” (Directory Numbers). By reviewing these messages, you can determine whether the Cisco CallManager has read the directory number from the database.

```
16:02:51.671 CCM|NodeId:    1, EventId: 1540 EventClass:  2 EventInfo: Call control started
16:02:51.843 CCM|ProcessDb -       Dn = 2XXX,      Line = 0,    Display = ,  RouteThisPattern,
NetworkLocation = OffNet,  DigitDiscardingInstruction = 1,   WhereClause =
16:02:51.859 CCM|Digit analysis: Add local pattern 2XXX , PID: 1,80,1
16:02:51.859 CCM|ForwardManager - Started
16:02:51.984 CCM|CallParkManager - Started
16:02:52.046 CCM|ConferenceManager - Started
```

In the following traces the Device Manager in Cisco CallManager is statically initializing two devices. The device with IP address 172.17.70.226 is a Gatekeeper and the device with IP address 172.17.70.245 is another Cisco CallManager in a different cluster. That Cisco CallManager is registered as an H.323 Gateway with this Cisco CallManager.

```
16:02:52.250 CCM|DeviceManager: Statically Initializing Device;  DeviceName=172.16.70.226
16:02:52.250 CCM|DeviceManager: Statically Initializing Device;  DeviceName=172.16.70.245
```

Cisco CallManager Registration Process

Another important part of the SDI trace is the registration process. When a device is powered up, it gets information via DHCP, connects to the TFTP server for its .cnf file, and then connects to the Cisco CallManager specified in the .cnf file. The device could be an MGCP Gateway, a Skinny Gateway, or a Cisco IP Phone. Therefore, it is important to be able to discover whether or not devices have successfully registered on the Cisco AVVID network.

In the following trace, Cisco CallManager has received new connections for registration. The registering devices are “MTP_nsa-cm1” (MTP services on CCM1), and “CFB_nsa-cm1” (Conference Bridge service on CCM1). These are software services running on Cisco CallManager but are treated internally as different external services and are therefore assigned a TCPHandle, socket number, and port number as well as a device name.

```
16:02:52.750 CCM|StationInit - New connection accepted. DeviceName=, TCPHandle=0x4fbaa00,
Socket=0x594, IPAddr=172.16.70.228, Port=3279, StationD=[0,0,0]
16:02:52.750 CCM|StationInit - New connection accepted. DeviceName=, TCPHandle=0x4fe05e8,
Socket=0x59c, IPAddr=172.16.70.228, Port=3280, StationD=[0,0,0]
16:02:52.781 CCM|StationInit - Processing StationReg. regCount: 1 DeviceName=MTP_nsa-cm1,
TCPHandle=0x4fbaa00, Socket=0x594, IPAddr=172.16.70.228, Port=3279, StationD=[1,45,2]
16:02:52.781 CCM|StationInit - Processing StationReg. regCount: 1 DeviceName=CFB_nsa-cm1,
TCPHandle=0x4fe05e8, Socket=0x59c, IPAddr=172.16.70.228, Port=3280, StationD=[1,96,2]
```

In the following trace, Skinny Station messages are sent between a Cisco IP Phone and Cisco CallManager. The Cisco IP Phone (172.16.70.231) is registering with Cisco CallManager. Refer to the descriptions of Skinny Station messages earlier in this section for more information. As soon as Cisco CallManager receives the registration request from a Cisco IP Phone, it assigns a TCPHandle number to this device. This number remains the same until the device or Cisco CallManager is restarted. Therefore, you can follow all the events related to a particular device by searching for or keeping track of the device's TCPHandle number, which appears in hex. Also, notice that Cisco CallManager provides the load ID to the Cisco IP Phone. Based on this load ID, the Cisco IP Phone runs the executable file (acquired from the TFTP server) that corresponds to the device.

```
16:02:57.000 CCM|StationInit - New connection accepted. DeviceName=, TCPHandle=0x4fbbc30,
Socket=0x5a4, IPAddr=172.16.70.231, Port=52095, StationD=[0,0,0]
16:02:57.046 CCM|NodeId:    1, EventId: 1703 EventClass:  2 EventInfo: Station Alarm, TCP Handle:
4fbbc30, Text: Name=SEP003094C26105  Load=AJ.30  Parms=Status/IPaddr LastTime=A P1:
2304(900) P2: -414838612(e74610ac)
16:02:57.046 CCM|StationInit - ***** InboundStim - AlarmMessageID tcpHandle=0x4fbbc30
Message="Name=SEP003094C26105  Load=AJ.30  Parms=Status/IPaddr LastTime=A" Parm1=2304
(900) Parm2=-414838612 (e74610ac)
16:02:57.093 CCM|StationInit - Processing StationReg. regCount: 1 DeviceName=SEP003094C26105,
TCPHandle=0x4fbbc30, Socket=0x5a4, IPAddr=172.16.70.231, Port=52095, StationD=[1,85,1]
16:02:57.093 CCM|StationInit - InboundStim - IpPortMessageID: 32715(0x7fcb) tcpHandle=0x4fbbc30
```

Cisco CallManager KeepAlive Process

Both the station, device, or service and the Cisco CallManager use the following messages to maintain a knowledge of the communications channel between them. The messages are used to begin the KeepAlive sequence that ensures that the communications link between the Cisco CallManager and the station remains active. The following messages can originate from either the Cisco CallManager or the station.

```
16:03:02.328 CCM|StationInit - InboundStim - KeepAliveMessage - Forward KeepAlive to StationD.
DeviceName=MTP_nsa-cm2, TCPHandle=0x4fa7dc0, Socket=0x568, IPAddr=172.16.70.229, Port=1556,
StationD=[1,45,1]
16:03:02.328 CCM|StationInit - InboundStim - KeepAliveMessage - Forward KeepAlive to StationD.
DeviceName=CFB_nsa-cm2, TCPHandle=0x4bf8a70, Socket=0x57c, IPAddr=172.16.70.229, Port=1557,
StationD=[1,96,1]
16:03:06.640 CCM|StationInit - InboundStim - KeepAliveMessage - Forward KeepAlive to StationD.
DeviceName=SEP0010EB001720, TCPHandle=0x4fbb150, Socket=0x600, IPAddr=172.16.70.230,
Port=49211, StationD=[1,85,2]
16:03:06.703 CCM|StationInit - InboundStim - KeepAliveMessage - Forward KeepAlive to StationD.
DeviceName=SEP003094C26105, TCPHandle=0x4fbbc30, Socket=0x5a4, IPAddr=172.16.70.231,
Port=52095, StationD=[1,85,1]
```

The messages in the following trace depict the KeepAlive sequence which indicates that the communications link between the Cisco CallManager and the station is active. Again, these messages can originate either by the Cisco CallManager or the station.

```
16:03:02.328 CCM|MediaTerminationPointControl - stationOutputKeepAliveAck tcpHandle=4fa7dc0
16:03:02.328 CCM|UnicastBridgeControl - stationOutputKeepAliveAck tcpHandle=4bf8a70
16:03:06.703 CCM|StationInit - InboundStim - IpPortMessageID: 32715(0x7fcb) tcpHandle=0x4fbbc30
16:03:06.703 CCM|StationD - stationOutputKeepAliveAck tcpHandle=0x4fbbc30
```

Cisco CallManager Intra-Cluster Call Flow Traces

The following SDI traces explore in detail the intra-cluster call flow. The Cisco IP Phones in the call flow can be identified by the directory number (dn), tcpHandle, and IP address. A Cisco IP Phone (dn: 1001, tcpHandle: 0x4fbbc30, IP address: 172.16.70.231) located in Cluster 2 is calling another Cisco IP Phone in the same Cluster (dn=1000, tcpHandle= 0x4fbb150, IP address= 172.16.70.230). Remember that you can follow a device through the trace by looking at the TCP handle value, time stamp, or name of the device. The TCP handle value for the device remains the same until the device is rebooted or goes offline.

The following traces show that the Cisco IP Phone (1001) has gone off-hook. The trace below shows the unique messages, TCP handle, and the called number, which are displayed on the Cisco IP Phone. There is no calling number at this point, because the user has not tried to dial any digits. The information below is in the form of Skinny Station messages between the Cisco IP Phones and the Cisco CallManager.

```
16:05:41.625 CCM|StationInit - InboundStim - OffHookMessageID tcpHandle=0x4fbbc30
16:05:41.625 CCM|StationD - stationOutputDisplayText tcpHandle=0x4fbbc30, Display= 1001
```

The next trace shows Skinny Station messages going from Cisco CallManager to a Cisco IP Phone. The first message is to turn on the lamp on the calling party's Cisco IP Phone.

```
16:05:41.625 CCM|StationD - stationOutputSetLamp stim: 9=Line instance=1 lampMode=LampOn
tcpHandle=0x4fbbc30
```

The stationOutputCallState message is used by Cisco CallManager to notify the station of certain call related information.

```
16:05:41.625 CCM|StationD - stationOutputCallState tcpHandle=0x4fbbc30
```

The stationOutputDisplayPromptStatus message is used by Cisco CallManager to cause a call-related prompt message to be displayed on the Cisco IP Phone.

```
16:05:41.625 CCM|StationD - stationOutputDisplayPromptStatus tcpHandle=0x4fbbc30
```

The stationOutputSelectSoftKey message is used by Cisco CallManager to cause the Skinny Station to select a specific set of soft keys.

```
16:05:41.625 CCM|StationD - stationOutputSelectSoftKeys tcpHandle=0x4fbbc30
```

The next message is used by Cisco CallManager to instruct the Skinny Station as to the correct line context for the display.

```
16:05:41.625 CCM|StationD - stationOutputActivateCallPlane tcpHandle=0x4fbbc30
```

In the following message, the digit analysis process is ready to identify incoming digits and check them for potential routing matches in the database. The entry, cn=1001, represents the calling party number. dd="" represents the dialed digit, which would show the called part number. Note that StationInit messages are sent by the phone, StationD messages are sent by Cisco CallManager, and digit analysis is performed by Cisco CallManager.

```
16:05:41.625 CCM|Digit analysis: match(fqcn="", cn="1001", pss="", dd="")
16:05:41.625 CCM|Digit analysis: potentialMatches=PotentialMatchesExist
```

The following debug message shows that the Cisco CallManager is providing inside dial tone to the calling party Cisco IP Phone.

```
16:05:41.625 CCM|StationD - stationOutputStartTone: 33=InsideDialTone tcpHandle=0x4fbbc30
```

Once Cisco CallManager detects an incoming message and recognizes the keypad button 1 has been pressed on the Cisco IP Phone, it immediately stops the output tone.

```
16:05:42.890 CCM|StationInit - InboundStim - KeypadButtonMessageID kpButton: 1
tcpHandle=0x4fbbc30
16:05:42.890 CCM|StationD - stationOutputStopTone tcpHandle=0x4fbbc30
16:05:42.890 CCM|StationD - stationOutputSelectSoftKeys tcpHandle=0x4fbbc30
16:05:42.890 CCM|Digit analysis: match(fqcn="", cn="1001", pss="", dd="1")
16:05:42.890 CCM|Digit analysis: potentialMatches=PotentialMatchesExist
16:05:43.203 CCM|StationInit - InboundStim - KeypadButtonMessageID kpButton: 0
tcpHandle=0x4fbbc30
16:05:43.203 CCM|Digit analysis: match(fqcn="", cn="1001", pss="", dd="10")
16:05:43.203 CCM|Digit analysis: potentialMatches=PotentialMatchesExist
16:05:43.406 CCM|StationInit - InboundStim - KeypadButtonMessageID kpButton: 0
tcpHandle=0x4fbbc30
16:05:43.406 CCM|Digit analysis: match(fqcn="", cn="1001", pss="", dd="100")
16:05:43.406 CCM|Digit analysis: potentialMatches=PotentialMatchesExist
16:05:43.562 CCM|StationInit - InboundStim - KeypadButtonMessageID kpButton: 0
tcpHandle=0x4fbbc30
16:05:43.562 CCM|Digit analysis: match(fqcn="", cn="1001", pss="", dd="1000")
```

Once the Cisco CallManager has received enough digits to match, it provides the digit analysis results in a table format. Any extra digits pressed on the phone after this point will be ignored by Cisco CallManager, since a match has already been found.

```
16:05:43.562 CCM|Digit analysis: analysis results
16:05:43.562 CCM||PretransformCallingPartyNumber=1001
|CallingPartyNumber=1001
|DialingPattern=1000
|DialingRoutePatternRegularExpression=(1000)
|PotentialMatches=PotentialMatchesExist
|DialingSdlProcessId=(1,38,2)
|PretransformDigitString=1000
|PretransformPositionalMatchList=1000
|CollectedDigits=1000
|PositionalMatchList=1000
|RouteBlockFlag=RouteThisPattern
```

The next trace shows that Cisco CallManager is sending out this information to a called party phone (the phone is identified by the tcpHandle number).

```
16:05:43.578 CCM|StationD - stationOutputCallInfo CallingPartyName=1001,  CallingParty=1001,
CalledPartyName=1000, CalledParty=1000, tcpHandle=0x4fbb150
```

The next trace indicates that Cisco CallManager is ordering the lamp to blink for incoming call indication on the called party's Cisco IP Phone.

```
16:05:43.578 CCM|StationD - stationOutputSetLamp stim: 9=Line instance=1 lampMode=LampBlink
tcpHandle=0x4fbb150
```

In the following traces, Cisco CallManager is providing ringer, display notification, and other call-related information to the called party's Cisco IP Phone. Again, you can see that all messages are directed to the same Cisco IP Phone because the same tcpHandle is used throughout the traces.

```
16:05:43.578 CCM|StationD - stationOutputSetRinger: 2=InsideRing tcpHandle=0x4fbb150
16:05:43.578 CCM|StationD - stationOutputDisplayNotify tcpHandle=0x4fbb150
16:05:43.578 CCM|StationD - stationOutputDisplayPromptStatus tcpHandle=0x4fbb150
16:05:43.578 CCM|StationD - stationOutputSelectSoftKeys tcpHandle=0x4fbb150
```

Notice that Cisco CallManager is also providing similar information to the calling party's Cisco IP Phone. Again, the tcpHandle is used to differentiate between Cisco IP Phones.

```
16:05:43.578 CCM|StationD - stationOutputCallInfo CallingPartyName=1001,  CallingParty=1001,
CalledPartyName=, CalledParty=1000, tcpHandle=0x4fbbc30
16:05:43.578 CCM|StationD - stationOutputCallInfo CallingPartyName=1001,  CallingParty=1001,
CalledPartyName=1000, CalledParty=1000, tcpHandle=0x4fbbc30
```

In the next trace, Cisco CallManager provides an alerting or ringing tone to the calling party's Cisco IP Phone, notifying that the connection has been established.

```
16:05:43.578 CCM|StationD - stationOutputStartTone: 36=AlertingTone tcpHandle=0x4fbbc30
16:05:43.578 CCM|StationD - stationOutputCallState tcpHandle=0x4fbbc30
16:05:43.578 CCM|StationD - stationOutputSelectSoftKeys tcpHandle=0x4fbbc30
16:05:43.578 CCM|StationD - stationOutputDisplayPromptStatus tcpHandle=0x4fbbc30
```

At this point, the called party's Cisco IP Phone goes off-hook. Therefore, Cisco CallManager stops generating the ringer tone to calling party.

```
16:05:45.140 CCM|StationD - stationOutputStopTone tcpHandle=0x4fbbc30
```

In the following messages, Cisco CallManager causes the Skinny Station to begin receiving a Unicast RTP stream. To do so, Cisco CallManager provides the IP address of the called party as well as codec information, and packet size in msec (milliseconds). PacketSize is an integer containing the sampling time in milliseconds used to create the RTP packets. NOTE: normally this value is set to 30msec. In this case it is set to 20msec.

```
16:05:45.140 CCM|StationD - stationOutputOpenReceiveChannel tcpHandle=0x4fbbc30 myIP: e74610ac
(172.16.70.231)
16:05:45.140 CCM|StationD - ConferenceID: 0 msecPacketSize: 20
compressionType:(4)Media_Payload_G711Ulaw64k
```

Similarly, Cisco CallManager provides information to the called party (1000).

```
16:05:45.140 CCM|StationD - stationOutputOpenReceiveChannel tcpHandle=0x4fbb150 myIP: e64610ac
(172.16.70.230)
16:05:45.140 CCM|StationD - ConferenceID: 0 msecPacketSize: 20
compressionType:(4)Media_Payload_G711Ulaw64k
```

Cisco CallManager has received the acknowledgment message from called party for establishing the open channel for RTP stream, as well as the IP address of the called party. This message is to inform the Cisco CallManager of two pieces of information about the Skinny Station. First, it contains the status of the open action. Second, it contains the receive port address and number for transmission to the remote end. The IP address of the transmitter (calling part) of the RTP stream is ipAddr, and PortNumber is the IP port number of the RTP stream transmitter (calling party).

```
16:05:45.265 CCM|StationInit - InboundStim - StationOpenReceiveChannelAckID tcpHandle=0x4fbb150,
Status=0, IpAddr=0xe64610ac, Port=17054, PartyID=2
```

The following messages are used by Cisco CallManager to order the station to begin transmitting the audio stream to the indicated remote Cisco IP Phone's IP address and port number.

```
16:05:45.265 CCM|StationD - stationOutputStartMediaTransmission tcpHandle=0x4fbbc30 myIP:
e74610ac (172.16.70.231)
16:05:45.265 CCM|StationD - RemoteIpAddr: e64610ac (172.16.70.230) RemoteRtpPortNumber: 17054
msecPacketSize: 20 compressionType:(4)Media_Payload_G711Ulaw64k
```

In the following traces, the previously explained messages are sent to the called party. These messages are followed by the messages indicating the RTP media stream has been started between the called and calling party.

```
16:05:45.312 CCM|StationD - stationOutputStartMediaTransmission tcpHandle=0x4fbb150 myIP:
e64610ac (172.16.70.230)
16:05:45.328 CCM|StationD - RemoteIpAddr: e74610ac (172.16.70.231) RemoteRtpPortNumber: 18448
msecPacketSize: 20 compressionType:(4)Media_Payload_G711Ulaw64k
16:05:46.203 CCM|StationInit - InboundStim - OnHookMessageID tcpHandle=0x4fbbc30
```

The calling party's Cisco IP Phone finally goes on-hook, which terminates all the control messages between the Skinny Station and Cisco CallManager as well as the RTP stream between Skinny Stations.

```
16:05:46.203 CCM|StationInit - InboundStim - OnHookMessageID tcpHandle=0x4fbbc30
```

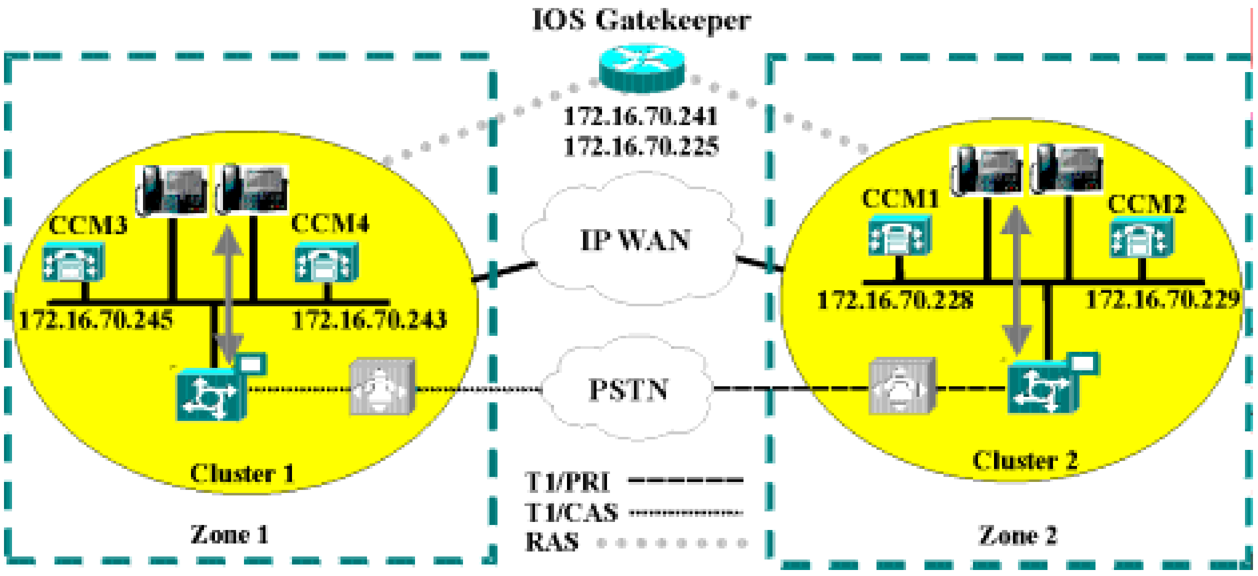
Appendix B – Troubleshooting Case Study 2

Cisco IP Phone-to-Cisco IOS Gateway Calls

In the previous case study, the call flow and troubleshooting techniques of an intra-cluster call was discussed in detail. This case study examines a Cisco IP Phone calling through a Cisco IOS Gateway to a phone hanging off of a local PBX or somewhere on the PSTN. Conceptually, when the call reaches the Cisco IOS Gateway, the gateway will forward the call either to a phone hanging off of its FXS port, or to the PBX. If the call is forwarded to the PBX, it could terminate to a phone hanging off of a local PBX or the PBX will forward it over the PSTN and the call will terminate somewhere on the PSTN.

Sample Topology

The following diagram shows the sample topology for this case study. Calls are routed through Cisco IOS Gateways and the interface to the PSTN or PBX is either T1/CAS or T1/PRI. The gateways can be models 26XX, 36XX, 53XX or 6K.



Call Flow Traces

This section discusses call flow through examples from the Cisco CallManager trace file CCM000000000. Refer to the previous section for the location of the file. The traces in this case study focus only on the call flow itself, as the more detailed trace information has already been explained in the previous case study (initialization, registration, KeepAlive mechanism, and so on).

In this call flow, a Cisco IP Phone (directory number 1001) located in the cluster 2 is calling a phone (directory number 3333) located somewhere on the PSTN. Remember that you can follow a device through the trace by looking at the TCP handle value, time stamp, or name of the device. The TCP handle value for the device remains the same until the device is rebooted or goes offline.

In the following traces, the Cisco IP Phone (1001) has gone off-hook. The trace shows the unique messages, TCP handle, and the calling number, which is displayed on the Cisco IP Phone. There is no called number at this point, because the user has not tried to dial any digits.

```
16:05:46.37515:20:18.390 CCM|StationInit - InboundStim – OffHookMessageID tcpHandle=0x5138d98

15:20:18.390 CCM|StationD - stationOutputDisplayText tcpHandle=0x5138d98, Display=1001
```

In the following traces, the user is dialing the 3333, one digit at a time. The number 3333 is the destination number of the phone, which is located somewhere on the PSTN network. The digit analysis process of the Cisco CallManager is currently active and is analyzing the digits to discover where the call needs to get routed. A more detailed explanation of the digit analysis was provided in the previous case study.

```
15:20:18.390 CCM|Digit analysis: match(fqcn="", cn="1001", pss="", dd="")
15:20:19.703 CCM|Digit analysis: match(fqcn="", cn="1001", pss="", dd="3")
15:20:20.078 CCM|Digit analysis: match(fqcn="", cn="1001", pss="", dd="33")
15:20:20.718 CCM|Digit analysis: match(fqcn="", cn="1001", pss="", dd="333")
15:20:21.421 CCM|Digit analysis: match(fqcn="", cn="1001", pss="", dd="3333")
15:20:21.421 CCM|Digit analysis: analysis results
```

In the following traces, the digit analysis has been completed, calling and called party has been matched, and the information has been parsed.

```
|CallingPartyNumber=1001
|DialingPattern=3333
|DialingRoutePatternRegularExpression=(3333)
|PretransformDigitString=3333
|PretransformPositionalMatchList=3333
|CollectedDigits=3333
|PositionalMatchList=3333
```

In the following traces, the number 0 indicates the originating location, and the number 1 indicates the destination location. The bandwidth of the originating location is determined by BW = –1. The value –1 implies that the bandwidth is infinite. The bandwidth is infinite because the call was originated from a Cisco IP Phone located in a LAN environment. The bandwidth of the destination location is determined by BW = 64. The call destination is to a phone located in a PSTN, and the codec type is used is G.711 (64Kbps).

```
15:20:21.421 CCM|Locations:Orig=0 BW=-1 Dest=1 BW=64 (-1 implies infinite bw available)
```

The following traces show the calling and called party information. In this example, the calling party name and number is the same because the administrator has not configured a display name, such as “John Smith.”

```
15:20:21.421 CCM|StationD - stationOutputCallInfo CallingPartyName=1001,  CallingParty=1001,
CalledPartyName=, CalledParty=3333, tcpHandle=0x5138d98
```

Before reviewing the following traces, it is important to understand the meaning of the term H.323. By way of a brief explanation, there are several protocols that are used when establishing an H.323 session. One protocol is H.225, which is primarily used for call signaling and is a subset of Q.931. Another protocol is H.245, which is used for capability exchange. One of the more important functions of H.245 is the Compressor/Decompressor (codec) type negotiation, such as G.711, G.729, and so on, between the calling and called side. Once the capability exchange is complete, the next important function of H.245 is performing a UDP port negotiation between the calling and called sides.

The following trace shows that the H.323 code has been initialized and is sending an H.225 setup message. You can also see the traditional HDLC SAPI messages, the IP address of the called side in hex, and the port numbers.

```
15:20:21.421 CCM|Out Message -- H225SetupMsg -- Protocol= H225Protocol
15:20:21.421 CCM|MMan_Id= 1. (iep=  0 dsl=  0 sapi=  0 ces=  0 IpAddr=e24610ac IpPort=47110)
```

The following trace shows the calling and called party information as well as the H.225 alerting message. Also shown is the mapping of a Cisco IP Phone’s hex value to the IP address. 172.16.70.231 is the IP address of the Cisco IP Phone (1001).

```
15:20:21.437 CCM|StationD - stationOutputCallInfo CallingPartyName=1001,  CallingParty=1001,
CalledPartyName=, CalledParty=3333, tcpHandle=0x5138d98
15:20:21.453 CCM|In  Message -- H225AlertMsg -- Protocol= H225Protocol
15:20:21.953 CCM|StationD - stationOutputOpenReceiveChannel tcpHandle=0x5138d98 myIP:
e74610ac (172.16.70.231)
```

The following trace shows the compression type used for this call (G.711 µ-law).

```
15:20:21.953 CCM|StationD - ConferenceID: 0 msecPacketSize: 20
compressionType:(4)Media_Payload_G711Ulaw64k
```

Once the H.225 alert message has been sent, the next part of H.323 is initialize: H.245. The following trace shows the calling and called party information, and the H.245 messages. Notice the TCP handle value is the same as before, indicating this is the continuation of the same call.

```
15:20:22.062 CCM|H245Interface(3) paths established ip = e74610ac, port = 23752
15:20:22.062 CCM|H245Interface(3) OLC outgoing confirm ip = e24610ac, port = 16758
15:20:22.062 CCM|MediaManager - wait_AuConnectInfo - received response, forwarding
```

The following trace shows the H.225 connect message, as well as other information that was explained earlier. When the H.225 connect message is received, the call has been connected.

```
15:20:22.968 CCM|In  Message -- H225ConnectMsg -- Protocol= H225Protocol
15:20:22.968 CCM|StationD - stationOutputCallInfo CallingPartyName=1001,  CallingParty=1001,
CalledPartyName=, CalledParty=3333, tcpHandle=0x5138d98
15:20:22.062 CCM|MediaCoordinator - wait_AuConnectInfoInd
```

```
15:20:22.062 CCM|StationD - stationOutputStartMediaTransmission tcpHandle=0x5138d98 myIP:
e74610ac (172.16.70.231)
15:20:22.062 CCM|StationD - RemoteIpAddr: e24610ac (172.16.70.226) RemoteRtpPortNumber: 16758
msecPacketSize: 20 compressionType:(4)Media_Payload_G711Ulaw64k
15:20:22.062 CCM|Locations:Orig=0 BW=-1Dest=1 BW=6(-1 implies infinite bw available)
```

The following message shows that an on-hook message from the Cisco IP Phone (1001) is being received. As soon as an on-hook message is received, the H.225 and Skinny disconnect messages are sent and the entire H.225 message is seen. This final message indicates the call has been terminated.

```
15:20:27.296 CCM|StationInit - InboundStim – OnHookMessageID tcpHandle=0x5138d98
15:20:27.296 CCM|ConnectionManager -wait_AuDisconnectRequest (16777247,16777248): STOP
SESSION
15:20:27.296 CCM|MediaManager - wait_AuDisconnectRequest - StopSession sending disconnect to
(64,5) and remove connection from list
15:20:27.296 CCM| Device SEP003094C26105 , UnRegisters with SDL Link to monitor NodeID= 1
15:20:27.296 CCM|StationD - stationOutputCloseReceiveChannel tcpHandle=0x5138d98 myIP:
e74610ac (172.16.70.231)
15:20:27.296 CCM|StationD - stationOutputStopMediaTransmission tcpHandle=0x5138d98 myIP:
e74610ac (172.16.70.231)
15:20:28.328 CCM|In  Message -- H225ReleaseCompleteMsg -- Protocol= H225Protocol
```

Debug Messages and Show Commands on the Cisco IOS Gatekeeper

In the previous section, the Cisco CallManager SDI trace was discussed in detail. In the topology for this case study, “debug ras” has been turned on in the Cisco IOS Gatekeeper.

The following debug messages show that the Cisco IOS Gatekeeper is receiving the admission request (ARQ) for the Cisco CallManager (172.16.70.228), followed by other successful RAS messages. Finally, the Cisco IOS Gatekeeper sends an admission confirmed (ACF) message to the Cisco CallManager.

```
*Mar 12 04:03:57.181: RASLibRASRecvData ARQ (seq# 3365) rcvd from [172.16.70.228883] on sock
[0x60AF038C]
*Mar 12 04:03:57.181: RASLibRAS_WK_TInit ipsock [0x60A7A68C] setup successful
*Mar 12 04:03:57.181: RASlibras_sendto msg length 16 from 172.16.70.2251719 to 172.16.70.228883
*Mar 12 04:03:57.181: RASLibRASSendACF ACF (seq# 3365) sent to 172.16.70.228
```

The following debug messages show that the call is in progress.

```
*Mar 12 04:03:57.181: RASLibRASRecvData successfully rcvd message of length 55 from
172.16.70.228883
```

The following debug messages shows that the Cisco IOS Gatekeeper has received a disengaged request (DRQ) from the Cisco CallManager (172.16.70.228), and the Cisco IOS Gatekeeper has sent a disengage confirmed (DCF) to the Cisco CallManager.

```
*Mar 12 04:03:57.181: RASLibRASRecvData DRQ (seq# 3366) rcvd from [172.16.70.228883] on sock
[0x60AF038C]
```

```
*Mar 12 04:03:57.181: RASlibras_sendto msg length 3 from 172.16.70.2251719 to 172.16.70.228883
*Mar 12 04:03:57.181: RASLibRASSendDCF DCF (seq# 3366) sent to 172.16.70.228
*Mar 12 04:03:57.181: RASLibRASRecvData successfully rcvd message of length 124 from
172.16.70.228883
```

The command “show gatekeeper endpoints” on the Cisco IOS Gatekeeper shows that all four Cisco CallManagers are registered with the Cisco IOS Gatekeeper. Remember that in the topology for this case study, there are four Cisco CallManagers, two in each cluster. This Cisco IOS Gatekeeper has two zones and each zone has two Cisco CallManagers.

```
R2514-1#show gatekeeper endpoints
                GATEKEEPER ENDPOINT REGISTRATION
                ================================
CallSignalAddr  Port  RASSignalAddr   Port  Zone Name       Type
--------------- ----- --------------- ----- ---------       ----   --
172.16.70.228      2     172.16.70.228      1493  gka.cisco.com     VOIP-GW
   H323-ID: ac1046e4->ac1046f5
172.16.70.229      2     172.16.70.229      3923  gka.cisco.com     VOIP-GW
   H323-ID: ac1046e5->ac1046f5
172.16.70.245      1     172.16.70.245      1041  gkb.cisco.com     VOIP-GW
   H323-ID: ac1046f5->ac1046e4
172.16.70.243       1     172.16.70.243      2043  gkb.cisco.com     VOIP-GW
   H323-ID: ac1046f5->ac1046e4
Total number of active registrations = 4
```

Debug Messages and Show Commands on the Cisco IOS Gateway

In the previous section, the Cisco IOS Gatekeeper show commands and debug outputs were discussed in detail. This section focuses on the debug output and show commands on the Cisco IOS Gateway. In the topology for this case study, calls are going through the Cisco IOS Gateways. The Cisco IOS Gateway interfaces to the PSTN or PBX with either T1/CAS or T1/PRI interfaces. Debug output of commands such as debug voip ccapi inout, debug h225 events and debug h225 asn1, are shown below.

In the following debug output, the Cisco IOS Gateway is accepting the TCP connection request from Cisco CallManager (172.16.70.228) on port 2328 for H.225.

```
*Mar 12 04:03:57.169: H225Lib::h225TAccept: TCP connection accepted from 172.16.70.228:2328 on
socket [1]
*Mar 12 04:03:57.169: H225Lib::h225TAccept: Q.931 Call State is initialized to be [Null].
*Mar 12 04:03:57.177: Hex representation of the received TPKT03000065080000100
```

The following debug output shows that the H.225 data is coming from the Cisco CallManager on this TCP session. Important to notice in this debug output is the protocolIdentifier, which indicates the H.323 version being used. The following debug shows that H.323 version 2 is being used. The called and calling party numbers are also shown.

```
- Source Address H323-ID
- Destination Address e164
```

```
*Mar 12 04:03:57.177:      H225Lib::h225RecvData: Q.931 SETUP received from socket [1]value H323-
UserInformation ::=
*Mar 12 04:03:57.181: {
*Mar 12 04:03:57.181:   h323-uu-pdu
*Mar 12 04:03:57.181:   {
*Mar 12 04:03:57.181:     h323-message-body setup :
*Mar 12 04:03:57.181:       {
*Mar 12 04:03:57.181:         protocolIdentifier { 0 0 8 2250 0 2 },
*Mar 12 04:03:57.181:         sourceAddress
*Mar 12 04:03:57.181:         {
*Mar 12 04:03:57.181:           h323-ID : "1001"
*Mar 12 04:03:57.181:         },
*Mar 12 04:03:57.185:         destinationAddress
*Mar 12 04:03:57.185:         {
*Mar 12 04:03:57.185:           e164 : "3333"
*Mar 12 04:03:57.185:         },
*Mar 12 04:03:57.189:      H225Lib::h225RecvData: State changed to [Call Present].
```

The following is debug output for Call Control Application Programming interface (CCAPi). Call control APi is indicating an incoming call. Called and calling party information can also be seen in the following output. CCAPi matches the dial peer 0, which is the default dial peer. It is matching dial peer 0 because the CCAPi could not find any other dial peer for the calling number, so it is using the default dial peer.

```
*Mar 12 04:03:57.189: cc_api_call_setup_ind (vdbPtr=0x616C9F54, callInfo={called=3333, calling=1001,
fdest=1 peer_tag=0}, callID=0x616C4838)
*Mar 12 04:03:57.193: cc_process_call_setup_ind (event=0x617A2B18) handed call to app "SESSION"
*Mar 12 04:03:57.193: sess_appl: ev(19=CC_EV_CALL_SETUP_IND), cid(17), disp(0)
*Mar 12 04:03:57.193: ccCallSetContext (callID=0x11, context=0x61782BBC)
Mar 12 04:03:57.193: ssaCallSetupInd finalDest cllng(1001), clled(3333)
*Mar 12 04:03:57.193: ssaSetupPeer cid(17) peer list:  tag(1)
*Mar 12 04:03:57.193: ssaSetupPeer cid(17), destPat(3333), matched(4), prefix(), peer(6179E63C)
*Mar 12 04:03:57.193: ccCallSetupRequest (peer=0x6179E63C, dest=, params=0x61782BD0 mode=0,
*callID=0x617A87C0)
*Mar 12 04:03:57.193: callingNumber=1001, calledNumber=3333, redirectNumber=
*Mar 12 04:03:57.193: accountNumber=,finalDestFlag=1,
guid=0098.89c8.9233.511d.0300.cddd.ac10.46e6
```

The CCAPi matches the dial-peer 1 with the destination pattern, which is the called number 3333. Keep in mind that peer_tag means dial peer. Notice the calling and called party number in the request packet.

```
*Mar 12 04:03:57.193: peer_tag=1
*Mar 12 04:03:57.197: ccIFCallSetupRequest: (vdbPtr=0x617BE064, dest=, callParams={called=3333,
calling=1001, fdest=1, voice_peer_tag=1}, mode=0x0)
```

The following debug output shows the H.225 Alerting messages are returning to the Cisco CallManager.

```
*Mar 12 04:03:57.197: ccCallSetContext (callID=0x12, context=0x61466B30)
*Mar 12 04:03:57.197: ccCallProceeding (callID=0x11, prog_ind=0x0)
*Mar 12 04:03:57.197: cc_api_call_proceeding(vdbPtr=0x617BE064, callID=0x12, prog_ind=0x0)
*Mar 12 04:03:57.197: cc_api_call_alert(vdbPtr=0x617BE064, callID=0x12, prog_ind=0x8, sig_ind=0x1)
```

```
*Mar 12 04:03:57.201: sess_appl: ev(17=CC_EV_CALL_PROCEEDING), cid(18), disp(0)
*Mar 12 04:03:57.201: ssa: cid(18)st(1)oldst(0)cfid(-1)csize(0)in(0)fDest(0)-cid2(17)st2(1)oldst2(0)
*Mar 12 04:03:57.201: ssaIgnore cid(18), st(1),oldst(1), ev(17)
*Mar 12 04:03:57.201: sess_appl: ev(7=CC_EV_CALL_ALERT), cid(18), disp(0)
*Mar 12 04:03:57.201: ssa: cid(18)st(1)oldst(1)cfid(-1)csize(0)in(0)fDest(0)-cid2(17)st2(1)oldst2(0)
*Mar 12 04:03:57.201: ssaFlushPeerTagQueue cid(17) peer list: (empty)
*Mar 12 04:03:57.201: ccCallAlert (callID=0x11, prog_ind=0x8, sig_ind=0x1)
*Mar 12 04:03:57.201: ccConferenceCreate (confID=0x617A8808, callID1=0x11, callID2=0x12, tag=0x0)
*Mar 12 04:03:57.201: cc_api_bridge_done (confID=0x7, srcIF=0x616C9F54, srcCallID=0x11,
dstCallID=0x12, disposition=0, tag=0x0)value H323-UserInformation
*Mar 12 04:03:57.201: {
*Mar 12 04:03:57.201:   h323-uu-pdu
*Mar 12 04:03:57.201:   {
*Mar 12 04:03:57.201:     h323-message-body alerting :
*Mar 12 04:03:57.201:       {
*Mar 12 04:03:57.201:         protocolIdentifier { 0 0 8 2250 0 2 },
*Mar 12 04:03:57.205:         destinationInfo
*Mar 12 04:03:57.205:         {
*Mar 12 04:03:57.205:           mc FALSE,
*Mar 12 04:03:57.205:           undefinedNode FALSE
*Mar 12 04:03:57.205:         },
```

Notice in this packet that Cisco IOS is also sending the H.245 address and port number to Cisco CallManager. Sometimes the Cisco IOS Gateway will send the unreachable address, which could cause either no audio or one-way audio.

```
*Mar 12 04:03:57.205:         h245Address ipAddress :
*Mar 12 04:03:57.205:           {
*Mar 12 04:03:57.205:             ip 'AC1046E2'H,
*Mar 12 04:03:57.205:             port 011008
*Mar 12 04:03:57.205:           },
*Mar 12 04:03:57.213: Hex representation of the ALERTING TPKT to send.0300003D0100
*Mar 12 04:03:57.213:
*Mar 12 04:03:57.213:       H225Lib::h225AlertRequest: Q.931 ALERTING sent from socket [1]. Call state
changed to [Call Received].
*Mar 12 04:03:57.213: cc_api_bridge_done (confID=0x7, srcIF=0x617BE064, srcCallID=0x12,
dstCallID=0x11, disposition=0, tag=0x0)
```

The following debug output shows that the H.245 session is coming up. You can see the capability indication for codec negotiation, as well as how many bytes will be present in each voice packet.

```
*Mar 12 04:03:57.217: cc_api_caps_ind (dstVdbPtr=0x616C9F54, dstCallId=0x11, srcCallId=0x12,
caps={codec=0xEBFB, fax_rate=0x7F, vad=0x3, modem=0x617C5720 codec_bytes=0, signal_type=3})
*Mar 12 04:03:57.217: sess_appl: ev(23=CC_EV_CONF_CREATE_DONE), cid(17), disp(0)
*Mar 12 04:03:57.217: ssa: cid(17)st(3)oldst(0)cfid(7)csize(0)in(1)fDest(1)-cid2(18)st2(3)oldst2(1)
*Mar 12 04:03:57.653: cc_api_caps_ind (dstVdbPtr=0x617BE064, dstCallId=0x12, srcCallId=0x11,
caps={codec=0x1, fax_rate=0x2, vad=0x2, modem=0x1, codec_bytes=160, signal_type=0})
```

The following debug output shows that both parties negotiated correctly and agreed on G.711 codec with 160 bytes of data.

```
*Mar 12 04:03:57.653: cc_api_caps_ack (dstVdbPtr=0x617BE064, dstCallId=0x12, srcCallId=0x11,
caps={codec=0x1, fax_rate=0x2, vad=0x2, modem=0x1, codec_bytes=160, signal_type=0})
*Mar 12 04:03:57.653: cc_api_caps_ind (dstVdbPtr=0x617BE064, dstCallId=0x12, srcCallId=0x11,
caps={codec=0x1, fax_rate=0x2, vad=0x2, modem=0x, codec_bytes=160, signal_type=0})
*Mar 12 04:03:57.653: cc_api_caps_ack (dstVdbPtr=0x617BE064, dstCallId=0x12, srcCallId=0x11,
caps={codec=0x1, fax_rate=0x2, vad=0x2, modem=0x1, codec_bytes=160, signal_type=0})
*Mar 12 04:03:57.657: cc_api_caps_ack (dstVdbPtr=0x616C9F54, dstCallId=0x11, srcCallId=0x12,
caps={codec=0x1, fax_rate=0x2, vad=0x2, modem=0x1, codec_bytes=160, signal_type=0})
*Mar 12 04:03:57.657: cc_api_caps_ack (dstVdbPtr=0x616C9F54, dstCallId=0x11, srcCallId=0x12,
caps={codec=0x1, fax_rate=0x2, vad=0x2, modem=0x1, codec_bytes=160, signal_type=0})
```

The H.323 connect and disconnect messages can be seen below.

```
*Mar 12 04:03:59.373: cc_api_call_connected(vdbPtr=0x617BE064, callID=0x12)
*Mar 12 04:03:59.373: sess_appl: ev(8=CC_EV_CALL_CONNECTED), cid(18), disp(0)
*Mar 12 04:03:59.373: ssa: cid(18)st(4)oldst(1)cfid(7)csize(0)in(0)fDest(0)-cid2(17)st2(4)oldst2(3)
*Mar 12 04:03:59.373: ccCallConnect (callID=0x11)
*Mar 12 04:03:59.373: {
*Mar 12 04:03:59.373:   h323-uu-pdu
*Mar 12 04:03:59.373:   {
*Mar 12 04:03:59.373:     h323-message-body connect :
*Mar 12 04:03:59.373:       {
*Mar 12 04:03:59.373:         protocolIdentifier { 0 0 8 2250 0 2 },
*Mar 12 04:03:59.373:         h245Address ipAddress :
*Mar 12 04:03:59.373:           {
*Mar 12 04:03:59.377:             ip 'AC1046E2'H,
*Mar 12 04:03:59.377:             port 011008
*Mar 12 04:03:59.377:           },
*Mar 12 04:03:59.389: Hex representation of the CONNECT TPKT to send.03000052080
*Mar 12 04:03:59.393: H225Lib::h225SetupResponse: Q.931 CONNECT sent from socket [1]
*Mar 12 04:03:59.393: H225Lib::h225SetupResponse: Q.931 Call State changed to [Active].
*Mar 12 04:04:08.769: cc_api_call_disconnected(vdbPtr=0x617BE064, callID=0x12, cause=0x10)
*Mar 12 04:04:08.769: sess_appl: ev(12=CC_EV_CALL_DISCONNECTED), cid(18), disp(0)
```

Cisco IOS Gateway with T1/PRI Interface

As explained earlier, there are two types of calls going through the Cisco IOS Gateways: the Cisco IOS Gateway interfaces to the PSTN or PBX with either T1/CAS or T1/PRI interfaces. The following are the debug outputs when the Cisco IOS Gateways use T1/PRI interface.

The "debug isdn q931" command on the Cisco IOS Gateway has been turned on, enabling Q.931, a Layer Three signaling protocol for D-channel in the ISDN environment. Each time a call is placed out of the T1/PRI interface, a setup packet must be sent. The setup packet always has (protocol descriptor) pd = 8 and it generates a random hex value for the callref. The callref is used to track the call. For example, if two calls are placed, the callref value can determine the call for which the RX (received) message is intended. Bearer capability 0x8890 means a 64kb/s data call. If it were a 0x8890218F, then it would be a 56kb/s data call and 0x8090A3 if it is a voice call. In the debug output below, the bearer capability is 0x8090A3, which is for voice. Called and calling party numbers are also shown.

The callref uses a different value for the first digit (to differentiate between TX and RX) and the second value is the same (SETUP had a 0 for the last digit and CONNECT_ACK also has a 0). The router is completely dependent upon the PSTN or PBX to assign a Bearer channel (B-channel). If the PSTN or PBX doesn't assign a channel to the router, the call won't be routed. In our case, a CONNECT message is received from the switch with the same reference number as was received for ALERTING (0x800B). Finally, you can see the exchange of the DISCONNECT message followed by RELEASE and RELEASE _COMP messages as the call is being disconnected. RELEASE_COMP messages are followed by a cause ID for the call rejection. The cause ID is a hex value. The meaning of the cause can be found by decoding the hex value and following up with your provider.

```
 *Mar  1 225209.694 ISDN Se115 TX ->  SETUP pd = 8  callref = 0x000B
 *Mar  1 225209.694        Bearer Capability i = 0x8090A3
 *Mar  1 225209.694        Channel ID i = 0xA98381
 *Mar  1 225209.694        Calling Party Number i = 0x2183, '1001'
 *Mar  1 225209.694        Called Party Number i = 0x80, '3333'
 *Mar  1 225209.982 ISDN Se115 RX <-  ALERTING pd = 8  callref = 0x800B
 *Mar  1 225209.982        Channel ID i = 0xA98381
 *Mar  1 225210.674 ISDN Se115 RX <-  CONNECT pd = 8  callref = 0x800B
 *Mar  1 225210.678 ISDN Se115 TX ->  CONNECT_ACK pd = 8  callref = 0x000B
 *Mar  1 225215.058 ISDN Se115 RX <-  DISCONNECT pd = 8  callref = 0x800B
 *Mar  1 225215.058        Cause i = 0x8090 - Normal call clearing  225217 %ISDN-6
DISCONNECT Int S10  disconnected from unknown , call lasted 4 sec
 *Mar  1 225215.058 ISDN Se115 TX ->  RELEASE pd = 8  callref = 0x000B
 *Mar  1 225215.082 ISDN Se115 RX <-  RELEASE_COMP pd = 8  callref = 0x800B
*Mar  1 225215.082  Cause i = 0x829F - Normal, unspecified or Special  intercept, call blocked group
restriction
```

Cisco IOS Gateway with T1/CAS Interface

As explained earlier, there are two types of calls going through the Cisco IOS Gateways: the Cisco IOS Gateway interface to the PSTN or PBX with either T1/CAS or T1/PRI interfaces. The following are the debug outputs when the Cisco IOS Gateways has T1/CAS interface. The "debug cas" on the Cisco IOS Gateway has been turned on.

The following debug message shows that the Cisco IOS Gateway is sending an off-hook signal to the switch.

```
Apr  5 17:58:21.727: from NEAT(0): (0/15): Tx LOOP_CLOSURE (ABCD=1111)
```

The following debug message indicates the switch is sending wink after receiving the loop closure signal from the Cisco IOS Gateway.

```
Apr  5 17:58:21.859: from NEAT(0): (0/15): Rx LOOP_CLOSURE (ABCD=1111)
Apr  5 17:58:22.083: from NEAT(0): (0/15): Rx LOOP_OPEN (ABCD=0000)
```

The following debug message indicates the Cisco IOS Gateway is going off-hook.

```
Apr  5 17:58:23.499: from NEAT(0): (0/15): Rx LOOP_CLOSURE (ABCD=1111)
```

The following is the output of the "show call active voice brief" on the Cisco IOS Gateway when the call is in progress. The called and calling party number and other useful information are also shown.

```
R5300-5#show call active voice brief
<ID>: <start>hs.<index> +<connect> pid:<peer_id> <dir> <addr> <state> tx:<packets>/<bytes>
rx:<packets>/<bytes> <state>
 IP <ip>:<udp> rtt:<time>ms pl:<play>/<gap>ms lost:<lost>/<early>/<late>  delay:<last>/<min>/<max>ms
<codec>
 FR <protocol> [int dlci cid] vad:<y/n> dtmf:<y/n> seq:<y/n> sig:<on/off> <codec> (payload size)
 Tele <int>: tx:<tot>/<v>/<fax>ms <codec> noise:<l> acom:<l> i/o:<l>/<l> dBm
511D : 156043737hs.1 +645 pid:0 Answer 1001 active
 tx:1752/280320 rx:988/158080
 IP172.16.70.228:18888 rtt:0ms pl:15750/80ms lost:0/0/0 delay:25/25/65ms g711ulaw
511D : 156043738hs.1 +644 pid:1 Originate 3333 active
 tx:988/136972 rx:1759/302548
 Tele 1/0/0 (30): tx:39090/35195/0ms g711ulaw noise:-43 acom:0  i/0:-36/-42 dBm
```

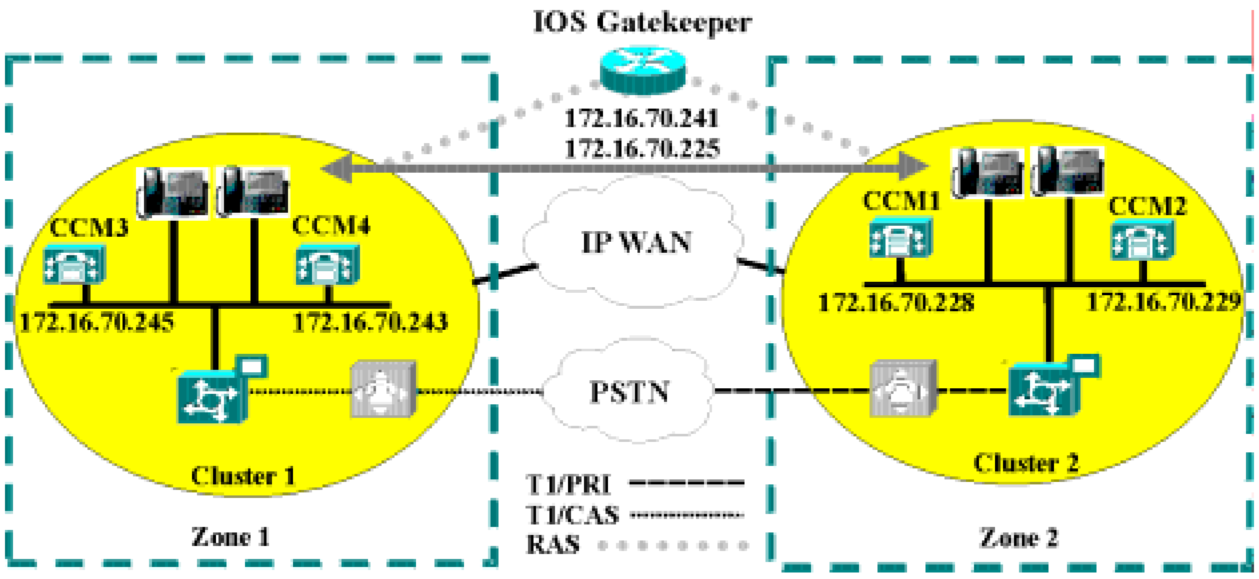
Appendix C – Troubleshooting Case Study 3

Inter-Cluster Cisco IP Phone-to-Cisco IP Phone Calls

In the previous case studies, the call flow and troubleshooting techniques of an intra-cluster call and a Cisco IP Phone call through a Cisco IOS Gateway to a phone hanging off of a local PBX or somewhere on the PSTN have been discussed in detail. This case study examines a Cisco IP Phone calling another Cisco IP Phone located in a different cluster. This type of call is also known as an inter-cluster Cisco IP Phone call.

Sample Topology

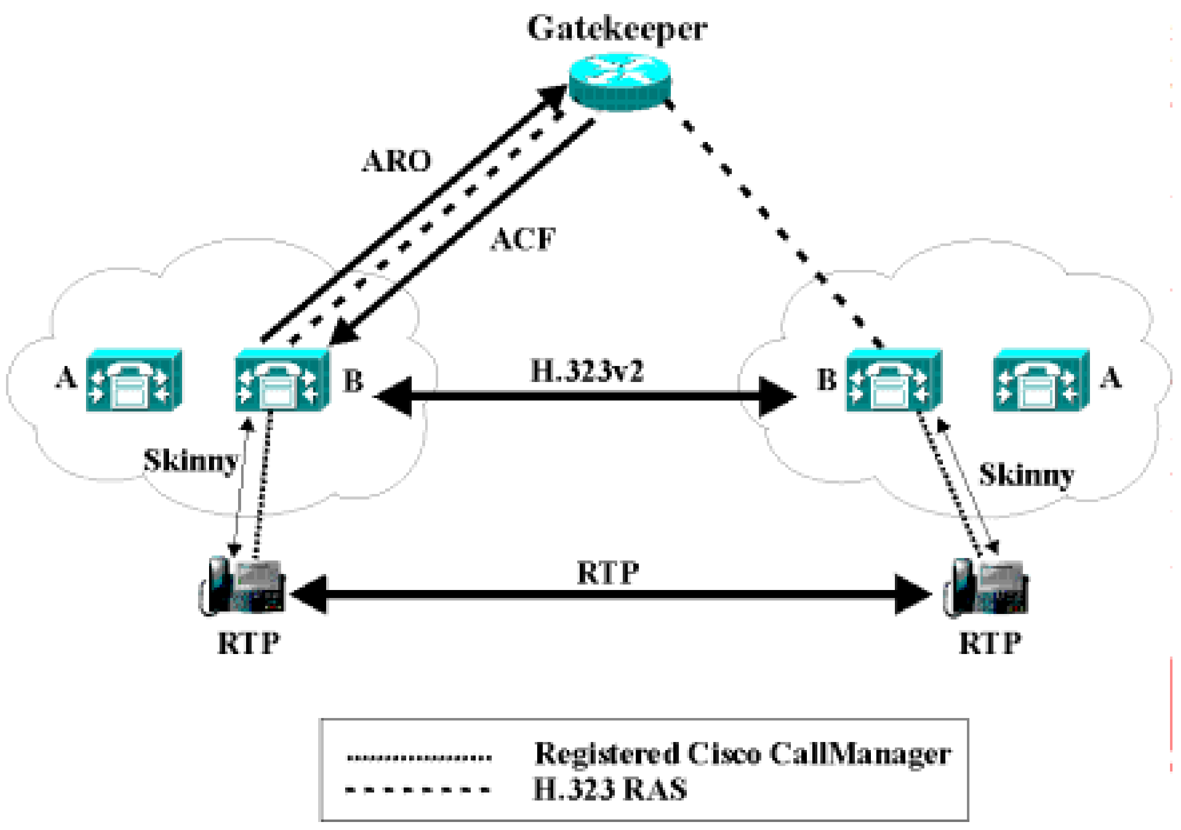
The following is the sample topology is used in this case study. There are two clusters, each having two Cisco CallManagers. There are also Cisco IOS Gateways and a Cisco IOS Gatekeeper in place.



Inter-Cluster H.323 Communication

As you can see in the topology, the Cisco IP Phone in Cluster 1 is making a call to the Cisco IP Phone in Cluster 2. Inter-cluster Cisco CallManager communication takes place using the H.323 Version 2 protocol. There is also a Cisco IOS Gatekeeper for admission control. The detailed explanation of the debug output and show commands, and the interaction between the Cisco IOS Gatekeeper and Cisco IOS Gateway and Cisco CallManager devices can be reviewed in the previous sections.

The call flow process is shown in the diagram below. The Cisco IP Phone can talk to the Cisco CallManager via Skinny Station protocol, and the Cisco CallManager can talk with the Cisco IOS Gatekeeper using the H.323 RAS protocol. The Admission Request message (ARQ) is sent to the Cisco IOS Gatekeeper, which sends the Admission Confirmed message (ACF) after making sure the inter-cluster call can be made using H.323 version 2 protocol. Once done, the audio path is made using the RTP protocol between Cisco IP Phones in different clusters.



Call Flow Traces

This section discusses the call flow using SDI trace examples captured in the CCM000000000 file. The location of this file can be found in the previous section. The traces discussed in this case study focus only on the call flow itself, because the more detailed trace information has already been explained in the previous case study (initialization, registration, KeepAlive mechanism, and so on.)

In this call flow, a Cisco IP Phone (2002) located in Cluster 2 is calling a Cisco IP Phone (1001) located in Cluster 1. Remember that you can follow a device through the trace by looking at the TCP handle value, time stamp, or name of the device. The TCP handle value for the device remains the same until the device is rebooted or goes offline.

In the following traces, the Cisco IP Phone (2002) has gone off-hook. The trace shows the unique messages, TCP handle, and the calling number, which is displayed on the Cisco IP Phone. The called number (1001), H.225 connect, and H.245 confirm messages can be seen in the following debug output. The codec type is G.711 µ-law.

```
16:06:13.921 CCM|StationInit - InboundStim - OffHookMessageID tcpHandle=0x1c64310
16:06:13.953 CCM|Out Message -- H225ConnectMsg -- Protocol= H225Protocol
16:06:13.953 CCM|Ie - H225UserUserIe IEData= 7E 00 37 05 02 C0 06
16:06:13.953 CCM|StationD - stationOutputCallInfo CallingPartyName=,  CallingParty=2002,
CalledPartyName=1001, CalledParty=1001, tcpHandle=0x1c64310
16:06:14.015 CCM|H245Interface(2) OLC indication chan number = 2
16:06:14.015 CCM|StationD - stationOutputOpenReceiveChannel tcpHandle=0x1c64310 myIP:
e74610ac (172.16.70.231)
16:06:14.015 CCM|StationD - ConferenceID: 0 msecPacketSize: 20
compressionType:(4)Media_Payload_G711Ulaw64k
16:06:14.062 CCM|StationInit - InboundStim - StationOpenReceiveChannelAckID tcpHandle=0x1c64310,
Status=0, IpAddr=0xe74610ac, Port=20444, PartyID=2
16:06:14.062 CCM|H245Interface(2) paths established ip = e74610ac, port = 20444
16:06:14.187 CCM|H245Interface(2) OLC outgoing confirm ip = fc4610ac, port = 29626
```

The calling and called party number, which is associated with a IP address and a hex value, can be seen in the following traces.

```
16:06:14.187 CCM|StationD - stationOutputStartMediaTransmission tcpHandle=0x1c64310 myIP:
e74610ac (172.16.70.231)
16:06:14.187 CCM|StationD - RemoteIpAddr: fc4610ac (172.16.70.252)
```

The following traces show the packet sizes and the MAC address of the Cisco IP Phone (2002). These traces are followed by the disconnect, then on-hook messages.

```
RemoteRtpPortNumber: 29626 msecPacketSize: 20 compressionType:(4)Media_Payload_G711Ulaw64k
16:06:16.515 CCM| Device SEP003094C26105 , UnRegisters with SDL Link to monitor NodeID= 1
16:06:16.515 CCM|StationD - stationOutputCloseReceiveChannel tcpHandle=0x1c64310 myIP:
e74610ac (172.16.70.231)
16:06:16.515 CCM|StationD - stationOutputStopMediaTransmission tcpHandle=0x1c64310 myIP:
e74610ac (172.16.70.231)
16:06:16.531 CCM|In  Message -- H225ReleaseCompleteMsg -- Protocol= H225Protocol
16:06:16.531 CCM|Ie - Q931CauseIe -- IEData= 08 02 80 90
16:06:16.531 CCM|Ie - H225UserUserIe -- IEData= 7E 00 1D 05 05 80 06
16:06:16.531 CCM|Locations:Orig=1 BW=64       Dest=0 BW=-1 (-1 implies infinite bw available)
16:06:16.531 CCM|MediaManager - wait_AuDisconnectRequest - StopSession sending disconnect to
(64,2) and remove connection from list
16:06:16.531 CCM|MediaManager - wait_AuDisconnectReply - received all disconnect replies, forwarding
a reply for party1(16777219) and party2(16777220)
16:06:16.531 CCM|MediaCoordinator - wait_AuDisconnectReply - removing MediaManager(2) from
connection list
16:06:16.734 CCM|StationInit - InboundStim - OnHookMessageID tcpHandle=0x1c64310
```

Failed Call Flow

The following section describes an unsuccessful inter-cluster call flow, as seen in the SDI trace. In the traces below, the Cisco IP Phone (1001) has gone off-hook. A TCP handle is assigned to the Cisco IP Phone.

```
16:05:33.468 CCM|StationInit - InboundStim - OffHookMessageID tcpHandle=0x4fbbc30
16:05:33.468 CCM|StationD - stationOutputDisplayText tcpHandle=0x4fbbc30, Display= 1001
```

```
16:05:33.484 CCM|StationD - stationOutputSetLamp stim: 9=Line instance=1 lampMode=LampOn
tcpHandle=0x4fbbc30
```

In the following traces, the user is dialing the called number (2000) of the Cisco IP Phone, and the process of digit analysis is trying to match the number.

```
16:05:33.484 CCM|Digit analysis: match(fqcn="", cn="1001", pss="", dd="")
16:05:33.484 CCM|Digit analysis: potentialMatches=PotentialMatchesExist
16:05:35.921 CCM|Digit analysis: match(fqcn="", cn="1001", pss="", dd="2")
16:05:35.921 CCM|Digit analysis:potentialMatches=ExclusivelyOffnetPotentialMatchesExist
16:05:36.437 CCM|Digit analysis: match(fqcn="", cn="1001", pss="", dd="20")
16:05:36.437 CCM|Digit analysis:potentialMatches=ExclusivelyOffnetPotentialMatchesExist
16:05:36.656 CCM|Digit analysis: match(fqcn="", cn="1001", pss="", dd="200")
16:05:36.656 CCM|Digit analysis:potentialMatches=ExclusivelyOffnetPotentialMatchesExist
16:05:36.812 CCM|Digit analysis: match(fqcn="", cn="1001", pss="", dd="2000")
```

The digit analysis has now been completed and the results are shown in the following traces. It is important to note that the "PotentialMatches=NoPotentialMatchesExist" reference below indicates that the Cisco CallManager is unable to match this directory number. Finally a reorder tone is sent to the calling party (1001), which is followed by an on-hook message.

```
16:05:36.812 CCM|Digit analysis: analysis results
16:05:36.812 CCM||PretransformCallingPartyNumber=1001
|CallingPartyNumber=1001
|DialingPattern=2XXX
|DialingRoutePatternRegularExpression=(2XXX)
|PotentialMatches=NoPotentialMatchesExist
|CollectedDigits=2000
16:05:36.828 CCM|StationD - stationOutputCallInfo CallingPartyName=1001,  CallingParty=1001,
CalledPartyName=, CalledParty=2000, tcpHandle=0x4fbbc30
16:05:36.828 CCM|StationD - stationOutputStartTone: 37=ReorderTone tcpHandle=0x4fbbc30
16:05:37.953 CCM|StationInit - InboundStim - OnHookMessageID tcpHandle=0x4fbbc30
```

Appendix D

Call Detail Records (CDRs and CMRs)

This appendix provides detailed information about Call Detail Records (CDRs) and Call Management Records (CMRs, also known as Diagnostic CDRs).

CDR records are written to a database for use in post processing activities. These activities include many functions but will primarily be billing and network analysis.

The database is a Microsoft SQL Server 7.0 database. Access to the database can be made via Open DataBase Connectivity (ODBC).

Access is provided to all tables in the database in a read-only fashion, and to the CDR and CMR tables in a read/write fashion.

To use CDR record data, you may want to read other tables in the database in an effort to obtain information about the type of device the CDR is about. This correlation between devices in the Device table and the IP address listed in the CDR record is not straightforward and is listed as a known issue later in this appendix.

Writing Records

Cisco CallManager writes CDR records to the SQL database as calls are made in a manner consistent with the configuration of each individual Cisco CallManager. This configuration is made via the Service Parameters screen in Cisco CallManager Administration.

All records are written to the primary database for a cluster. If the primary database is not available, then they will be written to any of the other backup databases. Once the primary database becomes available, then writing new records will continue on the primary database and the locally written records will be moved to the primary.

Reading Records

The easiest way to read data from the SQL database may be to use ODBC. A good connection string would look like:

```
DRIVER={SQL Server};SERVER=machineX;DATABASE=CCM0300
```

Be sure to use the correct database name. If a Cisco CallManager Release 3.0(1) version of the software is installed over an existing installation, then the database might be migrated if called for by the new installation. In this case, the old database will still exist, and the new database will also exist. The names will differ by adding one to the number of the name. For instance, the original name is CCM0300. After a migration, the newer database name will be CCM0301. The highest number database should be used.

The primary database (machine and name) currently in use by the cluster can be found by clicking on the Details button of Cisco CallManager Administration (click Help to reach the Welcome screen where the Details button is located). The registry on machines hosting a database can also be checked. Look at the registry key: "\\HKEY_LOCAL_MACHINE\Software\Cisco Systems Inc.\DBL" for the item called DBConnection0. This string item contains a connection string similar to that shown above with the machine name and database name of the primary database.

Access is controlled by use of SQL Users. The following table specifies the UserID and password that should be used when accessing the Cisco CallManager database.

Tables	SQl UserID	Password	Capability
CallDetailRecord, CallDetailRecordDiagnostic	CiscoCCMCDR	dipsy	Read/Write
(Other)	CiscoCCMReader	cowboys	Read only

Removing Records

Since Cisco CallManager is relying on third party applications to post-process the CDR data, you should remove the CDR data when all applications are through with the data. Since this involves modifying the database, the CiscoCCMCDR user should be used.

If CDR records accumulate to a configured maximum (10,000,000 CDR records), then the oldest CDR records will be removed along with related CMR records once a day.

When removing CDR data after analysis, be sure to remove all related CMR records also.

Table Schema

Detailed information about the format and use of each field in the CDR is provided later in this appendix.

The main tables to be used are the CallDetailRecord table, which holds CDR records, and the CallDetailRecordDiagnostic table, which holds CMR records. The CallDetailRecord table is related to the CallDetailRecordDiagnostic table via the two GlobalCallID columns, GlobalCallID_callManagerId and GlobalCallID_callId. There may be more than one CMR per CDR.

The CallDetailRecord table holds information about the endpoints of the call and other call control/routing aspects of the call. The CallDetailRecordDiagnostic table holds information about the quality of the streamed audio of the call.

Known Issues

Cisco CallManager Release 3.0(1) has several known issues with the CDR data. A few of these are listed here.

IP to Device Name Translation

The CDR table lists IP addresses for the endpoints of a call. These IP addresses are not easily converted to device names so that the type of device can be determined.

OnNet vs. OffNet

It is difficult to know if the call stayed completely on the IP network, or at least internal to the local system. One clue is to check the device type of both ends of the call. If both are phones, then one can assume that it stayed OnNet. If one is a gateway, then more assumptions must be made. If the gateway is an Analog Access type of device with a POTS or station port, then the call might have just gone to a local analog phone, or might have gone out to the PSTN. Look at the number dialed and correlate this to the known dial plan to estimate if the call went OffNet. Otherwise, the call probably went OffNet.

OffNet Digits Dialed

If a call is placed out a gateway, the digits dialed to get to the gateway may not be the digits sent to the PSTN. The gateway may be intelligent and modify the directory number further. If this is the case, Cisco CallManager does not know, and the CDR will not reflect the actual digits sent OffNet.

Fields in a Call Detail Record

This section defines all fields in the current records. The field types are those used by Cisco CallManager, and not necessarily those defined in the CDR record in the database. The database field definitions are adequate to store the data, but the interpretation of the data should take into account the field types defined here.
All unsigned integers are 32bit unsigned integers.

Field Data Conversions

There are some fields that require conversion from decimal format to another format for displays. This appendix defines their values, and how to convert them or where to get information on how to convert them.

Time Values

All time values are represented as unsigned 32 bit integers. This unsigned integer value is displayed from the database as a signed integer.

This field is a time_t value that is obtained from the Windows NT (2000) system routines. The value is a coordinated universal time (UTC) value, and represents the number of seconds since Midnight (00:00:00) Jan. 1, 1970.

Deciphering the Time Stamp

Using Microsoft Excel, you can write a formula to make converting this time stamp a little easier. If the value is in cell A1, you can make another cell:

=A1/86400+DATE(1970,1,1)

There are 86400 seconds in a day.

Then format the resulting cell as a date/time field in Excel.

IP Addresses

All IP addresses are stored in the system as unsigned integers. The database displays them as signed integers. To convert the signed decimal value to an IP address, first convert the value to a Hex number (taking into consideration that it is really an unsigned number). The 32bit Hex value represents 4 bytes. The 4 bytes are in reverse order (Intel standard). To get the IP address, reverse the order of the bytes, and convert each byte to a decimal number. The resulting 4 bytes represent the 4-byte fields of the IP address in dotted notation.

Note: The database will display it as a negative number when the low byte of the IP address has the most significant bit set.

Converting IP Addresses

For example: IP Address 192.168.18.188 would be displayed as follows
Database Display = -1139627840.
This converts to a Hex value of 0xBC12A8C0.
Reverse the Hex bytes = C0A812BC
CO A8 12 BC
Bytes Converted from Hex to Decimal = 192 168 18 188 which would be displayed as 192.168.18.188.
Example2: IP Address 192.168.18.59
Database Display = 991078592
This converts to a Hex value of 0x3B12A8C0
Reverse Byte order = C0A8123B
C0 A8 12 3B
Bytes Converted from Hex to Decimal = 192 168 18 59 which would be displayed as 192.168.18.59.

CDR Field Definition

The following table provides field definitions for CDRs.

Field Definitions	
Field	**Definition**
cdrRecordType	**Type of this record** unsigned integer Specifies the type of this specific record. It could be a Start call record(0), End call record(1), or a CMR record(2).
globalCallIdentifier	**Global Call Identifier** The Global Call Identifier consists of two fields which are both unsigned integers. The values must be treated as unsigned integers. The two fields are Unsigned integer GlobalCallID_CallID Unsigned integer GlobalCallID_CallManagerID This is the call identifier that is assigned to the entire call. All records associated with a standard call will have the same global call identifier.
origLegCallIdentifier	**Origination leg call identifier** unsigned integer This is a unique identifier that is used to track the origination leg of a call. It is unique within a cluster.
dateTimeOrigination	**Date/time of call origination** unsigned integer This represents the time that the device originating the call went off hook, or the time that an outside call was first recognized by the system (it received the Setup message). The value is a coordinated universal time (UTC) value, and represents the number of seconds since Midnight (00:00:00) Jan. 1, 1970.
origNodeId	**Originator's node ID** unsigned integer This field represents the node within the Cisco CallManager cluster where the call originator was registered at the time of this call.
origSpan	**Originator's span or port** unsigned integer This field contains the originator's port or span number if the call originated through a gateway. If not, this field contains zero (0).
callingPartyNumber	**Calling party number** up to 25 characters This is the directory number of the device from which the call originated.
origIpPort	**Calling party's IP port** unsigned integer This field contains the IP Port of the device from which the call originated.
origIpAddr	**Calling party's IP address** unsigned integer This field contains the IP address of the device from which the call originated.

Field Definitions	
originalCallingPartyNumberPartition	**Calling party's partition** up to 50 characters This field contains the Partition associated with the calling party.
origCause_Location	**ISDN location value** unsigned integer This field contains the location value from the Cause information Element.
origCause_Value	**Calling party cause Of call termination** unsigned integer This cause represents why the call to the originating device was terminated. In the case of transfers, forwards, and so on, the cause of call termination may be different for the originating device and the termination device. Thus, there are two cause fields associated with each call. Usually they will be the same.
origMediaTransportAddress_IP	**The IP address for the originator's media connection** unsigned integer This is the destination IP Address to which the Media Stream from the originator was connected.
origMediaTransportAddress_Port	**The port for the originator's media connection** unsigned integer This is the destination port to which the Media Stream from the originator was connected.
origMediaCap_payloadCapability	**The codec type used by the originator** unsigned integer This field contains the Codec type (compression or payload type) that the originator used on the sending side during this call. It may be different than the codec type used on its receiving side.
origMediaCap_maxFramesPerPacket	**The number of milliseconds of data per packet** unsigned integer This field contains the number of milliseconds of data per packet sent to the destination, by the originator of this call. The actual data size depends on the codec type being used to generate the data.
origMediaCap_g723BitRate	**The bit rate to be used by G.723** unsigned integer Defines the bit rate to be used by G.723. There are two bit rate values. They are: 1 =5.3K bit rate, and 2 = 6.3K bit rate.
lastRedirectDn	**Directory number of the party that last redirected this call** up to 25 characters This is the directory number of the last device that redirected this call. This field applies only to calls that were redirected, such as conference calls, call forwarded calls, and so on.
lastRedirectDnPartition	**Partition of the phone that last redirected this call** up to 50 characters This is the Partition of the last device that redirected this call. This field applies only to calls that were redirected such as conference calls, call forwarded calls, and so on.
destLegIdentifier	**The call identifier for the destination leg of the call** unsigned integer This is a unique identifier that is used to track the destination leg of this call. It is unique within a cluster.

Field Definitions	
destNodeId	**The node identifier for the node where the destination of the call was registered** unsigned integer The node within the Cisco CallManager cluster where the destination device was registered at the time of this call.
dest Span	**The destination span or port** unsigned integer This field contains the destination port or span number if the call was terminated through a gateway. If not, this field contains a (0) zero.
destIpAddr	**The IP address to which the call was delivered** unsigned integer This field contains the IP address of the signaling connection on the device that terminated the call.
destIpPort	**The IP port to which the call was delivered** unsigned integer This field contains the IP port of the signaling connection on the device that terminated the call.
originalCalledPartyNumber	**The destination received from the call originator** up to 25 characters This field contains the Directory Number to which the call was originally extended based on the digits dialed by the originator of the call. If the call completes normally (meaning it was not forwarded), this Directory Number should always be the same as the "finalCalledPartyNumber". If the call was forwarded, this field contains the original destination of the call before it was forwarded.
originalCalledPartyNumberPartition	**Called party's partition** up to 50 characters This field contains the partition associated with the called party.
finalCalledPartyNumber	**The destination to which the call was delivered** up to 25 characters This field contains the Directory Number to which the call was actually extended. If the call completes normally (meaning it was not forwarded), this Directory Number should always be the same as the "originalCalledPartyNumber". If the call was forwarded, this field contains the Directory Number of the final destination of the call after all forwards were completed.
finalCalledPartyNumberPartition	**The partition associated with the final destination of the call.** up to 50 characters This field contains the partition associated with the destination to which the call was actually extended. In a normal call, this field should be the same as "originalCalledPartyNumberPartition". If the call was forwarded, this field contains the partition of the final destination of the call after all forwards were completed.
destCause_location	**Called party cause location** unsigned integer This is the ISDN Location value from the Cause Information Element.

Field Definitions	
destCause_value	**Called party cause of call termination** unsigned integer This cause represents why the call to the termination device was terminated. In the case of transfers, forwards, and so on, the cause of call termination may be different for the recipient of the call and the originator of the call. Thus, there are two cause fields associated with each call. Usually they will be the same. When an attempt is made to extend a call to a busy device that is forwarded, the cause code will reflect “Busy” even though the call was connected to a forward destination.
destMediaTransportAddress_IP	**The IP address for the destination outgoing media connection** unsigned integer This is the origination IP Address from which the Media Stream from the destination was connected.
origMediaTransportAddress_Port	**The port for the destination outgoing media connection** unsigned integer This is the originator’s port from which the Media Stream from the destination was connected.
destMediaCap_payloadCapability	**The codec type used by the destination on sending side** unsigned integer This field contains the Codec type (compression or payload type) that the destination used on its sending side during this call. It may be different than the codec type used on its receiving side.
destMediaCap_maxFramesPerPacket	**The number of milliseconds of data per packet** unsigned integer This field contains the number of milliseconds of data per packet sent to the originator, by the destination of this call. The actual data size depends on the codec type being used to generate the data.
destMediaCap_g723BitRate	**The bit rate to be used by G.723** unsigned integer Defines the bit rate to be used by G.723. There are two bit rate values. They are: 1 =5.3K bit rate, and 2 = 6.3K bit rate.
dateTimeConnect	**Date/time of connect** unsigned integer This is the date and time that the call was connected between the originating and terminating devices. This is the date and time that the call was connected between the originating and terminating devices. The value is a coordinated universal time (UTC) value, and represents the number of seconds since Midnight (00:00:00) Jan. 1, 1970.
dateTimeDisconnect	**Date/time of disconnect** unsigned integer This is the time that the call was disconnected between the originating and terminating devices, or when the call was torn down even if it was never connected. The value is a coordinated universal time (UTC) value, and represents the number of seconds since Midnight (00:00:00) Jan. 1, 1970.
duration	**Call duration** This is the number of seconds that the call was connected. It is the difference between the date/time of connect and the date/time of disconnect.

CMR Field Definitions

The following table provides field definitions for CMRs (diagnostic CDRs).

Field Definitions	
Field	**Definition**
cdrRecordType	**Type of this record** unsigned integer Specifies the type of this specific record. It will be set to CMR record.
globalCallIdentifier	**Global Call Identifier for this call** The Global Call Identifier consists of two fields which are both unsigned integers. The values must be treated as unsigned integers. The two fields are Unsigned integer GlobalCallID_CallID Unsigned integer GlobalCallID_CallManagerID This is the call identifier that is assigned to the entire call. All records associated with a standard call will have the same global call identifier.
nodeID	**The Cisco CallManager node identifier** The node within the Cisco CallManager cluster where this record was generated.
callIdentifier	**Call Identifier** unsigned integer This is a call leg identifier that identifies to which call leg this record pertains.
directoryNum	**Directory number used on this call** This is the directory number of the device from which these diagnostics were collected.
directoryNumPartition	**The partition associated with the directory number** This is the partition of the directory number in this record.
dateTimeStamp	**Date/time of call termination** This represents the approximate time that the device went on hook. The time is put into the record when the phone responds to a request for diagnostic information. This is a time_t value.
numberPacketsSent	**Number of packets sent** The total number of RTP data packets transmitted by the device since starting transmission on this connection. The value is zero if the connection was set in “receive only” mode.
numberOctetsSent	**Number of Octets (bytes) of data sent to the other party** The total number of payload octets (that is, not including header or padding) transmitted in RTP data packets by the device since starting transmission on this connection. The value is zero if the connection was set in “receive only” mode.
numberPacketsReceived	**The number of data packets received during this call** The total number of RTP data packets received by the device since starting reception on this connection. The count includes packets received from different sources if this is a multicast call. The value is zero if the connection was set in “send only” mode.

Field Definitions	
numberOctetsReceived	**The number of octets (bytes) of data received during this call** The total number of payload octets (that is, not including header or padding) received in RTP data packets by the device since starting reception on this connection. The count includes packets received from different sources, if this is a multicast call. The value is zero if the connection was set in "send only" mode.
numberPacketsLost	**Lost RTP packets during this connection** The total number of RTP data packets that have been lost since the beginning of reception. This number is defined as the number of packets expected less the number of packets actually received, where the number of packets received includes any that are late or duplicates. Thus, packets that arrive late are not counted as lost, and the loss may be negative if there are duplicates. The number of packets expected is defined to be the extended last sequence number received, as defined next, less the initial sequence number received. The value is zero if the connection was set in "send only" mode. (For details, see RFC 1889)
jitter	**The interarrival jitter during this connection** An estimate of the statistical variance of the RTP data packet interarrival time, measured in milliseconds and expressed as an unsigned integer. The interarrival jitter J is defined to be the mean deviation (smoothed absolute value) of the difference D in packet spacing at the receiver compared to the sender for a pair of packets. Detailed computation algorithms are found in RFC 1889. The value is zero if the connection was set in "send only" mode.
latency	**The latency experienced during this connection** The value is an estimate of the network latency, expressed in milliseconds. This is the average value of the difference between the NTP timestamp indicated by the senders of the RTCP messages and the NTP timestamp of the receivers, measured when these messages are received. The average is obtained by summing all the estimates, then dividing by the number of RTCP messages that have been received. (For details see RFC 1889)

Call Records Logged By Call Type

Each normal call between two parties logs one CDR End Call record. Each End Call record contains all fields identified above, but some fields may not be used. If a field is not used, it will be blank if it is an ASCII string field, or "0" if it is a numeric field. When supplementary services are involved in a call, more End Call records may be written.

In addition to the CDR End Call record, there may be up to one CMR record per endpoint involved in a call. In a normal call between two parties each using a Cisco IP Phone, there will be two CMR records written: one for the originator, and one for the destination of the call. This section describes the records written for different call types in the system.

Normal Calls (Cisco IP Phone-to-Cisco IP Phone)

Normal calls log three records per call. They are: EndCall plus two diagnostic records, one for each endpoint. In the EndCall record, all fields may contain valid information. The duration will always be non-zero unless the "CdrLogCallsWithZeroDurationFlag" flag is enabled (set to true). The "originalCalledPartyNumber" field will contain the same directory number as the "finalCalledPartyNumber" field.

Abandoned Calls

The logging of calls with zero duration is optional. Normally these records will not be logged. If logging calls with zero duration is enabled, the following things should be noted.

- If the call was abandoned (such as when a phone is taken off hook, and placed back on hook), various fields will not contain data. In this case, the "originalCalledPartyNumber," "finalCalledPartyNumber," the partitions associated with them, "destIpAddr," and the dateTimeConnect fields will be blank. All calls that were not connected will have a "duration" of zero seconds. When a call is abandoned, the cause code is "0".
- If the user dialed a directory number and then abandoned the call before it was connected, the "First Dest" and "Final Dest" fields and their associated partitions will contain the directory number and partition to which the call would have been extended. The "Dest Ip" field will be blank, and the duration will be zero.

Forwarded or Redirected Calls

The call records for forwarded calls will be the same as those for normal calls except for the "originalCalledPartyNumber" field, and the "originalCalledPartyNumberPartition" fields. These fields will contain the directory number and partition for the destination that was originally dialed by the originator of the call. If the call was forwarded, the "finalCalledPartyNumber" and "finalCalledpartyNumberPartition" fields will be different, and will contain the directory number and partition of the final destination of the call. Also, when a call is forwarded, the "lastRedirectDn" and "lastRedirectDnPartition" fields will contain the directory number and partition of the last phone that forwarded or redirected this call.

Calls With Busy or Bad Destinations

These calls will be logged as a normal call with all relevant fields containing data. The Called Party Cause field will contain a cause code indicating why the call was not connected, and the Called Party IP and Date/Time Connect fields will be blank. If the originator abandoned the call, the cause will be "NO_ERROR" (0). The Duration will always be zero seconds. These calls will not be logged unless "CdrLogCallsWithZeroDurationFlag" is enabled.

Call Management Records Logged By Call Type

Each normal call between two Cisco IP Phones logs exactly two CMR records. Each call CMR record contains all fields identified above. When supplementary services are involved in a call, more than one record may be written. This section describes when diagnostic records are written for different call types in the system.

Normal Calls

Normal calls log exactly two CMR records per call, one for each phone involved in the call. Currently only Cisco IP Phones and MGCP gateways are capable of responding to the diagnostic information request. All fields will contain valid information.

Abandoned Calls

If the call was abandoned (such as when a phone is taken off-hook and placed back on hook), all fields related to streaming data will be blank (zero). This is because no streaming connection was established, and therefore no data was transferred. All records with blank fields will not be logged if the “CdrLogCallsWithZeroDurationFlag” is disabled.

Forwarded Calls

The call records for forwarded calls will be the same as those for normal calls.

Calls With Busy or Bad Destinations

In the normal case, only records that represent calls that were actually connected will be logged. In order to log calls with bad destinations, you must enable “CdrLogCallsWithZeroDurationFlag.” If it is enabled, then all calls will be logged including the case where the user goes off-hook and then on-hook again.
If the calls are logged, they will be logged as normal calls with all relevant fields containing data. There will only be one record per call since the calls were never connected to a destination. The record will be for the originator of the call.

Codec Types (Compression / Payload types)

The following table provides values and descriptions for codec types.

Codec Description	
Codec	**Description**
1	NonStandard
2	G711A-law 64k
3	G711A-law 56k
4	G711µ-law 64k
5	G711µ-law 56k
6	G722 64k
7	G722 56k
8	G722 48k
9	G7231
10	G728
11	G729
12	G729AnnexA
13	Is11172AudioCap
14	Is13818AudioCap
15	G729AnnexB
32	Data 64k
33	Data 56k
80	GSM
81	ActiveVoice
82	G726_32K
83	G726_24K
84	G726_16K

Cause Codes

The following table provides a list of cause codes that may appear in the Cause fields.

Cause Code Descriptions	
Cause Code	**Description**
0	No error
1	Unallocated (unassigned) number
2	No route to specified transit network (national use)
3	No route to destination
4	Send special information tone
5	Misdialed trunk prefix (national use)
6	Channel unacceptable
7	Call awarded and being delivered in an established channel
8	Preemption
9	Preemption - circuit reserved for reuse
16	Normal call clearing
17	User busy
18	No user responding
19	No answer from user (user alerted)
20	Subscriber absent
21	Call rejected

Cause Code Descriptions	
22	Number changed
26	Non-selected user clearing
27	Destination out of order
28	Invalid number format (address incomplete)
29	Facility rejected
30	Response to STATUS ENQUIRY
31	Normal, unspecified
34	No circuit/channel available
38	Network out of order
39	Permanent frame mode connection out of service
40	Permanent frame mode connection operational
41	Temporary failure
42	Switching equipment congestion
43	Access information discarded
44	Requested circuit/channel not available
46	Precedence call blocked
47	Resource unavailable, unspecified
49	Quality of Service not available
50	Requested facility not subscribed
53	Service operation violated
54	Incoming calls barred
55	Incoming calls barred within CUG (Closed User Group)
57	Bearer capability not authorized
58	Bearer capability not presently available
62	Inconsistency in designated outgoing access information and subscriber class
63	Service or option not available, unspecified
65	Bearer capability not implemented
66	Channel type not implemented
69	Requested facility not implemented
70	Only restricted digital information bearer capability is available (national use)
79	Service or option not implemented, unspecified
81	Invalid call reference value
82	Identified channel does not exist
83	A suspended call exists, but this call identity does not
84	Call identity in use
85	No call suspended
86	Call having the requested call identity has been cleared
87	User not member of CUG (Closed User Group)
88	Incompatible destination
90	Destination number missing and DC not subscribed
91	Invalid transit network selection (national use)
95	Invalid message, unspecified
96	Mandatory information element is missing
97	Message type non-existent or not implemented
98	Message is not compatible with the call state, or the message type is non-existent or not implemented
99	An information element or parameter does not exist or is not implemented
100	Invalid information element contents
101	The message is not compatible with the call state
102	The call was terminated when a timer expired and a recovery routine was executed to recover from the error
103	Parameter non-existent or not implemented - passed on (national use)
110	Message with unrecognized parameter discarded

Cause Code Descriptions	
111	Protocol error, unspecified
126	Call split. This is a Cisco-specific code. It is used when a call is terminated during a transfer operation because it was split off and terminated (was not part of the final transferred call). This can help determine which calls were terminated as part of a transfer operation.
127	Interworking, unspecified

Alarms

An alarm is issued when CDR or Diagnostic data is enabled, and the system is unable to write the data into the database.

Unable to write CDR data. (Alarm # 1711 - Major Alarm)

The system attempted to open the database, and was unsuccessful. Probable causes include:

- Cisco CallManager does not have sufficient privileges to open the file for writing to the database. Make sure Cisco CallManager has privileges that will permit write operations.
- The path is not set up, or the database server is down.

Calling Cisco Technical Assistance Center (TAC)

If you have a problem that cannot be resolved through your own troubleshooting, please call the TAC for assistance. Before calling TAC, have the following information available:

- Cisco CallManager Details
- Topology
- Logs and Traces you have run during troubleshooting, including SDI and SDL traces
- Stackwalk.txt files from the WINNT\system32 directory and the Cisco CallManager Trace subdirectory
- “sh-tech” on Cisco IOS Gateways, if applicable
- “sh-tech” on Cisco CallManager
- If the problem is with calls through a Cisco IOS Gateway, please also provide:
    - “debug voice ccapi inout”
    - “debug isdn q931”
    - [AS5300 only] “sh vfc xboard”
    - [AS5300 only] “sh vfc x version dspware”
    - [AS5300 only] “sh vfc x version vcware”

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