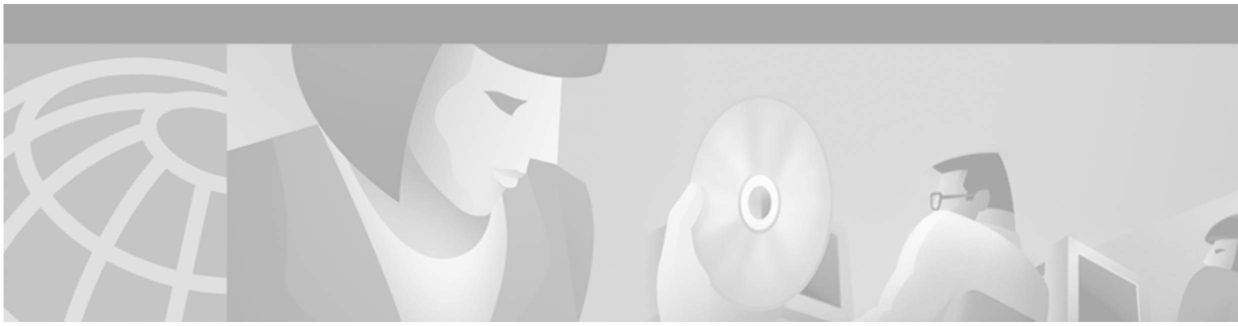


Cisco 2600 and 3600 Routers MGCP Voice Gateway Interoperability with Cisco CallManager

Document Update Alert

This document was originally produced for Cisco IOS Release 12.2(4)T. This feature has been updated in subsequent releases, and more recent documentation is available.

If you are using Cisco IOS Release 12.2(4)T or higher, refer to the following documentation in the *Cisco CallManager and Cisco IOS Interoperability Configuration Guide*, Cisco IOS Voice Configuration Library, Release 12.3:

- Configuring Cisco MGCP Gateways to Interoperate with Cisco CallManager

Feature History

Release	Modification
12.1(3)T	This feature was introduced with Cisco CallManager Version 3.0 and the Cisco Voice Gateway 200 (VG200).
12.2(2)XA	Support was added for Cisco CallManager Version 3.0(8), and Cisco 2600 series and Cisco 3600 series routers.
12.2(4)T	This feature was integrated into Cisco IOS Release 12.2(4)T.

This document describes the additional functionality and platforms offered by the Cisco 2600 and 3600 Routers Media Gateway Control Protocol (MGCP) Voice Gateway Interoperability with Cisco CallManager feature in Cisco IOS Release 12.2(4)T.

This document includes the following sections:

Feature Overview

MGCP voice gateway interoperability with Cisco CallManager allows modular access routers to act as redundant failover MGCP gateways. You can enable IP telephony and Cisco CallManager solutions using Cisco 2600 and Cisco 3600 series routers as voice gateways. This allows you to use the Cisco 2600 and 3600 platforms already in your networks as MGCP gateways within an IP telephony architecture.

An MGCP gateway handles the translation between audio signals and the packet network. The gateways interact with a call agent (also called a Media Gateway Controller or MGC) that performs signal and call processing on gateway calls.

In the MGCP configurations that Cisco IOS supports, the gateway can be any of the following:

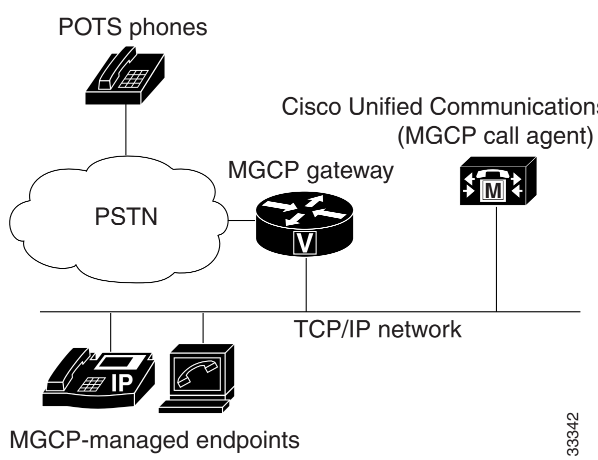
- Cisco router
- Access server
- Cable modem

The call agent is either of the following:

- A server from a third-party vendor
- Cisco CallManager

With MGCP, gateways are defined as secondary devices under control of the call agent. MGCP uses endpoints and connections to construct a call. Endpoints are sources of or destinations for data, and can be physical or logical locations in a device. Connections can be point-to-point or multipoint. The call agent manages connections between endpoints and controls how gateways function. (See Figure 1.)

Figure 1 MGCP



An MGCP gateway derives most of the configuration it requires from the call agent. To configure an MGCP gateway, you simply identify the call agent associated with the gateway and identify the gateway to the call agent.

MGCP optionally supports multiple call agents, which can eliminate a potential single point of failure in the voice network.

The Cisco IOS CLI allows you to set up MGCP on the gateway and to identify the Cisco CallManager. Cisco CallManager assumes control over establishing and tearing down connections between IP endpoints on your network and endpoints connected through the Public Switched Telephone Network (PSTN).

MGCP uses User Datagram Protocol (UDP) for establishing audio connections over IP networks. However, MGCP also uses hairpinning to return a call to the PSTN when the packet network is not available.

Creating a call connection involves a series of signals and events that make up the connection process. The signals might include such indicators as the off-hook status, a ringing signal, or a signal to play an announcement. These events and signals are specific to the type of endpoint involved in the call.

MGCP groups these events and signals into packages. A trunk package, for example, is a group of events and signals relevant to a trunking gateway, and an announcement package is a group of events and signals for an announcement server.

In an MGCP-enabled gateway, the out-of-band dual tone multi-frequency (DTMF) package is loaded upon startup. Once the out-of-band DTMF capabilities are configured in the Cisco CallManager MGCP gateway user interface, the router sends symbols over the UDP control channel to represent any DTMF tones it receives. Cisco CallManager interprets these symbols and passes on the DTMF signals, out of band, to the signaling endpoint.

The Cisco 2600 and 3600 Routers MGCP Voice Gateway Interoperability with Cisco CallManager feature includes the following:

- Supplementary Services, page 3
- Cisco CallManager Redundancy, page 4
- Cisco CallManager Switchback, page 4

When you are using MGCP with a Cisco 2600 series or 3600 series router, all configuration elements associated with dial-plans are controlled by Cisco CallManager and should not be configured in the Cisco 2600 series or 3600 series gateway for MGCP-managed endpoints.

Supplementary Services

Supplementary services includes call hold, call transfer when the line is busy or there is no answer, call forwarding, and three-party call conferencing to and from the PSTN or a private branch exchange (PBX).

Call hold is a function that places the handset into mute mode. Both the transmitter and receiver function are disengaged for a period of time until the hold button is pressed again to reconnect the parties.

Call transfer is a function that transfers a call to a third party through a pre-programmed button that performs the hookswitch and draw with what is called the recall dial tone. The receiver of the call then dials the third-party number, waits for the line to ring and for the new called party to answer, and then hangs up.

Call forwarding is a function that allows you to forward calls dialed from the original location to a remote location within or across the network.

Three-party call conferencing is similar to the transfer function, but rather than transferring the call to a third party, the third party called is added to the call. The conference feature allows the three parties to converse without worrying about cutting each other off.

Cisco CallManager Redundancy

Enabling MGCP and Cisco CallManager on the gateways provides optional redundancy or failover functionality. If the gateways loses communication with the primary Cisco CallManager due to a failure, services are switched to a backup Cisco CallManager. This capability allows existing connections to be preserved during the switchover.

Redundancy requires that you have two or three Cisco CallManagers available on your network. You identify the primary Cisco CallManager with the **mgcp call agent** command. Up to two backup Cisco CallManagers are added with the **ccm-manager redundant-host** command. If you do not configure a backup Cisco CallManager, redundancy is off.

If the Cisco CallManager redundancy feature is configured and the primary Cisco CallManager becomes unavailable, the first backup Cisco CallManager takes control of the devices that were registered with the primary Cisco CallManager. If you specified a second backup Cisco CallManager, it takes control of the devices if both the primary and first backup Cisco CallManagers fail. When the primary Cisco CallManager is brought back to service, control reverts back to that CallManager. This fallback can occur immediately, after a configurable amount of time, or only when all connected sessions have been released.

Cisco CallManager Switchback

Switchback is the way that the gateways reestablish communication with the primary Cisco CallManager when it becomes available again. The switchback mode can be immediately, at a specified time after the last active call ends, or after a specified length of time to ensure greater stability in the voice network. During the switchback, existing connections are not torn down.

Benefits

Supplementary Services

The hold, transfer, forward, and conference supplementary services are software applications. Therefore, you only have to upgrade the software on the server platform rather than upgrade hardware.

Redundancy

Redundancy allows you to designate up to two backup Cisco CallManagers to handle call processing for the disabled primary Cisco CallManager.

Cisco CallManager Switchback

Switchback eliminates a potential single point of failure in the VoIP network. Switchback lets you use redundant Cisco CallManagers so that your MGCP voice gateways can continue to work if the primary Cisco CallManager fails.

Restrictions

Caller ID is supported on Foreign Exchange Station (FXS) interfaces, but not on Foreign Exchange Office (FXO) interfaces in Cisco CallManager MGCP networks.

Related Features and Technologies

- Cisco CallManager
- Voice over IP

Related Documents

- *Cisco CallManager Administration Guide*, *Version 3.0:*
  http://www.cisco.com/univercd/cc/td/doc/product/voice/c_callmg/3_0/index.htm
- *Cisco IOS Interface Command Reference, Release 12.2:*
  http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/finter_r/index.htm
- *Cisco IOS IP Command Reference, Vol. 1 of 3: Addressing and Services, Release 12.2:*
  http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fipras_r/index.htm
- *Cisco IOS Voice, Video, and Fax Command Reference, Release 12.2:*
  http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fvvfax_r/index.htm
- *Cisco IOS Voice, Video, and Fax Configuration Guide, Release 12.2:*
  http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fvvfax_c/index.htm
- *MGCP CAS PBX AAL2 and PVC:*
  http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122newft/122t/122t2/ftmgcptk.htm
- *Release Notes for Cisco 2600 Series, IOS Release 12.2 XA:*
  http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122relnt/2600/rn2600xa.htm
- *Release Notes for Cisco 3600 Series, IOS Release 12.2 XA:*
  http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122relnt/3600/rn3600xa.htm

Supported Platforms

- Cisco 2600 series multiservice platforms
- Cisco 3600 series multiservice platforms

Supported Standards, MIBs, and RFCs

Standards

No new or modified standards are supported by this feature.

MIBs

No new or modified MIBs are supported by this feature.

To obtain lists of supported MIBs by platform and Cisco IOS release, and to download MIB modules, go to the Cisco MIB website on Cisco.com at the following URL:

http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml

RFCs

No new or modified RFCs are supported by this feature.

Prerequisites

- A Cisco 2600 series or Cisco 3600 series router that is running Cisco IOS software Release 12.2(2)XA or later and that supports MGCP
- Cisco CallManager, Version 3.0(8)
- 1-port network module—A 2-channel voice network module (Cisco product number NM-1V)
- 2-port network module—A 4-channel voice network module (Cisco product number NM-2V)
- 2-port FXS card—A 2-channel analog FXS voice interface (Cisco product number VIC-2FXS)
- 2-port FXO card—A 2-channel analog FXO voice interface (Cisco product number VIC-2FXO)

Voice network modules convert telephone voice signals into a form that can be transmitted over an IP network. These modules have no connectors.

Voice interface cards (2-port analog FXS and FXO) are installed in the voice network modules to provide the connection to the telephone equipment or network. You can install one voice interface card in a 2-channel voice network module and two voice interface cards in a 4-channel module.

Configuration Tasks

See the following sections for configuration tasks for the Cisco 2600 and 3600 Routers MGCP Voice Gateway Interoperability with Cisco CallManager feature. Each task in the list is identified as either required or optional.

- Configuring the Router's Ethernet Interface, page 7 (required)
- Configuring MGCP Globally, page 13 (required)
- Configuring MGCP to Control Cisco CallManager, page 14 (required)
- Configuring Dial Peers and Voice Ports for MGCP, page 17 (required)

Configuring the Router's Ethernet Interface

To configure an IP address on the router's Ethernet interface, use the following commands, beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)#` **`interface ethernet`** *`slot/port`*	Enters interface configuration mode so that you can configure the Ethernet interface. *slot*—Number of the slot being configured. Refer to the appropriate hardware manual for slot and port information. *port*—Number of the port being configured. Refer to the appropriate hardware manual for slot and port information.
Step 2	`Router(config-if)#` **`ip address`** *`ipaddress subnetmask`*	Configures an IP address and subnet mask on the router's Ethernet interface.

	Command	Purpose
Step 3	`Router(config-if)# no shut`	Activates the Ethernet port.
Step 4	`Router(config-if)# exit`	Exits interface configuration mode.

To display information about the Ethernet interface, enter the **show interfaces ethernet** command in EXEC mode. Example 1 illustrates a typical display that appears in response to this command.

Example 1 Output of the show interfaces ethernet Command

```
Router# show interfaces ethernet 4/2

Ethernet4/2 is up, line protocol is up
  Hardware is cxBus Ethernet, address is 0000.0c02.d0ce (bia 0000.0c02.d0ce)
  Internet address is 131.108.7.1, subnet mask is 255.255.255.0
  MTU 1500 bytes, BW 10000 Kbit, DLY 1000 usec, rely 255/255, load 1/255
  Encapsulation ARPA, loopback not set, keepalive set (10 sec)
  ARP type: ARPA, ARP Timeout 4:00:00
  Last input 0:00:00, output 0:00:09, output hang never
  Last clearing of "show interface" counters 0:56:40
  Output queue 0/40, 0 drops; input queue 0/75, 0 drops
  Five minute input rate 3000 bits/sec, 4 packets/sec
  Five minute output rate 0 bits/sec, 0 packets/sec
     4961 packets input, 715381 bytes, 0 no buffer
     Received 2014 broadcasts, 0 runts, 0 giants
     0 input errors, 0 CRC, 0 frame, 0 overrun, 0 ignored, 0 abort
     567 packets output, 224914 bytes, 0 underruns
     0 output errors, 168 collisions, 0 interface resets, 0 restarts
     0 babbles, 2 late collision, 7 deferred
     0 lost carrier, 0 no carrier
     0 output buffer failures, 0 output buffers swapped out
```

Table 1 describes the significant fields in this example.

Table 1 show interfaces ethernet Field Descriptions

Field	Description
Ethernet ... is up Ethernet ... is administratively down	Indicates whether the interface hardware is currently active and if it has been taken down by an administrator. “Disabled” indicates the router has received over 5000 errors in a keepalive interval, which is 10 seconds by default.
Line protocol is {up | down | administratively down}	Indicates whether the software processes that handle the line protocol consider the line usable or if it has been taken down by an administrator.
Hardware	Hardware type (for example, MCI Ethernet, SCI, cBus Ethernet) and address.
Internet address	Internet address followed by subnet mask.
MTU	Maximum transmission unit of the interface.
BW	Bandwidth of the interface in kilobits per second.
DLY	Delay of the interface in microseconds.

Table 1 show interfaces ethernet Field Descriptions (continued)

Field	Description
Rely	Reliability of the interface as a fraction of 255 (255/255 is 100 percent reliability), calculated as an exponential average over 5 minutes.
Load	Load on the interface as a fraction of 255 (255/255 is completely saturated), calculated as an exponential average over 5 minutes.
Encapsulation	Encapsulation method assigned to the interface.
Loopback	Indicates whether loopback is set or not.
Keepalive	Indicates whether keepalives are set or not.
ARP type	Type of Address Resolution Protocol assigned.
Last input	Number of hours, minutes, and seconds since the last packet was successfully received by an interface and processed locally on the router. Useful for knowing when a dead interface failed. This field is not updated by fast-switched traffic.
Output	Number of hours, minutes, and seconds since the last packet was successfully transmitted by the interface. Useful for knowing when a dead interface failed.
Output hang	Number of hours, minutes, and seconds (or never) since the interface was last reset because of a transmission that took too long. When the number of hours in any of the "last" fields exceeds 24, the number of days and hours is printed. If that field overflows, asterisks are printed.
Last clearing	Time at which the counters that measure cumulative statistics (such as number of bytes transmitted and received) shown in this report were last reset to 0. Variables that might affect routing (for example, load and reliability) are not cleared when the counters are cleared. In this field, *** indicates the elapsed time is too large to be displayed, and 0:00:00 indicates the counters were cleared more than 231 ms (and less than 232 ms ago).
Output queue, input queue, drops	Number of packets in output and input queues. Each number is followed by a slash, the maximum size of the queue, and the number of packets dropped because of a full queue.

Table 1 show interfaces ethernet Field Descriptions (continued)

Field	Description
Five minute input rate Five minute output rate	Average number of bits and packets transmitted per second in the last 5 minutes. If the interface is not in promiscuous mode, it senses only network traffic it sends and receives (rather than all network traffic). The 5-minute input and output rates should be used only as an approximation of traffic per second during a given 5-minute period. These rates are exponentially weighted averages with a time constant of 5 minutes. A period of four time constants must pass before the average is within 2 percent of the instantaneous rate of a uniform stream of traffic over that period.
Packets input	Total number of error-free packets received by the system.
Bytes	Total number of bytes, including data and MAC encapsulation, in the error-free packets received by the system.
No buffer	Number of received packets discarded because there was no buffer space in the main system. Compare with ignored count. Broadcast storms on Ethernets and bursts of noise on serial lines are often responsible for no input buffer events.
Received ... broadcasts	Total number of broadcast or multicast packets received by the interface.
Runts	Number of packets that are discarded because they are smaller than the minimum packet size of the medium. For instance, any Ethernet packet that is less than 64 bytes is considered a runt.
Giants	Number of packets that are discarded because they exceed the maximum packet size of the medium. For example, any Ethernet packet that is greater than 1518 bytes is considered a giant.
Input errors	Includes runts, giants, no buffer, CRC, frame, overrun, and ignored counts. Other input-related errors can also cause the input errors count to be increased, and some datagrams might have more than one error; therefore, this sum might not balance with the sum of enumerated input error counts.

Table 1 show interfaces ethernet Field Descriptions (continued)

Field	Description
CRC	Cyclic redundancy checksum generated by the originating LAN station or far-end device does not match the checksum calculated from the data received. On a LAN, this usually indicates noise or transmission problems on the LAN interface or the LAN bus itself. A high number of CRCs is usually the result of collisions or a station transmitting bad data.
Frame	Number of packets received incorrectly having a CRC error and a noninteger number of octets. On a LAN, this is usually the result of collisions or a malfunctioning Ethernet device.
Overrun	Number of times the receiver hardware was unable to hand received data to a hardware buffer because the input rate exceeded the receiver's ability to handle the data.
Ignored	Number of received packets ignored by the interface because the interface hardware ran low on internal buffers. These buffers are different from the system buffers mentioned previously in the buffer description. Broadcast storms and bursts of noise can cause the ignored count to be increased.
Abort	Number of packets whose receipt was aborted.
Packets output	Total number of messages transmitted by the system.
Bytes	Total number of bytes, including data and MAC encapsulation, transmitted by the system.
Underruns	Number of times that the transmitter has run faster than the router can handle. This might never be reported on some interfaces.
Output errors	Sum of all errors that prevented the final transmission of datagrams out of the interface being examined. This might not balance with the sum of the enumerated output errors because some datagrams might have more than one error, and others might have errors that do not fall into any of the specifically tabulated categories.
Collisions	Number of messages retransmitted because of an Ethernet collision. A retransmission of this type is usually the result of an overextended LAN (Ethernet or transceiver cable too long, more than two repeaters between stations, or too many cascaded multiport transceivers). A packet that collides is counted only once in output packets.

Table 1 show interfaces ethernet Field Descriptions (continued)

Field	Description
Interface resets	Number of times an interface has been completely reset. Such a reset can happen if packets queued for transmission were not sent within several seconds. On a serial line, this can be caused by a malfunctioning modem that is not supplying the transmit clock signal, or by a cable problem. If the system notices that the carrier detect line of a serial interface is up but the line protocol is down, it periodically resets the interface in an effort to restart it. Interface resets can also occur when an interface is looped back or shut down.
Restarts	Number of times a Type 2 Ethernet controller was restarted because of errors.
Babbles	The transmit jabber timer expired.
Late collision	Number of late collisions. Late collision happens when a collision occurs after the preamble is transmitted.
Deferred	Deferred indicates that the chip had to defer even though it was ready to transmit a frame because the carrier was asserted.
Lost carrier	Number of times the carrier was lost during transmission.
No carrier	Number of times the carrier was not present during transmission.
Output buffer failures	Number of failed buffers.
Output buffers swapped out	Number of buffers swapped out.

Configuring MGCP Globally

To configure MGCP globally, use the following commands in privileged EXEC mode:

	Command	Purpose
Step 1	`Router(config)#` **`hostname`** *`name`*	Assigns a unique name to the router so that the Cisco CallManager can identify it. The default name is Router.
Step 2	`Router(config)#` **`mgcp`**	Enables the MGCP protocol.
Step 3	`Router(config)#` **`mgcp call-agent`** `{`*`ipaddr`* `|` *`hostname`*`} [`**`service-type`** *`type`*`] [`**`version`** *`version-number`*`]`	(Required) Specifies the primary Cisco CallManager's IP address or domain name, and the port gateway service type and version number. *ipaddr*—Specifies the IP address of the Cisco CallManager. *hostname*—Specifies the Cisco CallManager's host name in the format *host.name.ext* **service-type** *type*—(Optional) Specifies the type of gateway control service supported by the Cisco CallManager. Valid values are **mgcp** or **sgcp**. For MGCP configurations, use **mgcp**. **version** *version-number*—(Optional) Specifies the version of service-type. For **mgcp**, the only valid value is 0.1. For **sgcp**, valid values are 1.1 and 1.5.
Step 4	`Router(config)#` **`mgcp dtmf-relay voip codec`** `{`**`all`** `|` **`low-bit-rate`**`}` **`mode`** `{`**`cisco`**`|` **`nse`** `|` **`out-of-band`**`}`	Selects the codec type and enables dual-tone multi-frequency (DTMF) relay. **voip**—(required) Specifies Voice over IP calls. **codec**—Specifies use of either a G.711 or a G.726 codec. **all**—Specifies use of any codec. **low-bit-rate**—Specifies any version of the G.729 low-bit-rate codecs. **cisco**—Removes DTMF tone from the voice stream and sends FRF.11 with a special payload 121 for DTMF digits. **nse**—Uses the NSE-based forwarding method. **out-of-band**—Removes DTMF tone from the voice stream and does not send FRF.11.

To verify the MGCP global configuration, enter the **show mgcp** command in privileged EXEC mode. Example 2 illustrates a typical display that appears in response to this command.

Example 2 Output of the show mgcp Command

```
Router# show mgcp

GCP Admin State ACTIVE, Oper State ACTIVE - Cause Code NONE
MGCP call-agent: 11.0.0.50 2427 Initial protocol service is MGCP, v. 0.1
MGCP block-newcalls DISABLED
MGCP send RSIP for SGCP is DISABLED
MGCP quarantine mode discard/step
MGCP quarantine of persistent events is ENABLED
MGCP dtmf-relay voip codec all mode out-of-band
MGCP dtmf-relay for VoAAL2 disabled for all codec types
MGCP voip modem passthrough mode: CISCO, codec: g711ulaw, redundancy: DISABLED,
MGCP voaal2 modem passthrough mode: NSE, codec: g711ulaw
MGCP TSE payload: 0
MGCP Network (IP/AAL2) Continuity Test timer: 200
MGCP 'RTP stream loss' timer: 5
MGCP request timeout 500, MGCP request retries 3
MGCP rtp unreachable timeout 1000
MGCP gateway port: 2427, MGCP maximum waiting delay 3000
MGCP restart delay 0, MGCP vad DISABLED
MGCP simple-sdp DISABLED
MGCP undotted-notation DISABLED
MGCP codec type g711ulaw, MGCP packetization period 20
MGCP JB threshold lwm 30, MGCP JB threshold hwm 150
MGCP LAT threshold lmw 150, MGCP LAT threshold hwm 300
MGCP PL threshold lwm 1000, MGCP PL threshold hwm 10000
MGCP CL threshold lwm 1000, MGCP CL threshold hwm 10000
MGCP playout mode is adaptive 60, 4, 200 in msec
MGCP IP ToS low delay disabled, MGCP IP ToS high throughput disabled
MGCP IP ToS high reliability disabled, MGCP IP ToS low cost disabled
MGCP IP RTP precedence 5, MGCP signaling precedence: 3
MGCP default package: line-package
MGCP supported packages: gm-package dtmf-package trunk-package line-package
hs-package rtp-package ms-package dt-package sst-packagc-package
MGCP VoAAL2 ignore-lco-codec DISABLED
```

Configuring MGCP to Control Cisco CallManager

To configure MGCP to control Cisco CallManager, use the following commands in global configuration mode:

	Command	Purpose
Step 1	`Router(config)#` **`ccm-manager MGCP`**	Enables support for Cisco CallManager within MGCP.
Step 2	`Router(config)#` **`ccm-manager redundant-host`** {*ip-address* | *DNS-name*} [*ip-address* | *DNS-name*]	(Optional) Identifies up to two backup Cisco CallManagers. *ip-address*—Internet protocol address of the backup Cisco CallManager. *DNS-name*—Domain name system of the backup Cisco CallManager.

	Command	Purpose
Step 3	`Router(config)# ccm-manager switchback {graceful	immediate	schedule-time hh:mmm	uptime -delay minutes}`	(Optional) If you configured one or two backup Cisco CallManagers, you can enter this command to configure switchback mode. This determines when the primary Cisco CallManager is used if it becomes available again while a backup Cisco CallManager is being used. The default is graceful. **graceful**—Completes all outstanding calls before returning the gateway to the control of the primary Cisco CallManager. **immediate**—Returns the gateway to the control of the primary Cisco CallManager without delay, as soon as the network connection to the Cisco CallManager is reestablished. **schedule-time** *hh*:*mmm*—Returns the gateway to the control of the primary Cisco Call Manager at the specified hour and minute, based on a 24-hour clock. If the specified time is earlier than the current time, the switchback will occur at the specified time on the following day. **uptime-delay** *minutes*—Returns the gateway to the control of the primary Cisco Call Manager when the primary Cisco CallManager has run for a specified number of minutes after a network connection is reestablished to the primary CallManager. Permitted values are from 1 to 1440 (1 minute to 24 hours).

To verify the configuration, enter the **show ccm-manager** command in privileged EXEC mode. Example 3 illustrates a typical display that appears in response to this command.

Example 3 Output of the show ccm-manager Command

```
Router# show ccm-manager

c3660A#show ccm-manager
MGCP Domain Name: c3660A.cisco.com
Total number of host: 2
Priority        Status         Host
===================================================
Primary         Registered     10.0.0.201
First backup    Backup polling 10.0.0.50
Second backup   Undefined

Current active Call Manager: 10.0.0.201
Current backup Call Manager: 10.0.0.50
Redundant link port:         2428
Failover Interval:           30 seconds
Keepalive Interval:          15 seconds
Last keepalive sent:         00:20:18 (elapsed time: 00:00:06)
Last MGCP traffic time:      00:20:18 (elapsed time: 00:00:06)
Last switchover time:        None
Switchback mode:             Not selected (Default:Graceful)
```

Example 2 describes the significant fields shown in the display.

Table 2 show ccm-manager Field Descriptions

Field	Description
MGCP Domain Name (System)	System used in the Internet for translating names of network nodes into IP addresses.
Total number of host	Number of Cisco CallManagers.
Priority	Priority of the Cisco CallManagers. The order of priority is primary, first backup, and second backup.
Status	Current usage of the Cisco CallManager. Possible values are registered, idle, backup polling, and down.
Host	Host address of the Cisco CallManager.
Current active Call Manager	Active Cisco CallManager. Can be the primary, first backup, or second backup Cisco CallManager.
Current backup Call Manager	Backup Cisco CallManager currently being used. Empty when no backup is available.
Redundant link port	Port that the Cisco CallManager will use.
Failover Interval	Maximum amount of time that can elapse without the gateway receiving messages from the currently active Cisco Call Manager, or the gateway switches to the backup Cisco Call Manager.
Keepalive Interval	If the gateway hasn't received any messages from the currently active Cisco CallManager within the specified amount of time, the gateway sends a keepalive message to the Cisco CallManager to be sure that it is OK.
Last keepalive sent	Indicates when the last keepalive message was sent.
Last switchover time	Last time switchover occurred.
Switchback mode	Displays the switchback mode configuration that determines when the primary Cisco CallManager will be used if it becomes available again while a backup Cisco CallManager is being used.

Configuring Dial Peers and Voice Ports for MGCP

To configure dial peers and voice ports for MGCP, use the following commands, beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)#` **`dial-peer voice`** *`number`* **`pots`**	Designates that the specified dial peer is a POTS dial peer using VoIP encapsulation.
Step 2	`Router(config-dial-peer)#` **`application MGCPAPP`**	Enables MGCP on the dial peer. **Note** In some Cisco IOS versions, the **application MGCPAPP** command is case-sensitive. Unless you know that your version is not case sensitive, always enter MGCPAPP in uppercase letters. You can check whether your version is case sensitive after configuring this command by looking at the output of the **show running-configuration** command.
Step 3	`Router(config-dial-peer)#` **`port`** *`slot-number/subunit-number/port`*	Binds the MGCP application to the specified voice port.
Step 4	`Router(config-dial-peer)#` **`exit`**	Exits dial-peer configuration mode.
Step 5	`Router(config)#` **`voice-port`** *`slot/subunit/port`*	Enters voice port configuration mode. Valid values vary by router platform.
Step 6	`Router(config-voiceport)#` **`no shut`**	Activates the voice port. If a voice port is not being used, shut the voice port down with the **shutdown** command. **Note** The **shutdown** and **no shutdown** commands affect both ports on a voice interface card (VIC). It is only necessary to enter the command on one of the ports.

The router is now ready to communicate with the Cisco CallManager. It periodically sends out messages attempting to establish a connection. When the Cisco CallManager configuration is complete, the connection should automatically establish itself. You should not have to make any further changes on the router.

To display configuration information for dial peers, use the **show dial-peer voice** command in privileged EXEC mode or the **show voice-port** command in privileged EXEC mode. The commands are described briefly below.

Command	Purpose
`Router# show dial-peer voice [number] [summary]`	Displays the configuration for all Voice over IP (VoIP) and POTS dial peers configured for the router. **number**—(Optional) A specific dial peer. This option displays configuration information for a single dial peer identified by the **number** argument. Valid entries are any integer that identifies a specific dial peer, from 1 to 32767. **summary**—(Optional) Displays a summary of all voice dial peers.
`Router# show voice port [slot/subunit/port | summary]`	**Note** This command format is applicable only to Cisco 2600 and 3600 series routers that have analog voice ports. (Optional) Displays configuration information for the analog voice port you specify with the *slot/subunit/port* designation. *slot*—Specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. *subunit*—Specifies a voice interface card (VIC) where the voice port is located. Valid entries are 0 and 1. (The VIC fits into the voice network module.) *port*—Specifies an analog voice port number. Valid entries are 0 and 1. **summary**—(Optional) Displays a summary of all voice ports.

Command	Purpose
Router# **show voice port** [*slot/port:ds0-group* | **summary**]	**Note** This command format is applicable only to Cisco 2600 and 3600 series routers that have digital voice ports (with T1 packet voice trunk network modules). (Optional) Displays information for the digital voice port you specify with the *slot/port:ds0-group* designation. *slot*—Specifies a router slot in which the packet voice trunk network module (NM) is installed. Valid entries are router slot numbers for the particular platform. *port*—Specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. (One VWIC fits in an NM.) *ds0-group*—Specifies a T1 or E1 logical port number. Valid entries are from 0 to 23 for T1 and from 0 to 30 for E1. **summary**—(Optional) Displays a summary of all voice ports.

Example 4 illustrates a typical display that appears in response to the **show dial-peer voice** command for a VoIP dial peer.

Example 4 Output of the show dial-peer voice Command

```
Router# show dial-peer voice 1000

c3660A#show dial-peer voice 1000
VoiceEncapPeer1000
information type = voice,
description = `',
tag = 1000, destination-pattern = `',
answer-address = `', preference=0,
numbering Type = `unknown'
group = 1000, Admin state is up, Operation state is down,
incoming called-number = `', connections/maximum = 0/unlimited,
DTMF Relay = disabled,
huntstop = disabled,
in bound application associated: 'mgcpapp'
out bound application associated: ''
dnis-map =
permission :both
incoming COR list:maximum capability
outgoing COR list:minimum requirement
type = pots, prefix = `',
forward-digits default
session-target = `', voice-port = `',
direct-inward-dial = disabled,
digit_strip = enabled,
register E.164 number with GK = TRUE

Connect Time = 0, Charged Units = 0,
Successful Calls=0, Failed Calls=0, Incomplete Calls=0
```

```
Accepted Calls = 0, Refused Calls = 0,
Last Disconnect Cause is "",
Last Disconnect Text is "",
Last Setup Time = 0.
```

Table 3 describes the significant fields in this example.

Table 3 show dial-peer voice Field Descriptions

Field	Description
destination-pattern	Destination pattern (telephone number) for this peer.
answer-address	Answer address configured for this dial peer.
group	Group number associated with this peer.
Admin state	Administrative state of this peer.
Operation state	Operational state of this peer.
incoming called-number	Indicates the incoming called number if it has been set by means of the **incoming-called number** command.
DTMF Relay	Indicates for this dial peer whether or not dual-tone multifrequency (DTMF) relay has been enabled, by using the **dtmf-relay** command.
huntstop	Indicates whether dial-peer hunting has been turned on, by means of the **huntstop** command, for this dial peer.
Permission	Configured permission level for this peer.
session-target	Session target of this peer.
Connect Time	Accumulated connect time to the peer since system startup for both incoming and outgoing calls. The value is given in hundredths of a second.
Charged Units	Total number of charged units applying to this peer since system startup. The value is given in hundredths of a second.
Failed Calls	Number of failed call attempts to this peer since system startup.
Accepted Calls	Number of calls accepted from this peer since system startup.
Refused Calls	Number of calls from this peer refused since system startup.
Last Disconnect Cause	Encoded network cause associated with the last call. This value is updated whenever a call is started or cleared and depends on the interface type and session protocol being used on this interface.

Table 3 show dial-peer voice Field Descriptions (continued)

Field	Description
Last Disconnect Text	ASCII text describing the reason for the last call termination.
Last Setup Time	Value of the system up time when the last call to this peer was started.

Example 5 illustrates sample output from the **show voice-ports** command for an FXS analog voice port on a Cisco 3600 router.

Example 5 Output of the show voice ports Command

```
Router# show voice port 1/0/0

c3660A#show voice port 1/0/0

Foreign Exchange Office 1/0/0 Slot is 1, Sub-unit is 0, Port is 0
Type of VoicePort is FXO
Operation State is DORMANT
Administrative State is UP
No Interface Down Failure
Description is not set
Noise Regeneration is enabled
Non Linear Processing is enabled
Non Linear Mute is disabled
Non Linear Threshold is -21 dB
Music On Hold Threshold is Set to -38 dBm
In Gain is Set to 0 dB
Out Attenuation is Set to 3 dB
Echo Cancellation is enabled
Echo Cancellation NLP mute is disabled
Echo Cancellation NLP threshold is -21 dB
Echo Cancel Coverage is set to 8 ms
Playout-delay Mode is set to default
Playout-delay Nominal is set to 60 ms
Playout-delay Maximum is set to 200 ms
Playout-delay Minimum mode is set to default, value 40 ms
Playout-delay Fax is set to 300 ms
Connection Mode is normal
Connection Number is not set
Initial Time Out is set to 10 s
Interdigit Time Out is set to 10 s
Call Disconnect Time Out is set to 60 s
Ringing Time Out is set to 180 s
Wait Release Time Out is set to 30 s
Companding Type is u-law
Region Tone is set for US

Analog Info Follows:
Currently processing none
Maintenance Mode Set to None (not in mtc mode)
Number of signaling protocol errors are 0
Impedance is set to 600r Ohm
Station name None, Station number None

Voice card specific Info Follows:
Signal Type is loopStart
Number Of Rings is set to 1
Supervisory Disconnect is inactive
```

```
Answer Supervision is inactive
Hook Status is On Hook
Ring Detect Status is inactive
Ring Ground Status is inactive
Tip Ground Status is inactive
Dial Type is dtmf
Digit Duration Timing is set to 100 ms
InterDigit Duration Timing is set to 100 ms
Pulse Rate Timing is set to 10 pulses/second
InterDigit Pulse Duration Timing is set to 750 ms
Percent Break of Pulse is 60 percent
GuardOut timer is 2000 ms
```

Table 4 describes the significant fields in this example.

Table 4 show voice ports Field Descriptions

Field	Description
Slot	Slot used in the VIC for this port.
Sub-unit	Subunit used in the VIC for this port.
Port	Port number for this interface associated with the VIC.
Type of VoicePort	Type of voice port: FXO, FXS, or E&M.
Operation State	Operation state of the port.
Administrative State	Administrative state of the voice port.
Description	Description of the voice port.
Noise Regeneration	Whether or not background noise should be played to fill silent gaps if voice activity detection (VAD) is activated.
Non-Linear Processing	Whether or not nonlinear processing is enabled for this port.
Music on Hold Threshold	Configured Music-On-Hold Threshold value for this interface.
In Gain	Amount of gain inserted at the receiver side of the interface.
Out Attenuation	Amount of attenuation inserted at the transmit side of the interface.
Echo Cancellation	Whether or not echo cancellation is enabled for this port.
Echo Cancel Coverage	Echo cancel coverage for this port.
Connection Mode	Connection mode of the interface.
Connection Number	Full E.164 telephone number used to establish a connection with the trunk or private line, automatic ringdown (PLAR) mode.
Initial Time Out	Amount of time the system waits for an initial input digit from the caller.
Interdigit Time Out	Amount of time the system waits for a subsequent input digit from the caller.

Table 4 show voice ports Field Descriptions (continued)

Field	Description
Ringing Time Out	Ringing timeout duration.
Wait Release Time Out	The length of time a voice port stays in the call-failure state while the Cisco MC3810 multiservice concentrator sends a busy tone, a reorder tone, or an out-of-service tone to the port.
Companding Type	Companding standard used to convert between analog and digital signals in PCM systems.
Region Tone	Configured regional tone for this interface.
Analog Information Fields	
Currently processing	Type of call currently being processed: none, voice, or fax.
Maintenance Mode	Maintenance mode of the voice port.
Number of signaling protocol errors	Number of signaling protocol errors.
Impedance	Configured terminating impedance for the E&M interface.
Voice Card Information Fields	
Signal Type	Type of signaling for a voice port: loop-start, ground-start, wink-start, immediate, or delay-dial.
Hook Status	Hook status of the FXO/FXS interface.
Ring Ground Status	Ring ground indication.
Tip Ground Status	Tip ground indication.
Dial Type	Out-dialing type of the voice port.
Digit Duration Timing	DTMF digit duration, in milliseconds.
InterDigit Duration Timing	DTMF interdigit duration, in milliseconds.
Pulse Rate Timing	Pulse dialing rate, in pulses per second (pps).
InterDigit Pulse Duration Timing	Pulse dialing interdigit timing, in milliseconds.

Troubleshooting Tips

All dial plan-related configuration elements are controlled by the Cisco CallManager, and should not be configured in the MGCP gateway for MGCP-managed endpoints (any endpoint with an **application mgcapp** command in its dial-peer statement). You should not use the **destination-pattern** or **session-target** dial-peer configuration commands, nor the **connection** voice-port configuration command.

Monitoring and Maintaining MGCP Support for the Cisco CallManager

Command	Purpose
`Router# debug ccm-manager [errors	packets	events]`	Displays Cisco CallManager errors. **errors**—Displays Cisco CallManager errors. **packets**—Displays errors related to Cisco CallManager packets. **events**—Displays errors related to Cisco CallManager events such as when the primary Cisco CallManager failed and control switched to the backup Cisco CallManager.
`Router# show ccm-manager [hosts	redundancy]`	Displays a list of Cisco CallManagers, as well as their status and availability. **hosts**—Displays each configured Cisco CallManager, its operational status, and its host address. **redundancy**—Displays failover mode and status information, including the redundant link port, the failover interval, keepalives, MGCP traffic time, switchover, and switchback.

Configuration Examples

This section provides the following configuration examples:

- Router's Ethernet Interface Configuration Example, page 24
- MGCP Global Configuration Example, page 25
- MGCP Control of Cisco CallManager Example, page 25
- Dial Peers and Voice Ports Configuration Example, page 25

Following the configuration examples there is a display of a current configuration. Go to Display of Configuration Example.

Router's Ethernet Interface Configuration Example

In the following example, there is a Cisco 3600 series router and one FastEthernet 10/100 port.

```
router(config)# interface fastethernet 0/0
router(config-if)# ip address 10.0.0.200 255.255.255.0
router(config-if)# no shut
```

MGCP Global Configuration Example

In the following example, the IP address for the Cisco CallManager is 10.0.0.201, the DTMF relay is enabled, and the router is communicating with a Cisco CallManager.

```
router(config)# hostname 3660A
3660A(config)# mgcp
3660A(config)# mgcp call-agent 10.0.0.201
3660A(config)# mgcp dtmf-relay voip codec all mode out-of-band
3660A(config)# ccm-manager mgcp
```

MGCP Control of Cisco CallManager Example

In the following example, the commands required to configure the gateway and redundancy are shown.

```
mgcp !Configures router to run MGCP
mgcp call-agent 10.0.0.201 service-type mgcp version 0.1 ! Defines Primary CallManager
mgcp dtmf-relay voip codec all mode out-of-band !Voice over IP calls, no DTMF
```

To configure redundancy, enter the following commands:

```
ccm-manager switchback graceful !After last call ends, use primary Cisco CallManager
ccm-manager redundant-host 10.0.0.50 !Defines redundant Cisco CallManager (first backup)
ccm-manager mgcp !Enables support for Cisco CallManager within MGCP
```

Dial Peers and Voice Ports Configuration Example

In the following example, voice port 0 is configured in voice interface card 1 with MGCP. There are two FXO ports, and two FXS ports. (Voice ports are always installed in slot 1 of the gateway, and slot and port numbering begins at 0.) The MGCP application is applied to a dial peer. The **voice-port** command specifies that the voice network module will be installed in router slot 1, the location of the VIC is 1, and the voice port is 0.

```
3660A(config)# dial-peer voice 1 pots
3660A(config)# application MGCPAPP
3660A(config)# port 1/0/0

/* FXO port */

3660A(config)# dial-peer voice 2 pots
3660A(config)# application MGCPAPP
3660A(config)# port 1/0/1

/* FXO port */

3660A(config)# dial-peer voice 3 pots
3660A(config)# application MGCPAPP
3660A(config)# port 1/1/0

/* FXS port */

3660A(config)# dial-peer voice 4 pots
3660A(config)# application MGCPAPP
3660A(config)# port 1/1/1

/* FXS port */

3660A(config)# voice-port 1/0/0
```

```
3660A(config-voiceport)# no shut
Both ports are in service

/* FXO port */

3660A(config)# voice-port 1/1/0
3660A(config-voiceport)# no shut
Both ports are in service

/* FXS port */
```

Display of Configuration Example

To see the current operating configuration, including changes you just made, enter the **show running-configuration** command, as follows.

```
Router# show running-configuration
```

The following command output displays the complete configuration:

```
Building configuration...

Current configuration : 1244 bytes
!
version 12.1
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 3660A
!
boot system flash
boot system rom
boot system tftp 3660A 10.0.0.201
no logging buffered
logging rate-limit console 10 except errors
no logging console
enable secret #####
enable password #####
!
ip subnet-zero
no ip finger
no ip domain-lookup
!
mgcp
mgcp call-agent 10.0.0.201
mgcp dtmf-relay codec all mode out-of-band
mgcp sdp simple
call rsvp-sync
!
ccm-manager switchback graceful
ccm-manager redundant-host 10.0.0.50
ccm-manager mgcp
!
!
interface FastEthernet0/0
 ip address 10.0.0.200 255.255.255.0
 no ip mroute-cache
 speed auto
 full-duplex
!
```

```
ip default-gateway 10.0.0.1
ip classless
no ip http server
!
snmp-server engineID local 00000009020000196983000
snmp-server community public RO
!
voice-port 1/0/0
!
voice-port 1/0/1
!
voice-port 1/1/0
!
voice-port 1/1/1
!
!
dial-peer voice 1 pots
 application mgcpapp
 port 1/0/0
!
dial-peer voice 2 pots
 application mgcpapp
 port 1/0/1
!
dial-peer voice 3 pots
 application mgcpapp
 port 1/1/0
!
dial-peer voice 4 pots
 application mgcpapp
 port 1/1/1
!
!
line con 0
 transport input none
line aux 0
line vty 0 4
 password ww
 login
!
end
```

Command Reference

This section documents the following modified commands:

- **ccm-manager application**
- **ccm-manager mgcp**
- **ccm-manager redundant-host**
- **ccm-manager switchback**
- **debug ccm-manager**
- **mgcp call-agent**
- **show ccm-manager**

All other commands used with this feature are documented in the Cisco IOS Release 12.2 command reference publications.

Note The commands, keywords, and arguments that you can use may differ slightly from those presented here, because they vary with the platform, Cisco IOS release, and configuration you are using. When in doubt, use Cisco IOS command help (command ?) to determine the syntax choices that are available.

ccm-manager application

To configure the port number for the redundant link application, use the **ccm-manager application** command in global configuration mode. To disable the configuration, use the **no** form of this command.

ccm-manager application redundant-link port *number*

no ccm-manager application

Syntax Description

number	Port number for the transport protocol. The protocol may be the User Data Protocol (UDP), Reliable User Datagram Protocol (RDUP), or Transmission Control Protocol (TCP). Permitted values are from 0 to 65535, and it must not be a well-known reserved port number.

Defaults

Port 2428

Command Modes

Global configuration

Command History

Release	Modification
12.1(3)T	This command was introduced with Cisco CallManager Version 3.0 and the Cisco Voice Gateway 200 (VG200).
12.2(2)XA	The command was implemented on Cisco 2600 series and 3600 series routers.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T.

Usage Guidelines

This command is optional. Use this command only when defining an application-specific port other than the default.

Examples

In the following example, the port number of the redundant link application is 2429:

```
Router(config)# ccm-manager application redundant-link port 2429
```

Related Commands

Command	Description
ccm-manager redundant-host	Configures the IP address or the DNS name of up to two backup Cisco CallManagers.
ccm-manager switchback	Configures the switchback mode that determines when the primary Cisco CallManager will be used if it becomes available again while a backup Cisco CallManager is being used.

ccm-manager mgcp

To allow the gateway to communicate with the Cisco CallManager by means of Media Gateway Control Protocol (MGCP) and supply redundant services, use the **ccc-manager mgcp** command in global configuration mode. To disable this command, use the **no** form of this command.

ccm-manager mgcp

no ccm-manager mgcp

Syntax Description

This command has no arguments or keywords.

Defaults

Cisco CallManager does not communicate with the gateway through MGCP.

Command Modes

Global configuration

Command History

Release	Modification
12.1(3)T	This command was introduced with Cisco CallManager Version 3.0 and the Cisco Voice Gateway 200 (VG200).
12.2(2)XA	The command was implemented on Cisco 2600 series and 3600 series routers.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T.

Usage Guidelines

This command sets the gateway to MGCP mode. In MGCP mode, the gateway can communicate with the Cisco CallManager through MGCP, and it can enable redundancy when a backup Cisco CallManager is available.

Examples

In the following example, support for Cisco CallManager and redundancy is enabled within MGCP:

```
Router(config)# ccm-manager mgcp
```

Related Commands

Command	Description
mgcp	Enables Media Gateway Control Protocol mode.
ccm-manager redundant-host	Configures the IP address or the DNS name of up to two backup Cisco CallManagers.
ccm-manager switchback	Configures the switchback mode that determines when the primary Cisco CallManager will be used if it becomes available again while a backup Cisco CallManager is being used.

ccm-manager redundant-host

To configure the IP address or the DNS name of up to two backup Cisco CallManagers, use the **ccm-manager redundant-host** command in global configuration mode. To disable the configuration of a backup Cisco CallManager, use the **no** form of this command.

ccm-manager redundant-host {*ip-address* | *DNS-name*} [*ip-address* | *DNS-name*]

no ccm-manager redundant-host {*ip-address* | *DNS-name*} [*ip-address* | *DNS-name*]

Syntax Description

ip-address	Internet protocol address of the backup Cisco CallManager.
DNS-name	Domain name system of the backup Cisco CallManager.

Defaults

If you do not configure a backup Cisco CallManager, the redundancy feature is considered off.

Command Modes

Global configuration

Command History

Release	Modification
12.1(3)T	This command was introduced with Cisco CallManager Version 3.0 and the Cisco Voice Gateway 200 (VG200).
12.2(2)XA	The command was implemented on Cisco 2600 series and 3600 series routers. The *DNS-name* argument was added.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T.

Usage Guidelines

You must configure one backup Cisco CallManager, and you can configure a maximum of two backup Cisco CallManagers. The list of IP addresses or DNS names is an ordered list. The Cisco CallManager defined in the **mgcp call-agent** command has the highest priority (that is, it is the primary Cisco CallManager). The gateway selects a Cisco CallManager based on the order in which it appears in this list.

Examples

In the following example, the IP address of the backup Cisco CallManager is 10.0.0.50:

Router(config)# **ccm-manager redundant-host 10.0.0.50**

Related Commands

Command	Description
ccm-manager application	Configures the port number for the redundant link application.
ccm-manager switchback	Configures the switchback mode that determines when the primary Cisco CallManager will be used if it becomes available again while a backup Cisco CallManager is being used.

ccm-manager switchback

When a backup Cisco CallManager is used because the primary Cisco CallManager is unavailable, use the **ccm-manager switchback** command in global configuration mode to specify when to use the primary Cisco CallManager once it becomes available again. To disable this command, use the **no** form of this command.

ccm-manager switchback {**graceful** | **immediate** | **schedule-time** *hh:mmm* | **uptime-delay** *minutes*}

no ccm-manager switchback

Syntax Description

graceful	After the last active call ends (when there is no voice call in setup mode on the gateway), control returns to the primary Cisco CallManager.
immediate	Regardless of the current conditions, when the TCP link to the primary Cisco CallManager is established, control switches back immediately to the primary Cisco CallManager.
schedule-time *hh:mm*	At a specified hour and minute, based on a 24-hour clock, control returns to the primary Cisco CallManager. If the specified time is earlier than the current time, the switchback occurs at the specified time on the following day.
uptime-delay *minutes*	When the primary Cisco CallManager has run for a specified number of minutes after a network connection has been reestablished to that CallManager, control is returned to the primary Cisco CallManager. Permitted values are from 1 to 1440 (1 minute to 24 hours).

Defaults

Graceful

Command Modes

Global configuration

Command History

Release	Modification
12.1(3)T	This command was introduced with Cisco CallManager Version 3.0 and the Cisco Voice Gateway 200 (VG200).
12.2(2)XA	The command was implemented on Cisco 2600 series and 3600 series routers.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T.

Usage Guidelines

This command allows you to configure switchback to the higher priority Cisco CallManager when it becomes available. Switchback allows call control to revert back to the original (primary) Cisco CallManager once service has been restored.

Examples

In the following example, the primary Cisco CallManager will be used as soon as it becomes available:

```
Router(config)# ccm-manager switchback immediate
```

Related Commands

Command	Description
ccm-manager application	Configures the port number for the redundant link application.
ccm-manager redundant-host	Configures the IP address or the DNS name of up to two backup Cisco CallManagers.
ccm-manager switchover-to-backup	Manually redirects a Cisco 2600 series or Cisco 3600 series router to the backup Cisco CallManager. The switchover occurs immediately. This command does not switch the gateway to the backup Cisco CallManager if you have the switchback option set to **immediate** and the primary Cisco CallManager is still running.

debug ccm-manager

To display Cisco CallManager debug messages, use the **debug ccm-manager** command in privileged EXEC mode. To disable this command, use the **no** form of this command.

debug ccm-manager [**errors** | **packets** | **events**]

no debug ccm-manager

Syntax Description

errors	Displays Cisco CallManager errors.
packets	Displays Cisco CallManager packets.
events	Displays Cisco CallManager events, such as when the primary Cisco CallManager failed and control switched to the backup Cisco CallManager.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(3)T	This command was introduced with Cisco CallManager Version 3.0 and the Cisco Voice Gateway 200 (VG200).
12.2(2)XA	The command was implemented on Cisco 2600 series and Cisco 3600 series routers.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T.

Examples

The following is sample output from the **debug ccm-manager events** command.

```
Router# debug ccm-manager events

c3660A#
00:22:31: CMAPP: Proc Keepalive - remain_wait:2000 ipaddr:10.0.0.50
00:22:33: CMAPP: Proc Keepalive - remain_wait:13000 ipaddr:10.0.0.50
00:22:33: CMAPP: Send Keepalive - last_traffic_time:15000 ipaddr:10.0.0.50
00:22:34: CMAPP: TCP open failed for 10.0.0.201, calling callback.
00:22:34: CMAPP-redunlink: cmapp_redun_link_callback(signal=3,sessionID=-2107756412)
00:22:34: CMAPP-redunlink: cmapp_conn_refused(sessionID=-2107756412)
00:22:34: CMAPP-redunlink: conn_refused for [1]:10.0.0.201
00:22:34: CMAPP-conn_refused: cmapp_host_table[0].link_state=3
00:22:34: CMAPP-conn_refused: cmapp_host_table[1].link_state=1
00:22:34: CMAPP: Freeing link record with address 825E3084 for 10.0.0.201. c3660A#
0:22:44: CMAPP-redunlink: cmapp_mgr_exec_redun_link_state.
active_cm_idx=0, backup_cm_idx=-1, ping_cm_idx=-1
00:22:44: CMAPP-redunlink: xgcp_oper_state=2,
```

```
0=MGCP_STATE_DOWN,1=MGCP_STATE_BLOCK_NEW_CALL,2=MGCP_STATE_ACTIVE
00:22:44: CMAPP-redunlink: cmapp_open_new_redun_link (for_active_cm=0,idx=-1)
00:22:44: CMAPP-redunlink: [0]:ipaddr=10.0.0.50,session=-2107758936,link=3,host=5
00:22:44: CMAPP-redunlink: [1]:ipaddr=10.0.0.201,session=0,link=1,host=2
00:22:44: CMAPP: PROCESSED REDUNLINK TIMER
```

Table 6 describes the significant fields shown in the sample output from the **debug ccm-manager events** command.

Table 5 debug ccm-manager Field Descriptions

Field	Description
nn:*nn*:*nn*:	Timestamp indicating when the event occurred.
CMAPP: *error message*	Cisco CallManager routine in which the error occurred.

Related Commands

Command	Description
show ccm-manager	Displays a list of Cisco CallManagers, as well as their status and availability.

mgcp call-agent

To configure the primary or default Cisco CallManager and to designate the optional destination UDP port number specifications to the specified Cisco CallManager, use the **mgcp call-agent** command in global configuration mode. To unconfigure the Cisco CallManager address, use the **no** form of this command.

mgcp call-agent *ip-address* [*port*] [**service-type** {**sgcp** | **mgcp**}]

no mgcp call-agent

Syntax Description

ip-address	Specifies the IP address or domain name of the Cisco CallManager.
port	(Optional) Specifies the UDP port that the Cisco CallManager will use. Valid values are 1025 through 65535.
service-type	(Optional) Specifies the type of gateway control service to be supported by the Cisco CallManager.
sgcp	(Optional) Simple Gateway Control Protocol.
mgcp	(Optional) Media Gateway Control Protocol.

Defaults

If the destination UDP port is not configured, port number 2427 is used. The default service type is **mgcp**, Version 0.1.

Command Modes

Global configuration

Command History

Release	Modification
12.1(1)T	This command was introduced for the Cisco AS5300.
12.1(3)T	The **service-type** keyword was added to the command.
12.2(2)XA	The command was implemented on Cisco 2600 series and 3600 series routers.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T.

Usage Guidelines

This is a required command that enables the functionality of MGCP and Cisco CallManager features. Use this command on any platform and media gateway.

When **service-type** is sent to **mgcp**, the call agent processes the Restart In Progress (RSIP) error messages sent by the gateway. When **service-type** is sent to **sgcp**, the call agent ignores the RSIP messages.

If you do not enter the **mgcp call-agent** command, no part of the system is controlled by any Cisco CallManager.

Examples

The following examples illustrate two formats for specifying the call agent (use either of the formats):

```
Router(config)# mgcp call-agent 209.165.200.225 5530 service-type mgcp
Router(config)# mgcp call-agent igloo 2009 service-type mgcp
```

Related Commands

Command	Description
debug mgcp events	Displays debug messages for MGCP events.
mgcp	Enables Media Gateway Control Protocol mode.

show ccm-manager

To display a list of Cisco CallManagers, as well as their status and availability, use the **show ccm-manager** command in privileged EXEC mode.

show ccm-manager [**hosts** | **redundancy**]

Syntax Description

hosts	Displays each configured Cisco CallManager, its operational status, and its host address.
redundancy	Displays failover mode and status information, including the redundant link port, the failover interval, keepalives, MGCP traffic time, switchover, and switchback.

Defaults

If you omit both keywords, **hosts** and **redundancy** information is displayed.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(3)T	This command was introduced with Cisco CallManager Version 3.0 and the Cisco Voice Gateway 200 (VG200).
12.2(2)XA	The command was implemented on Cisco 2600 series and 3600 series routers.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T.

Usage Guidelines

Use this command to determine whether your primary or backup Cisco CallManager is down, idle, or backup polling.

Examples

The following is sample output from the **show ccm-manager** command for displaying the status and availability of the primary and backup Cisco CallManagers.

```
Router# show ccm-manager

c3660A#show ccm-manager
MGCP Domain Name: c3660A.cisco.com
Total number of host: 2
Priority        Status         Host
===================================================
Primary         Registered     10.0.0.201
First backup    Backup polling 10.0.0.50
Second backup   Undefined

Current active Call Manager: 10.0.0.201
Current backup Call Manager: 10.0.0.50
Redundant link port:         2428
Failover Interval:           30 seconds
```

```
Keepalive Interval:          15 seconds
Last keepalive sent:         00:20:18 (elapsed time: 00:00:06)
Last MGCP traffic time:      00:20:18 (elapsed time: 00:00:06)
Last switchover time:        None
Switchback mode:             Not selected (Default:Graceful)
```

Table 6 describes the significant fields shown in the display.

Table 6 show ccm-manager Field Descriptions

Field	Description
MGCP Domain Name (System)	System used in the Internet for translating names of network nodes into IP addresses.
Total number of host	Number of Cisco CallManagers.
Priority	Priority of the Cisco CallManagers. Possible priorities are primary, first backup, and second backup.
Status	Current usage of the Cisco CallManager. Possible values are registered, idle, backup polling, and undefined.
Host	Host address of the Cisco CallManager.
Current active Call Manager	Active Cisco CallManager. Can be the primary, first backup, or second backup Cisco CallManager.
Current backup Call Manager	Backup Cisco CallManager currently being used. Empty when no backup is available.
Redundant link port	Port that the Cisco CallManager will use.
Failover Interval	Maximum amount of time that can elapse without the gateway receiving messages from the currently active Cisco Call Manager, or the gateway switches to the backup Cisco Call Manager.
Keepalive Interval	If the gateway hasn't received any messages from the currently active Cisco CallManager within the specified amount of time, the gateway sends a keepalive message to the Cisco CallManager to be sure that it is OK.
Last keepalive sent	When the last keepalive message was sent.
Last switchover time	Last time switchover occurred.
Switchback mode	Displays the switchback mode configuration that determines when the primary Cisco CallManager will be used if it becomes available again while a backup Cisco CallManager is being used.

Related Commands

Command	Description
show mgcp	Displays the router's global parameters for MGCP.

Glossary

call leg—A discrete segment of a call connection that lies between two points in the connection. An end-to-end call consists of four call legs, two from the perspective of the source access server, and two from the perspective of the destination access server.

Cisco CallManager—The Cisco CallManager serves as the software-based call-processing component of the Cisco IP telephony solution. The Cisco CallManager system extends enterprise telephony features and functions to packet telephony network devices such as IP phones, media processing devices, Voice-over-IP (VoIP) gateways, and multimedia applications. The Cisco CallManager system includes a suite of integrated voice applications that perform voice conferencing and manual attendant console functions.

Cisco CallManager server—Cisco's high-availability server platform on which Cisco CallManager software comes preinstalled.

cluster—Set of Cisco CallManagers that share the same database.

codec—A DSP software algorithm that compresses/decompresses speech or audio signals.

dial peer—Defines the characteristics associated with a call leg. Dial peers are used to apply attributes to call legs and to identify the call origin and destination. In Voice over IP, there are two types of dial peers: POTS and VoIP. Use the **dial-peer** voice command to define dial peers and to switch to dial-peer configuration mode.

digital signal processor—See DSP.

DNS—Domain Name System. A system used in the Internet for translating names of network nodes into IP addresses.

domain name system—See DNS.

DSP—Digital signal processor. A specialized computer chip designed to perform speedy and complex operations on digitized waveforms. It is useful in processing sound, such as voice phone calls, and video. A DSP segments the voice signal into frames and stores them in voice packets.

DTMF— Dual tone multifrequency. A system used by touch tone telephones where one high and one low frequency, or tone, is assigned to each touch tone button on a phone. DTMF digits can be detected by the voice ports after the call setup is complete and are also trapped by the session application at either end of the connection and carried over the IP network encapsulated in Real Time Conferencing Protocol (RTCP) by using the RTCP APP extension mechanism.

dual tone multi-frequency—See DTMF.

E&M—The "ear and mouth" interface (also called the "earth and magnet" interface, or the "recEive and transMit" interface). Trunk circuits connect telephone switches to one another; they do not connect end-user equipment to the network. The most common form of analog trunk circuit is the E&M interface, which uses special signaling paths that are separate from the trunk's audio path to convey information about calls. The signalling paths are known as the E-lead and the M-lead. E&M connections from routers to telephone switches or to PBXs are preferable to FXS/FXO connections, because E&M provides better answer and disconnect supervision.

FXO—Foreign Exchange Office interface. A connection between a POTS telephone and a digital telephony switching system.

FXS—Foreign Exchange Station interface. A connection between a digital telephony switching system and a POTS telephone.

gateway—A special purpose device that performs an application-layer conversion of information from one protocol stack to another. To connect an IP telephony device to the Public Switched Telephone Network (PSTN), you must use an intermediary device, called a gateway. A VoIP gateway allows users of IP phones and PC-based soft phones to exchange calls with users of plain old telephone service (POTS) phones on the PSTN. The gateway translates between the signals used on the PSTN and the IP packets used to transmit data on a TCP/IP network.

H.323—A collection of protocols that define standard methods for interconnecting H.323 endpoints and POTS devices. It is a method for converting between voice and data transmission formats and for managing connections between telephony endpoints.

IP address—Internet protocol address. A 32-bit address assigned to hosts using TCP/IP. An IP address belongs to one of five classes (A, B, C, D, or E) and is written as 4 octets separated by periods (dotted decimal format). Each address consists of a network number, an optional subnetwork number, and a host number. The network and subnetwork numbers together are used for routing, and the host number is used to address an individual host within the network or subnetwork. A subnet mask is used to extract network and subnetwork information from the IP address. Also known as an Internet address. See also subnet mask.

MGCP—Media Gateway Control Protocol. MGCP makes possible external control and management of data communications equipment operating at the edge of multi-service packet networks (known as media gateways) by software programs. The software programs are known as “call agents” or “media gateway controllers.”

Media Gateway Control Protocol—See MGCP.

PBX—private branch exchange. Digital or analog telephone switchboard located on the subscriber premises, typically with an attendant console, and used to connect private and public telephone networks. A PBX is a small, privately owned version of the phone company's larger central switching office. It is connected to one or more central offices by trunks, and provides service to a number of individual phones. It can be used, for example, in a hotel, business, or government office. On a PBX, an outside line is normally accessed by the dialing of an access digit, such as 9.

POTS—Plain old telephone service. Dial peer describing the characteristics of a traditional telephony network connection. POTS dial peers point to a particular voice port on a voice network device. To configure a POTS dial peer, you must configure the associated telephone number and the logical interface.

private branch exchange—See PBX.

protocol—A set of rules or conventions that govern the format and relative timing of data movement in a communications network. There are three basic types of protocols: character-oriented, byte-oriented, and bit-oriented. The protocols for data communications cover such things as framing, error handling, transparency, and line control. Ethernet is an example of a LAN protocol.

PSTN—Public Switched Telephone Network. General term referring to the variety of telephone networks and services in place worldwide.

public switched telephone network—See PSTN.

SGCP—Simple Gateway Control Protocol. A protocol that controls Voice over IP gateways by an external element (called a call-agent).

simple gateway control protocol—See SGCP.

subnet mask—A 32-bit address mask used in IP to indicate the bits of an IP address that are being used for the subnet address. A subnet mask is used to extract network and subnetwork information from the IP address.

T1—Trunk Level 1. A digital transmission link that has a total signaling speed of 1.544 Mbps. Uses time-division multiplexing (TDM) to divide the available bandwidth into 24 64-kbps timeslots. T1 transmits through the telephone-switching network using AMI or B8ZS coding. T1 is a standard for digital transmission in North America. A T1 device combines the output of up to 24 regular telephone lines for transmission over a digital network. Also known as T-1.

TCP—Transmission Control Protocol. A connection-oriented transport layer protocol that provides reliable full-duplex data transmission. TCP is a part of the TCP/IP protocol stack.

telephony—Telephony is the science of converting sound to electrical signals and transmitting it between widely removed points.

transmission control protocol—See TCP.

UDP—User Datagram Protocol. A connectionless messaging protocol for delivery of data packets. A simple protocol that exchanges datagrams without acknowledgments or guaranteed delivery, requiring that error processing and retransmission be handled by other protocols.

User Datagram Protocol—See UDP.

Voice over IP—See VoIP.

VoIP—Voice over IP. Enables users to have voice communications over a data network using the Internet Protocol (IP). VoIP can consolidate voice and data traffic for more efficient use of bandwidth, reduce toll charges, and make possible alternatives to expensive and proprietary PBX systems. VoIP dial peers point to specific VoIP devices. To configure a VoIP dial peer, you must configure the associated destination telephone number and a destination IP address. Use the destination-pattern command to define the destination telephone number associated with a VoIP peer. Use the session target command to specify a destination IP address for a VoIP peer.