



# Interworking of Cisco MGCP Voice Gateways and Cisco CallManager Version 3.1

## Feature History

Release	Modification
12.1(3)T	This feature was introduced with Cisco CallManager Version 3.0 and the Cisco Voice Gateway 200 (Cisco VG200).
12.1(5)XM	H.323 support was added for E1 and T1 PRI, E&M, E1-CAS and BRI. Analog support with MGCP was added and analog DID was also added.
12.2(2)XA	Support was added for Cisco CallManager Version 3.0(8) and the Cisco 2600 and Cisco 3600 series routers but only for analog Foreign Exchange Stations (FXSs) and Foreign Exchange Offices (FXOs).
12.2(2)XN	Support for Media Gateway Control Protocol (MGCP) voice gateway interoperability was added to Cisco CallManager Version 3.1, Cisco 2600 series routers, Cisco 3600 series routers, and the Cisco VG200. New MGCP features are also supported including T1 CAS, ISDN PRI Backhauling, Single-Point Configuration, Gateway Fallback, and Multicast Music-on-Hold (MOH).

This document describes Cisco IOS support for the interworking of Cisco MGCP voice gateways and Cisco CallManager Version 3.1 in Cisco IOS Release 12.2(2)XN. This document also describes new command-line interface (CLI) commands that enable you to configure Cisco IP telephony devices for operation in a Voice over IP (VoIP) network.

This document contains the following sections:

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# Feature Overview

This section briefly describes the the hardware platforms, communications protocols, and the Cisco IOS software available to support IP telephony MGCP gateway functionality in a Cisco AVVID (Architecture for Voice, Video, and Integrated Data) network. It contains the following sections:

- [Cisco 2600 Series Routers, page 2](#)
- [Cisco 3600 Series Routers, page 2](#)
- [Cisco Voice Gateway 200, page 3](#)
- [Media Gateway Control Protocol, page 3](#)
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- [Benefits of Cisco VoIP Technology Using MGCP, page 10](#)
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## Cisco 2600 Series Routers

Cisco 2600 series routers are key members of the current Cisco data, voice, and video integration portfolio. This family of modular routers enables network managers and service providers to meet a broad range of needs in branch offices for end-to-end IP and Frame Relay based packet telephony solutions. A typical branch office could need the following:

- Multiservice voice and data integration services
- Analog and digital dial access services
- Internet and intranet access with firewall security
- Virtual Private Networks (VPNs)
- Inter-VLAN routing services
- Routing with bandwidth management

When you use MGCP with a Cisco 2600 series router, all configuration elements associated with dial plans are controlled by the Cisco CallManager and need not be configured on the routers for MGCP-managed endpoints.

## Cisco 3600 Series Routers

Cisco 3600 series routers are also key members of the Cisco data, voice, and video integration portfolio that constitute a family of modular, multiservice access platforms intended for use in medium- and large-sized offices and by smaller Internet service providers.

Compared to Cisco 2600 series routers, Cisco 3600 series routers generally provide higher port densities, greater performance, and greater expansion capabilities, while also taking advantage of the enhanced interoperability features offered through Cisco CallManager Version 3.1.

As with the Cisco 2600 series routers, when you use MGCP with a Cisco 3600 series router, all configuration elements associated with dial plans are controlled by the Cisco CallManager and need not be configured on the routers for MGCP-managed endpoints.

## Cisco Voice Gateway 200

The Cisco VG200 is designed primarily for use in branch offices that do not require support for integrated WAN data traffic management.

The Cisco VG200 incorporates a single slot that can accommodate a voice network module for connecting an analog or digital interface to the Public Switched Telephone Network (PSTN). A voice network module can accommodate up to two voice interface cards (VICs); each card provides up to two ports.

The Cisco VG200 has a Fast Ethernet LAN port to facilitate connection to a VoIP network.

## Media Gateway Control Protocol

MGCP is used in combination with a call agent (see the section “[Cisco CallManager Version 3.1, page 4](#)”) to enable the remote control and management of voice and data communications devices at the edge of multiservice IP packet networks (see the “[MGCP Voice Gateways](#)” section on page 3).

As a communication protocol, MGCP overcomes the distributed configuration and administration problems inherent in the use of protocols such as the H.323 standard. Using a centralized architecture, MGCP greatly simplifies the configuration and administration of voice gateways that operate in multiservice IP packet networks.

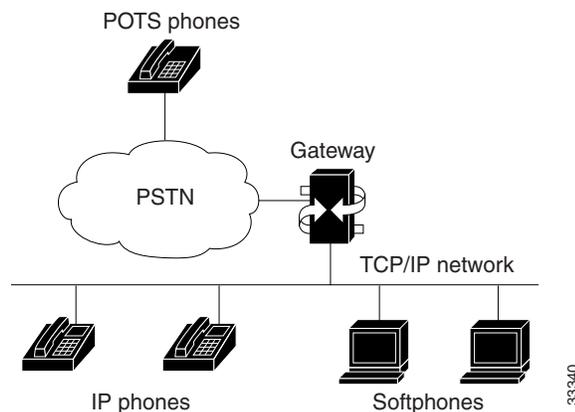
In addition, MGCP supports the presence of multiple (redundant) call agents in a VoIP network, thereby eliminating the potential for a single point of failure in controlling the MGCP gateways in the network.

In effect, MGCP functions as a master/slave protocol to ensure that the MGCP voice gateways in a VoIP network properly receive and execute the configuration, control, and management commands that are issued to the gateways by Cisco CallManager.

## MGCP Voice Gateways

Cisco MGCP voice gateways provide simple and inexpensive interfaces between LAN and WAN data networks and the PSTN. As illustrated in [Figure 1](#), however, an intermediary device called a gateway is required to provide connectivity between the IP phones and softphones in a TCP/IP-based network and the plain old telephone service (POTS) phones in the PSTN.

**Figure 1** Typical VoIP Network with an MGCP Voice Gateway



This combination of technologies and telephony services forms a larger functional domain that is commonly referred to as a VoIP network.

An MGCP voice gateway enables users of IP phones and PC-based softphones to exchange calls with users of POTS phones in the PSTN. The gateway enables this interaction by translating between the signals used in the PSTN (for connecting the communicating endpoints) and the information carried by IP packets (for transmitting traffic across a TCP/IP network).

An MGCP gateway derives most of its configuration information from Cisco CallManager. Initially, through Cisco IOS CLI commands, you configure the gateway to operate under MGCP control and, additionally, you identify the gateway to the Cisco CallManager server for gateway registration purposes. Once the MGCP gateway is registered with a Cisco CallManager server, the latter assumes control over establishing and tearing down connections between the IP endpoints in your VoIP network and the communicating endpoints in the PSTN connected by means of your VoIP network.

Using MGCP and Cisco CallManager, you can deploy a wide range of IP telephony solutions in a VoIP network with the following Cisco platforms functioning as MGCP voice gateways:

- Cisco 2600 series routers
- Cisco 3600 series routers
- Cisco VG200

Typically, IOS gateways that act as MGCP voice gateways to Cisco CallManager include IP packet networks that handle the translation of signaling information used in setting up, maintaining, and tearing down voice circuits. The circuits pass through an integrated VoIP network as part of an overall, end-to-end communication path.

## Cisco CallManager Version 3.1

Cisco CallManager Version 3.1 provides a central point of configuration, administration, and control for the MGCP voice gateways in a VoIP network. By means of MGCP, Cisco CallManager controls the setting up and tearing down of connections between the endpoints in a VoIP network, as well as controlling the communicating endpoints in the PSTN.

When using MGCP in combination with Cisco MGCP voice gateways, Cisco CallManager controls all the dial-plan related configuration elements. Such elements do not need to be configured on a gateway in order to manage the communicating endpoints.

The following features and services are supported by Cisco CallManager Version 3.1 and described briefly in the following sections:

- [ISDN PRI Backhaul and T1 CAS, page 4](#)
- [MGCP Gateway Fallback, page 5](#)
- [Single-Point Configuration of MGCP Gateways in AVVID Networks, page 8](#)
- [Multicast Music-on-Hold, page 9](#)

## ISDN PRI Backhaul and T1 CAS

ISDN PRI backhaul provides a method for transporting complete IP telephony signaling information from an ISDN PRI interface of an MGCP voice gateway to Cisco CallManager through a highly reliable TCP connection.

This feature works by terminating all the ISDN PRI Layer 2 (Q.921) signaling functions in the Cisco IOS code on the MGCP voice gateway while, at the same time, packaging all the ISDN PRI Layer 3 (Q.931) signaling information into packets for transmission to the Cisco CallManager through an IP tunnel over a highly reliable TCP connection. This methodology ensures the integrity of the Q.931 signaling information being passed through the network for managing IP telephony devices.

A rich set of user-side and network-side ISDN PRI calling functions is supported by the ISDN PRI backhaul feature. A single TCP connection is used by the gateway to backhaul all the ISDN D channels to Cisco CallManager. The “SAP/Channel ID” parameter in the header of each message identifies individual D channels. In addition to carrying the backhaul traffic, the inherent TCP keepalive mechanism is also used to determine MGCP voice gateway connectivity to an available call agent.

The MGCP voice gateway also establishes a TCP link to the backup (secondary) Cisco CallManager server. In the event of Cisco CallManager switchover, the ISDN PRI backhaul functions are assumed by the secondary Cisco CallManager server. During this switchover, all active ISDN PRI calls are preserved, and the affected MGCP gateway is registered with the new Cisco CallManager server through a Restart-in-Progress (RSIP) message to ensure continued gateway operation.

T1 CAS is supported in non-backhaul fashion and supported CAS signaling types on the Cisco CallManager are E&M, wink-start, and E&M delay-dial. E1 CAS is not supported.

## MGCP Gateway Fallback

MGCP gateway fallback is designed to improve the reliability of PSTN interfaces on MGCP voice gateways in a VoIP network. This feature provides basic call processing support on an MGCP voice gateway when it loses connectivity to all of the Cisco CallManager servers that are configured for the gateway. A prioritized list of Cisco CallManager servers is configured on the gateway, thus making each Cisco CallManager server potentially available for use as a backup call agent.

On startup, the MGCP voice gateway attempts to establish a TCP connection to the highest order Cisco CallManager server on the configured list. If successful, the gateway registers itself with the primary (highest priority) call agent.

If no call agent in this prioritized list is accessible, the gateway, in effect, “falls back” onto itself and uses its default H.323 session application (Version 2) to perform basic call-handling functions in support of the following types of interfaces on the gateway:

- FXS analog interfaces—For connecting to the PSTN or analog phones
- FXO analog interfaces—For connecting to the PSTN PBXs
- T1-CAS digital interfaces—For connecting to the PSTN or PBXs
- T1 and E1 PRI digital interfaces—For connecting to PBXs and central offices (COs)

Under the control of Cisco CallManager Version 3.1, the following platforms as well as others can function in a VoIP network in support of the MGCP gateway fallback feature:

- Cisco 2600 series routers
- Cisco 3600 series routers
- Cisco VG200

These platforms provide connectivity in a Cisco VoIP network to traditional telephone trunks linked to the PSTN.

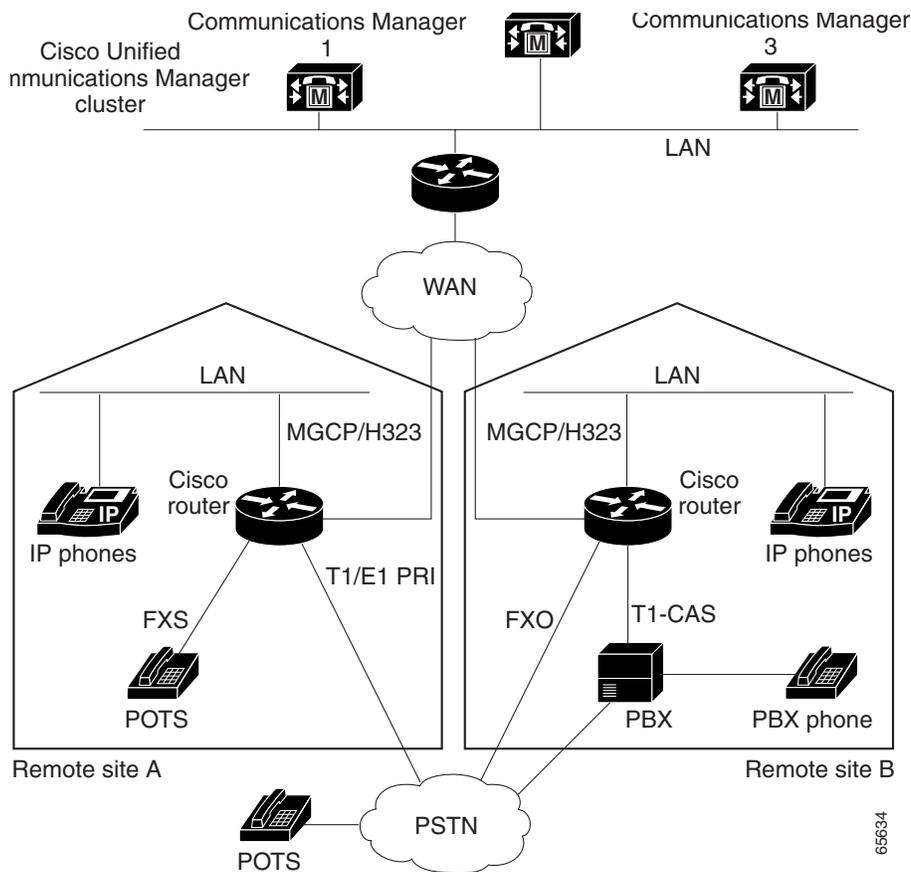
In the setting of a remote branch office and a centralized CallManager cluster, the MGCP voice gateways in the remote site communicate through a WAN TCP connection to a host Cisco CallManager server in your VoIP network. The gateway maintains this connection by sending empty MGCP Notify (NTFY) keepalive messages to Cisco CallManager at 15-second intervals. If the active Cisco CallManager fails to acknowledge receipt of the keepalive message within 30 seconds, the gateway attempts to switch over to the next highest order Cisco CallManager server that is available.

If none of the Cisco CallManager servers respond, the gateway switches into fallback mode and reverts to its default H.323 session application for basic call control support of IP telephony activity in the network.

H.323 is a standardized communication protocol that enables dissimilar devices to communicate with each other using a common set of codecs, call setup and negotiating procedures, and basic data transport methods.

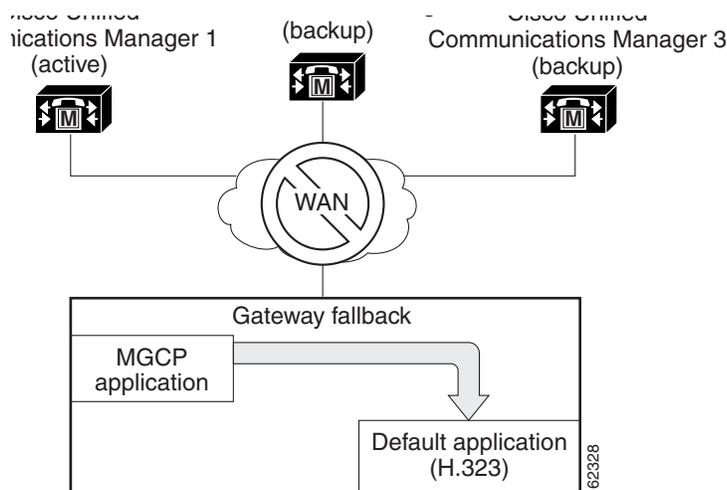
Figure 2 illustrates a typical VoIP network topology in which MGCP gateway fallback is supported.

**Figure 2 Typical VoIP Network Topology Supporting the MGCP Gateway Fallback Feature**



Cisco CallManager Version 3.1 provides the following functionality:

- Gateway fallback support—Figure 3 depicts the fallback process to the default H.323 session application that occurs when the WAN TCP connection to the primary Cisco CallManager server is lost and no backup Cisco CallManager server is available.

**Figure 3 MGCP Gateway Fallback Transition to the Default H.323 Session Application**

All active MGCP analog and T1 CAS calls are maintained during the fallback transition. Callers are unaware of the fallback transition, and these active MGCP calls are cleared only when the communicating callers hang up. Active MGCP PRI backhaul calls are released during fallback.

Any transient MGCP calls (that is, calls that are not in the connected state) are cleared at the onset of the fallback transition and must be attempted again later.

- **Basic connection services in fallback mode**—Provides basic connection services for IP telephony traffic that passes through the gateway.

When the local MGCP gateway transitions into fallback mode, the default H.323 session application assumes responsibility for handling new calls. Only basic two-party voice calls are supported during the fallback period.

Except for ISDN T1 and E1 PRI calls, all the MGCP calls that are active at the time of fallback are preserved, while transient calls are released. When a user completes (hangs up) an active MGCP call, the MGCP application handles the on-hook event and clears all call resources.

- **Rehome support**—Provides a rehome function while in the gateway fallback mode that detects the restoration of a WAN TCP connection to the primary Cisco CallManager server.

When the fallback mode is in effect, the affected MGCP gateway repeatedly tries to open a TCP connection to a Cisco CallManager server in the prioritized list of call agents. This process continues until one of the Cisco CallManager servers in the prioritized list responds.

On response, the TCP open request from the MGCP gateway is honored, and the gateway reverts to MGCP mode and sends a Restart-in-Progress (RSIP) message to effect registration of the gateway with the responding Cisco CallManager.

Except for ISDN T1 and E1 PRI calls, all currently active calls that are initiated and set up during the fallback period are maintained by the default H.323 session application while transient calls are released. After rehome occurs, the new Cisco CallManager assumes responsibility for controlling new IP telephony activity.

## Single-Point Configuration of MGCP Gateways in AVVID Networks

When you configure MGCP voice gateways and Cisco CallManager Version 3.1 in the Cisco Architecture for Voice, Video, and Integrated Data (AVVID) networks, you can use a centralized tftpboot directory on a host device in your network to automatically download most of the configuration. The same CallManager server can be concurrently configured as a TFTP server. The downloaded configuration information that the gateways require for operability in a VoIP network can be created and maintained in a unified database.

To enable this feature, you first configure MGCP in your VoIP network through the Cisco CallManager Administrator (the web-based graphical user interface [GUI] of Cisco CallManager). You can also use this facility to automatically issue Cisco IOS CLI configuration commands (in the form of XML files generated from the GUI) to designated MGCP voice gateways in your network. Such commands can be issued by Cisco CallManager to MGCP voice gateways at startup time or whenever a desired configuration change on an MGCP gateway needs to be made.

Each MGCP gateway type in your VoIP network has an associated gateway-specific configuration schema that is stored in the centralized tftpboot directory, from which a tailored XML file can be created (per the appropriate schema) and transmitted through TFTP to a designated MGCP voice gateway.

When the network administrator or other responsible individual at the console of the Cisco CallManager changes the configuration information in the database, a notification message is sent by Cisco CallManager to the affected MGCP voice gateways or IP phones in the network, instructing these devices to download a new XML configuration file.

Each such device incorporates an XML parser or configurator that interprets the XML file content according to its device-specific requirements. In the case of Cisco IOS based MGCP voice gateways, for example, the gateway translates the XML file content into specific Cisco IOS CLI commands for local execution.

When an MGCP voice gateway or IP phone is first started, it has already been preconfigured with or has obtained through Dynamic Host Configuration Protocol (DHCP) the following information, enabling the device to retrieve an XML file from the TFTP server:

- A unique device identifier, which can be either of the following:
  - A specific device name (on Cisco IOS based MGCP voice gateways only)
  - The MAC address of the device (for gateways and IP phones that are not based on Cisco IOS software)
- IP address of the TFTP server in the network
- Sufficient information to configure an IP interface on the device
- Routing information required to access the TFTP server

With this configuration information available at startup, the MGCP gateway proceeds to download the XML file from the TFTP server. The gateway parses the XML file, converts the information to appropriate Cisco IOS CLI configuration commands, and configures itself to run in the VoIP environment. Finally, the gateway registers itself with Cisco CallManager through an RSIP message, thereby readying itself for service in the network.

When an MGCP gateway configuration in the tftpboot directory is changed using the Cisco CallManager Administrator GUI, the GUI user has the option to reset the gateway or to restart it. If either option is selected, the Cisco CallManager Administrator triggers a configuration update, which causes a special message to be sent to the MGCP gateway notifying it that a new configuration download is required. Thus, with the capabilities supported by this single-point configuration feature, you need not manually synchronize the configuration process between an MGCP voice gateway and the Cisco CallManager.

If a configuration change made using the Cisco CallManager Administrator is limited to a single interface on an MGCP voice gateway, an interface-specific XML configuration file is created, and an interface-specific download is indicated to the gateway. In this case, only the configuration information for that interface is affected, and only those IP telephony calls that pass through that interface are affected.

## Multicast Music-on-Hold

The MOH feature provides the functionality to stream music from an MOH server to the voice interfaces of on-net and off-net callers that have been placed on hold.

This integrated multicast capability of Cisco CallManager 3.1 is implemented through the H.323 signaling plane in Cisco CallManager.

In an MOH environment, whenever caller A places caller B on hold, Cisco CallManager requests the MOH server to stream RTP packets to the “on-hold” interface through the preconfigured multicast address. In this way, RTP packets can be relayed to appropriately configured voice interfaces in a VoIP network that have been placed on hold.

Multiple MOH servers can be present in the same network, but each server must have a different Class D IP address, and the address must be preconfigured in Cisco CallManager and the Cisco IOS MGCP voice gateways.

The MOH feature enables you to subscribe to a music streaming service when using a Cisco IOS MGCP voice gateway. By means of a preconfigured multicast address on a gateway, the gateway can “listen for” Real-Time Transport Protocol (RTP) packets that are broadcast from a default router in the network and can relay the packets to designated voice interfaces in the network.

RTP is the Internet-standard protocol for transporting real-time data across a network, including audio and video information. Thus, RTP is well suited for media on demand and interactive services, such as IP telephony.

The default router in the network for handling multicast traffic must have the following enabled:

- Multicast routing
- A multicast routing protocol, for example Protocol Independent Multicast (PIM) or Distance Vector Multicast Routing Protocol (DVMRP)
- An IP routing protocol, for example Routing Information Protocol (RIP) or Open Shortest Path First (OSPF)

When you configure a multicast address on a gateway, the gateway sends an Internet Gateway Management Protocol (IGMP) “join” message to the default router, indicating to the default router that the gateway is to receive RTP multicast packets.

## Benefits of Cisco VoIP Technology Using MGCP

Cisco VoIP telephony products enable live voice traffic to be carried over a multiservice IP packet data network by means of MGCP. Thus, voice and data traffic can be integrated for transmission across an IP network infrastructure. This combination of VoIP technology and networking resources yields the following user benefits:

- Ensures more efficient use of available network bandwidth.
- Employs a broader range of hardware platforms in support of IP telephony functionality.
- Reduces toll charges for voice communications.
- Provides a viable alternative to costly, proprietary PBX switching systems.

## Restrictions

- Voice interfaces on the NM-HDA and the AIM-VOICE-30 are not supported.
- Integrated access is not supported when voice is controlled using MGCP from a CallManager. Integrated access means that the channels on a T1 and E1 interface are divided between a group used for voice and another group used for WAN access.
- T1 and E1 protocols, such as QSIG, E1 R2, T1 FGD, and PRI NFAS, are not supported with MGCP only with H.323.

## Related Documents

- *Cisco CallManager Administration Guide*
- *Cisco IOS Interface Command Reference*, Release 12.2
- *Cisco IOS IP Command Reference, Vol. 1 of 3: Addressing and Services*, Release 12.2
- *Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.2
- *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2
- *Cisco 2600 and 3600 Routers MGCP Voice Gateway Interoperability with Cisco CallManager*, Release 12.2(2)XA

## Supported Platforms

- Cisco 2600 series
- Cisco 3600 series
- Cisco VG200

You can configure any of these platforms to operate as MGCP voice gateways in a VoIP network, enabling you to take advantage of the new and enhanced gateway interoperability features now supported by Cisco CallManager Version 3.1 (see the [“Cisco CallManager Version 3.1” section on page 4](#)).

# 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

- Cisco IOS Release 12.2(2)XN or a later release, IP Plus or better. To determine the version of Cisco IOS software on the Cisco VG200, Cisco 2600 series router, Cisco 3600 series router, log into the router and enter the **show version** command in EXEC mode.
- The following are memory requirements:
  - For the Cisco 2600 series router: 16 MB Flash memory and 64 MB DRAM
  - For the Cisco 3620 router and Cisco 3640 router: 16 MB Flash memory and 64 MB DRAM
  - For the Cisco 3660 router: 16 MB Flash memory and 128 MB DRAM
  - For the Cisco VG200: 8 MB Flash memory and 64 MB DRAM
- Cisco CallManager Version 3.1.
- NM-HDV and NM-2V hardware.

## Configuration Tasks

This section presents the following configuration tasks for enabling MGCP support for Cisco CallManager Version 3.1 and the Cisco IOS based MGCP voice gateways. Each task in the list is identified as either required or optional.

- [Single-Point Configuration of MGCP Voice Gateways in AVVID Networks, page 12](#) (required)
- [Configuring MGCP Control of Dial Peers and Voice Ports, page 14](#) (required)
- [Configuring MGCP Globally in a VoIP Network, page 16](#) (required)
- [Configuring MGCP Control of the Cisco CallManager, page 18](#) (required)
- [Configuring MGCP T1 CAS, page 21](#) (optional)
- [Configuring ISDN Signaling Backhaul, page 23](#) (optional)
- [Enabling MGCP Gateway Fallback Support, page 27](#) (optional)
- [Enabling Multicast Music-on-Hold, page 28](#) (optional)

## Single-Point Configuration of MGCP Voice Gateways in AVVID Networks

When using a Cisco IOS voice gateway in conjunction with MGCP and Cisco CallManager Version 3.1, you can complete the necessary configuration for a given gateway on the Cisco CallManager server and download the configuration to that gateway through a TFTP server.

Before performing the steps in this section, you should complete the basic configuration of your MGCP voice gateway by using the initial configuration dialog and the CLI. Your gateway configuration, as well as that of other network devices, must provide connectivity between your target gateway and the TFTP server.


**Note**

The IP host name should match the gateway name that is specified in the CallManager configuration.

After the required basic configuration of your MGCP voice gateway is completed and the gateway is reset, the configuration file, formatted in XML, is downloaded automatically from the Cisco CallManager server to your local gateway through a TFTP server. On receipt, the local gateway parses this file, converts it to Cisco IOS CLI commands, and uses it to update its active configuration.

After the initial download is completed, the local gateway saves the configuration file. If the gateway is restarted or reset from the Cisco CallManager GUI and the specified TFTP server is not available, the gateway keeps trying to download the file and does not alter the current configuration.


**Note**

Downloading MGCP gateway configuration information overrides any existing MGCP local configuration information. For this reason, you should remove any existing MGCP settings or port configuration information before proceeding with the following steps.

To enable the MGCP configuration download feature, use the following commands in EXEC configuration mode:

Command	Purpose
Router# <b>ccm-manager config</b>	Enables single-point configuration of MGCP voice gateway.

To obtain information about the status of the configuration download feature, use the following privileged EXEC command on the MGCP voice gateway:

```
Router# show ccm-manager config-download
```

The system displays the current status of the download feature, as shown in the following example.

```
Configuration Auto-download Information
=====
Current version-id: {1645327B-F59A-4417-8E01-7312C61216AE}
Last config-downloaded:00:00:49
Current state: Waiting for commands
Configuration Download statistics:
      Download Attempted           : 4
      Download Successful          : 4
      Download Failed              : 0
      Configuration Attempted     : 1
      Configuration Successful     : 1
      Configuration Failed(Parsing): 0
      Configuration Failed(config) : 0
Last config download command: New Registration
```

**Note**

For a description of each of the above output display fields, refer to the **show ccm-manager config-download** command reference page.

## Verifying the MGCP Gateway Configuration Settings

To view the current MGCP gateway configuration parameters, use the **show mgcp** command in privileged EXEC mode. These settings determine how the gateway will behave in an MGCP environment. The following is sample output from the **show mgcp** command:

```
Router# show mgcp

MGCP Admin State ACTIVE, Oper state ACTIVE - Cause Code NONE
MGCP call-agent: 172.20.71.44 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 for VoIP disabled for all codec types
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 disabled
MGCP request timeout 500, MGCP request retries 3
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 lwm 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 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
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
ms-package dt-package gc-package
MGCP VoAAL2 ignore-lco-codec DISABLED>
```

**Note**

For a description of each of the above output display fields, refer to the **show mgcp** command reference page.

## Configuring MGCP Control of Dial Peers and Voice Ports



### Note

All dial plan configuration elements are controlled by Cisco CallManager and should not be configured on the MGCP voice gateway for MGCP-managed endpoints (that is, any endpoint with an **application mgcpapp** command in its associated dial-peer statement). Also, you should not use the **destination-pattern** and **session target** dial-peer configuration commands or use the **connection** voice-port configuration command on the MGCP voice gateway, unless you are configuring MGCP gateway fallback. To configure MGCP gateway fallback, the H.323 dial peers must be configured with the **destination-pattern** and **session target** dial-peer configuration commands.

To configure dial peers and voice ports for communication with the Cisco CallManager under MGCP control, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# <b>dial-peer voice</b> <i>number</i> <b>pots</b>	Designates the specified dial peer as a POTS device using VoIP encapsulation.
Step 2	Router(config-dial-peer)# <b>application mgcpapp</b>	Enables MGCP on the dial peer.
Step 3	Router(config-dial-peer)# <b>exit</b>	Exits dial-peer configuration mode.
Step 4	Router(config)# <b>voice-port</b> <i>slot/subunit/port</i>	Enables voice port configuration mode and binds the MGCP application to the specified voice port. Valid values for the <i>slot</i> , <i>subunit</i> , and <i>port</i> arguments vary by router platform.
Step 5	Router(config-voiceport)# <b>no shutdown</b>	Activates the voice port. If a voice port is not used, use the <b>shutdown</b> command to shut it down.  <b>Note</b> The <b>shutdown</b> and <b>no shutdown</b> commands need to be entered on only one of the ports.

The gateway is now ready to communicate with the Cisco CallManager under MGCP control. The gateway periodically sends out messages attempting to establish a connection to the Cisco CallManager.

When the Cisco CallManager configuration is complete, the connection with the Cisco CallManager should automatically be established. You should not have to perform any further configuration tasks.

## Verifying the Dial Peer and Voice Port Configurations

To verify the configuration information for the dial peers, use the **show dial-peer voice** command in privileged EXEC mode. The following is sample output for a VoIP dial peer:

```
Router# 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.

```

**Note**


---

For a description of each of the above output display fields, refer to the **show dial-peer voice** command reference page.

---

To verify the configuration information for the voice ports, use the **show voice port** command in privileged EXEC mode. The following is sample output for a Cisco 3600 series router with a Foreign Exchange Station (FXS) analog voice port.

```

Router# 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 information as 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 information as 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

```



**Note** For a description of each of the above output display fields, refer to the **show voice port** command reference page.

## Configuring MGCP Globally in a VoIP Network

To configure MGCP globally in a VoIP network, use the following commands in global configuration mode:

	Command	Purpose
Step 1	Router(config)# <b>hostname</b> <i>name</i>	Assigns a unique name to a network router with the <i>name</i> argument enabling the Cisco CallManager to identify the device. The default device name is Router.  <b>Note</b> MGCP control takes effect only if MGCP is configured and enabled globally.
Step 2	Router(config)# <b>mgcp call-agent</b> { <i>ip-address</i>   <i>host-name</i> } [ <i>port</i> ] [ <b>service-type</b> <i>type</i> ] [ <b>version</b> <i>version-number</i> ]	Specifies the primary Cisco CallManager IP address or domain name, as well as the port gateway service type and version number.  The keywords and arguments are as follows: <ul style="list-style-type: none"> <li>• <i>ip-address</i>—Specifies the IP address of the Cisco CallManager.</li> <li>• <i>host-name</i>—Specifies the name of the Cisco CallManager server host in the format <i>host.name.ext</i>.</li> </ul>

Command	Purpose
	<ul style="list-style-type: none"> <li>• <i>port</i> (Optional)—Enables the MGCP protocol in a VoIP network and specifies the port number for Cisco CallManager to use. The valid values are from 1025 through 65,535. The default port number is 2427.</li> <li>• <b>service-type</b> <i>type</i> (Optional)—Specifies the type of gateway control service supported by the Cisco CallManager. The valid value is mgcp.</li> <li>• <b>version</b> <i>version-number</i> (Optional)— Specifies the version of the specified service type. The valid value is 0.1.</li> </ul>
<b>Step 3</b> Router(config)# <b>mgcp dtmf-relay voip codec {all   low-bit-rate} mode {cisco  nse   out-of-band}</b>	Selects the codec type and enables dual tone multifrequency (DTMF) relay services. The keywords are as follows: <ul style="list-style-type: none"> <li>• <b>voip</b>—Specifies VoIP calls.</li> <li>• <b>codec</b>—Specifies use of a codec device. <ul style="list-style-type: none"> <li>– <b>all</b>—Specifies use of all codecs.</li> <li>– <b>low-bit-rate</b>—Specifies any version of the G.729 low-bit-rate codec devices.</li> </ul> </li> <li>• <b>mode</b>—Specifies the mode for DTMF or NSE-based forwarding. <ul style="list-style-type: none"> <li>– <b>cisco</b>—Removes DTMF tone from the voice stream and sends FRF.11 with a special payload 121 for DTMF digits.</li> <li>– <b>nse</b>—Uses the NSE-based forwarding method.</li> <li>– <b>out-of-band</b>—Removes DTMF tone from the voice stream and does not send FRF.11.</li> </ul> </li> </ul>

## Verifying the Global MGCP Configuration

To display the global parameters configured on the router or gateway for MGCP, use the **show mgcp** command in privileged EXEC mode. The following is sample output from the command:

```
Router# show mgcp

GCP Admin State ACTIVE, Oper State ACTIVE - Cause Code NONE
MGCP call-agent: 10.0.0.21 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 lwm 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-packagec-package
MGCP VoAAL2 ignore-lco-codec DISABLED
```



### Note

For a description of each of the above output display fields, refer to the **show mgcp** command reference page.

## Configuring MGCP Control of the Cisco CallManager

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

	Command	Purpose
Step 1	Router(config)# <b>ccm-manager mgcp</b>	Enables MGCP support for controlling the Cisco CallManager.

Command	Purpose
<b>Step 2</b> Router(config)# <b>ccm-manager redundant-host</b> { <i>ip-address</i>   <i>DNS-name</i> } [ <i>ip-address</i>   <i>DNS-name</i> ]	(Optional) Identifies up to two backup Cisco CallManager server hosts.  The arguments are as follows: <ul style="list-style-type: none"> <li>• <i>ip-address</i>—Specifies the IP address of the host backup Cisco CallManager server.</li> <li>• <i>DNS-name</i>—Specifies the Domain Name System (DNS) name of the host backup Cisco CallManager server.</li> </ul>
<b>Step 3</b> Router(config)# <b>ccm-manager switchback</b> { <b>graceful</b>   <b>immediate</b>   <b>schedule-time</b> <i>hh:mm</i>   <b>uptime-delay</b> <i>minutes</i> }	(Optional) Configures the gateway switchback function and specifies how the gateway performs if the primary Cisco CallManager server becomes unavailable and later becomes available. The router ignores the return of a higher-level Cisco CallManager server.  The keywords are as follows: <ul style="list-style-type: none"> <li>• <b>graceful</b>—Completes all outstanding calls before returning the gateway to the control of the primary Cisco CallManager server.</li> <li>• <b>immediate</b>—Returns the gateway to the control of the primary Cisco CallManager server without delay, as soon as the network connection to the primary Cisco CallManager server is reestablished.</li> <li>• <b>schedule-time</b> <i>hh:mm</i>—Returns the gateway to the control of the primary Cisco CallManager server at the specified hour and minute, based on a 24-hour clock. If the specified time is earlier than the current time, the switchback to the primary Cisco CallManager server will occur at the specified time on the following day.</li> <li>• <b>uptime-delay</b> <i>minutes</i>—Returns the gateway to the control of the primary Cisco CallManager server when it has run for a specified number of minutes after a network connection to it has been reestablished. Permitted values for this argument range from 1 to 1440 (1 minute to 24 hours).</li> </ul>
<b>Step 4</b> Router(config)# <b>exit</b>	Exits global configuration mode and enters privileged EXEC mode.
<b>Step 5</b> Router# <b>ccm-manager switchover-to-backup</b>	Manually redirects the gateway to the backup Cisco CallManager server. The switchover to the backup Cisco CallManager server occurs immediately.  This command does not switch the gateway to the backup Cisco CallManager server if you have the switchback option set to <b>immediate</b> and the primary Cisco CallManager server is still running.

## Verifying the Configuration of MGCP Control of the Cisco CallManager

To verify the configuration of MGCP control of the Cisco CallManager, use the **show ccm-manager** command in privileged EXEC mode. The following is sample output for this command:

```
c3660A# show ccm-manager

MGCP Domain Name: c3660A.cisco.com
Total number of host: 2
Priority          Status          Host
=====
Primary          Registered      172.20.71.38
First backup     Backup Ready   172.20.71.44
Second backup    None

Current active Call Manager: 172.20.71.38
Backhaul/Redundant link port: 2428
Failover Interval: 30 seconds
Keepalive Interval: 15 seconds
Last keepalive sent: 00:10:33 (elapsed time: 02:39:29)
Last MGCP traffic time: 02:49:57 (elapsed time: 00:00:05)
Last failover time 02:49:56 from (172.20.71.38)
Switchback mode: Immediate
MGCP fallback mode: Not Selected
Last MGCP fallback start time: 00:00:00
Last MGCP fallback end time: 00:00:00

PRI Backhaul Link info:
Link Protocol: TCP
Remote Port Number: 2428
Remote IP Address: 172.20.71.38
Current Link State: OPEN
Statistics:
Packets recvd: 0
Recv failures: 0
Packets xmitted: 0
Xmit failures: 0
PRI Ports being backhauled:
Slot 1, port 1

Configuration Auto-Download Information
=====
No configurations downloaded
Current state: Automatic Configuration Download feature is disabled
Configuration Error History:
FAX mode: cisco
```



### Note

For a description of each of the above output display fields, refer to the **show ccm-manager** command reference page.

## Configuring MGCP T1 CAS

To configure T1 CAS for MGCP support, use the following commands beginning in global configuration mode:

	Command	Purpose
<b>Step 1</b>	Router(config)# <b>controller</b> {t1   e1} slot/port	Enters controller configuration mode.  <b>Note</b> MGCP E1 CAS is not supported.
<b>Step 2</b>	Router(config-controller)# <b>framing</b> {esf   sf   crc4   no-crc4   mp-crc4} [australia]	Specifies the framing type on a DS1 link for T1 and E1 PRI. The keywords are as follows: <ul style="list-style-type: none"> <li>• <b>esf</b>—Specifies extended Super Frame as the T1 frame type.</li> <li>• <b>sf</b>—Specifies Super Frame as the T1 frame type. This is the default.</li> <li>• <b>crc4</b>—Specifies CRC4 frame as the E1 frame type. This is the default for Australia.</li> <li>• <b>no-crc4</b>—Specifies no CRC4 frame as the E1 frame type.</li> <li>• <b>australia</b>—(Optional) Specifies the E1 frame type used in Australia.</li> </ul>
<b>Step 3</b>	Router(config-controller)# <b>linecode</b> {ami   b8zs   hdb3}	Specifies the line encoding method for a DS1 link. The keywords are as follows: <ul style="list-style-type: none"> <li>• <b>ami</b>—Specifies alternate mark inversion (AMI) as the line-code type. Valid for T1 or E1 controllers. This is the default for T1 lines.</li> <li>• <b>b8zs</b>—Specifies B8ZS as the line-code type. Valid for T1 controller only.</li> <li>• <b>hdb3</b>—Specifies high-density bipolar 3 (hdb3) as the line-code type. Valid for E1 controller only. This is the default for E1 lines.</li> </ul>

Command	Purpose
<b>Step 4</b> Router(config-controller)# <b>ds0-group</b> <i>ds0-group-no</i> <b>timeslots</b> <i>timeslot-list</i> <b>type</b> { <b>e&amp;m-delay-dial</b>   <b>e&amp;m-fgd</b>   <b>e&amp;m-immediate-start</b>   <b>e&amp;m-wink-start</b>   <b>fgd-eana</b>   <b>fxo-ground-start</b>   <b>fxo-loop-start</b>   <b>fxs-ground-start</b>   <b>fxs-loop-start</b> }	<p>Specifies the DS0 time slots that make up a logical voice port on a T1 controller and to specifies the signaling type.</p> <p><b>Note</b> Only E&amp;M wink-start and E&amp;M delay-dial are supported.</p> <p>The arguments and keywords are as follows:</p> <ul style="list-style-type: none"> <li>• <i>ds0-group-no</i>—Specifies a value from 0 to 23 that identifies the DS0 group.</li> <li>• <b>timeslots</b> <i>timeslot-list</i>—Specifies a single time-slot number, a single range of numbers, or multiple ranges of numbers separated by commas. For T1 signaling, allowable values are from 1 to 24. Examples are as follows:             <ul style="list-style-type: none"> <li>– 2</li> <li>– 1-15,17-24</li> <li>– 1-23</li> <li>– 2,4,6-12</li> </ul> </li> <li>• <b>type</b> <ul style="list-style-type: none"> <li>– <b>e&amp;m-delay-dial</b>—The originating endpoint sends an off-hook signal and then waits for an off-hook signal followed by an on-hook signal from the destination.</li> <li>– <b>e&amp;m-fgd</b>—E&amp;M Type II Feature Group D.</li> <li>– <b>e&amp;m-immediate-start</b>—E&amp;M immediate start.</li> <li>– <b>e&amp;m-wink-start</b>—E&amp;M Mercury Exchange Limited Channel-Associated Signaling (MELCAS) wink-start signaling support.</li> <li>– <b>fgd-eana</b>—Feature Group D exchange access in North America.</li> <li>– <b>fxo-ground-start</b>—FXO ground-start signaling support.</li> <li>– <b>fxo-loop-start</b>—FXO loop-start signaling support.</li> <li>– <b>fxs-ground-start</b>—FXS ground-start signaling support.</li> <li>– <b>fxs-loop-start</b>—FXS loop-start signaling support.</li> </ul> </li> </ul>
<b>Step 5</b> Router(config-controller)# <b>exit</b>	Returns to global configuration mode.

	Command	Purpose
Step 6	Router(config)# <b>dial-peer voice tag</b> {pots   vofr   voip}	Enters dial-peer configuration mode and specifies a dial peer.  <b>Note</b> The dial-peer <b>vofr</b> keyword is not supported for MGCP.
Step 7	Router(config-dial-peer)# <b>application application-name</b> [out-bound]	Enables a specific application (in this case MGCP) on a dial peer.  The argument and keyword are as follows: <ul style="list-style-type: none"> <li>• <i>application-name</i>—Indicates the name of the predefined application to enable on the dial peer. For MGCP the application name is mgcpapp.</li> <li>• <b>out-bound</b>—(Optional) Named application handles the MMOIP dial peer in the outgoing mode for store-and-forward fax.</li> </ul>

## Configuring ISDN Signaling Backhaul

To configure ISDN to backhaul Q.931 signaling, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# <b>controller</b> {t1   e1} slot/port	Enters controller configuration mode.
Step 2	Router(config-controller)# <b>clock source</b> [internal   line]	Configures the clock source used by the T1 or E1 controller.  The keywords are as follows: <ul style="list-style-type: none"> <li>• <b>internal</b> (Optional)—Specifies that the clocking source is obtained from the port adapter line.</li> <li>• <b>line</b> (Optional)—Specifies that the clocking source is obtained from the network.</li> </ul>
Step 3	Router(config-controller)# <b>cablelength long</b> {0db   -7.5db   -15db   -22.5db}	Specifies the cable length longer than 600 feet for a T1 link. The cable length must conform to the actual cable length that you are using. For example, if you attempt to enter the <b>cablelength short</b> command on a long-haul T1 link, the command is rejected.  The keywords are as follows: <ul style="list-style-type: none"> <li>• <b>0db</b>—Specifies the decibel pulse level at 0 dB. This is the default pulse rate.</li> <li>• <b>-7.5db</b>—Specifies the decibel pulse level at -7.5 dB.</li> <li>• <b>-15db</b>—Specifies the decibel pulse level at -15 dB.</li> <li>• <b>-22.5db</b>—Specifies the decibel pulse level at -22.5 dB.</li> </ul>

Command	Purpose
or Router(config-controller)# <b>cablelength short</b> {110ft   220ft   330ft   440ft   550ft   600ft}	Specifies a cable length 600 feet or less for a T1 link. The keywords are as follows. <ul style="list-style-type: none"> <li>• <b>110ft</b>—Specifies a cable length from 0 to 110 feet.</li> <li>• <b>220ft</b>—Specifies a cable length from 111 to 220 feet.</li> <li>• <b>330ft</b>—Specifies a cable length from 221 to 330 feet.</li> <li>• <b>440ft</b>—Specifies a cable length from 331 to 440 feet.</li> <li>• <b>550ft</b>—Specifies a cable length from 441 to 550 feet.</li> <li>• <b>600ft</b>—Specifies a cable length from 551 to 600 feet.</li> </ul> If you do not set the cable length, the system defaults to a setting of cablelength long 0 dB.
<b>Step 4</b> Router(config-controller)# <b>pri-group timeslots</b> <i>list-of-timeslots service mgcp</i>	Specifies MGCP as the control protocol used for backhaul. The controller time slots cannot be shared between backhaul and other Layer 3 protocols.
<b>Step 5</b> Router(config-controller)# <b>framing {esf   sf  </b> <b>crc4   no-crc4   mp-crc4}</b> [australia]	Specifies the framing type on a DS1 link for T1 and E1 PRI. The keywords are as follows: <ul style="list-style-type: none"> <li>• <b>esf</b>—Specifies extended Super Frame as the T1 frame type.</li> <li>• <b>sf</b>—Specifies Super Frame as the T1 frame type. This is the default.</li> <li>• <b>crc4</b>—Specifies CRC4 frame as the E1 frame type. This is the default for Australia.</li> <li>• <b>no-crc4</b>—Specifies no CRC4 frame as the E1 frame type.</li> <li>• <b>australia</b>—(Optional) Specifies the E1 frame type used in Australia.</li> </ul>
<b>Step 6</b> Router(config-controller)# <b>linecode {ami  </b> <b>b8zs   hdb3}</b>	Specifies the line encoding method for a DS1 link. The keywords are as follows: <ul style="list-style-type: none"> <li>• <b>ami</b>—Specifies alternate mark inversion (AMI) as the line-code type. Valid for T1 or E1 controllers. This is the default for T1 lines.</li> <li>• <b>b8zs</b>—Specifies B8ZS as the line-code type. Valid for T1 controller only.</li> <li>• <b>hdb3</b>—Specifies high-density bipolar 3 (hdb3) as the line-code type. Valid for E1 controller only. This is the default for E1 lines.</li> </ul>
<b>Step 7</b> Router(config-controller)# <b>exit</b>	Returns to global configuration mode.

Command	Purpose
<b>Step 8</b> Router(config)# <b>interface serial</b> <i>slot/port:timeslot number</i>	Enters serial interface configuration mode.  The arguments and keywords are as follows: <ul style="list-style-type: none"> <li>• <i>slot/port</i>—Specifies the slot number and port number where the channelized E1 or T1 controller is located.</li> <li>• <i>timeslot</i>—Specifies, for ISDN, the D channel time slot, which is the 23 channel for T1 and the 15 channel for E1. PRI time slots are range from 0 to 23 for channelized T1 and range from 0 to 30 for channelized E1. On a dual port card, it is possible to run channelized on one port and primary rate on the other port.</li> </ul> <b>Note</b> The colon (:) is required. <ul style="list-style-type: none"> <li>• <i>number</i>—Specifies the channelized E1 or T1 controller number.</li> </ul>
<b>Step 9</b> Router(config-if)# <b>isdn switch-type</b> [primary-4ess   primary-5ess   primary-dms100   primary-ni   primary-net5   primary-ntt   primary-ts014]	Configures the ISDN switch type. This configuration can be done in global configuration mode or interface configuration mode.  The keywords are as follows: <ul style="list-style-type: none"> <li>• <b>primary-4ess</b> (Optional)—Specifies electronic switching system (ESS) 4.</li> <li>• <b>primary-5ess</b> (Optional)—Specifies ESS 5 that works with T1.</li> <li>• <b>primary-dms-100</b> (Optional)—Specifies the DMS 100 switch that works with T1 and E1 PRI.</li> <li>• <b>primary-ni</b> (Optional)—Specifies an NI switch that works with T1.</li> <li>• <b>primary-net5</b> (Optional)—Specifies a Net5 switch that works with E1.</li> <li>• <b>primary-ntt</b> (Optional)—Specifies the Japanese T1 and E1 PRI switch.</li> <li>• <b>primary-ts014</b> (Optional)—Specifies the Australian T1 and E1 PRI switch.</li> </ul>
<b>Step 10</b> Router(config-if)# <b>isdn bind-L3 ccm-manager</b>	Configures ISDN to backhaul Q.931. Repeat Step 1 through Step 8 for each T1 or E1 interface that will use backhaul.
<b>Step 11</b> Router(config-if)# <b>isdn protocol-emulate</b> {user   network}	Specifies the ISDN protocol emulation. The default is the user-side ISDN protocol. The keywords are as follows: <ul style="list-style-type: none"> <li>• <b>user</b>—Specifies Layer 2 and Layer 3 port protocol operation as TE (port functions as QSIG slave).</li> <li>• <b>network</b>—Specifies Layer 2 and Layer 3 port protocol operation as NT (port functions as QSIG master).</li> </ul>

## Verifying the Backhaul Configurations

To verify the backhaul configurations, use the **show isdn status** command. In this example, the Layer 2 protocol is Q.921, and the Layer 3 protocol is CCM-MANAGER. This output verifies that the Layer 2 and Layer 3 protocols are configured to backhaul ISDN. If you are connected to a live line, you should see Layer 1 status as active and Layer 2 as MULTIPLE\_FRAME\_ESTABLISHED.

```
Router# show isdn status
*00:03:34.423 UTC Sat Jan 1 2000
Global ISDN Switchtype = primary-net5
ISDN Serial1:23 interface
 dsl 0, interface ISDN Switchtype = primary-net5
L2 Protocol = Q.921 L3 Protocol(s) = CCM-MANAGER
Layer 1 Status:
ACTIVE
Layer 2 Status:
TEI = 0, Ces = 1, SAPI = 0, State = MULTIPLE_FRAME_ESTABLISHED
Layer 3 Status:
NLCB:callid=0x0, callref=0x0, state=31, ces=0 event=0x0
NLCB:callid=0x0, callref=0x0, state=0, ces=1 event=0x0
0 Active Layer 3 Call(s)
Activated dsl 0 CCBS = 0
Number of active calls = 0
Number of available B-channels = 23
Total Allocated ISDN CCBS = 0
```

The following is sample output from the **show ccm-manager backhaul** command. The output shows the PRI backhaul link information.

```
Router# show ccm-manager backhaul

PRI Backhaul Link info:
Link Protocol:          TCP
Remote Port Number:    2428
Remote IP Address:     172.20.71.38
Current Link State:    OPEN
Statistics:
  Packets recvd:        0
  Recv failures:        0
  Packets xmitted:      21
  Xmit failures:        0
PRI Ports being backhauled:
Slot 1, port 1
```



### Note

For a description of each of the above output display fields, refer to the appropriate command reference page.

## Enabling MGCP Gateway Fallback Support

The MGCP gateway fallback feature operates as an H.323 gateway. The router takes over the functions of the Cisco CallManager if the link to the Cisco CallManager is lost, such as during an outage in a network segment that contains the Cisco CallManager server.

To enable gateway fallback support for an MGCP gateway, use the following command in global configuration mode:

Command	Purpose
Router(config)# <b>ccm-manager fallback-mgcp</b>	Enables fallback support for an MGCP voice gateway in a Cisco VoIP network.

### Verifying the Status of the MGCP Gateway Fallback Feature

To verify MGCP gateway fallback status, use the **show ccm-manager fallback-mgcp** command in privileged EXEC mode. The following is sample output that shows MGCP fallback status and, if enabled, whether the feature is active or not (ON or OFF):

```
Router# show ccm-manager fallback-mgcp

Current active Call Manager:    172.20.71.29
MGCP Fallback mode:            Enabled/OFF
Last MGCP Fallback start time:  00:00:00
Last MGCP Fallback end time:    00:00:00
```

To display general configuration information about the Cisco CallManager, use the **show ccm-manager** command on the Cisco VG200, Cisco 2600 series routers, and Cisco 3600 series routers. The following is sample output that shows the Cisco CallManager configuration:

```
Router# show ccm-manager

MGCP Domain Name: Cisco
Priority          Status                Host
=====
Primary          Registered             172.20.71.29
First backup     None
Second backup    None

Current active Call Manager:    172.20.71.29
Backhaul/Redundant link port:   2428
Failover Interval:              30 seconds
Keepalive Interval:            15 seconds
Last keepalive sent:            02:51:20 (elapsed time: 00:00:25)
Last MGCP traffic time:         02:51:20 (elapsed time: 00:00:25)
Last switchover time:           None
Switchback mode:                Immediate
MGCP Fallback mode:            Enabled/OFF
Last MGCP Fallback start time:  00:00:00
Last MGCP Fallback end time:    00:00:00
```



#### Note

For a description of each of the above output display fields, refer to the appropriate command reference page.

## Enabling Multicast Music-on-Hold

Cisco CallManager Version 3.1 supports the capability to place callers on hold with music being supplied from a streaming multicast music-on-hold (MOH) server. This section describes how to enable the MOH feature on Cisco MGCP voice gateways so that they can support the multicast streaming of music to callers while on hold.

To configure multicast MOH on a Cisco MGCP voice gateway, use the following command in global configuration mode:

Command	Purpose
Router(config)# <b>ccm-manager music-on-hold</b>	Enables music-on-hold.

## Verifying MOH Status

To verify the MOH status and display information about the current configuration, use the **show ccm-manager music-on-hold** command in privileged EXEC mode. The following is sample output from this command:

```
Router# show ccm-manager music-on-hold
Multicast      RTP      Packets      Call      ID      Incoming
Address        Port     In/Out       ID        Protocol Interface
209.165.200.224 16256   3000/3000    1         IGMP    fe0/0
```



### Note

For a description of each of the above output display fields, refer to the **show ccm-manager music-on-hold** command reference page.

## Monitoring and Maintaining Cisco CallManager and MGCP

To monitor and maintain Cisco CallManager and MGCP support, use the following commands in EXEC mode as needed:

Command	Purpose
Router# <b>debug ccm-manager {backhaul {events   packets}   config-download {all   errors   events   packets   xml}   errors   events   music-on-hold {errors   events   packets}   packets}</b>	<p>Displays the CCM error information.</p> <p>The keywords are as follows:</p> <ul style="list-style-type: none"> <li>• <b>backhaul</b>—Displays CCM backhaul events and packets. <ul style="list-style-type: none"> <li>– <b>events</b>—Displays CCM backhaul events.</li> <li>– <b>packets</b>—Displays CCM backhaul packets.</li> </ul> </li> </ul>

Command	Purpose
	<ul style="list-style-type: none"> <li>• <b>config-download</b>—Displays debug information for CCM configuration download errors, events, packets, XML parser, or all of these. <ul style="list-style-type: none"> <li>- <b>all</b>—Displays all errors.</li> <li>- <b>errors</b>—Displays download errors.</li> <li>- <b>events</b>—Displays errors related to the download events.</li> <li>- <b>packets</b>—Displays errors related to the download packets.</li> <li>- <b>xml</b>—Displays XML errors.</li> </ul> </li> <li>• <b>errors</b>—Displays CCM errors.</li> </ul>
	<ul style="list-style-type: none"> <li>• <b>events</b>—Displays CCM events.</li> <li>• <b>music-on-hold</b>—Displays CCM MOH errors, events, and packets. <ul style="list-style-type: none"> <li>- <b>errors</b>—Displays MOH errors.</li> <li>- <b>events</b>—Displays MOH events.</li> <li>- <b>packets</b>—Displays MOH packets.</li> </ul> </li> <li>• <b>packets</b>—Displays CCM packets.</li> </ul>
<pre>Router# show ccm-manager [backhaul   config-download   fallback-mgcp   hosts   music-on-hold   redundancy]</pre>	<p>Displays a list of Cisco CallManager servers and presents information related to their status and availability.</p> <p>The optional keywords are as follows:</p> <ul style="list-style-type: none"> <li>• <b>backhaul</b>—Displays CallManager backhaul information.</li> <li>• <b>config-download</b>—Displays automated configuration download information.</li> <li>• <b>fallback-mgcp</b>—Displays MGCP CallManager fallback information and status.</li> <li>• <b>hosts</b>—Displays the host name of each configured Cisco CallManager server, its operational status, and the address of the host.</li> <li>• <b>music-on-hold</b>—Displays information about the current MOH configuration and status.</li> <li>• <b>redundancy</b>—Displays information related to the failover mode and presents status information, including the redundant link port, the last failover interval, the last keepalive interval for the Cisco CallManager server, the MGCP traffic time, and the switchback mode.</li> </ul>

Command	Purpose
Router# <b>debug mgcp</b> [ <b>all</b>   <b>errors</b>   <b>events</b>   <b>packets</b>   <b>parser</b> ]	<p>Turns on debugging for the gateway.</p> <p>The keywords are as follows:</p> <ul style="list-style-type: none"> <li>• <b>all</b>—(Optional) Debugs errors, events, packets, and the parser for MGCP modules.</li> <li>• <b>errors</b>—(Optional) Debugs errors for MGCP modules.</li> <li>• <b>events</b>—(Optional) Debugs events for MGCP modules.</li> <li>• <b>packets</b>—(Optional) Debugs packets for MGCP modules.</li> <li>• <b>parser</b>—(Optional) Debugs the parser for MGCP modules.</li> </ul>

## Configuration Examples

This section provides the following configuration examples:

- [MGCP Control of Dial Peers and Voice Ports Configuration Example, page 30](#)
- [MGCP Control and Cisco CallManager Configuration Examples, page 31](#)
- [T1 CAS and PRI Backhaul Configuration Example, page 31](#)
- [MGCP Gateway Fallback for H.323 T1 CAS Support Configuration Examples, page 34](#)
- [MGCP Fallback to H.323 T1 CAS Configuration Example, page 35](#)
- [MGCP Fallback to H.323 for ISDN T1 PRI Configuration Example, page 37](#)
- [MGCP Fallback to H.323 for ISDN E1 PRI Configuration Example, page 38](#)
- [Single-Point Configuration Example, page 39](#)
- [Multicast Music-on-Hold Configuration Example, page 41](#)
- [Running Configuration Example, page 42](#)

## MGCP Control of Dial Peers and Voice Ports Configuration Example

The following example shows that voice port 0 on voice interface card 1 has been configured to run under MGCP control. This configuration includes two FXO ports and one FXS port that are configured to run under MGCP control. Slot and port numbering begin at 0.

```
dial-peer voice 1 pots
  application mgcpapp
  port 1/0/0
! FXO port
dial-peer voice 2 pots
  application mgcpapp
  port 1/0/1
! FXO port
dial-peer voice 3 pots
  application mgcpapp
  port 1/1/0
```

```

! FXS port
dial-peer voice 4 pots
  application mgcpapp
  port 1/1/1

```

## MGCP Control and Cisco CallManager Configuration Examples

The following example shows the configuration of a router to run MGCP control using the primary Cisco CallManager server:

```

mgcp
! Configures the router to run MGCP.
mgcp call-agent 10.0.0.201 service-type mgcp version 0.1
! Defines the primary Cisco CallManager server.
mgcp dtmf-relay voip codec all mode out-of-band
! Enables VoIP calls without DTMF.

```

The following example shows the configuration of the Cisco CallManager redundancy functions:

```

ccm-manager switchback graceful
! After the last call ends, use the primary Cisco CallManager.
ccm-manager redundant-host 10.0.0.21
! Defines the redundant Cisco CallManager server (first backup).
ccm-manager mgcp
! Enables support for the Cisco CallManager within MGCP.

```

## T1 CAS and PRI Backhaul Configuration Example

The following example shows T1 CAS and PRI backhaul configured for an MGCP voice gateway:

```

Current configuration : 3533 bytes
!
version 12.2
no parser cache
no service single-slot-reload-enable
service timestamps debug datetime localtime
service timestamps log datetime localtime
no service password-encryption
!
hostname uut5-c3660
!
logging rate-limit console 10 except errors
!
username all
!
voice-card 3
!
voice-card 5
!
ip subnet-zero
!
ip domain-name tertis.com
!
no ip dhcp-client network-discovery
mgcp
mgcp call-agent 10.0.0.21 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp modem passthrough voip mode cisco
mgcp modem passthrough voip codec g711alaw
mgcp modem passthrough voip redundancy

```

```

mgcp package-capability dtmf-package
mgcp package-capability mf-package
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp default-package line-package
mgcp timer net-cont-test 3000
isdn switch-type primary-ni
call rsvp-sync
!
ccm-manager fallback-mgcp
ccm-manager redundant-host 10.0.0.21
ccm-manager mgcp
ccm-manager music-on-hold
fax interface-type vfc
mta receive maximum-recipients 0
!
controller T1 3/0
  framing esf
  clock source internal
  linecode b8zs
  ds0-group 0 timeslots 1 type e&m-wink-start
  ds0-group 1 timeslots 2 type e&m-wink-start
  ds0-group 2 timeslots 3 type e&m-wink-start
  ds0-group 3 timeslots 4 type e&m-wink-start
  ds0-group 4 timeslots 5 type e&m-wink-start
  ds0-group 5 timeslots 6 type e&m-wink-start
!
controller T1 3/1
  framing esf
  clock source internal
  linecode b8zs
  ds0-group 0 timeslots 1-12 type e&m-wink-start
!
controller E1 5/0
  pri-group timeslots 1-31 service mgcp
!
interface FastEthernet0/0
  ip address 10.0.0.21 255.255.255.224
  speed 10
  half-duplex
!
interface Ethernet2/0
  ip address 10.0.0.2 255.255.255.224
  half-duplex
!
interface Serial2/0
  ip address 10.0.0.1 255.255.255.224
  encapsulation frame-relay
  shutdown
  no fair-queue
  frame-relay interface-dlci 1005
!
interface Ethernet2/1
  no ip address
  shutdown
  half-duplex
!
interface Serial15/0:15
  no ip address
  no logging event link-status
  isdn switch-type primary-ts014
  isdn incoming-voice voice
  isdn T306 60000
  isdn bind-L3 ccm-manager

```

```
no cdp enable
!
interface Ethernet6/0
no ip address
shutdown
half-duplex
!
ip classless
ip route 10.0.0.4 255.255.255.224 Ethernet2/0
ip http server
!
snmp-server manager
!
voice-port 1/0/0
!
voice-port 1/0/1
!
voice-port 3/0:0
!
voice-port 3/0:1
!
voice-port 3/0:2
!
voice-port 3/0:3
!
voice-port 3/0:4
!
voice-port 3/0:5
!
voice-port 3/1:0
!
voice-port 4/1/0
!
voice-port 4/1/1
!
voice-port 5/0:15
!
dial-peer cor custom
!
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 20 pots
application mgcpapp
port 4/1/0
!
dial-peer voice 30 pots
application mgcpapp
port 4/1/1
!
dial-peer voice 500 pots
application mgcpapp
port 3/0:0
!
dial-peer voice 200 pots
application mgcpapp
port 5/0:15
!
dial-peer voice 501 pots
```

```

application mgcpapp
port 3/0:1
!
dial-peer voice 502 pots
application mgcpapp
port 3/0:2
!
dial-peer voice 503 pots
application mgcpapp
port 3/0:3
!
dial-peer voice 504 pots
application mgcpapp
port 3/0:4
!
dial-peer voice 505 pots
application mgcpapp
port 3/0:5
!
line con 0
exec-timeout 0 0
line aux 0
exec-timeout 0 0
line vty 0 4
exec-timeout 0 0
login
end

```

## MGCP Gateway Fallback for H.323 T1 CAS Support Configuration Examples

The following example shows a specific H.323 configuration on MGCP voice gateways for VoIP calls, when using the fallback feature:

```

dial-peer voice 555 voip
application mgcpapp
destination pattern 555...
incoming-called-number 444...
session-target ipv4:172.20.21.8
codec g711ulaw

```



### Note

When configuring MGCP gateway fallback support, the pots dial-peer statement includes the **application** command with the **mgcpapp** keyword and must specify the voice port. In order for the default session application to take over during fallback, you must also configure a destination pattern.

The following example shows a specific H.323 configuration on MGCP voice gateways for T1 CAS with e&m-wink-start emulation type using the fallback feature:

```

ccm-manager switchback immediate
ccm-manager fallback-mgcp
ccm-manager mgcp
controller T1 1/0
framing esf
linecode b8zs
ds0-group 1 timeslots 1-24 type e&m-wink-start
!
voice-port 1/0:1
!
dial-peer voice 1 pots
application mgcpapp
destination-pattern 91.....

```

```
port 1/0:1
```

The following example shows a specific H.323 configuration on MGCP voice gateways for FXS ports, using the fallback feature:

```
dial-peer voice 1 pots
 application mgcpapp
 destination-pattern 555-1212
 port 1/0/0
```

The following example shows a specific H.323 configuration on MGCP voice gateways for FXO ports, using the fallback feature:

```
dial-peer voice 2 pots
 application mgcpapp
 destination-pattern 527....
 prefix 527
 port 1/1/1
```

## MGCP Fallback to H.323 T1 CAS Configuration Example

The following example shows a specific configuration for MGCP fallback to H.323 T1 CAS on a voice gateway:

```
Current configuration : 2181 bytes
!
version 12.2
no service single-slot-reload-enable
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname Tertis-3640a
!
logging rate-limit console 10 except errors
!
memory-size iomem 25
voice-card 3
!
ip subnet-zero
!
no ip domain-lookup
ip domain-name tertis.com
!
no ip dhcp-client network-discovery
frame-relay switching
mgcp
mgcp call-agent 10.0.0.21 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000
mgcp package-capability rtp-package
no mgcp timer receive-rtcp
call rsvp-sync
!
ccm-manager switchback immediate
ccm-manager fallback-mgcp
ccm-manager redundant-host 10.0.0.21
ccm-manager mgcp
!
controller T1 3/0
 framing esf
 linecode b8zs
```

```

ds0-group 1 timeslots 1 type e&m-wink-start
!
controller T1 3/1
  framing sf
  linecode ami
!
interface FastEthernet0/0
  ip address 10.0.0.21 255.255.255.224
  duplex auto
  speed auto
!
interface Serial0/0
  ip address 10.0.0.21 255.255.255.224
  encapsulation frame-relay
  no keepalive
  frame-relay interface-dlci 300
!
interface Serial0/1
  no ip address
  shutdown
  clockrate 2000000
!
interface Ethernet2/0
  ip address 10.0.0.21 255.255.255.224
  half-duplex
!
interface TokenRing2/0
  no ip address
  shutdown
  ring-speed 16
!
ip classless
ip route 10.0.0.21 255.255.255.0 14.0.0.1
ip route 10.0.0.21 255.255.255.255 Ethernet2/0
ip route 10.0.0.21 255.255.255.255 Ethernet2/0
no ip http server
!
snmp-server manager
!
voice-port 1/0/0
!
voice-port 1/0/1
!
voice-port 1/1/0
!
voice-port 1/1/1
!
voice-port 3/0:1
!
dial-peer cor custom
!
dial-peer voice 44 pots
  application mgcpapp
  destination-pattern 4301
  port 1/1/0
!
dial-peer voice 55 pots
  application mgcpapp
  destination-pattern 4302
  port 1/1/1
!

```

```

dial-peer voice 85 voip
 destination-pattern 805....
 session target ipv4:10.0.0.21
 codec g711ulaw
!
dial-peer voice 33 pots
 application mgcpapp
 destination-pattern 807....
 port 3/0:1
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 4
 login
end

```

## MGCP Fallback to H.323 for ISDN T1 PRI Configuration Example

The following example shows a specific H.323 configuration on MGCP voice gateways for T1 PRI ports, using the fallback feature:

```

fallback-mgcp
 ccm-manager redundant-host CM-B
 ccm-manager mgcp
 ccm-manager music-on-hold
 ccm-manager config server cm-a
!
controller T1 1/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24 service mgcp
!
controller T1 1/1
 framing esf
 linecode b8zs
 pri-group timeslots 1-24 service mgcp
!
interface Serial1/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-ni
 isdn incoming-voice voice
 isdn T306 30000
 isdn bind-l3 ccm-manager
 no cdp enable
!
interface Serial1/1:23
 no ip address
 no logging event link-status
 isdn switch-type primary-ni
 isdn incoming-voice voice
 isdn T306 30000
 isdn bind-l3 ccm-manager
 no cdp enable
!
dial-peer voice 9991023 pots
 application mgcpapp
 direct-inward-dial
 port 1/0:23
!
dial-peer voice 9991123 pots

```

```

application mgcpapp
direct-inward-dial
port 1/1:23
!
dial-peer voice 1640001 pots
destination-pattern 16.....
direct-inward-dial
port 1/0:23
!
dial-peer voice 1740001 pots
destination-pattern 17.....
direct-inward-dial
port 1/1:23
!

```

**Note**


---

If the **ccm-manager config** command is enabled and you separate the MGCP and the H.323 dial peers under different tags, make sure that the MGCP dial peers are configured before the H.323 dial peers.

---

## MGCP Fallback to H.323 for ISDN E1 PRI Configuration Example

The following example shows a specific H.323 configuration on MGCP voice gateways for E1 PRI ports, using the fallback feature:

```

!
ccm-manager fallback-mgcp
ccm-manager redundant-host CM-B
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server cm-a
!
controller E1 1/0
pri-group timeslots 1-31
!
controller E1 1/1
pri-group timeslots 1-31
!
interface Serial1/0:15
no ip address
no logging event link-status
isdn switch-type primary-net5
isdn incoming-voice voice
isdn T310 30000
no cdp enable
!
interface Serial1/1:15
no ip address
no logging event link-status
isdn switch-type primary-net5
isdn incoming-voice voice
isdn T310 30000
no cdp enable
!
dial-peer voice 9991015 pots
application mgcpapp
direct-inward-dial
port 1/0:15
!
dial-peer voice 9991115 pots
application mgcpapp

```

```

direct-inward-dial
port 1/1:15
!
dial-peer voice 1840001 pots
description vg200-e1 port 0
destination-pattern 18.....
direct-inward-dial
port 1/0:15
!
dial-peer voice 1940001 pots
description vg200-e1 port 1
destination-pattern 19.....
direct-inward-dial
port 1/1:15
!

```

**Note**

If the **ccm-manager config** command is enabled and you separate the MGCP and the H.323 dial peers under different tags, make sure that the MGCP dial peers are configured before the H.323 dial peers. DID is required for E1 PRI dial peers.

## Single-Point Configuration Example

The following example shows a single-point configuration for an MGCP voice gateway:

```

version 12.2
no parser cache
no service single-slot-reload-enable
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname vs-3640
!
logging rate-limit console 10 except errors
!
memory-size iomem 10
voice-card 1
!
ip subnet-zero
!
ip domain-name tertis.com
!
no ip dhcp-client network-discovery
mgcp
mgcp call-agent 10.10.1.10 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000
mgcp modem passthrough voip mode cisco
mgcp package-capability rtp-package
mgcp package-capability sst-package
isdn switch-type primary-ni
call rsvp-sync
!
ccm-manager config
ccm-manager switchback immediate
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server 10.10.1.10
!
controller T1 1/0

```

```

framing esf
linecode b8zs
ds0-group 1 timeslots 1-12 type e&m-wink-start
!
controller T1 1/1
framing esf
linecode b8zs
ds0-group 1 timeslots 1-12 type e&m-wink-start
!
interface Ethernet0/0
ip address 10.0.0.21 255.255.255.0
half-duplex
!
interface Ethernet0/1
ip address 10.0.0.21 255.0.0.0
half-duplex
!
ip classless
ip route 10.0.0.21 255.255.255.255 Ethernet0/1
ip http server
!
snmp-server manager
!
voice-port 1/0:1
!
voice-port 1/1:1
!
voice-port 2/0/0
timing hookflash-out 50
!
voice-port 2/0/1
timing hookflash-out 50
!
voice-port 2/1/0
timing hookflash-out 50
signal groundStart
!
voice-port 2/1/1
timing hookflash-out 50
signal groundStart
!
dial-peer cor custom
!
dial-peer voice 999200 pots
application mgcpapp
port 2/0/0
!
dial-peer voice 999201 pots
application mgcpapp
port 2/0/1
!
dial-peer voice 999210 pots
application mgcpapp
port 2/1/0
!
dial-peer voice 999211 pots
application mgcpapp
port 2/1/1
!
dial-peer voice 999101 pots
application mgcpapp
port 1/0:1
!
dial-peer voice 999111 pots

```

```

    application mgcpapp
    port 1/1:1
    !
    line con 0
    line aux 0
    line vty 0 4
        login
    !
end

```

## Multicast Music-on-Hold Configuration Example

The following example shows multicast MOH configured for an MGCP voice gateway:

```

version 12.2
no parser cache
no service single-slot-reload-enable
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname vs-3640
!
logging rate-limit console 10 except errors
!
memory-size iomem 10
voice-card 1
!
ip subnet-zero
!
ip domain-name tertis.com
!
no ip dhcp-client network-discovery
mgcp
mgcp call-agent 10.0.0.21 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000
mgcp modem passthrough voip mode cisco
mgcp package-capability rtp-package
mgcp package-capability sst-package
no mgcp timer receive-rtcp
call rsvp-sync
!
ccm-manager redundant-host 10.0.0.21
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server 10.0.0.21
!
controller T1 2/0
    framing sf
    linecode ami
    ds0-group 0 timeslots 1 type e&m-wink-start
!
controller T1 2/1
    framing sf
    linecode ami
!
interface FastEthernet0/0
    ip address 10.0.0.21 255.255.255.0
    no ip mroute-cache
    duplex auto
    speed auto

```

```

no cdp enable
voice-port 1/0/0
!
voice-port 1/0/1
!
voice-port 2/0:0
!
dial-peer cor custom
!
dial-peer voice 125 pots
  application mgcpapp
  port 1/0/0
!
dial-peer voice 150 pots
  application mgcpapp
  port 2/0:0
!
line con 0
  exec-timeout 0 0
line aux 0
line vty 0 4
  login
!
no scheduler max-task-time
scheduler allocate 4000 4000
!
end

```

## Running Configuration Example

To display the running configuration, including any configuration changes recently made, use the **show running-config** command. The following sample output shows the current configuration:

```

Current configuration : 1244 bytes
!
version 12.2
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 10.0.0.200 255.255.0.0
 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 000000090200000196983000
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 new and modified commands. All other commands used with this feature are documented in the Cisco IOS Release 12.2 command reference publications.

## New Commands

- [ccm-manager config](#)
- [ccm-manager fallback-mgcp](#)
- [ccm-manager music-on-hold](#)
- [ccm-manager switchover-to-backup](#)
- [isdn bind-l3 ccm-manager](#)

## Modified Commands

- [application](#)
- [cablelength long](#)
- [cablelength short](#)
- [ccm-manager mgcp](#)
- [ccm-manager redundant-host](#)
- [ccm-manager switchback](#)
- [clock source](#)
- [controller](#)
- [debug ccm-manager](#)
- [debug mgcp](#)
- [dial-peer voice](#)
- [ds0-group](#)
- [framing \(E1/T1 controller\)](#)
- [hostname](#)
- [interface serial](#)
- [isdn protocol-emulate](#)
- [isdn switch-type \(PRI\)](#)
- [linecode](#)
- [mgcp call-agent](#)
- [mgcp dtmf-relay](#)
- [pri-group](#)
- [show ccm-manager](#)
- [show dial-peer voice](#)
- [show isdn status](#)
- [show mgcp](#)
- [show voice port](#)
- [voice-port](#)

# application

To enable a specific application on a dial peer, use the **application** command in dial peer configuration mode. To remove the application from the dial peer, use the **no** form of this command.

**application** *application-name* [**out-bound**]

**no application** *application-name* [**out-bound**]

## Syntax Description

<i>application-name</i>	Indicates the name of the predefined application you wish to enable on the dial peer. For H.323 networks, the application is defined by a Tool Command Language/interactive voice response (TCL/IVR) filename and location. Incoming calls using plain old telephone service (POTS) dial peers and outgoing calls using Multimedia Mail over IP (MMoIP) dial peers are handed off to this application. For Media Gateway Control Protocol (MGCP) or Simple Gateway Control Protocol (SGCP) networks, see the usage guidelines below for valid application names.
<b>out-bound</b>	(Optional) Named application handles the MMOIP dial peer in the outgoing mode for store-and-forward fax.

## Defaults

No default behavior or values.

## Command Modes

Dial peer configuration

## Command History

Release	Modification
11.3(6)NA2	This command was introduced on the Cisco 2500 series routers, 3600 series routers, and Cisco AS5300 universal access server.
12.0(5)T	The SGCPAPP application was supported initially on the Cisco AS5300 universal access server in a private release that was not generally available.
12.0(7)XK	Support for the SGCPAPP application was extended to the Cisco MC3810 multiservice concentrator and the Cisco 3600 series routers (except for the Cisco 3620) in a private release that was not generally available.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.1(3)T	The MGCPAPP application was supported initially on the Cisco AS5300 universal access server.
12.1(3)XI	The <b>out-bound</b> keyword was added for the store-and-forward fax feature on the Cisco AS5300 universal access server.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

**Usage Guidelines**

Use this command to associate a predefined session application with an incoming POTS dial peer or an outgoing MMoIP dial peer. Calls using this incoming POTS dial peer or this outgoing MMoIP dial peer will be handed to the predefined specified session application.

The SGCP application and MGCP application can be applied only to POTS dial peers. Neither application uses dial peer hunting.



**Note** In Cisco IOS Release 12.2, you cannot mix SGCP and non-SGCP endpoints in the same T1 controller. You also cannot mix SGCP and non-SGCP endpoints in the same DS0 group.

**Examples**

The following example shows how to define an application and how to apply it to an outbound MMoIP dial peer for the fax onramp operation:

```
call application voice fax_on_vfc_onramp http://santa/username/clid_4digits_npw_3.tcl
dial-peer voice 3 mmoip
  application fax_on_vfc_onramp out-bound
  destination-pattern 57108..
  session target mailto:$d$@mail-server.cisco.com
```

The following example shows how to apply the MGCP application to a dial peer:

```
dial-peer voice 1 pots
  application mgcpapp
```

**Related Commands**

Command	Description
<b>call application voice</b>	Defines the name to be used for an application and indicates the location of the appropriate IVR script to be used with this application.
<b>mgcp</b>	Starts the MGCP daemon.
<b>sgcp</b>	Starts and allocates resources for the SCGP daemon.
<b>sgcp call-agent</b>	Defines the IP address of the default SGCP call agent.

## cablelength long

To increase the pulse of a signal at the receiver and decrease the pulse from the transmitter using pulse equalization and line build-out for a T1 cable, use the **cablelength long** command in controller configuration mode. To return the pulse equalization and line build-out values to their default settings, use the **no** form of this command.

**cablelength long** *dbgain-value dbloss-value*

**no cablelength long**

Syntax Description	
<i>dbgain-value</i>	Number of decibels (dB) by which the receiver signal is increased. Use one of the following values: <ul style="list-style-type: none"> <li>• gain26</li> <li>• gain36</li> </ul> The default is 26 dB.
<i>dbloss-value</i>	Number of decibels by which the transmit signal is decreased. Use one of the following values: <ul style="list-style-type: none"> <li>• 0db</li> <li>• -7.5db</li> <li>• -15db</li> <li>• -22.5db</li> </ul> The default is 0 dB.

**Defaults** Receiver gain of 26 dB and transmitter loss of 0 dB.

**Command Modes** Controller configuration mode

Command History	Release	Modification
	11.2	This command was introduced.
	11.3	The following choices were added: <b>gain26, gain36, 0db, -7.5db, -15db, -22.5db.</b>
	12.0(5)T and 12.0(5)XK	This command was modified to include support as an ATM interface configuration command for the Cisco 2600 and 3600 series routers and as a controller configuration command for the Cisco AS5800 universal access server.
	12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

**Usage Guidelines**

A pulse equalizer regenerates a signal that has been attenuated and filtered by a cable loss. Pulse equalization does not produce a simple gain, but it filters the signal to compensate for complex cable loss. A gain26 receiver gain compensates for a long cable length equivalent to 26 dB of loss, while a gain36 compensates for 36 dB of loss.

The lengthening or *building out* of a line is used to control far-end crosstalk. Line build-out attenuates the stronger signal from the customer installation transmitter so that the transmitting and receiving signals have similar amplitudes. A signal difference of less than 7.5 dB is ideal. Line build-out does not produce simple flat loss (also known as *resistive* flat loss). Instead, it simulates a cable loss of 7.5 dB, 15 dB, or 22.5 dB so that the resulting signal is handled properly by the receiving equalizer at the other end.

On the Cisco 2600 and Cisco 3600 series routers, this command is supported on T1 long-haul links only. If you enter the **cablelength long** command on a DSX-1 (short haul) interface, the command is rejected.

The transmit attenuation value is best obtained by experimentation. If the signal received by the far-end equipment is too strong, reduce the transmit level by entering additional attenuation.

**Examples**

The following example specifies a pulse gain of 36 decibels and a decibel pulse rate of -7.5 decibels:

```
interface atm 0/2
 cablelength long gain36 -7.5db
```

**Related Commands**

Command	Description
<b>cablelength short</b>	Sets a cable length 655 feet or shorter for a DS1 link.

## cablelength short

To set a cable length 655 feet or shorter for a DS1 link on the Cisco 2600 and 3600 series routers, use the **cablelength short** command in controller configuration mode. This command is supported on T1 controllers only. To delete the **cablelength short** value, use the **no** form of this command. To set cable lengths longer than 655 feet, use the **cablelength long** command.

**cablelength short** *length*

**no cablelength short**

<b>Syntax Description</b>	<i>length</i>	Specifies a cable length. Use one of the following values to specify this value: <ul style="list-style-type: none"> <li>133—Specifies a cable length from 0 to 133 feet.</li> <li>266—Specifies a cable length from 134 to 266 feet.</li> <li>399—Specifies a cable length from 267 to 399 feet.</li> <li>533—Specifies a cable length from 400 to 533 feet.</li> <li>655—Specifies a cable length from 534 to 655 feet.</li> </ul>
---------------------------	---------------	---

**Defaults** There is no default value or behavior.

**Command Modes** Controller configuration mode

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	11.3(2)AA	This command was introduced.
	12.0(5)T and 12.0(5)XK	This command was modified to include support as an ATM interface command for the Cisco 2600 and 3600 series routers and as a controller configuration command for the Cisco AS5800 universal access server.
	12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

**Usage Guidelines** This command is supported on T1 short-haul links only. If you enter the **cablelength short** command on a long-haul interface, the command is rejected.

**Examples** On a Cisco 2600 or 3600 series router, the following example specifies a cable length from 0 to 133 feet:

```
interface atm 0/2
cablelength short 133
```

**cablelength short**

<b>Related Commands</b>	<b>Command</b>	<b>Description</b>
	<b>cablelength long</b>	Increases the pulse of a signal at the receiver and decreases the pulse from the sender using pulse equalization and line build-out.

## ccm-manager config

To configure the local Media Gateway Control Protocol (MGCP) voice gateway with a TFTP server IP address or logical name for download of the XML configuration file and to enable the download of the configuration, use the **ccm-manager config** command in global configuration mode. To disable the dial-peer and server configurations, use the **no** form of this command.

```
ccm-manager config { dialpeer-prefix | server { ip-address | name } }
```

```
no ccm-manager config { dialpeer-prefix | server { ip-address | name } }
```

Syntax Description		
<b>dialpeer-prefix</b>	Dial peer created for a voice dial-peer tag. Range is from 1 to 2147483647. The default is 999.	
<b>server</b>	IP address or logical name of the TFTP server from which the XML configuration files are downloaded.	<ul style="list-style-type: none"> <li><i>ip-address</i>—IP address of the TFTP server from which to download the XML configuration files to the local MGCP voice gateway.</li> <li><i>name</i>—Logical (symbolic) name of the TFTP server from which to download XML configuration files to the local MGCP voice gateway.</li> </ul>

**Defaults**  
The configuration download is disabled.  
dialpeer-prefix: 999

**Command Modes**  
Global configuration

Command History	Release	Modification
	12.2(2)XN	This command was introduced on the Cisco 2600 series, Cisco 3600 series and the Cisco VG200.

**Usage Guidelines**  
The **ccm-manager config** command is optional. If you separate the MGCP and H.323 dial peers under different dial-peer tags, ensure that the MGCP dial peers are configured before the H.323 dial peers. Direct-inward-dial (DID) is required for E1 PRI dial peers.

**Examples**  
The following example shows the configuration on the command:  
ccm-manager config

In the following example, the IP address of the TFTP server from which a configuration file is downloaded is identified:

```
ccm-manager config server 10.0.0.21
! Enter configuration commands, one per line.
ctrl z
```

---

**Related Commands**

<b>Command</b>	<b>Description</b>
<b>debug ccm-manager config</b>	Displays dialog during configuration download from the Cisco CallManager to the gateway.
<b>show ccm-config</b>	Displays whether or not the ccm-manager config is enabled.

# ccm-manager fallback-mgcp

To enable the gateway fallback feature on a Media Gateway Control Protocol (MGCP) voice gateway, use the **ccm-manager fallback-mgcp** command in global configuration mode. To disable fallback on the MGCP voice gateway, use the **no** form of this command.

**ccm-manager fallback-mgcp**

**no ccm-manager fallback-mgcp**

**Syntax Description** This command has no keywords or arguments.

**Defaults** Enabled

**Command Modes** Global configuration

Command History	Release	Modification
	12.2(2)XN	This command was introduced for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

**Usage Guidelines** The **ccm-manager fallback-mgcp** command must be enabled to cause the gateway to fall back. The mode and timing are set by default.

**Examples** The following example enables the gateway fallback feature on an MGCP voice gateway.

```
ccm-manager fallback-mgcp
```

Related Commands	Related Command	Purpose
	<b>show ccm-manager fallback-mgcp</b>	Displays the status of the MGCP gateway fallback feature.

## ccm-manager mgcp

To enable the gateway to communicate with the Cisco CallManager (CCM) through the Media Gateway Control Protocol (MGCP) and to supply redundant control agent services, use the **ccm-manager mgcp** command in global configuration mode. To disable communication with the CCM and redundant control agent services, use the **no** form of this command.

**ccm-manager mgcp**

**no ccm-manager mgcp**

**Syntax Description** This command has no keywords or arguments.

**Defaults** The 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 for Cisco CallManager Version 3.0 and the Cisco Voice Gateway 200 (Cisco VG200).
	12.2(2)XA	This command was implemented on the Cisco 2600 series and 3600 series routers.
	12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

**Usage Guidelines** This command enables the gateway to communicate with Cisco CallManager through MGCP. This command also enables control agent redundancy when a backup Cisco CallManager server is available.

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

```
ccm-manager mgcp
ccm-manager redundant-host 10.0.0.22
```

**Related Commands**

<b>Command</b>	<b>Description</b>
<b>ccm-manager redundant-host</b>	Configures the IP address or the DNS name of up to two backup Cisco CallManager servers for control agent redundancy.
<b>ccm-manager switchback</b>	Configures the switchback mode that determines when the primary Cisco CallManager server will be used if it becomes available again while a backup Cisco CallManager server is being used.
<b>mgcp</b>	Enables MGCP mode.

## ccm-manager music-on-hold

To enable the multicast music-on-hold (MOH) feature on Media Gateway Control Protocol (MGCP) voice gateways, use the **ccm-manager music-on-hold** command in global configuration mode. To disable the MOH feature on the voice gateway, use the **no** form of this command.

**ccm-manager music-on-hold**

**no ccm-manager music-on-hold**

**Syntax Description** This command has no arguments or keywords.

**Defaults** Disabled

**Command Modes** Global configuration

Command History	Release	Modification
	12.2(2)XN	This command was introduced for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

**Examples** The following example shows multicast MOH configured for an MGCP voice gateway:

```
mgcp call-agent 10.0.0.21 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000
mgcp modem passthrough voip mode cisco
mgcp package-capability rtp-package
mgcp package-capability sst-package
no mgcp timer receive-rtcp
call rsvp-sync
!
ccm-manager redundant-host 10.0.0.21
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server 10.0.0.21
!
```

Related Commands	Command	Description
	<b>show ccm-manager music-on-hold</b>	Displays the MOH information
	<b>debug ccm-manager music-on-hold events   packets</b>	Displays debugging information for MOH.

## ccm-manager redundant-host

To configure the IP address or the Domain Name System (DNS) name of up to two backup Cisco CallManager servers as call agents, use the **ccm-manager redundant-host** command in global configuration mode. To disable the backup CCM servers as call agents, 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

<i>ip-address</i>	Specifies the IP address of the backup Cisco CallManager server.
<i>DNS-name</i>	Specifies the DNS name of the backup Cisco CallManager server.

### Defaults

If you do not configure a backup Cisco CallManager server, redundancy is disabled.

### Command Modes

Global configuration

### Command History

Release	Modification
12.1(3)T	This command was introduced for Cisco CallManager Version 3.0 and the Cisco Voice Gateway 200 (Cisco VG200).
12.2(2)XA	This command was implemented on the Cisco 2600 series and 3600 series routers, and the <i>DNS-name</i> argument was added.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

### Usage Guidelines

You can configure up to two backup Cisco CallManager servers. The list of IP addresses or DNS names is an ordered and prioritized list. The Cisco CallManager server defined with the **mgcp call-agent** command has the highest priority (that is, it is the primary Cisco CallManager server). The gateway selects a Cisco CallManager server on the basis of the order of its appearance in this list.

### Examples

In the following example, the IP address of the backup Cisco CallManager server is 10.0.0.21:

```
ccm-manager redundant-host 10.0.0.21
```

Related Commands	Command	Description
	<b>ccm-manager switchback</b>	Configures the switchback mode that determines when the primary Cisco CallManager server will be used if it becomes available again while a backup Cisco CallManager server is being used.
	<b>ccm-manager switchover-to-backup</b>	Redirects (manually and immediately) a Cisco 2600 series router or Cisco 3600 series router to the backup Cisco CallManager server.
	<b>mgcp call-agent</b>	Defines the Cisco CallManager server as the highest priority.

## ccm-manager switchback

To specify the time at which control is to be returned to the primary Cisco CallManager server once it becomes available, use the **ccm-manager switchback** command in global configuration mode. To disable the specified time at which the primary server takes control, use the **no** form of this command.

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

```
no ccm-manager switchback
```

### Syntax Description

<b>graceful</b>	Specifies that control is returned to the primary Cisco CallManager server after the last active call ends (when there is no voice call in active setup mode on the gateway).
<b>immediate</b>	Specifies an immediate switchback to the primary Cisco CallManager server when the TCP link to the primary Cisco CallManager server is established regardless of current call conditions.
<b>schedule-time</b> <i>hh:mm</i>	Specifies an hour and minute, based on a 24-hour clock, when control is returned to the primary Cisco CallManager server. If the specified time is earlier than the current time, the switchback occurs at the specified time on the following day.
<b>uptime-delay</b> <i>minutes</i>	Specifies the number of minutes the primary Cisco CallManager server has run after the TCP link to it has been reestablished and control is returned to that primary call agent. Valid values are from 1 to 1440 (1 minute to 24 hours).

### Defaults

A graceful return to the primary Cisco CallManager server is the default switchback action.

### Command Modes

Global configuration

### Command History

Release	Modification
12.1(3)T	This feature was introduced for Cisco CallManager Version 3.0 and the Cisco Voice Gateway 200 (Cisco VG200).
12.2(2)XA	This command was implemented on the Cisco 2600 series and 3600 series routers.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

### Usage Guidelines

This command effects a switchback to the higher priority Cisco CallManager server when it becomes available. In other words, call control reverts to the original (primary) Cisco CallManager server once service to that call agent has been restored.

**Examples**

In the following example, the primary Cisco CallManager server is used as soon as it becomes available:

```
ccm-manager switchback immediate
```

**Related Commands**

<b>Command</b>	<b>Description</b>
<b>ccm-manager application redundant-link</b>	Configures the port number for the redundant link application (that is, for the secondary Cisco CallManager server).
<b>ccm-manager redundant-host</b>	Configures the IP address or the DNS name of up to two backup Cisco CallManager servers.
<b>ccm-manager switchover-to-backup</b>	Manually and immediately redirects a Cisco 2600 series or Cisco 3600 series router to the backup Cisco CallManager server.

# ccm-manager switchover-to-backup

To manually redirect the gateway to the backup Cisco CallManager (CCM) server, use the `ccm-manager switchover-to-backup` command in privileged EXEC mode.

## `ccm-manager switchover-to-backup`

**Syntax Description** This command has no arguments or keywords.

**Defaults** No default behaviors or values.

**Command Modes** Privileged EXEC

Command History	Release	Modification
	12.2(2)XN	This command was introduced on the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

**Usage Guidelines** The switchover to the backup CCM server occurs immediately. This command does not switch the gateway to the backup Cisco CallManager server if you have the `switchback` command option set to `immediate` and the primary Cisco CallManager server is still running.

**Examples** In the following example, the backup CCM server is used as soon as it becomes available:

```
ccm-manager switchover-to-backup
```

Related Commands	Command	Description
	<code>ccm-manager application redundant-link</code>	Configures the port number for the redundant link application (that is, for the secondary Cisco CallManager server).
	<code>ccm-manager redundant-host</code>	Configures the IP address or the DNS name of up to two backup Cisco CallManager servers.
	<code>ccm-manager switchback</code>	Specifies the time at which control is returned to the primary Cisco CallManager server once the server is available.

# clock source

To configure the clock source of a DS1 link, enter the **clock source** command in interface configuration, controller configuration, or ATM interface configuration mode. To restore the default **line** setting, use the **no** form of this command.

**clock source { line | internal | loop-timed }**

**no clock source**

## Syntax Description

<b>line</b>	Specifies that the T1/E1 link uses the recovered clock from the line. This is the default.
<b>internal</b>	Specifies that the T1/E1 link uses the internal clock from the interface.
<b>loop-timed</b>	Specifies that the T1/E1 interface takes the clock from the Rx (line) and uses it for Tx.

## Defaults

The default value is line.

## Command Modes

Controller configuration mode

## Command History

Release	Modification
10.3	This command was introduced.
11.1 CA	This command was modified to support the E1-G.703/G.704 serial port adapter, PA-E3 serial port adapters, and Cisco 7200 series routers.
11.3 MA	This command was introduced as a controller configuration command for the Cisco MC3810.
12.0(5)T and 12.0(5)XK	The command was introduced as an ATM interface configuration command for the Cisco 2600 and 3600 series routers.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

## Usage Guidelines

This command sets clocking for individual T1/E1 links.

Make sure that you specify the clock source correctly for each link, even if you are planning to specify that a certain link will provide clocking for all the links in an IMA group. Because links may be taken in and out of service, requiring that the system select another link for common clocking, any link in an IMA group may provide the common clock.

If the ATM interface is part of an IMA group, you can use the **loop-timed** keyword to specify that the clock source is the same as the IMA group clock source.

---

**Examples**

On a Cisco 2600 or 3600 series router, the following example specifies an internal clock source for the link:

```
interface atm 0/2
 clock source internal
```

---

**Related Commands**

<b>Command</b>	<b>Description</b>
<b>ima clock-mode</b>	Sets the transmit clock mode for an ATM IMA group.

---

# controller

To configure a T1 or E1 controller and enter controller configuration mode, use the **controller** command in global configuration mode.

**controller** {**t1** | **e1**} *slot/port*

Syntax Description		
<b>t1</b>		T1 controller.
<b>e1</b>		E1 controller.
<i>slot/port</i>		Backplane slot number and port number on the interface. See your hardware installation manual for the specific values and slot numbers.

**Defaults** No T1 or E1 controller is configured.

**Command Modes** Global configuration

Command History	Release	Modification
	10.0	This command was introduced.
	10.3	The <b>e1</b> keyword was added.
	12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

**Usage Guidelines** This command is used in configurations where the router or access server is intended to communicate with a T1 or E1 fractional data line. Additional parameters for the T1 or E1 line must be configured for the controller before the T1 or E1 circuits can be configured using the **interface** command in global configuration mode.

**Examples** The following example configures the MIP in slot 4, port 0 as a T1 controller:

```
controller t1 4/0
```

The following example configures NIM 0 as a T1 controller:

```
controller t1 0
```

**Related Commands**

<b>Command</b>	<b>Description</b>
<b>channel-group</b>	Defines the time slots that belong to each T1 or E1 circuit.
<b>clear controller</b>	Resets the T1 or E1 controller.
<b>clock source line</b>	Sets the E1 line clock source for the Cisco AS5200.
<b>framing</b>	Selects the frame type for the T1 or E1 data line.
<b>linecode</b>	Selects the linecode type for T1 or E1 line.
<b>show controllers e1</b>	Displays information about the E1 links supported by the NPM (Cisco 4000) or MIP (Cisco 7500 series).
<b>show controllers t1 call-counters</b>	Displays the total number of calls and call durations on a T1 controller.

## debug ccm-manager

To display debugging information about the Cisco CallManager (CCM), use the **debug ccm-manager** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

```
debug ccm-manager { backhaul { events | errors } | config-download { all | errors | events | packets | xml } | errors | events | music-on-hold { errors | events | packets } | packets }
```

```
no debug ccm-manager
```

Syntax Description		
<b>backhaul</b>	Enables debugging of the CCM backhaul. The keywords are as follows:	<ul style="list-style-type: none"> <li>• <b>events</b>—Displays CCM backhaul events.</li> <li>• <b>errors</b>—Displays CCM backhaul errors.</li> </ul>
<b>config-download</b>	Enables debugging of the CCM configuration download. The keywords are as follows:	<ul style="list-style-type: none"> <li>• <b>all</b>—Displays all CCM configuration parameters.</li> <li>• <b>errors</b>—Displays CCM configuration errors.</li> <li>• <b>events</b>—Displays CCM configuration events.</li> <li>• <b>packets</b>—Displays CCM configuration packets.</li> <li>• <b>xml</b>—Displays the CCM configuration XML parser.</li> </ul>
<b>errors</b>	Displays errors related to CCM.	
<b>events</b>	Displays CCM events, such as when the primary CCM server fails and control is switched to the backup CCM server.	
<b>music-on-hold</b>	Displays music-on-hold (MOH). The keywords are as follows:	<ul style="list-style-type: none"> <li>• <b>errors</b>—Displays MOH errors.</li> <li>• <b>events</b>—Displays MOH events.</li> <li>• <b>packets</b>—Displays MOH packets.</li> </ul>
<b>packets</b>	Displays CCM packets.	

**Command Modes** Privileged EXEC

Command History	Release	Modification
	12.1(3)T	This feature was introduced for Cisco CallManager Version 3.0 and the Cisco Voice Gateway 200 (Cisco VG200).
	12.2(2)XA	This command was implemented on Cisco 2600 series and Cisco 3600 series routers.
	12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

**Examples**

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

```
debug ccm-manager events
```

```
*Feb 28 22:56:05.873: cmapp_mgcpapp_go_down: Setting mgc status to NO_RESPONSE
*Feb 28 22:56:05.873: cmapp_host_fsm: New state DOWN for host 0 (172.20.71.38)
*Feb 28 22:56:05.873: cmapp_mgr_process_ev_active_host_failed: Active host 0
(172.20.71.38) failed
*Feb 28 22:56:05.873: cmapp_mgr_check_hostlist: Active host is 0 (172.20.71.38)
*Feb 28 22:56:05.877: cmapp_mgr_switchover: New actv host will be 1 (172.20.71.44)
*Feb 28 22:56:05.877: cmapp_host_fsm: Processing event GO_STANDBY for host 0
(172.20.71.38) in state DOWN
*Feb 28 22:56:05.877: cmapp_open_new_link: Open link for [0]:172.20.71.38
*Feb 28 22:56:05.877: cmbh_open_tcp_link: Opening TCP link with Rem IP 172.20.71.38, Local
IP 172.20.71.19, port 2428
*Feb 28 22:56:05.881: cmapp_open_new_link: Open initiated OK: Host 0 (172.20.71.38),
session_id=8186DEE4
*Feb 28 22:56:05.881: cmapp_start_open_link_tmr: Host 0 (172.20.71.38), tmr 0
*Feb 28 22:56:05.881: cmapp_host_fsm: New state STANDBY_OPENING for host 0 (172.20.71.38)
*Feb 28 22:56:05.881: cmapp_host_fsm: Processing event GO_ACTIVE for host 1 (172.20.71.44)
in state STANDBY_READY
*Feb 28 22:56:05.885: cmapp_mgr_send_rehome: new addr=172.20.71.44,port=2427
*Feb 28 22:56:05.885: cmapp_host_fsm: New state REGISTERING for host 1 (172.20.71.44)
```

Table 11 describes the significant fields shown in the display.

**Table 11** *debug ccm-manager events Field Descriptions*

Field	Description
<i>nn:nn:nn:</i>	Time stamp that indicates when the Cisco CallManager event occurred.
CMAPP: <i>error message</i>	The Cisco CallManager routine in which the error event occurred.

**Related Commands**

Command	Description
<b>show ccm-manager</b>	Displays a list of Cisco CallManager servers, their current status, and their availability.

# debug mgcp

To enable debug traces for errors, events, packets, or the parser for MGCP modules, use the **debug mgcp** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

**debug mgcp** [**all** | **errors** | **events** | **packets** | **parser**]

**no debug mgcp** [**all** | **errors** | **events** | **packets** | **parser**]

## Syntax Description

<b>all</b>	(Optional) Debugs errors, events, packets, and the parser for MGCP modules.
<b>errors</b>	(Optional) Debugs errors for MGCP modules.
<b>events</b>	(Optional) Debugs events for MGCP modules.
<b>packets</b>	(Optional) Debugs packets for MGCP modules.
<b>parser</b>	(Optional) Debugs the parser for MGCP modules.

## Command Modes

Privileged EXEC

## Command History

Release	Modification
12.1(1)T	This command was introduced for the Cisco AS5300 universal access server.
12.1(3)T	Additional information was displayed for the gateways.
12.1(5)XM	The output was modified to display parameters for the MGCP CAS PBX and AAL2 PVC features.
12.2(2)T	Support for this command was introduced on the Cisco 7200 routers.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

## Examples

The following is sample output from the **debug mgcp all** command:

```
debug mgcp all

Media Gateway Control Protocol all debugging is on.

00:04:52: MGC stat - 00.0.0.10, total=120, succ=119, failed=0
00:04:52: MGCP Packet received -
DLCX 422361185 S1/ds1-1/1 MGCP 0.1
S: dt/rel
R: dt/sup
X: 183936595
00:04:52: -- mgcp_parse_packet() - call mgcp_parse_header
- mgcp_parse_header()- Request Verb FOUND DLCX
- mgcp_parse_packet() - out mgcp_parse_header
SUCCESS: mgcp_parse_packet()-MGCP Header parsing was OK
- mgcp_parse_parameter_lines(), code_str:: dt/rel, code_len:2, str:S: dt/rel
- mgcp_parse_parameter_lines(str:S: dt/rel) -num_toks: 19
- mgcp_parse_parameter_lines() check NULL str(dt/rel), in_ptr(S: dt/rel)
```

```

- mgcp_parse_parameter_lines() return Parse function in mgcp_parm_rules_array[8]
- mgcp_parse_sig_requests(in_ptr:dt/rel, len: 6)
- mgcp_parse_sig_requests()- protocol is MGCP_V01 call get_package_name buf_ptr:dt/rel
- mgcp_parse_sig_requests()- call get_event_name() buf_ptr:rel
- mgcp_get_event_name()- event_type is MGCP_SIGNAL_EVT
- mgcp_get_event_name looking for event string rel in event table

00
- mgcp_parse_req_event()- protocol is MGCP_V01 call get_package_name buf_ptr:dt/sup

MGC stat - 00.0.0.10, total=129, succ=128, failed=0:04:52: mgcp_need_to_parse_params:
event = 69, param-is-optional = 0- mgcp_parse_sig_requests()-call mgcp-associate_pkg_evt()
- mgcp_associate_pkg_evt
- mgcp_parse_sig_requests()- mgcp-associate_pkg_evt(SUCC) buf_ptr:, len:0
- mgcp_parse_sig_requests() - NOT mp_emb_syntax_checking
- mgcp_parse_sig_requests() - add evt_node to rsd_- mgcp_parse_req_event()- call
get_event_name()buf_ptr:sup
- mgcp_get_event_name()- event_type is MGCP_REQUEST_EVT
- mgcp_get_event_name looking for event string sup in event table

00:04:52: mgcp_need_to_parse_params: event = 86, param-is-optional = 0-
mgcp_associate_pkg_evt
- mgcp_parse_req_event() -Check action: buf_ptr:
[additional display text omitted]

```

**Related Commands**

Command	Description
mgcp	Initiates the MGCP daemon.

## dial-peer voice

To enter dial-peer configuration mode and to specify the method of voice encapsulation, use the **dial-peer voice** command in global configuration mode. To disable a defined dial peer, use the **no** form of this command.

```
dial-peer voice tag {pots | voatm | vofr | voip}
```

```
no dial-peer voice tag {pots | voatm | vofr | voip}
```

### Syntax Description

<i>tag</i>	Digits that define a particular dial peer. Valid entries are from 1 to 2,147,483,647.
<b>mmoip</b>	Indicates that this is a multimedia mail peer using IP encapsulation on the IP backbone.
<b>pots</b>	Indicates that this is a plain old telephone service (POTS) peer using Voice over IP encapsulation on the IP backbone.
<b>voatm</b>	(Cisco 3600 series routers and Cisco MC3810 multiservice concentrators only) Specifies that this is a Voice over ATM dial peer using the real-time AAL5 voice encapsulation on the ATM backbone network.
<b>vofr</b>	Specifies that this is a Voice over Frame Relay dial peer using FRF.11 encapsulation on the Frame Relay backbone network.
<b>voip</b>	Indicates that this is a VoIP peer using voice encapsulation on the POTS network.

### Defaults

No default behavior or values.

### Command Modes

Global configuration

### Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series routers.
11.3(1)MA	This command was first supported on the Cisco MC3810 multiservice concentrator, with support for the <b>pots</b> , <b>voatm</b> , <b>vofr</b> , and <b>vohdlc</b> keywords.
12.0(3)T	This command was first supported on the AS5300, with support for the <b>pots</b> and <b>voip</b> keywords.
12.0(3)XG	The <b>vofr</b> keyword was added for the Cisco 2600 series routers and Cisco 3600 series platforms.
12.0(4)T	The <b>vofr</b> keyword was integrated into Cisco IOS Release 12.0(4)T. The <b>vofr</b> keyword was added to the Cisco 7200 series routers platform.
12.0(4)XJ	The <b>mmoip</b> keyword was added for the Cisco AS5300 universal access server platform. Also, the <b>dial-peer voice</b> command was implemented for store and forward fax.

Release	Modification
12.0(7)XK	The <b>voip</b> keyword was added for the Cisco MC3810 multiservice concentrator, and the <b>voatm</b> keyword was added for the Cisco 3600 series routers router. Support for <b>vohdlc</b> on the Cisco MC3810 multiservice concentrator was removed in this release.
12.1(1)	The <b>mmoip</b> keyword addition in Cisco IOS Release 12.0(4)XJ was integrated into Cisco IOS Release 12.1(1). The <b>dial-peer voice</b> implementation for store and forward fax was also integrated into this mainline release.
12.1(2)T	The keyword changes in Cisco IOS Release 12.0(7)XK were integrated into Cisco IOS Release 12.1(2)T.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

### Usage Guidelines

Use the **dial-peer voice** global configuration command to switch to dial-peer configuration mode from global configuration mode and to define a particular dial peer. Use the **exit** command to exit dial-peer configuration mode and return to global configuration mode.

After you have created a dial peer, that dial peer remains defined and active until you delete that particular dial peer. To delete a dial peer, use the **no** form of this command. To disable a dial peer, use the **no shutdown** command in dial-peer configuration mode.

### Examples

The following example shows how to access dial-peer configuration mode and configure a POTS peer identified as dial peer 10 and an MMoIP dial peer identified as dial peer 20:

```
dial-peer voice 10 pots
dial-peer voice 20 mmoip
```

The following example deletes the MMoIP peer identified as dial peer 20:

```
no dial-peer voice 20 mmoip
```

Related Commands	Command	Description
	<b>codec (dial-peer)</b>	Specifies the voice coder rate of speech for a Voice over Frame Relay dial peer.
	<b>destination-pattern</b>	Specifies the prefix, the full E.164 telephone number, or an ISDN directory number (depending on the dial plan) to be used for a dial peer.
	<b>dtmf-relay (Voice over Frame Relay)</b>	Enables the generation of FRF.11 Annex A frames for a dial peer.
	<b>preference</b>	Indicates the preferred order of a dial peer within a rotary hunt group.
	<b>sequence-numbers</b>	Enables the generation of sequence numbers in each frame generated by the DSP for Voice over Frame Relay applications.
	<b>session protocol</b>	Establishes a session protocol for calls between the local and remote routers via the packet network.
	<b>session target</b>	Specifies a network-specific address for a specified dial peer or destination gatekeeper.
	<b>voice-port</b>	Enters voice-port configuration mode.

## ds0-group

To specify the DS0 time slots that make up a logical voice port on a T1 controller and to specify the signaling type by which the router communicates with the PBX or Public Switched Telephone Network (PSTN), use the **ds0-group** command in controller configuration mode. To remove the group and signaling setting, use the **no** form of this command.

```
ds0-group ds0-group-no timeslots timeslot-list type {e&m-delay-dial | e&m-fgd |
e&m-immediate-start | e&m-wink-start | fgd-eana | fxo-ground-start | fxo-loop-start |
fxs-ground-start | fxs-loop-start}
```

```
no ds0-group ds0-group-no
```

Syntax Description	
<i>ds0-group-no</i>	A value from 0 to 23 that identifies the DS0 group.
<b>timeslots</b> <i>timeslot-list</i>	Time slot <i>timeslot-list</i> is a single time-slot number, a single range of numbers, or multiple ranges of numbers separated by commas. For T1, allowable values are from 1 to 24. Examples are as follows: <ul style="list-style-type: none"> <li>• 2</li> <li>• 1-15,17-24</li> <li>• 1-23</li> <li>• 2,4,6-12</li> </ul>
<b>type</b>	The signaling method selection for the <b>type</b> keyword depends on the connection that you are making. The ear and mouth (E&M) interface allows connection for PBX trunk lines (tie lines) and telephone equipment. The Foreign Exchange Station (FXS) interface allows connection of basic telephone equipment and PBX. The Foreign Exchange Office (FXO) interface is for connecting the central office (CO) to a standard PBX interface where permitted by local regulations; it is often used for off-premise extensions (OPXs). Types are the following: <ul style="list-style-type: none"> <li>• <b>e&amp;m-delay-dial</b>—The originating endpoint sends an off-hook signal and then waits for an off-hook signal followed by an on-hook signal from the destination.</li> <li>• <b>e&amp;m-fgd</b>—E&amp;M Type II Feature Group D.</li> <li>• <b>e&amp;m-immediate-start</b>—E&amp;M immediate start.</li> </ul> <p><b>Note</b> Only the E&amp;M immediate start and E&amp;M delay-dial signaling protocol is supported.</p> <ul style="list-style-type: none"> <li>• <b>e&amp;m-wink-start</b>—E&amp;M Mercury Exchange Limited Channel-Associated Signaling (MELCAS) wink-start signaling support.</li> <li>• <b>fgd-eana</b>—Feature Group D exchange access North American.</li> </ul>

- **fxo-ground-start**—FXO ground-start signaling support.
- **fxo-loop-start**—FXO loop-start signaling support.
- **fxs-ground-start**—FXS ground-start signaling support.
- **fxs-loop-start**—FXS loop-start signaling support.

**Defaults**

No DS0 group is defined.

**Command Modes**

Controller configuration

**Command History**

Release	Modification
11.2	This command was introduced for the Cisco AS5300 universal access server as the <b>cas-group</b> command.
11.3(1)MA	The command was introduced as the <b>voice-group</b> command for the Cisco MC3810 multiservice concentrator.
12.0(1)T	The <b>cas-group</b> command was introduced for the Cisco 3600 series routers.
12.0(5)T	The command was renamed <b>ds0-group</b> on the Cisco AS5300 and Cisco 2600 and 3600 series routers. Some keyword modifications were implemented.
12.0(5)XE	This command was introduced for the Cisco 7200 series.
12.0(7)XK	Support for this command was extended to the Cisco MC3810 multiservice concentrator. When the <b>ds0-group</b> command became available on the Cisco MC3810 multiservice concentrator, the <b>voice-group</b> command was removed and no longer supported. The <b>ext-sig</b> keyword replaced the <b>ext-sig-master</b> and <b>ext-sig-slave</b> keywords that were available with the <b>voice-group</b> command.
12.0(7)XR	The <b>mgcp</b> service type was added.
12.1(1)T	The <b>ds0-group</b> command was implemented for the Cisco 7200 series.
12.1(2)XH	The <b>e&amp;m-fgd</b> and <b>fgd-eana</b> keywords were added for Feature Group D signaling.
12.1(3)T	The command was modified for the Cisco 7500 series routers. The <b>fgd-os</b> signaling type and the <b>voice</b> service type were added.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

**Usage Guidelines**

The **ds0-group** command automatically creates a logical voice port that is numbered on Cisco 2600 series routers and Cisco 3600 series routers as: slot/port:ds0-group-no.

Although only one voice port is created for each group, applicable calls are routed to any channel in the group.

**Examples**

The following example shows ranges of T1 controller time slots configured for FXS ground-start and FXO loop-start signaling on a Cisco 2600 or 3600 series router:

```
T1 1/0
 framing esf
 linecode b8zs
 ds0-group 1 timeslots 1-10 type fxs-ground-start
 ds0-group 2 timeslots 11-24 type fxo-loop-start
```

**Related Commands**

<b>Command</b>	<b>Description</b>
<b>codec</b>	Specifies the voice coder rate of speech for a dial peer.
<b>codec complexity</b>	Specifies call density and codec complexity based on the codec standard you are using.

## framing (E1/T1 controller)

To select the frame type for the E1 or T1 data line, use the **framing** command in controller configuration mode.

### T1 Lines

```
framing {sf | esf}
```

### E1 Lines

```
framing {crc4 | no-crc4} [australia]
```

Syntax Description	Command	Description
	<b>sf</b>	Specifies Super Frame as the T1 frame type. This is the default.
	<b>esf</b>	Specifies extended Super Frame as the T1 frame type.
	<b>crc4</b>	Specifies CRC4 frame as the E1 frame type. This is the default for Australia.
	<b>no-crc4</b>	Specifies no CRC4 frame as the E1 frame type.
	<b>australia</b>	(Optional) Specifies the E1 frame type used in Australia.

### Defaults

Super frame is the default on a T1 line.

CRC4 frame is the default on an E1 line.

### Command Modes

Controller configuration

### Usage Guidelines

Use this command in configurations where the router or access server is intended to communicate with T1 or E1 fractional data lines. The service provider determines the framing type (**sf**, **esf**, or **crc4**) required for your T1/E1 circuit.

This command does not have a **no** form.

### Examples

The following example selects extended Super Frame as the T1 frame type:

```
framing esf
```

### Related Commands

Command	Description
<b>cablelength</b>	Specifies the distance of the cable from the routers to the network equipment.
<b>linecode</b>	Selects the linecode type for T1 or E1 line.

# hostname

To specify or modify the host name for the network server, use the **hostname** global configuration command.

**hostname** *name*

## Syntax Description

<i>name</i>	New host name for the network server.
-------------	---------------------------------------

## Defaults

The factory-assigned default host name is Router.

## Command Modes

Global configuration

## Command History

Release	Modification
10.0	This command was introduced.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

## Usage Guidelines

The host name is used in prompts and default configuration filenames.

Do not expect case to be preserved. Upper- and lowercase characters look the same to many internet software applications. It may seem appropriate to capitalize a name the same way you might do in English, but conventions dictate that computer names appear all lowercase. For more information, refer to RFC 1178, *Choosing a Name for Your Computer*.

The name must also follow the rules for ARPANET host names. They must start with a letter, end with a letter or digit, and have as interior characters only letters, digits, and hyphens. Names must be 63 characters or fewer. For more information, refer to RFC 1035, *Domain Names—Implementation and Specification*.

## Examples

The following example changes the host name to “sandbox”:

```
hostname sandbox
```

## Related Commands

Command	Description
<b>setup</b>	Enables you to make major enhancements to your configurations, for example, adding a protocol suit, making major addressing scheme changes, or configuring newly installed interfaces.

# interface serial

To specify a serial interface created on a channelized E1 or channelized T1 controller (for ISDN PRI, channel-associated signalling, or robbed-bit signalling), use the **interface serial** command in global configuration mode.

```
interface serial slot/port:timeslot
```

Syntax Description	
<i>slot/port</i>	Slot number and port number where the channelized E1 or T1 controller is located.
<i>:timeslot</i>	For ISDN, the D channel time slot, which is <b>:23</b> channel for channelized T1 and the <b>:15</b> for channelized E1. PRI time slots are in the range 0 to 23 for channelized T1 and in the range 0 to 30 for channelized E1.  For channel-associated signalling or robbed-bit signalling, the channel group number.  The colon (:) is required.  On a dual port card, it is possible to run channelized on one port and primary rate on the other port.

**Defaults** No default behavior or values.

**Command Modes** Global configuration

Command History	Release	Modification
	10.0	This command was introduced.
	12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

**Usage Guidelines** You must explicitly specify a serial interface. The D channel is always the 23 channel for T1 and the 15 for E1.

**Examples**

The following example configures channel groups on time slots 1 to 11 and ISDN PRI on time slots 12 to 24 of T1 controller 0. Then the examples configures the first two channel groups as serial interfaces 0:0 and 0:1.

```
controller t1 0
channel-group 0 timeslot 1-6
channel-group 1 timeslot 7
channel-group 2 timeslot 8
channel-group 3 timeslot 9-11
pri-group timeslots 12-24
!
interface serial 0:0
ip address 172.18.13.2 255.255.255.0
encapsulation ppp
!
interface serial 0:1
ip address 172.18.13.3 255.255.255.0
encapsulation ppp
```

The following example configures ISDN PRI on T1 controller 4/1 and then configures the D channel on the resulting serial interface 4/1:23:

```
controller t1 4/1
framing crc4
linecode hdb3
pri-group timeslots 1-24

interface serial 4/1:23
ip address 172.18.13.1 255.255.255.0
encapsulation ppp
```

**Related Commands**

Command	Description
<b>controller</b>	Configures a T1 or E1 controller and enters controller configuration mode.
<b>show controllers t1 call-counters</b>	Displays the total number of calls and call durations on a T1 controller.
<b>show interfaces</b>	Displays statistics for all interfaces configured on the router or access server.

# isdn protocol-emulate

To configure the Layer 2 and Layer 3 port protocol of a BRI voice port or a PRI interface to emulate NT (network) or TE (user) functionality, use the **isdn protocol-emulate** command in interface configuration mode. To restore the default (user), use the **no** form of this command.

**isdn protocol-emulate** {user | network}

**no isdn protocol-emulate**

## Syntax Description

<b>user</b>	Layer 2 and Layer 3 port protocol operation as TE (port functions as QSIG slave).
<b>network</b>	Layer 2 and Layer 3 port protocol operation as NT (port functions as QSIG master).

## Defaults

Port functions as Q.SIG slave.

## Command Modes

Interface configuration

## Command History

Release	Modification
12.0(3)XG	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco MC3810 concentrator.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

## Usage Guidelines

If you use the **no isdn protocol-emulate** command, the Layer 2 and Layer 3 protocol emulation defaults to user.

## Examples

The following example configures the Layer 2 and Layer 3 function of T1 PRI interface 23 to act as the Q.SIG master (NT):

```
interface serial 1:23
 isdn protocol-emulate network
```

The following example configures the Layer 2 and Layer 3 function of an E1 PRI interface to operate as Q.SIG slave (TE):

```
interface serial 1:15
 isdn protocol-emulate user
```

**Related Commands**

<b>Command</b>	<b>Description</b>
<b>isdn switch-type (PRI)</b>	Specifies the central office switch type on the ISDN PRI interface.
<b>network-clock-priority</b>	Specifies the clock-recovery priority for the BRI voice ports in a BVM.
<b>pri-group nec-fusion</b>	Configures your NEC PBX to support FCCS
<b>show cdapi</b>	Displays the CDAPI.
<b>show rawmsg</b>	Displays the raw messages owned by the required component.

## isdn switch-type (PRI)

To specify the central office switch type on the ISDN interface, or to configure the Cisco MC3810 PRI interface to support QSIG signalling, use the **isdn switch-type** command in global or interface configuration mode. To disable the switch or QSIG signalling on the ISDN interface, use the **no** form of this command.

**isdn switch-type** *switch-type*

**no isdn switch-type** *switch-type*

Syntax Description	<i>switch-type</i>	Service provider switch type; see <a href="#">Table 12</a> for a list of supported switches.
--------------------	--------------------	--

Defaults	The switch type defaults to <b>none</b> , which disables the switch on the ISDN interface.
----------	--

Command Modes	Global configuration or interface configuration
---------------	---



### Note

This command can be entered in either global configuration mode or in interface configuration mode. When entered in global configuration mode, the setting applies to the entire Cisco MC3810. When entered in interface configuration mode, the setting applies only to the T1/E1 interface specified. The interface configuration mode setting overrides the global configuration setting.

Command History	Release	Modification
	9.21	This command was introduced as a global command.
	11.3 T	This command was introduced as an interface command.
	12.0(2)T	The <b>primary-qsig-slave</b> and <b>primary-qsig master</b> switch type options were added to support PRI QSIG signalling.
	12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

Usage Guidelines	<p>You have a choice of configuring the <b>isdn-switch-type</b> command to support QSIG at either the global configuration level or at the interface configuration level. For example, if you have a QSIG connection on one line as well as on the BRI port, you can configure the ISDN switch type in one of the following combinations:</p> <ul style="list-style-type: none"> <li>Set the global <b>isdn-switch-type</b> command to support QSIG, and set the interface <b>isdn-switch-type</b> command for the <b>interface bri 0</b> command to a BRI setting such as 5ess.</li> <li>Set the global <b>isdn-switch-type</b> command to support BRI 5ess, and set the interface <b>isdn-switch-type</b> command for the <b>interface serial 1:23</b> command to support QSIG.</li> </ul>
------------------	--

- Configure the global **isdn-switch-type** command to another setting (such as switch type VN3), and then set the interface **isdn-switch-type** command for the **interface bri 0** command to a BRI setting, and set the interface **isdn-switch-type** command for the **interface serial 1:23** command to support QSIG.

The voice-port **codec** command must be configured before any calls can be placed over the connection to the PINX. The default codec type is G729a.

To disable the switch on the ISDN interface, specify the **isdn switch-type none** command.

Table 12 lists supported PRI switch types by geographic area.



#### Note

If you are using the Multiple ISDN Switch Types feature to apply the ISDN switch types to different interfaces, refer to the chapter “Setting Up Basic ISDN Service” in the *Cisco IOS Dial Technologies Configuration Guide* for additional details.

**Table 12 ISDN Service Provider PRI Switch Types**

Keywords by Area	Switch Type
<b>Voice/PBX Systems</b>	
<b>primary-qsig</b>	Supports QSIG signaling per Q.931. Network side functionality is assigned with the <b>isdn protocol-emulate</b> command.
<b>Australia and Europe</b>	
<b>primary-net5</b>	NET5 ISDN PRI switch types for Asia, Australia, and New Zealand; ETSI-compliant switches for Euro-ISDN E-DSS1 signaling system.
<b>Japan</b>	
<b>primary-ntt</b>	Japanese ISDN PRI switch.
<b>North America</b>	
<b>primary-4ess</b>	AT&T 4ESS switch type for the United States.
<b>primary-5ess</b>	AT&T 5ESS switch type for the United States.
<b>primary-dms100</b>	NT DMS-100 switch type for the United States.
<b>primary-ni</b>	National ISDN switch type.
<b>All users</b>	
<b>none</b>	No switch defined.

#### Examples

The following example demonstrates the Multiple ISDN Switch Type Feature. The global ISDN switch type setting is basic-net3. The PRI interface (channelized T1 controller), is configured to use the **isdn switch-type primary-net5** command.

```
isdn switch-type basic-net3
!
interface serial0:23
 isdn switch-type primary-net5
 ip address 172.21.24.85 255.255.255.0
```

## ■ isdn switch-type (PRI)

The following example configures T1 interface 23 to support Q.SIG signaling:

```
interface serial 1:23
 isdn switch-type primary-qsig
```

**Related Commands**

<b>Command</b>	<b>Description</b>
<b>isdn protocol-emulate</b>	Configures the Layer 2 and Layer 3 port protocol of a BRI voice port or a PRI interface to emulate NT (network) or TE (user) functionality.
<b>pri-group nec-fusion</b>	Configures your NEC PBX to support FCCS.
<b>show cdapi</b>	Displays the CDAPI.
<b>show rawmsg</b>	Displays the raw messages owned by the required component.

## isdn bind-l3 ccm-manager

To bind Layer 3 of the ISDN PRI interface of the Media Gateway Control Protocol (MGCP) voice gateway to the Cisco CallManager for PRI Q.931 signaling backhaul support, use the **isdn bind-L3 ccm-manager** command in interface configuration mode. To disable the binding of the ISDN PRI Layer 3 interface of the MGCP, use the **no** form of this command.

### isdn bind-l3 ccm-manager

**Syntax Description** This command has no arguments or keywords.

**Defaults** Disabled

**Command Modes** Interface configuration

Command History	Release	Modification
	12.2(2)XN	This command was introduced for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

**Usage Guidelines** This command enables the ISDN PRI backhaul on an MGCP-enabled voice gateway.

**Examples** The following example shows the configuration of the PRI Layer 3:

```
isdn bind-l3 ccm-manager
```

# linecode

To select the line-code type for T1 or E1 lines, use the **linecode** command in controller configuration mode.

**linecode** { **ami** | **b8zs** | **hdb3** }

## Syntax Description

<b>ami</b>	Specifies alternate mark inversion (AMI) as the line-code type. Valid for T1 or E1 controllers. This is the default for T1 lines.
<b>b8zs</b>	Specifies B8ZS as the line-code type. Valid for T1 controller only.
<b>hdb3</b>	Specifies high-density bipolar 3 (hdb3) as the line-code type. Valid for E1 controller only. This is the default for E1 lines.

## Defaults

The **ami** keyword is the default for T1 lines.

High-density bipolar 3 is the default for E1 lines.

## Command Modes

Controller configuration

## Command History

Release	Modification
10.3	This command was introduced.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

## Usage Guidelines

Use this command in configurations in which the router or access server must communicate with T1 fractional data lines. The T1 service provider determines which line-code type, either **ami** or **b8zs**, is required for your T1 circuit. Likewise, the E1 service provider determines which line-code type, either **ami** or **hdb3**, is required for your E1 circuit.

This command does not have a **no** form.

## Examples

The following example specifies B8ZS as the line-code type:

```
linecode b8zs
```

## mgcp call-agent

To configure the IP address for the primary or default Cisco CallManager server and to designate the optional destination User Datagram Protocol (UDP) port number for the specified Cisco CallManager server, use the **mgcp call-agent** command in global configuration mode. To disable the specified Cisco CallManager server, use the **no** form of this command.

```
mgcp call-agent {ip-address | host-name} [port] [service-type type] [version version-number]
```

```
no mgcp call-agent
```

Syntax Description		
	<i>ip-address</i>	Specifies the IP address of the Cisco CallManager server.
	<i>host-name</i>	Specifies the domain name of the Cisco CallManager server.
	<i>port</i>	(Optional) Specifies the UDP port. Valid values are from 1025 to 65,535. The default port is 2427.
	<b>service-type</b> <i>type</i>	(Optional) Specifies the type of gateway control service to be supported by the call agent. The valid value is <b>mgcp</b> , which is the default.
	<b>version</b> <i>version-number</i>	(Optional) Specifies the version of <b>service-type</b> . The valid value is 0.1.

### Defaults

port: 2427

service-type: mgcp

### Command Modes

Global configuration

### Command History

Release	Modification
12.1(1)T	This command was introduced for the Cisco AS5300 universal access server.
12.1(3)T	The <b>service-type</b> parameter was added to the command.
12.1(5)XM	This command was implemented on Cisco MC3810 multiservice concentrators and Cisco 3600 series routers, and the <b>version</b> keyword was added.
12.2(2)T	This command was integrated into Cisco IOS Release 12.2(2)T.
12.2(2)XA	This command was implemented on Cisco 2600 and 3600 series routers.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

### Usage Guidelines

Use this command on any platform and media gateway.

If you do not specify a UDP port from the command line, Media Gateway Control Protocol (MGCP) will use 2427 as the default call agent UDP port.

The service type mgcp supports the RSIP error messages sent by the gateway if **mgcp restart notify** command is enabled.

---

**Examples**

The following examples illustrate several formats for specifying the call agent and service type (use any one of these formats):

```
mgcp call-agent 255.255.255.225 5530 service-type mgcp
mgcp call-agent igloo 2009 service-type mgcp version 0.1
```

---

**Related Commands**

Command	Description
<b>mgcp</b>	Starts the MGCP daemon.

## mgcp dtmf-relay

To ensure accurate forwarding of digits on compressed codecs, use the **mgcp dtmf-relay** command in global configuration mode. To disable the forwarding of digits on compressed codecs, use the **no** form of this command.

```
mgcp dtmf-relay voip codec {all | low-bit-rate} mode {cisco | nse | out-of-band}
```

```
no mgcp dtmf-relay voip
```

Syntax Description		
<b>voip codec</b>	Dual Tone Multifrequency (DTMF) for Voice over IP (VoIP) calls. The keywords are as follows:	<ul style="list-style-type: none"> <li>• <b>all</b>—All codecs.</li> <li>• <b>low-bit-rate</b>—Any version of the G.729 low-bit-rate codecs.</li> </ul>
<b>mode</b>	DTMF mode. The keywords are as follows:	<ul style="list-style-type: none"> <li>• <b>cisco</b>—DTMF tone from the voice stream is removed and FRF.11 is sent with a special payload of 121 for DTMF digits.</li> <li>• <b>nse</b>—NSE-based forwarding method.</li> <li>• <b>out-of-band</b>—DTMF tone from the voice stream is removed and FRF.11 is not sent.</li> </ul>

**Defaults** No DTMF relay for all codecs is specified.

**Command Modes** Controller configuration

Command History	Release	Modification
	12.1(3)T	This command was added to Media Gateway Control Protocol (MGCP).
	12.1(5)XM	Support for this command was expanded to include the Cisco MC3810 multiservice concentrator.
	12.2(2)T	This command was integrated into the Cisco IOS Release 12.2(2)T.
	12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

**Usage Guidelines** Only VoIP supports the **mode** options for forwarding digits on codecs.

**Examples** The following example shows how to configure a low-bit-rate codec using VoIP in NSE mode:

```
mgcp dtmf-relay voip codec low-bit-rate mode nse
```

# pri-group

To specify an ISDN PRI on a channelized T1 or E1 controller, use the **pri-group** command in controller configuration mode. To remove the ISDN PRI configuration, use the **no** form of this command.

**pri-group timeslots** *timeslot-range*

**no pri-group**

## Syntax Description

**timeslots** *timeslot-range* Specifies a single range of values. For T1, the allowable range is from 1 to 23. For E1, the allowable range is from 1 to 15.

## Defaults

There is no ISDN-PRI group configured.

## Command Modes

Controller configuration

## Command History

Release	Modification
11.0	This command was introduced.
12.0(2)T	This command was introduced for the Cisco MC3810 multiservice concentrator.
12.0(7)XK	This command was introduced for the Cisco 2600 and 3600 series router.
12.1(2)T	The command was integrated into Cisco IOS Release 12.1(2)T.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

## Usage Guidelines

Before you enter the **pri-group** command, you must specify an ISDN-PRI switch type and an E1 or T1 controller.



### Note

Only one PRI group can be configured on a controller.

## Examples

The following example configures ISDN-PRI on all time slots of controller E1 on a Cisco 2600 series router:

```
controller E1 4/1
pri-group timeslots 1-7,16
```

Related Commands	Command	Description
	isdn switch-type	Configures the Cisco 2600 series router router PRI interface to support QSIG signaling.

## show ccm-manager

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

```
show ccm-manager [backhaul | config-download | fallback-mgcp | hosts | music-on-hold |
  redundancy]
```

Syntax	Description
<b>backhaul</b>	(Optional) Displays PRI backhaul link information only.
<b>config-download</b>	(Optional) Displays information about the status of Media Gateway Control Protocol (MGCP) configuration download.
<b>fallback-mgcp</b>	(Optional) Displays the status of the MGCP gateway fallback feature.
<b>hosts</b>	(Optional) Displays a list of each configured Cisco CallManager server in the network, together with its operational status and host IP address.
<b>music-on-hold</b>	(Optional) Displays information about all the multicast music-on-hold (MOH) sessions in the gateway at any given point in time.
<b>redundancy</b>	(Optional) Displays failover mode and status information for hosts, including the redundant link port, failover interval, keepalive interval, MGCP traffic time, switchover time, and switchback mode.

**Defaults** If you omit any keywords in this command, information related to all keywords is displayed by default.

**Command Modes** Privileged EXEC

Command History	Release	Modification
	12.1(3)T	This feature was introduced with Cisco CallManager Version 3.0 and the Cisco Voice Gateway 200 (Cisco VG200).
	12.2(2)XA	This command was implemented on Cisco 2600 series and 3600 series routers.
	12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

**Usage Guidelines** Use this command to determine whether your primary or backup Cisco CallManager server is down, idle, or polling the backup Cisco CallManager server.

**Examples**

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

```
show ccm-manager

MGCP Domain Name: c3660A.cisco.com
Priority          Status                      Host
=====
Primary          Registered                     IOS-38 (172.20.71.38)
First Backup     Backup Ready                    ios-44 (172.20.71.44)
Second Backup    None

Current active Call Manager: 172.20.71.38
Backhaul/Redundant link port: 2428
Failover Interval: 30 seconds
Keepalive Interval: 15 seconds
Last keepalive sent: 03:06:24 (elapsed time: 00:00:06)
Last MGCP traffic time: 03:06:24 (elapsed time: 00:00:06)
Last failover time: 02:56:35 from (172.20.71.44)
Switchback mode: Graceful
MGCP fallback mode: Enabled/OFF
Last MGCP fallback start time: 00:00:00
Last MGCP fallback end time: 00:00:00

PRI Backhaul Link info:
  Link Protocol: TCP
  Remote Port Number: 2428
  Remote IP Address: 172.20.71.38
  Current Link State: OPEN
  Statistics:
    Packets recvd: 1
    Recv failures: 0
    Packets xmitted: 3
    Xmit failures: 0
  PRI Ports being backhauled:
    Slot 1, port 1
MGCP Download Tones: Enabled

Configuration Auto-Download Information
=====
Current version-id: {1645327B-F59A-4417-8E01-7312C61216AE}
Last config-downloaded:00:00:49
Current state: Waiting for commands
Configuration Download statistics:
  Download Attempted : 6
  Download Successful : 6
  Download Failed : 0
  Configuration Attempted : 1
  Configuration Successful : 1
  Configuration Failed(Parsing): 0
  Configuration Failed(config) : 0
Last config download command: New Registration
Configuration Error History:
FAX mode: cisco
```

Table 13 describes the significant fields shown in the display.

**Table 13** show ccm-manager Field Descriptions

Field	Description
MGCP Domain Name (system)	System used in the Internet for translating domain names of network nodes into IP addresses.
Priority	Priority of the Cisco CallManager servers present in the network. Possible priorities are primary, first backup, and second backup.
Status	Current usage of the Cisco CallManager server. Possible values are registered, idle, backup polling, and undefined.
Host	Host IP address of the Cisco CallManager server.
Current active Call Manager	Active Cisco CallManager server. This field can be any one of the following: primary, first backup, and second backup.
Backhaul/Redundant link port	Port that the Cisco CallManager server is to use.
Failover Interval	Maximum amount of time that can elapse without the gateway receiving messages from the currently active Cisco Call Manager before the gateway switches to the backup Cisco Call Manager.
Keepalive Interval	If the gateway has not received any messages from the currently active Cisco CallManager server within the specified amount of time, the gateway sends a keepalive message to the Cisco CallManager server to determine if it is operational.
Last keepalive sent	The time at which the last keepalive message was sent.
Last MGCP traffic time	The time as which the last MGCP traffic message was sent.
Switchback mode	Displays the switchback mode configuration that determines when the primary Cisco CallManager server will be used if it becomes available again while a backup Cisco CallManager server is being used. Possible values that can appear in this field include the following: <ul style="list-style-type: none"> <li>• graceful</li> <li>• immediate</li> <li>• schedule-time</li> <li>• uptime-delay</li> </ul>
MGCP Fallback mode	Displays the MGCP fallback mode configuration. If “Not Selected” displays, then fallback is not configured. If “Enabled/OFF” displays, then fallback is configured but not in effect. If “Enabled/ON” displays, then fallback is configured and in effect.
Last MGCP Fallback start time	Starts the timestamp of the last fallback.
Lasts MGCP Fallback end time	End the timestamp of the last fallback.

The following is sample output from the **show ccm-manager config** command that displays :

```
show ccm-manager config

Configuration Auto-Download Information
=====
Current version-id:{4171F93A-D8FC-49D8-B1C4-CE33FA8095BF}
Last config-downloaded:00:00:47
Current state:Waiting for commands
Configuration Download statistics:
    Download Attempted           :6
    Download Successful          :6
    Download Failed              :0
    Configuration Attempted     :1
    Configuration Successful     :1
    Configuration Failed(Parsing):0
    Configuration Failed(config) :0
Last config download command:New Registration
```

Table 14 describes the significant fields shown in the display.

**Table 14** *show ccm-manager config Field Descriptions*

Field	Description
Current state	Current configuration state.
Download Attempted	The number of times the gateway has tried to download the configuration file. The number of successes and failures is displayed.
Configuration Attempted	The number of times the gateway has tried to configure the gateway based on the configuration file. The number of successes and failures is displayed.

The following is sample output from the **show ccm-manager fallback-mgcp** command that displays:

```
show ccm-manager fallback-mgcp

Current active Call Manager: 172.20.71.38
MGCP Fallback mode:         Enabled/OFF
Last MGCP Fallback start time: 00:14:35
Last MGCP Fallback end time: 00:17:25
```

Table 15 describes the significant fields shown in the display.

**Table 15** show ccm-manager fallback-mgcp Field Descriptions

Field	Description
MGCP Fallback mode	The following are displayed: <ul style="list-style-type: none"> <li>• Not Selected—Fallback is not configured.</li> <li>• Enabled/OFF—Fallback is configured but not in effect.</li> <li>• Enabled/ON—Fallback is configured and in effect.</li> </ul>
Last MGCP Fallback start time	Start time stamp of last fallback.
Last MGCP Fallback end time	End time stamp of last fallback.

The following is sample output from the **show ccm-manager music-on-hold** that displays:

```
show ccm-manager music-on-hold
```

```
Current active multicast sessions :1
Multicast      RTP port  Packets      Call   Codec   Incoming
Address        number    in/out       id     id      Interface
=====
172.20.71.38   2428     5/5         99    g711
```

**Table 16** describes the significant fields shown in the sample output above from the **show ccm-manager music-on-hold** command.

**Table 16** show ccm-manager music-on-hold Field Descriptions

Field	Description
Current active multicast sessions	Number of active calls on hold.
Multicast Address	A valid class D address from which the gateway is getting the RTP streams.
RTP port number	A valid RTP port number on which the gateway receives the RTP packets.
Packets in/out	Number of RTP packets received and sent to the digital signal processor (DSP).
Call id	Call ID of the call that is on hold.
Codec	Codec number.
Incoming Interface	The interface through which the gateway is receiving the RTP stream.

**Related Commands**

Command	Description
<b>show mgcp</b>	Displays the MGCP configuration information.

# show dial-peer voice

To display configuration information for dial peers, use the **show dial-peer voice** command in privileged EXEC mode.

**show dial-peer voice** [*number*] [**summary**]

Syntax Description	
<i>number</i>	(Optional) A specific dial peer. This option displays configuration information for a single dial peer identified by the <i>number</i> argument. Valid entries are any integers that identify a specific dial peer, from 1 to 32,767.
<b>summary</b>	(Optional) Displays a summary of all voice dial peers.

**Command Modes** Privileged EXEC

Command History	Release	Modification
	11.3(1)T	This command was introduced.
	11.3(1)MA	The <b>summary</b> keyword was added for the Cisco MC3810 multiservice concentrator.
	12.0(3)XG	This command was modified to support Voice over Frame Relay (VoFR) for the Cisco 2600 series and Cisco 3600 series routers.
	12.0(4)T	Support was added for VoFR for the Cisco 7200 series routers.
	12.1(3)T	This command was modified for Modem Passthrough over Voice over IP on the Cisco AS5300 universal access server.
	12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

**Usage Guidelines** Use the **show dial-peer voice** privileged EXEC command to display the configuration for all Voice over IP (VoIP) and plain old telephone service (POTS) dial peers configured for the router. To show configuration information for only one specific dial peer, use the argument *number* to identify the dial peer.

**Examples** The following is sample output from the **show dial-peer voice** command for a POTS dial peer:

```
show dial-peer voice 1

VoiceEncapPeer1
  tag = 1, dest-pat = `+1xxxxxxxxxxx',
  answer-address = `',
  group = 0, Admin state is up, Operation state is down
  Permission is Both,
  session-target = `', voice port =
  Connect Time = 0, Charged Units = 0
  Successful Calls = 0, Failed Calls = 0
```

■ **show dial-peer voice**

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

The following is sample output from the **show dial-peer voice** command for a VoIP dial peer:

```
show dial-peer voice 10

VoiceOverIpPeer10
  tag = 10, dest-pat = `',
  incall-number = `+14087',
  group = 0, Admin state is up, Operation state is down
  Permission is Answer,
  session-protocol = cisco, req-qos = bestEffort,
  acc-qos = bestEffort,
  fax-rate = voice, codec = g729r8,
  Expect factor = 10,Icpif = 30, VAD = disabled, Poor QOV Trap = disabled,
  Connect Time = 0, Charged Units = 0
  Successful Calls = 0, Failed Calls = 0
  Accepted Calls = 0, Refused Calls = 0
  Last Disconnect Cause is ""
  Last Disconnect Text is ""
  Last Setup Time = 0
```

Table 17 provides an alphabetical listing of the **show dial-peer voice** output fields and a description of each field.

**Table 17** *show dial-peer voice Field Descriptions*

Field	Description
Accepted Calls	Number of calls accepted from this peer since system startup.
acc-qos	Lowest acceptable quality of service configured for calls for this peer.
Admin state	Administrative state of this peer.
answer-address	Answer address configured for this dial peer.
Charged Units	Total number of charging units applying to this peer since system startup. The unit of measure for this field in hundredths of a second.
codec	Default voice coder rate of speech for this peer.
Connect Time	Accumulated connect time to the peer since system startup for both incoming and outgoing calls. The unit of measure for this field is in hundredths of a second.
dest-pat	Destination pattern (telephone number) for this peer.
DTMF Relay	Indicates whether or not dual-tone multifrequency (DTMF) relay has been enabled, by using the <b>dtmf-relay</b> command, for this dial peer.
Expect factor	User-requested Expectation Factor of voice quality for calls through this peer.
Failed Calls	Number of failed call attempts to this peer since system startup.
fax-rate	Fax transmission rate configured for this peer.
group	Group number associated with this peer.
huntstop	Indicates whether dial-peer hunting has been turned on, by using the <b>huntstop</b> command, for this dial peer.

**Table 17** show dial-peer voice Field Descriptions (continued)

Field	Description
Icpif	Configured Calculated Planning Impairment Factor (ICPIF) value for calls sent by a dial peer.
incall-number	Full E.164 telephone number to be used to identify the dial peer.
incoming called-number	Indicates the incoming called number if it has been set by using the <b>incoming-called number</b> command.
information type	Information type for this call; for example, voice or fax.
Last Disconnect Cause	Encoded network cause associated with the last call. This value will be updated whenever a call is started or cleared and depends on the interface type and session protocol being used on this interface.
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.
Modem passthrough	Modem pass-through signaling method is named signaling event (NSE).
Operation state	Operational state of this peer.
Payload type	NSE payload type.
Permission	Configured permission level for this peer.
Poor QOV Trap	Whether Poor Quality of Voice trap messages have been enabled or disabled.
Redundancy	Packet redundancy (RFC 2198) for modem traffic.
Refused Calls	Number of calls from this peer refused since system startup.
req-qos	Configured requested quality of service for calls for this dial peer.
session-target	Session target of this peer.
sess-proto	Session protocol to be used for Internet calls between local and remote routers through the IP backbone.
Successful Calls	Number of completed calls to this peer.
tag	Unique dial peer ID number.
VAD	Whether voice activation detection (VAD) is enabled for this dial peer.

**Related Commands**

Command	Description
<b>show call active voice</b>	Displays the Voice over IP active call table.
<b>show call history voice</b>	Displays the Voice over IP call history table.
<b>show num-exp</b>	Displays how the number expansions are configured in Voice over IP.
<b>show voice port</b>	Displays configuration information about a specific voice port.

## show isdn status

To display the information about memory, Layer 2 and Layer 3 timers, and the status of PRI channels, use the **show isdn** command in EXEC mode.

**show isdn status** [*dsl* | *serial number*]

<b>Syntax Description</b>	[ <i>dsl</i>   <i>serial number</i> ]	Displays the status of all ISDN interfaces or, optionally, a specific digital signal link (DSL) or a specific ISDN PRI interface (created and configured as a serial interface). Values of the argument <i>dsl</i> range from 0 to 15.
---------------------------	---------------------------------------	--

<b>Command Modes</b>	EXEC
----------------------	------

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	11.1	This command was introduced.
	12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

**Examples** The following is sample output from the **show isdn** command with the **status** keyword when no calls are active for a Cisco 3600 with 8 BRI's and 1 E1 PRI:

```
show isdn status

Global ISDN Switchtype = basic-5ess
ISDN BRI0 interface
  dsl 0, interface ISDN Switchtype = basic-5ess
  Layer 1 Status:
    ACTIVE
  Layer 2 Status:
    TEI = 64, Ces = 1, SAPI = 0, State = MULTIPLE_FRAME_ESTABLISHED
  Layer 3 Status:
    0 Active Layer 3 Call(s)
  Activated dsl 0 CCBs = 0
ISDN BRI1 interface
  dsl 1, interface ISDN Switchtype = basic-5ess
  Layer 1 Status:
    DEACTIVATED
  Layer 2 Status:
    Layer 2 NOT Activated
  Layer 3 Status:
    0 Active Layer 3 Call(s)
  Activated dsl 1 CCBs = 0
ISDN BRI2 interface
  dsl 2, interface ISDN Switchtype = basic-5ess
  Layer 1 Status:
    DEACTIVATED
  Layer 2 Status:
    Layer 2 NOT Activated
  Layer 3 Status:
```

```
    0 Active Layer 3 Call(s)
  Activated dsl 2 CCBs = 0
ISDN BRI3 interface
  dsl 3, interface ISDN Switchtype = basic-5ess
  Layer 1 Status:
    ACTIVE
  Layer 2 Status:
    TEI = 75, Ces = 1, SAPI = 0, State = MULTIPLE_FRAME_ESTABLISHED
  Layer 3 Status:
    0 Active Layer 3 Call(s)
  Activated dsl 3 CCBs = 0
ISDN BRI4 interface
  dsl 4, interface ISDN Switchtype = basic-5ess
  Layer 1 Status:
    DEACTIVATED
  Layer 2 Status:
    Layer 2 NOT Activated
  Layer 3 Status:
    0 Active Layer 3 Call(s)
  Activated dsl 4 CCBs = 0
ISDN BRI5 interface
  dsl 5, interface ISDN Switchtype = basic-5ess
  Layer 1 Status:
    DEACTIVATED
  Layer 2 Status:
    Layer 2 NOT Activated
  Layer 3 Status:
    0 Active Layer 3 Call(s)
  Activated dsl 5 CCBs = 0
ISDN BRI6 interface
  dsl 6, interface ISDN Switchtype = basic-5ess
  Layer 1 Status:
    DEACTIVATED
  Layer 2 Status:
    Layer 2 NOT Activated
  Layer 3 Status:
    0 Active Layer 3 Call(s)
  Activated dsl 6 CCBs = 0
ISDN BRI7 interface
  dsl 7, interface ISDN Switchtype = basic-5ess
  Layer 1 Status:
    DEACTIVATED
  Layer 2 Status:
    Layer 2 NOT Activated
  Layer 3 Status:
    0 Active Layer 3 Call(s)
  Activated dsl 7 CCBs = 0
ISDN Serial0:15 interface
  dsl 8, interface ISDN Switchtype = primary-ni
  Layer 1 Status:
    ACTIVE
  Layer 2 Status:
    TEI = 0, Ces = 1, SAPI = 0, State = MULTIPLE_FRAME_ESTABLISHED
  Layer 3 Status:
    0 Active Layer 3 Call(s)
  Activated dsl 8 CCBs = 0
  Total Allocated ISDN CCBs = 0
```

The following is sample output from the **show isdn** command with the **status** keyword, with one active call:

```
show isdn status

The current ISDN Switchtype = ntt
ISDN BRI0 interface
  Layer 1 Status:
    ACTIVE
  Layer 2 Status:
    TEI = 64, State = MULTIPLE_FRAME_ESTABLISHED
  Layer 3 Status:
    1 Active Layer 3 Call(s)
  Activated dsl 0 CCBs = 1
    CCB:callid=8003, callref=0, sapi=0, ces=1, B-chan=1
  Number of active calls = 1
  Number of available B-channels = 1
  Total Allocated ISDN CCBs = 1
```

Table 18 describes the significant fields shown in the **show isdn status** display.

**Table 18** *show isdn status Field Descriptions*

Field	Description
<b>Layer 1 Status</b>	
ACTIVE	Status of ISDN Layer 1.
<b>Layer 2 Status</b>	
TEI = 65, State = MULTIPLE_FRAME_ESTABLISHED	Status of ISDN Layer 2. Terminal endpoint identifier number and multiframe structure state.
<b>Spid Status</b>	
TEI 65, ces = 1, state = 5(init)	Terminal endpoint identifier number and state.
<b>Layer 3 Status</b>	
1 Active Layer 3 Call(s)	Number of active calls.
Activated dsl 0 CCBs =	Number of the Digital Signal Links activated. Number of call control blocks in use.
CCB:callid=8003, callref=0, sapi=0, ces=1, B-chan=1	Information about the active call.
Number of active calls =	Number of active calls.
Number of available B-channels =	Number of B channels that are not being used.
Total Allocated ISDN CCBs =	Number of ISDN call control blocks that are allocated.

# show mgcp

To display Media Gateway Control Protocol (MGCP) configuration information, use the **show mgcp** command in EXEC mode.

**show mgcp [connection | endpoint | statistics]**

Syntax Description	connection	(Optional) Displays the active MGCP-controlled connections.
	endpoint	(Optional) Displays the MGCP-controlled endpoints.
	statistics	(Optional) Displays MGCP statistics regarding received and transmitted network messages.

**Command Modes** EXEC

Command History	Release	Modification
	12.1(1)T	This command was introduced for the Cisco AS5300 universal access server.
	12.1(3)T	Output was updated to show additional gateway and platform information.
	12.1(5)XM	Output was updated to show additional gateway and platform information.
	12.2(2)T	Support for this command was introduced on the Cisco 7200 series routers.
	12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

**Examples** The following is sample output from the **show mgcp** command:

```
show mgcp

MGCP Admin State ACTIVE, Oper State ACTIVE - Cause Code NONE
MGCP call-agent: 00.0.0.00 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 for VoIP disabled for all codec types
MGCP dtmf-relay for VoAAL2 disabled for all codec types
MGCP voip modem passthrough mode: NSE, codec: g711ulaw, redundancy: DISABLED,
MGCP voaal2 modem passthrough mode: NSE, codec: g711ulaw
MGCP TSE payload: 100
MGCP Network (IP/AAL2) Continuity Test timer: 200
MGCP 'RTP stream loss' timer: 5
MGCP request timeout 500, MGCP request retries 3
MGCP gateway port: 2427, MGCP maximum waiting delay 3000
MGCP restart delay 0, MGCP vad DISABLED
MGCP simple-sdp ENABLED
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
```

```
show mgcp
```

```
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: gm-package
MGCP supported packages: gm-package dtmf-package mf-package trunk-package
  line-package hs-package rtp-package ms-package dt-package
  mo-package
MGCP VoAAL2 ignore-lco-codec DISABLED
```

Table 19 describes the significant fields shown in the display.

**Table 19** *show mgcp Field Descriptions*

Field	Description
Initial protocol service is...	Indicates the protocol initiated for this session.
MGCP Admin State...Oper State	The administrative and operational state of the MGCP daemon. The administrative state controls starting and stopping the application using the <b>mgcp</b> and <b>mgcp block-newcalls</b> commands. The operational state controls normal MGCP operations.
MGCP block-newcalls enabled	The state of the <b>mgcp block-newcalls</b> command.
MGCP call-agent	The address of the call agent specified in the <b>mgcp</b> command.
MGCP cisco fgdos	For Cisco use only.
MGCP codec type	The setting for the <b>mgcp codec</b> command.
MGCP default package	The <b>default-package</b> parameter setting for the <b>mgcp default-package</b> command.
MGCP dtmf-relay	The setting for the <b>mgcp dtmf-relay</b> command.
MGCP gateway port	The UDP port specification.
MGCP IP precedence	The <b>precedence</b> parameter setting for the <b>mgcp ip-tos</b> command.
MGCP IP RTP precedence	The <b>rtp precedence</b> parameter setting for the <b>mgcp ip-tos</b> command.
MGCP IP ToS high reliability	The <b>high-reliability</b> parameter setting for the <b>mgcp ip-tos</b> command.
MGCP IP ToS high throughput	The <b>high-throughput</b> parameter setting for the <b>mgcp ip-tos</b> command.
MGCP IP ToS low cost	The <b>low-cost</b> parameter setting for the <b>mgcp ip-tos</b> command.
MGCP IP ToS low delay	The <b>low-delay</b> parameter setting for the <b>mgcp ip-tos</b> command.
MGCP JB threshold hwm	The <b>jitter buffer maximum threshold</b> parameter setting for the <b>mgcp quality-threshold</b> command.

**Table 19** show mgcp Field Descriptions (continued)

Field	Description
MGCP JB threshold lwm	The <b>jitter buffer minimum threshold</b> parameter setting for the <b>mgcp quality-threshold</b> command.
MGCP LAT threshold hwm	The latency maximum threshold parameter setting for the <b>mgcp quality-threshold</b> command.
MGCP LAT threshold lwm	The <b>latency minimum threshold</b> parameter setting for the <b>mgcp quality-threshold</b> command.
MGCP maximum waiting delay	The setting for the <b>mgcp max-waiting-delay</b> command.
MGCP modem passthrough	Indicates whether a call agent will be involved in relaying modem data.
MGCP Network (IP/AAL2) Continuity Test timer	The setting for the <b>net-cont-test</b> option in the <b>mgcp timer</b> command.
MGCP packetization period	The <b>packetization period</b> parameter setting for the <b>mgcp codec</b> command.
MGCP PL threshold hwm	The <b>packet loss maximum threshold</b> parameter setting for the <b>mgcp quality-threshold</b> command.
MGCP PL threshold lwm	The <b>packet loss minimum threshold</b> parameter setting for the <b>mgcp quality-threshold</b> command.
MGCP playout mode	The jitter buffer packet size type and size.
MGCP quarantine mode	Indicates the protocol initiated for this session.
MGCP quarantine of persistent events	Indicates whether the persistent events will be handled by the quarantine buffer.
MGCP request retries	The setting for the <b>mgcp request retries</b> command.
MGCP request timeout	The setting for the <b>mgcp request timeout</b> command.
MGCP 'RTP stream loss' timer	The setting for the <b>receive-rtcp</b> option in the <b>mgcp timer</b> command.
MGCP restart delay	The setting for the <b>mgcp restart-delay</b> command.
MGCP sdp simple	Indicates whether the simple sdp protocol is being used.
MGCP send RSIP for SGCP	The setting for the <b>mgcp sgcp restart notify</b> command.
MGCP signaling precedence	The <b>rtp precedence</b> parameter setting for the <b>mgcp ip-tos</b> command.
MGCP supported packages	The packages supported in this session.
MGCP TSE payload	The settings for the <b>mgcp tse payload</b> command.
MGCP vad	The setting for the <b>mgcp vad</b> command.

**Table 19** *show mgcp Field Descriptions (continued)*

Field	Description
MGCP voip modem passthrough	Indicates the settings for mode, codec, and redundancy from the <b>mgcp modem passthrough mode</b> , <b>mgcp modem passthrough codec</b> , and <b>mgcp modem passthrough voip redundancy</b> commands.
MGCP voaal2 modem passthrough	Indicates the settings for mode, codec, and redundancy from the <b>mgcp modem passthrough mode</b> and <b>mgcp modem passthrough codec</b> commands.

The following is sample output for Voice over IP (VoIP) connections:

```
show mgcp connection
```

```
Endpoint  Call_ID(C) Conn_ID(I) (P)ort (M)ode (S)tate (C)odec (E)vent[SIFL] (R)esult[EA]
1. S0/DS1-0/1 C=103,23,24 I=0x8 P=16586,16634 M=3 S=4,4 C=5 E=2,0,0,2 R=0,0
2. S0/DS1-0/2 C=103,25,26 I=0x9 P=16634,16586 M=3 S=4,4 C=5 E=0,0,0,0 R=0,0
3. S0/DS1-0/3 C=101,15,16 I=0x4 P=16506,16544 M=3 S=4,4 C=5 E=2,0,0,2 R=0,0
4. S0/DS1-0/4 C=101,17,18 I=0x5 P=16544,16506 M=3 S=4,4 C=5 E=0,0,0,0 R=0,0
5. S0/DS1-0/5 C=102,19,20 I=0,6 P=16572,16600 M=3 S=4,4 C=5 E=2,0,0,2 R=0,0
6. S0/DS1-0/6 C=102,21,22 I=0x7 P=16600,16572 M=3 S=4,4 C=5 E=0,0,0,0 R=0,0
```

```
Total number of active calls 6
```

[Table 20](#) describes the significant fields shown in the display.

**Table 20** *show mgcp connection Field Descriptions*

Field	Description
Endpoint	The endpoint for each call shown in the digital endpoint naming convention of slot number (S0) and digital line (DS1-0) number (1).
Call_ID(C)	The MGCP call ID sent by the call agent, the internal Call Control Application Programming Interface (CCAPI) call ID for this endpoint, and the peer call legs CCAPI call ID. (CCAPI is an API that provides call control facilities to applications.)
Conn_ID(I)	The connection ID generated by the gateway and sent in the ACK message.
(P)ort	The ports used for this connection. The first port is the local UDP port. The second port is the remote UDP port.

**Table 20** *show mgcp connection Field Descriptions (continued)*

Field	Description
(M)ode	<p>The call mode, where:</p> <ul style="list-style-type: none"> <li>• 0—Indicates an invalid value for mode.</li> <li>• 1—Indicates that the gateway should send only packets.</li> <li>• 2—Indicates that the gateway should receive only packets.</li> <li>• 3—Indicates that the gateway can send and receive packets.</li> <li>• 4—Indicates that the gateway should neither send nor receive packets.</li> <li>• 5—Indicates that the gateway should place the circuit in loopback mode.</li> <li>• 6—Indicates that the gateway should place the circuit in test mode.</li> <li>• 7—Indicates that the gateway should use the circuit for network access for data.</li> <li>• 8—Indicates that the gateway should place the connection in network loopback mode.</li> <li>• 9—Indicates that the gateway should place the connection in network continuity test mode.</li> <li>• 10—Indicates that the gateway should place the connection in conference mode.</li> </ul> <p>All other values are used for internal debugging.</p>
(S)tate	The call state. The values are used for internal debugging purposes.
(C)odec	The codec identifier. The values are used for internal debugging purposes.
(E)vent [SIFL]	Used for internal debugging.
(R)esult [EA]	Used for internal debugging.

The following is sample output for VoIP and VoAAL2 statistics:

```
show mgcp statistics

UDP pkts rx 8, tx 9
Unrecognized rx pkts 0, MGCP message parsing errors 0
Duplicate MGCP ack tx 0, Invalid versions count 0
CreateConn rx 4, successful 0, failed 0
DeleteConn rx 2, successful 2, failed 0
DeleteConn tx 0, successful 0, failed 0
NotifyRequest rx 0, successful 4, failed 0
AuditConnection rx 0, successful 0, failed 0
AuditEndpoint rx 0, successful 0, failed 0
RestartInProgress tx 1, successful 1, failed 0
Notify tx 0, successful 0, failed 0
ACK tx 8, NACK tx 0
ACK rx 0, NACK rx 0
IP address based Call Agents statistics:
IP address 10.24.167.3, Total msg rx 8, successful 8, failed 0
```

Table 21 describes the significant fields shown in the display.

**Table 21** *show mgcp statistics Field Descriptions*

Field	Description
Unrecognized rx pkts	The number of unrecognized UDP packets received by the MGCP application.
MGCP message parsing errors	The number of MGCP messages received with parsing errors.
Duplicate MGCP ack tx	The number of duplicate MGCP acknowledgment messages transmitted to the call agents.
Invalid versions count	The number of MGCP messages received with an invalid MGCP version.
CreateConn rx	The number of Create Connection (CRCX) messages received by the gateway, the number that were successful, and the number that failed.
DeleteConn rx	The number of Delete Connection (DLCX) messages received by the gateway, the number that were successful, and the number that failed.
NotifyRequest rx	The number of Notify Request (RQNT) messages received by the gateway, the number that were successful, and the number that failed.
AuditConnection rx	The number of Audit Connection (AUCX) messages received by the gateway, the number that were successful, and the number that failed.
AuditEndpoint rx	The number of Audit Endpoint (AU EP) messages received by the gateway, the number that were successful, and the number that failed.
RestartInProgress tx	The number of Restart in Progress (RSIP) messages transmitted by the gateway, the number that were successful, and the number that failed.
Notify tx	The number of Notify (NTFY) messages transmitted by the gateway, the number that were successful, and the number that failed.
ACK tx, NACK tx	The number of Acknowledgment and Negative Acknowledgment messages transmitted by the gateway.

**Table 21** show mgcp statistics Field Descriptions (continued)

Field	Description
ACK rx, NACK rx	The number of Acknowledgment and Negative Acknowledgment messages received by the gateway.
IP address based Call Agents statistics: IP address, Total msg rx	IP address of the call agent, the total number of MGCP messages received from that call agent, the number of messages that were successful, and the number of messages that failed.

The following example shows how endpoints are configured:

```
show mgcp endpoint
```

```

      ENDPOINT-NAME      V-PORT  SIG-TYPE  ADMIN
ds1-0/1@sblab116      0:1     fxs-gs    up
ds1-0/2@sblab116      0:1     fxs-gs    up
ds1-0/3@sblab116      0:1     fxs-gs    up
ds1-0/4@sblab116      0:1     fxs-gs    up
ds1-0/5@sblab116      0:1     fxs-gs    up
ds1-0/6@sblab116      0:1     fxs-gs    up
ds1-0/7@sblab116      0:1     fxs-gs    up
ds1-0/8@sblab116      0:1     fxs-gs    up
ds1-0/9@sblab116      0:1     fxs-gs    up
ds1-0/10@sblab116     0:1     fxs-gs    up
ds1-0/11@sblab116     0:1     fxs-gs    up
ds1-0/12@sblab116     0:1     fxs-gs    up
ds1-0/13@sblab116     0:1     fxs-gs    up
ds1-0/14@sblab116     0:1     fxs-gs    up
ds1-0/15@sblab116     0:1     fxs-gs    up
ds1-0/16@sblab116     0:1     fxs-gs    up
ds1-0/17@sblab116     0:1     fxs-gs    up
ds1-0/18@sblab116     0:1     fxs-gs    up
ds1-0/19@sblab116     0:1     fxs-gs    up
ds1-0/20@sblab116     0:1     fxs-gs    up
ds1-0/21@sblab116     0:1     fxs-gs    up
ds1-0/22@sblab116     0:1     fxs-gs    up
ds1-0/23@sblab116     0:1     fxs-gs    up
ds1-0/24@sblab116     0:1     fxs-gs    up

```

Table 22 describes the significant fields shown in the display.

**Table 22** show mgcp endpoints Field Descriptions

Field	Description
ENDPOINT-NAME	The MGCP endpoint name.
V-PORT	The voice port number.
SIG-TYPE	The type of signaling and can be as follows: <ul style="list-style-type: none"> <li>e&amp;m-dly—E&amp;M delay dial.</li> <li>e&amp;m-wnk—E&amp;M wink start.</li> <li>fxs-gs—FXS, ground start.</li> </ul> <p><b>Note</b> If the signaling is PRI that is backhauled, then “none” displays in the field.</p>
ADMIN	The administrative state that can be up or down.

■ show mgcp

Related Commands	Command	Description
	mgcp	Starts the MGCP daemon.

# show voice port

To display configuration information about a specific voice port, use the **show voice port** command in EXEC mode.

## Analog Voice Ports

```
show voice port [slot/subunit/port | summary]
```

## Digital Voice Ports with T1 Packet Voice Trunk Network Modules

```
show voice port [slot/port:ds0-group | summary]
```

### Syntax Description

#### Analog Voice Ports

<i>slot/subunit/port</i>	(Optional) Displays information for the analog voice port you specify with the <i>slot/subunit/port</i> designation. <ul style="list-style-type: none"> <li><i>slot</i>—Specifies a router slot in which a voice network module (VNM) is installed. Valid entries are router slot numbers for the particular platform.</li> <li><i>subunit</i>—Specifies a voice interface card (VIC) in which the voice port is located. Valid entries are 0 and 1. (The VIC fits into the voice network module.)</li> <li><i>port</i>—Specifies an analog voice port number. Valid entries are 0 and 1.</li> </ul>
<b>summary</b>	(Optional) Displays a summary of all voice ports.

#### Digital Voice Ports

<i>slot/port:ds0-group</i>	(Optional) Displays information for the digital voice port you specify with the <i>slot/port:ds0-group</i> designation. <ul style="list-style-type: none"> <li><i>slot</i>—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.</li> <li><i>port</i>—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.)</li> <li><i>ds0-group</i>—Specifies a T1 or E1 logical port number. Valid entries are from 0 to 23 for T1 and from 0 to 30 for E1.</li> </ul>
<b>summary</b>	(Optional) Displays a summary of all voice ports.

### Command Modes

EXEC

**Command History**

Release	Modification
11.3(1) T	This command was introduced.
11.3(1)MA	Port-specific values for the Cisco MC3810 multiservice concentrator were added.
12.0(3)T	Port-specific values for the Cisco MC3810 multiservice concentrator were added.
12.0(5)XK	The <i>ds0-group</i> argument was added for the Cisco 2600 and Cisco 3600 series routers.
12.0(5)XE	Additional syntax was created for digital voice to allow specification of the DS0 group. This command applies to Voice over IP (VoIP) on the Cisco 7200 series routers.
12.0(7)T	The additions from Cisco IOS Release 12.0(5)XE were integrated into Cisco IOS Release 12.0(7)T.
12.0(7)XK	The <b>summary</b> keyword was added for the Cisco 2600 and Cisco 3600 series routers. The <i>ds0-group</i> argument was added for the Cisco MC3810 multiservice concentrator.
12.1(2)T	This command was integrated into the Cisco IOS Release 12.1(2)T release.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

**Usage Guidelines**

Use the **show voice port EXEC** command to display configuration and VIC-specific information about a port.

This command applies to Voice over IP (VoIP), Voice over Frame Relay (VoFR), and Voice over ATM (VoATM).

The **ds0-group** command automatically creates a logical voice port that is numbered as follows on the Cisco 2600 and Cisco 3600 series routers: slot/port:ds0-group-no. Although only one voice port is created for each group, applicable calls are routed to any channel in the group.

**Examples**

The following is sample output from the **show voice port** command for an E&M analog voice port on a Cisco 3600:

```
show voice port

E&M Slot is 1, Sub-unit is 0, Port is 0
Type of VoicePort is E&M
Operation State is unknown
Administrative State is unknown
The Interface Down Failure Cause is 0
Alias is NULL
Noise Regeneration is disabled
Non Linear Processing is disabled
Music On Hold Threshold is Set to 0 dBm
In Gain is Set to 0 dB
Out Attenuation is Set to 0 dB
Echo Cancellation is disabled
Echo Cancel Coverage is set to 16ms
```

```
Connection Mode is Normal
Connection Number is
Initial Time Out is set to 0 s
Interdigit Time Out is set to 0 s
Analog Info Follows:
Region Tone is set for northamerica
Currently processing none
Maintenance Mode Set to None (not in mtc mode)
Number of signaling protocol errors are 0

Voice card specific Info Follows:
Signal Type is wink-start
Operation Type is 2-wire
Impedance is set to 600r Ohm
E&M Type is unknown
Dial Type is dtmf
In Seizure is inactive
Out Seizure is inactive
Digit Duration Timing is set to 0 ms
InterDigit Duration Timing is set to 0 ms
Pulse Rate Timing is set to 0 pulses/second
InterDigit Pulse Duration Timing is set to 0 ms
Clear Wait Duration Timing is set to 0 ms
Wink Wait Duration Timing is set to 0 ms
Wink Duration Timing is set to 0 ms
Delay Start Timing is set to 0 ms
Delay Duration Timing is set to 0 ms
```

The following is sample output from the **show voice port** command for a Foreign Exchange Station (FXS) analog voice port on a Cisco 3600 series router:

```
show voice port

Foreign Exchange Station 1/0/0 Slot is 1, Sub-unit is 0, Port is 0
Type of VoicePort is FXS
Operation State is DORMANT
Administrative State is UP
The Interface Down Failure Cause is 0
Alias is NULL
Noise Regeneration is enabled
Non Linear Processing is enabled
Music On Hold Threshold is Set to 0 dBm
In Gain is Set to 0 dB
Out Attenuation is Set to 0 dB
Echo Cancellation is enabled
Echo Cancel Coverage is set to 16ms
Connection Mode is Normal
Connection Number is
Initial Time Out is set to 10 s
Interdigit Time Out is set to 10 s
Analog Info Follows:
Region Tone is set for northamerica
Currently processing none
Maintenance Mode Set to None (not in mtc mode)
Number of signaling protocol errors are 0
Voice card specific Info Follows:
Signal Type is loopStart
Ring Frequency is 25 Hz
Hook Status is On Hook
Ring Active Status is inactive
Ring Ground Status is inactive
Tip Ground Status is inactive
Digit Duration Timing is set to 100 ms
InterDigit Duration Timing is set to 100 ms
```

■ **show voice port**

Hook Flash Duration Timing is set to 600 ms

The following is sample output from the **show voice port** command for an E&M digital voice port on a Cisco 3600 series router:

```
show voice port

recEive and transMit Slot is 1, Sub-unit is 0, Port is 1
Type of VoicePort is E&M
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
Music On Hold Threshold is Set to -38 dBm
In Gain is Set to 0 dB
Out Attenuation is Set to 0 dB
Echo Cancellation is enabled
Echo Cancel Coverage is set to 8 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
Region Tone is set for US
```

Table 23 describes the significant fields shown in the display.

**Table 23** *show voice port Field Descriptions*

Field	Description
Administrative State	Administrative state of the voice port.
Alias	User-supplied alias for the voice port.
Analog interface A-D gain offset	Offset of the gain for analog-to-digital conversion.
Analog interface D-A gain offset	Offset of the gain for digital-to-analog conversion.
Clear Wait Duration Timing	Time of inactive seizure signal to declare call cleared.
Coder Type	Voice compression mode used.
Companding Type	Companding standard used to convert between analog and digital signals in PCM systems.
Connection Mode	Connection mode of the interface.
Connection Number	Full E.164 telephone number used to establish a connection with the trunk or PLAR mode.
Currently Processing	Type of call currently being processed: none, voice, or fax.
Delay Duration Timing	Maximum delay signal duration for delay dial signaling.
Delay Start Timing	Timing of generation of delayed start signal from detection of incoming seizure.
Description	Description of the voice port.
Dial Type	Out-dialing type of the voice port.
Digit Duration Timing	DTMF digit duration in milliseconds.
E&M Type	Type of E&M interface.
Echo Cancel Coverage	Echo cancel coverage for this port.

**Table 23** *show voice port Field Descriptions (continued)*

<b>Field</b>	<b>Description</b>
Echo Cancellation	Whether or not echo cancellation is enabled for this port.
Hook Flash Duration Timing	Maximum length of hook flash signal.
Hook Status	Hook status of the FXO/FXS interface.
Impedance	Configured terminating impedance for the E&M interface.
In Gain	Amount of gain inserted at the receiver side of the interface.
In Seizure	Incoming seizure state of the E&M interface.
Initial Time Out	Amount of time the system waits for an initial input digit from the caller.
InterDigit Duration Timing	DTMF interdigit duration, in milliseconds.
InterDigit Pulse Duration Timing	Pulse dialing interdigit timing, in milliseconds.
Interdigit Time Out	Amount of time the system waits for a subsequent input digit from the caller.
Maintenance Mode	Maintenance mode of the voice port.
Maximum Payout Delay	The amount of time before the Cisco MC3810 multiservice concentrator DSP starts to discard voice packets from the digital signal processor (DSP) buffer.
Music On Hold Threshold	Configured Music-On-Hold Threshold value for this interface.
Noise Regeneration	Whether or not background noise should be played to fill silent gaps if VAD is activated.
Nominal Payout Delay	The amount of time the Cisco MC3810 multiservice concentrator DSP waits before starting to play out the voice packets from the DSP buffer.
Non-Linear Processing	Whether or not nonlinear processing is enabled for this port.
Number of signaling protocol errors	Number of signaling protocol errors.
Operations State	Operation state of the port.
Operation Type	Operation of the E&M signal: two-wire or four-wire.
Out Attenuation	Amount of attenuation inserted at the transmit side of the interface.
Out Seizure	Outgoing seizure state of the E&M interface.
Port	Port number for this interface associated with the voice interface card.
Pulse Rate Timing	Pulse dialing rate, in pulses per second (pps).
Region Tone	Configured regional tone for this interface.
Ring Active Status	Ring active indication.
Ring Cadence	Configured ring cadence for this interface.
Ring Frequency	Configured ring frequency for this interface.
Ring Ground Status	Ring ground indication.
Ringing Time Out	Ringing timeout duration.

**Table 23** *show voice port Field Descriptions (continued)*

<b>Field</b>	<b>Description</b>
Signal Type	Type of signaling for a voice port: loop-start, ground-start, wink-start, immediate, and delay-dial.
Slot	Slot used in the voice interface card for this port.
Sub-unit	Subunit used in the voice interface card for this port.
Tip Ground Status	Tip ground indication.
Type of VoicePort	Type of voice port: FXO, FXS, or E&M.
The Interface Down Failure Cause	Text string describing why the interface is down,
Voice Activity Detection	Whether Voice Activity Detection is enabled or disabled.
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.
Wink Duration Timing	Maximum wink duration for wink start signaling.
Wink Wait Duration Timing	Maximum wink wait duration for wink start signaling.

# voice-port

To enter voice-port configuration mode, use the **voice-port** command in global configuration mode.

```
voice-port {slot-number/subunit-number/port} | {slot/port:ds0-group-no}
```

Syntax Description		
<i>slot-number</i>		Slot number in the Cisco router in which the VIC is installed. Valid entries are from 0 to 3, depending on the slot where it has been installed.
<i>subunit-number</i>		Subunit on the VIC in which the voice port is located. Valid entries are 0 or 1.
<i>port</i>		Voice port number. Valid entries are 0 or 1.
<i>slot</i>		The router location in which the voice port adapter is installed. Valid entries are from 0 to 3.
<i>port</i>		Indicates the voice interface card location. Valid entries are 0 or 3.
<i>ds0-group-no</i>		Indicates the defined DS0 group number. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.

**Defaults** No default behavior or values.

**Command Modes** Global configuration

Command History	Release	Modification
	11.3(1)T	This command was introduced.
	11.3(3)T	Support for Cisco 2600 series routers was added.
	12.0(3)T	Support for the Cisco AS5300 access server was added.
	12.0(7)T	Support for the Cisco AS5800 universal access server, the Cisco 7200 series router, and the Cisco 1750 router was added. Arguments for the Cisco 2600 and Cisco 3600 series router were added.
	12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series routers, 3600 series routers, and the Cisco Voice Gateway 200 (Cisco VG200).

**Usage Guidelines** Use the **voice-port** global configuration command to switch to voice-port configuration mode from global configuration mode. Use the **exit** command to exit voice-port configuration mode and return to global configuration mode.

---

**Examples**

The following example accesses voice-port configuration mode for port 0, located on subunit 0 on a voice interface card installed in slot 1 for the Cisco 3600 series:

```
voice-port 1/0/0
```

---

**Related Commands**

<b>Command</b>	<b>Description</b>
<b>dial-peer voice</b>	Enters dial-peer configuration mode and specifies the method of voice encapsulation.

---

# 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 agent for a Cisco IP telephony solution. The Cisco CallManager extends enterprise telephony features and functions to packet telephony network devices such as IP phones, media processing devices, VoIP gateways, and multimedia applications. The Cisco CallManager includes a suite of integrated voice applications that perform voice conferencing and manual attendant console functions.

**Cisco CallManager server**—The Cisco high-availability host platform on which Cisco CallManager software is preinstalled.

**cluster**—A set of Cisco CallManager servers that share the same MGCP voice gateway configuration database.

**codec**—A software algorithm that compresses and 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 VoIP transmission, there are two types of dial peers: POTS and VoIP. You use the **dial-peer** voice command to define dial peers and to switch to dial-peer configuration mode.

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

**DTMF**—dual tone multifrequency. Frequencies in a touch tone telephone in which one high and one low tone is assigned to each touch tone button on a phone. DTMF digits can be detected by the voice ports after 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 means of the RTCP APP extension mechanism.

**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 audio path of the trunk to convey information about calls. The signaling 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 call answer and disconnect supervision.

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

**FXS**—Foreign Exchange Station. 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 PSTN, you must use an intermediary device called a gateway. A VoIP gateway allows users of IP phones and PC-based softphones to exchange calls with users of POTS phones on the PSTN. The gateway translates between the signals used on the PSTN and the IP packets used to transmit data over 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 enables external control and management of data communications equipment operating at the edge of multiservice packet networks (known as media gateways) by means of software programs. Such software programs are referred to as “call agents” or “media gateway controllers.”

**PBX**—A 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 a 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 dialing an access digit, such as 9.

**POTS**—plain old telephone service. A 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.

**protocol**—A set of rules or conventions that governs 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 aspects as framing, error handling, transparency, and line control. Ethernet is an example of a LAN protocol.

**PSTN**—Public Switched Telephone Network. A generic term that refers to the wide variety of telephone networks and telephony services in existence worldwide.

**SGCP**—Simple Gateway Control Protocol. A protocol that controls VoIP gateways through an external element (called a call agent).

**subnet mask**—A 32-bit address mask used in IP addressing 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. T1 uses time-division multiplexing (TDM) to divide the available bandwidth into 24 64-kbps time slots. 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 referred to as T-1.

**TCP**—A connection-oriented transport layer protocol that supports 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 the signals between widely separated endpoints.

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

**VoIP**—Voice over IP. Enables users to conduct voice communications over a data network using the IP. VoIP can consolidate voice and data traffic for more efficient use of bandwidth, reducing toll charges and providing 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.

