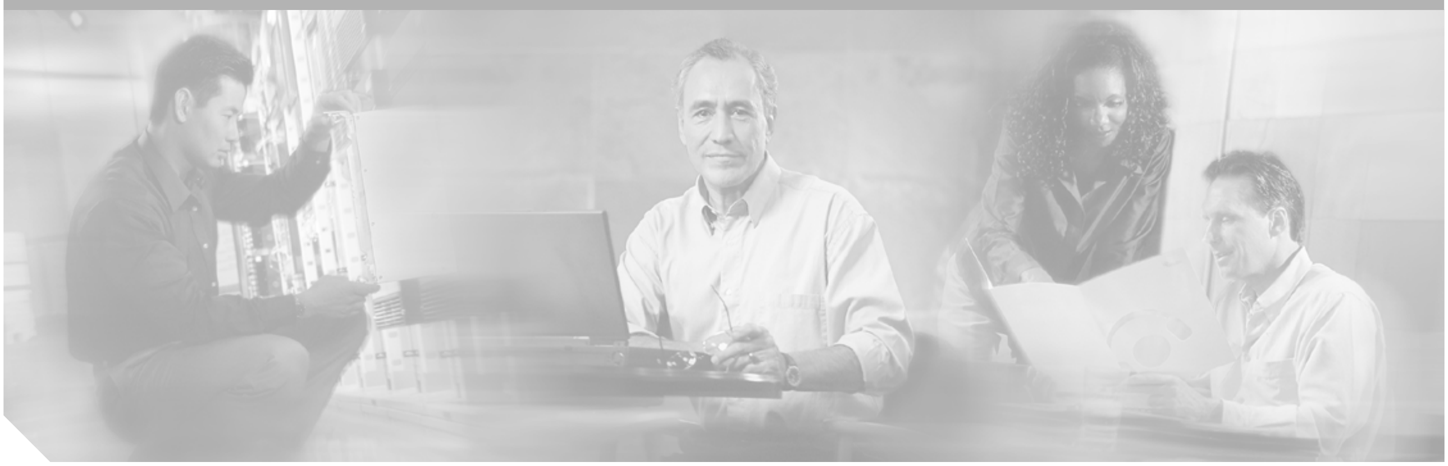

Cisco IOS MGCP and Related Protocols Configuration Guide

Cisco IOS Release 12.4T
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Corporate Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
http://www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

CONTENTS

Preface

The preface contains the following sections:

Audience

This guide is written for developers and network administrators who are installing, configuring, and maintaining Media Gateway Control Protocol (MGCP) applications. This guide describes the configuration tasks related to the Cisco IOS software implementation of these applications.

Developers using this guide may be interested in joining the Cisco Developer Support Program. This program was created to provide you with a consistent level of support that you can depend on while leveraging Cisco interfaces in your development projects.

Tips A signed developer-support agreement is required to participate in this program. For more information and to access this agreement, go to http://www.cisco.com/go/developersupport or contact developer-support@cisco.com.

Structure of This Guide

Title	Contents
"Preface"	Contains overview and support information.
"MGCP Features Roadmap"	Lists MGCP features (Cisco IOS Release 12.3 and later) and the location of feature documentation. Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support.
"Overview of MGCP and Related Protocols"	Describes MGCP concepts.
"Basic MGCP Configuration"	Describes basic MGCP configuration procedures.
"Configuring MGCP 1.0 Including NCS 1.0 and TGCP 1.0 Profiles"	Describes the concepts and configuration procedures for the MGCP 1.0 including Network-based Call Signaling (NCS) 1.0 profiles feature in Cisco IOS Release 12.3.
"Configuring MGCP Basic CLASS and Operator Services"	Describes the concepts and configuration procedures for the MGCP Basic Custom Local Area Subscriber Services (CLASS) and Operator Services (BCOS) features in Cisco IOS Release 12.3.
"Configuring NAS Package for MGCP"	Describes the concepts and configuration procedures for the Network Access Server Package for MGCP feature in Cisco IOS Release 12.3.
"Configuring SGCP RSIP and AUEP Enhancements"	Describes the concepts and configuration procedures for the Simple Gateway Control Protocol (SGCP) Restart In Progress (RSIP) and Audit Endpoints (AUEP) Enhancements feature in Cisco IOS Release 12.3.
"Configuring MGCP Gateway Support for the mgcp bind Command"	Describes the concepts and configuration procedures for the MGCP Gateway Support for the mgcp bind command feature in Cisco IOS Release 12.3.
"Configuring MGCP CAS MD Package"	Describes the concepts and configuration procedures for the MGCP CAS MD Package feature in Cisco IOS Release 12.4(4)T.
"Appendix A: Configuring MGCP CAS PBX and AAL2 PVC"	Describes the concepts and the configuration procedures for the MGCP CAS PBX and AAL2 PVC feature in Cisco IOS Release 12.3.
Glossary	Definitions of the terms in this document.

Additional References for MGCP and SGCP

The following sections provide additional references for MGCP and SGCP:

- Related Features and Technologies, page xi
- Related Documents, page xi
- Supported Platforms, page xii
- Supported Standards, MIBs, and RFCs, page xii

Related Features and Technologies

- Voice over IP (VoIP) (all platforms)
- VoAAL2-PVC (Cisco MC3810 only)

Related Documents

Related Topic	Document Title
Cisco documentation	Bookmark the Cisco documentation home page for easy access to all Cisco technical documentation at http://www.cisco.com/univercd/home/home.htm
Cisco IOS application, configuration, and command guides	*Cisco IOS MGCP and Related Protocols*, Release 12.3 (this document)
	Cisco IOS Fax Services over IP, Release 12.3
	Cisco IOS Release 12.3 Configuration Guides and Command References
	Cisco IOS Trunk Connections and Conditioning Features, Release 12.3
Cisco IOS feature documentation	*G.Clear, GSMFR, and G.726 Codecs and Modem and Fax Passthrough*, Release 12.2(11)T
	MGCP Voice on Cisco AS5850 Universal Gateway, Release 12.2(11)T
Product documentation	Cisco 2600 Series Routers
	Cisco 3600 Series Routers
	Cisco AS5350
	Cisco AS5400
	Cisco AS5850
	Cisco CVA120 Series Cable Voice Adapters
	Cisco MC3810 Series Multiservice Access Concentrators
	Cisco uBR925 Cable Access Router

Supported Platforms

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at http://www.cisco.com/go/fn. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

Supported Standards, MIBs, and RFCs

Standards

Standards	Title
IETF MGCP draft	*IETF MGCP draft version 0.1* dated November 9, 1998
IETF Informational Internet-Draft	*NAS Packages for MGCP*
PacketCable	*Network-Based Call Signaling (NCS) Protocol Specification* PKT-SP-EC-MGCP-I02-991201, December 1, 1999.
	PSTN Gateway Call Signaling Protocol Specification (TGCP) PKT-SP-TGCP-D02-991028, December 1, 1999.
RTP and RTCP 1889 and 1890	RFC 1889: *RTP: A Transport Protocol for Real-Time Applications* and RFC 1890: *RTP Profile for Audio and Video Conferences with Minimal Control*

MIBs

MIBs	MIBs Link
• Dial Control MIB • RTP MIB • XGCP-MIB	To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

RFCs

RFC	Title
RFC 1889	*RTP: A Transport Protocol for Real-Time Applications*
RFC 1890	*RTP Profile for Audio and Video Conferences with Minimal Control*

RFC	Title
RFC 2327	*SDP: Session Description Protocol*
RFC 2705	*Media Gateway Control Protocol (MGCP) version 1.0*
RFC 2833	*RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*
RFC 3064	*MGCP CAS Packages*

Obtaining Documentation

Cisco documentation and additional literature are available on Cisco.com. Cisco also provides several ways to obtain technical assistance and other technical resources. These sections explain how to obtain technical information from Cisco Systems.

Cisco.com

You can access the most current Cisco documentation at this URL:

http://www.cisco.com/univercd/home/home.htm

You can access the Cisco website at this URL:

http://www.cisco.com

You can access international Cisco websites at this URL:

http://www.cisco.com/public/countries_languages.shtml

Ordering Documentation

You can find instructions for ordering documentation at this URL:

http://www.cisco.com/univercd/cc/td/doc/es_inpck/pdi.htm

You can order Cisco documentation in these ways:

- Registered Cisco.com users (Cisco direct customers) can order Cisco product documentation from the Ordering tool:

  http://www.cisco.com/en/US/partner/ordering/index.shtml

- Nonregistered Cisco.com users can order documentation through a local account representative by calling Cisco Systems Corporate Headquarters (California, USA) at 408 526-7208 or, elsewhere in North America, by calling 800 553-NETS (6387).

Documentation Feedback

You can send comments about technical documentation to bug-doc@cisco.com.

You can submit comments by using the response card (if present) behind the front cover of your document or by writing to the following address:

Cisco Systems
Attn: Customer Document Ordering
170 West Tasman Drive
San Jose, CA 95134-9883

We appreciate your comments.

Obtaining Technical Assistance

For all customers, partners, resellers, and distributors who hold valid Cisco service contracts, Cisco Technical Support provides 24-hour-a-day, award-winning technical assistance. The Cisco Technical Support Website on Cisco.com features extensive online support resources. In addition, Cisco Technical Assistance Center (TAC) engineers provide telephone support. If you do not hold a valid Cisco service contract, contact your reseller.

Cisco Technical Support Website

The Cisco Technical Support Website provides online documents and tools for troubleshooting and resolving technical issues with Cisco products and technologies. The website is available 24 hours a day, 365 days a year at this URL:

http://www.cisco.com/techsupport

Access to all tools on the Cisco Technical Support Website requires a Cisco.com user ID and password. If you have a valid service contract but do not have a user ID or password, you can register at this URL:

http://tools.cisco.com/RPF/register/register.do

Submitting a Service Request

Using the online TAC Service Request Tool is the fastest way to open S3 and S4 service requests. (S3 and S4 service requests are those in which your network is minimally impaired or for which you require product information.) After you describe your situation, the TAC Service Request Tool automatically provides recommended solutions. If your issue is not resolved using the recommended resources, your service request will be assigned to a Cisco TAC engineer. The TAC Service Request Tool is located at this URL:

http://www.cisco.com/techsupport/servicerequest

For S1 or S2 service requests or if you do not have Internet access, contact the Cisco TAC by telephone. (S1 or S2 service requests are those in which your production network is down or severely degraded.) Cisco TAC engineers are assigned immediately to S1 and S2 service requests to help keep your business operations running smoothly.

To open a service request by telephone, use one of the following numbers:

Asia-Pacific: +61 2 8446 7411 (Australia: 1 800 805 227)
EMEA: +32 2 704 55 55
USA: 1 800 553 2447

For a complete list of Cisco TAC contacts, go to this URL:

http://www.cisco.com/techsupport/contacts

Definitions of Service Request Severity

To ensure that all service requests are reported in a standard format, Cisco has established severity definitions.

Severity 1 (S1)—Your network is "down," or there is a critical impact to your business operations. You and Cisco will commit all necessary resources around the clock to resolve the situation.

Severity 2 (S2)—Operation of an existing network is severely degraded, or significant aspects of your business operation are negatively affected by inadequate performance of Cisco products. You and Cisco will commit full-time resources during normal business hours to resolve the situation.

Severity 3 (S3)—Operational performance of your network is impaired, but most business operations remain functional. You and Cisco will commit resources during normal business hours to restore service to satisfactory levels.

Severity 4 (S4)—You require information or assistance with Cisco product capabilities, installation, or configuration. There is little or no effect on your business operations.

Obtaining Additional Publications and Information

Information about Cisco products, technologies, and network solutions is available from various online and printed sources.

- Cisco Marketplace provides a variety of Cisco books, reference guides, and logo merchandise. Visit Cisco Marketplace, the company store, at this URL:

  http://www.cisco.com/go/marketplace/

- The Cisco *Product Catalog* describes the networking products offered by Cisco Systems, as well as ordering and customer support services. Access the Cisco Product Catalog at this URL:

  http://cisco.com/univercd/cc/td/doc/pcat/

- *Cisco Press* publishes a wide range of general networking, training and certification titles. Both new and experienced users will benefit from these publications. For current Cisco Press titles and other information, go to Cisco Press at this URL:

  http://www.ciscopress.com

- *Packet* magazine is the Cisco Systems technical user magazine for maximizing Internet and networking investments. Each quarter, Packet delivers coverage of the latest industry trends, technology breakthroughs, and Cisco products and solutions, as well as network deployment and troubleshooting tips, configuration examples, customer case studies, certification and training information, and links to scores of in-depth online resources. You can access Packet magazine at this URL:

  http://www.cisco.com/packet

- *iQ Magazine* is the quarterly publication from Cisco Systems designed to help growing companies learn how they can use technology to increase revenue, streamline their business, and expand services. The publication identifies the challenges facing these companies and the technologies to help solve them, using real-world case studies and business strategies to help readers make sound technology investment decisions. You can access iQ Magazine at this URL:

  http://www.cisco.com/go/iqmagazine

- *Internet Protocol Journal* is a quarterly journal published by Cisco Systems for engineering professionals involved in designing, developing, and operating public and private internets and intranets. You can access the Internet Protocol Journal at this URL:

  http://www.cisco.com/ipj

- World-class networking training is available from Cisco. You can view current offerings at this URL:

  http://www.cisco.com/en/US/learning/index.html

MGCP Features Roadmap

This chapter contains a list of MGCP features (Cisco IOS Release 12.3 and later) and the location of feature documentation.

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at http://www.cisco.com/go/fn. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

Release	Features in That and Later Releases	Feature Description	Feature Documentation
12.4(4)T	MGCP CAS MD Package	Adds support for Feature Group D (FGD) Exchange Access North American (EANA) protocol signaling and automatic number identification (ANI) and dialed number identification service (DNIS) digits.	Configuring MGCP CAS MD Package chapter of this guide.
12.3(4)XQ	MGCP-Controlled Backhaul of BRI Signaling in Conjunction with Cisco CallManager	Provides MGCP service to remote-office media gateways that connect by means of ISDN BRI trunks to a centralized Cisco CallManager media gateway controller for the purpose of call processing. This feature is supported on the Cisco 1700 series.	*MGCP-Controlled Backhaul of BRI Signaling in Conjunction with Cisco CallManager*
12.2(13)T	MGCP Gateway Support for the mgcp bind Command	Unifies all prior SGCP/MGCP support and adds CAS and AAL2 PVC features.	Configuring MGCP Gateway Support for the mgcp bind Command chapter of this guide
12.2(11)T	SGCP RSIP and AUEP Enhancements	Adds enhancements to SGCP for disconnected RSIP and audit endpoints requested by call agents.	Configuring SGCP RSIP and AUEP Enhancements chapter of this guide
12.2(2)T	MGCP Basic (CLASS) and Operation Services	Provides xGCP support for three-way calling on residential and trunking gateways.	Configuring MGCP Basic CLASS and Operator Services chapter of this guide

Release	Features in That and Later Releases	Feature Description	Feature Documentation
12.2(2)XB	Network Access Server (NAS) Package for MGCP	Adds support for the MGCP NAS package on universal gateways. Data calls can be terminated on a trunking media gateway that is serving as a NAS. Trunks on the NAS are controlled and managed by a call agent supporting MGCP for both voice and data calls. The call agent must support the MGCP NAS package.	Configuring NAS Package for MGCP chapter of this guide
12.2(2)XA	MGCP 1.0 Including NCS 1.0 and TGCP 1.0 Profiles	Implements MGCP 1.0, NCS 1.0, and TGCP 1.0 support in existing MGCP stacks.	Configuring MGCP 1.0 Including NCS 1.0 and TGCP 1.0 Profiles chapter of this guide
12.1(5)XM	MGCP CAS PBX and AAL2 PVC	Provides CAS connectivity to traditional PBXs.	Appendix A: Configuring MGCP CAS PBX and AAL2 PVC of this guide

Overview of MGCP and Related Protocols

This chapter provides overview information on Media Gateway Control Protocol (MGCP) and related protocols.

Note For more information about Cisco IOS voice features, see the entire Cisco IOS Voice Configuration Library—including library preface and glossary, feature documents, and troubleshooting information—at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vcl.htm.

Contents

Prerequisites for Configuring MGCP and Related Protocols

- Configure IP routing.
- Configure voice ports.
- Configure VoIP.
- Configure the call agent. (For information on setting up call agents, see the documentation that accompanies the call agents used in your network configuration.)

Information About MGCP and Related Protocols

MGCP is an extension of the earlier version of the protocol Simple Gateway Control Protocol (SGCP) and supports SGCP functionality in addition to several enhancements. Systems using SGCP can easily migrate to MGCP, and MGCP commands are available to enable SGCP capabilities.

An MGCP gateway handles translation between audio signals and the packet network. Gateways interact with a call agent (CA)—also called a media gateway controller (MGC)—that performs signal and call processing on gateway calls. In the MGCP configurations that Cisco IOS supports, a gateway can be a Cisco router, access server, or cable modem, and the CA is a server from a third-party vendor.

Configuration commands for MGCP define the path between the call agent and the gateway, the type of gateway, and the type of calls handled by the gateway.

MGCP uses endpoints and connections to construct a call. Endpoints are sources of or destinations for data, and can be physical or logical locations in a device. Connections can be point-to-point or multipoint.

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

Package Types

A call connection involves a series of events and signals—such as off-hook status, a ringing signal, or a signal to play an announcement—that are specific to the type of endpoint involved in the call.

MGCP groups these events and signals into packages. A trunk package, for example, is a group of events and signals relevant to a trunking gateway; an announcement package is a group of events and signals relevant to an announcement server. MGCP supports the following seven package types:

- Trunk
- Line
- Dual-tone multifrequency (DTMF)
- Generic media
- Real-Time Transport Protocol (RTP)
- Announcement server
- Script

The trunk package and line package are supported by default on certain types of gateways. Although configuring a gateway with additional endpoint package information is optional, you may want to specify packages for your endpoints to add to or override the defaults.

Protocol Benefits

MGCP provides the following benefits:

- Alternative dial tone for VoIP environments—Deregulation in the telecommunications industry gives competitive local-exchange carriers (CLECs) opportunities to provide toll bypass from the incumbent local-exchange carriers (ILECs) by means of VoIP. MGCP enables a VoIP system to control call setup and teardown and Custom Local Area Subscriber Services (CLASS) features for less sophisticated gateways.
- Simplified configuration for static VoIP network dial peers—When you use MGCP as the call agent in a VoIP environment, you need not configure static VoIP network dial peers. The MGCP call agent provides functions similar to VoIP-network dial peers.

  Note Plain old telephone service (POTS) dial peer configuration is still required.

- Migration paths—Systems using earlier versions of the protocol can migrate easily to MGCP.
- Varied network needs supported for the following:
  - Interexchange carriers (IXCs) who have no legacy time-division multiplexing (TDM) equipment in their networks and want to deploy a fully featured network that offers both long-distance services to corporate customers and connectivity to local exchange carriers or other IXCs with traditional TDM equipment.

- IXCs who have TDM equipment in their networks and want to relieve network congestion using data technologies to carry voice traffic or to cap the growth of TDM ports. In these situations, the packet network provides basic switched trunking without services or features.
- Competitive competitive local-exchange carriers (CLECs) who want to provide residential and enhanced services.
- Dial-access customers who want enhanced Signaling System 7 (SS7) access capabilities and increased performance, reliability, scalability, and economy.

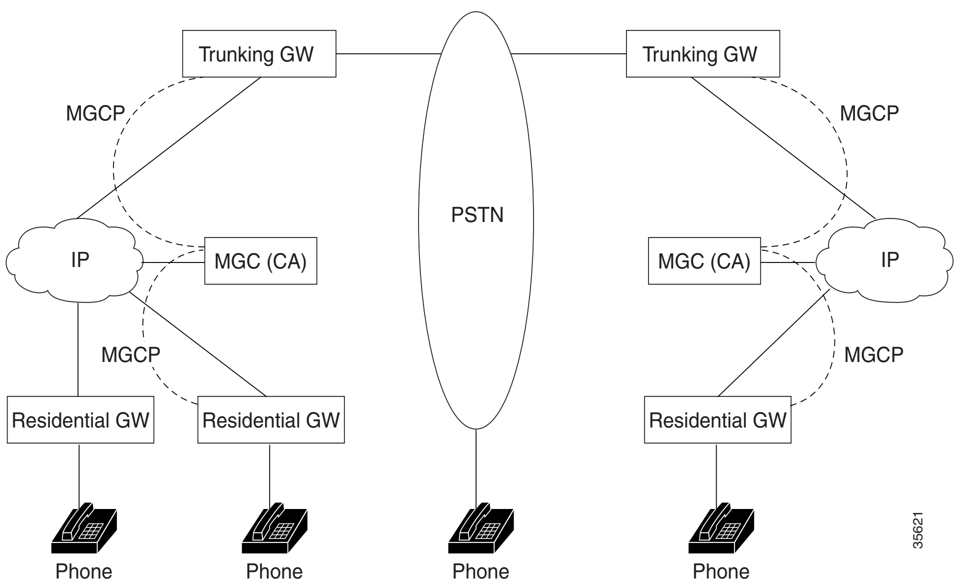
Supported Gateways

MGCP supports both residential and trunking gateways.

Residential Gateway

A residential gateway (RGW) provides an interface between analog (RJ-11) calls from a telephone and the VoIP network. Examples of RGWs include cable modems and Cisco 2600 series routers. Figure 1 shows an RGW configuration.

Figure 1 Residential and Trunking Gateways



RGW functionality supports analog POTS calls for both SGCP and MGCP on the Cisco 2600 series routers and Cisco uBR924 cable access router as shown in Table 1.

Table 1 RGW Functionality

Functionality	Platform	
	Cisco 2600 Series	**Cisco uBR924**
Call waiting	Yes	Yes
Default call-agent address specifiable for each foreign exchange station (FXS) port	—	Yes

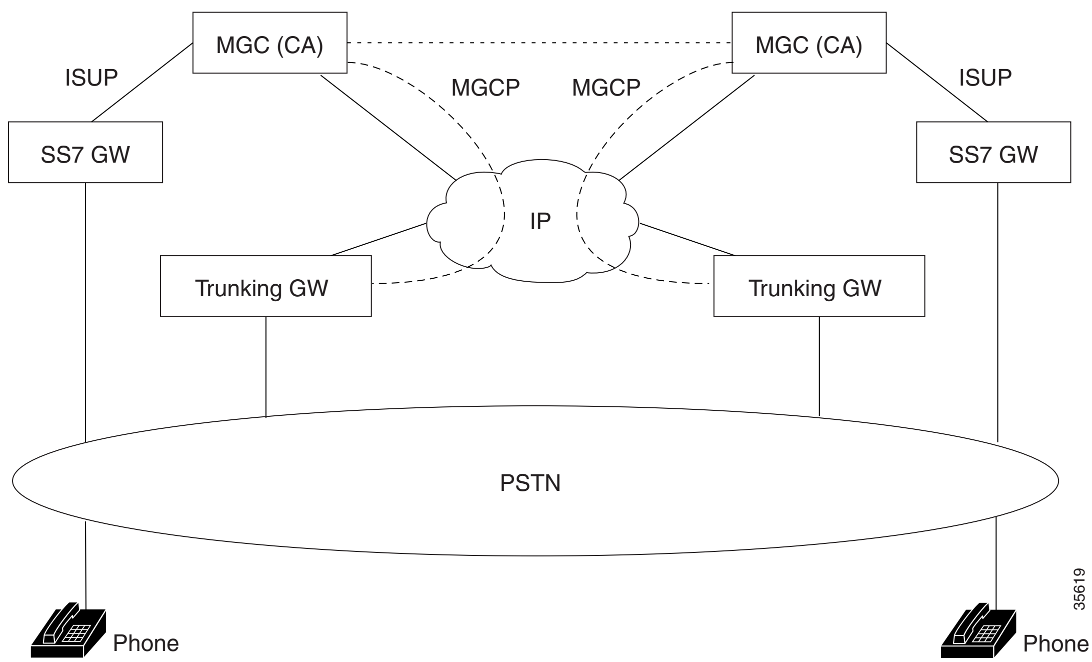
Table 1 RGW Functionality (continued)

Functionality	Platform	
	Cisco 2600 Series	Cisco uBR924
Distinctive ringing	—	Yes
Fax and modem calls	Yes	Yes
On-hook caller identification (ID)	—	Yes
Ring splash	—	Yes
Stutter dial tone	Yes	Yes

Trunking Gateway

A trunking gateway (TGW) provides an interface between PSTN trunks and a VoIP network. A trunk can be a DS0, T1, or E1 line. Examples of TGWs include access servers and routers. Figure 2 shows a TGW configuration.

Figure 2 Trunking Gateways



TGW functionality supports SGCP and MGCP as shown in Table 2.

Table 2 TGW Functionality

Functionality	Platform	
	Cisco AS5300	Cisco 3660
911 outgoing calls on T1 lines	Yes[1]	
Fax and modem calls	Yes	Yes

Table 2 TGW Functionality (continued)

Functionality	Platform	
	Cisco AS5300	**Cisco 3660**
PRI/ISDN signaling (calls are backhauled to the call agent)	Yes	
SS7	Yes	Yes
T1 and E1 interfaces	Yes	Yes

1. Server must have SGCP 1.1+ protocol for Feature Group D Operator Services (FGD-OS)

Additional References

The following sections provide references related to <<technology area>>.

Related Documents

Related Topic	Document Title
Cisco IOS configuration examples	Cisco Systems Technologies website at http://cisco.com/en/US/tech/index.html. Select a technology category and subsequent hierarchy of subcategories. Click **Technical Documentation > Configuration Examples**.
Cisco IOS debug command reference	*Cisco IOS Debug Command Reference, Release 12.3T* at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123tcr/123dbr/index.htm
Cisco IOS troubleshooting information	*Cisco IOS Voice Troubleshooting and Monitoring Guide* at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vvfax_c/voipt_c/index.htm
Cisco IOS voice command reference	*Cisco IOS Voice Command Reference, Release 12.3T* at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123tcr/123tvr/index.htm
Cisco IOS Voice Configuration Library preface and glossary	Cisco IOS Voice Configuration Library at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vcl.htm

MIBs

MIBs	MIBs Link
•	To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

Technical Assistance

Description	Link
Technical Assistance Center (TAC) home page, containing 30,000 pages of searchable technical content, including links to products, technologies, solutions, technical tips, and tools. Registered Cisco.com users can log in from this page to access even more content.	http://www.cisco.com/public/support/tac/home.shtml

Basic MGCP Configuration

This chapter provides basic configuration information for Media Gateway Control Protocol (MGCP) and related protocols.

Note For more information about related Cisco IOS voice features, see the following:

- "Overview of MGCP and Related Protocols" on page 3
- Entire Cisco IOS Voice Configuration Library—including library preface and glossary, other feature documents, and troubleshooting documentation—at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vcl.htm.

Contents

Prerequisites for Configuring MGCP and Related Protocols

Prerequisites are described in the "Prerequisites for Configuring MGCP and Related Protocols" section on page 3.

How to Configure MGCP and Related Protocols

To configure MGCP, perform the tasks in the following sections. Each task is identified as either optional or required.

- Do at least one of the following tasks, depending on your network configuration (required):
  - Configuring a TGW for MGCP, page 10
  - Configuring a TGW for SGCP, page 11
  - Configuring an RGW, page 12
- Verifying the TGW or RGW Configuration, page 13 (required)
- Blocking New Calls, page 13 (optional)

Note RGWs are configured only with MGCP.

Configuring a TGW for MGCP

To configure a trunking gateway (TGW) for MGCP, use the following commands beginning in global configuration mode:

SUMMARY STEPS

1. **mgcp**
2. **mgcp call-agent** [*ipaddr*|*hostname*] [*port*] **service-type mgcp**
3. **controller t1** *number*
4. **ds0-group** *channel-number* **timeslots** *range* **type none service mgcp**
5. **exit**
6. **mgcp restart-delay** *value*
7. **mgcp package-capability** {**s-package** | **dtmf-package** | **gm-package** | **lcs-package** | **rtp-package** | **trunk-package** | **script-package**}
8. **mgcp default-package** {**as-package** | **dtmf-package** | **gm-package** | **rtp-package** | **trunk-package**}
9. **mgcp dtmf-relay** {**codec** | **low-bit-rate**} **mode** {**cisco** | **out-of-band**}
10. **mgcp modem passthru** {**cisco** | **ca**}
11. **gcp sdp simple**
12. **exit**

DETAILED STEPS

	Command	Purpose
Step 1	Router(config)# **mgcp**	Initiates the MGCP application.
Step 2	Router(config)# **mgcp call-agent** [*ipaddr*|*hostname*] [*port*] **service-type mgcp**	Specifies the call agent's IP address or domain name, the port, and gateway control service type.
Step 3	Router(config)# **controller t1** *number*	Specifies the channel number of the T1 trunk to be used for analog calls and enters controller configuration mode.
Step 4	Router(config-controller)# **ds0-group** *channel-number* **timeslots** *range* **type none service mgcp**	Configures the channelized T1 time slots to accept the analog calls.
Step 5	Router(config-controller)# **exit**	Exits the current mode.
Step 6	Router(config)# **mgcp restart-delay** *value*	(Optional) Specifies the delay value sent in the RSIP graceful teardown method, in seconds. Range is from 0 to 600. Default is 0.
Step 7	Router(config)# **mgcp package-capability** {**trunk-package** | **dtmf-package** | **gm-package** | **lcs-package** | r**tp-package** | **as-package**}	(Optional) Specifies the event packages that are supported on the trunking gateway. Default is **trunk-package**.
Step 8	Router(config)# **mgcp default-package** {**as-package** | **dtmf-package** | **gm-package** | **rtp-package** | **trunk-package**}	(Optional) Specifies the default event package. Overrides the **mgcp package-capability** default package.
Step 9	Router(config)# **mgcp dtmf-relay** {**codec** | **low-bit-rate**} **mode** {**cisco** | **out-of-band**}	(Optional) Used for relaying digits through the IP network. Default is **no mgcp dtmf-relay** for all codecs.
Step 10	Router(config)# **mgcp modem passthru** {**cisco** | **ca**}	(Optional) Configures the gateway for modem and fax data.
Step 11	Router(config)# **mgcp sdp simple**	(Optional) Specifies use of a subset of the session description protocol (SDP). Some call agents require this subset to send data through the network. Default is **no mgcp sdp simple**.
Step 12	Router(config)# **exit**	Exits the current mode.

Configuring a TGW for SGCP

To configure a trunking gateway (TGW) for Simple Gateway Control Protocol (SGCP), use the following commands beginning in global configuration mode:

SUMMARY STEPS

1. **mgcp**
2. **mgcp call-agent** [*ipaddr* | *hostname*] [*port*] **service-type sgcp**
3. **controller t1** *number*
4. **ds0-group** *channel-number* **timeslots** *range* **type** {**none** | **fgdos**} [*tone_type*] [*addr_info*] **service** {**sgcp** | **voice**}
5. **exit**

DETAILED STEPS

	Command	Purpose
Step 1	`Router(config)# mgcp`	Initiates the MGCP application.
Step 2	`Router(config)# mgcp call-agent [ipaddr | hostname] [port] service-type sgcp`	Specifies the call agent's IP address or domain name, the port, and gateway control service type.
Step 3	`Router(config)# controller t1 number`	Specifies the channel number of the T1 trunk to be used for analog calls and enters controller configuration mode.
Step 4	`Router(config-controller)# ds0-group channel-number timeslots range type {none | fgdos} [tone_type] [addr_info] service {sgcp | voice}`	Configures the channelized T1 time slots to accept the analog calls. For type **none**, use **service sgcp**. For type **fgdos**, use **service voice**.
Step 5	`Router(config-controller)# exit`	Exits the current mode.

Configuring an RGW

To configure a residential gateway (RGW), use the following commands beginning in global configuration mode:

SUMMARY STEPS

1. **mgcp**
2. **mgcp call-agent** [*ipaddr* | *hostname*] [*port*] **service-type mgcp**
3. **dial-peer voice** *number* **pots**
4. **application MGCPAPP**
5. **exit**
6. **mgcp package-capability** {**line-package** | **dtmf-package** | **gm-package** | **rtp-package**}
7. **mgcp default-package** [**line-package** | **dtmf-package** | **gm-package**]
8. **exit**

DETAILED STEPS

	Command	Purpose
Step 1	`Router(config)# mgcp`	Initiates the MGCP application. **Note** RGWs are configured only with MGCP.
Step 2	`Router(config)# mgcp call-agent [ipaddr | hostname] [port] service-type mgcp`	Specifies the call-agent IP address or domain name, port, and gateway control service type.
Step 3	`Router(config)# dial-peer voice number pots`	Sets up the dial peer for a voice port.
Step 4	`Router(config-dial-peer)# application MGCPAPP`	Selects the MGCP application to run on the voice port.
Step 5	`Router(config-dial-peer)# exit`	Exits the current mode.
Step 6	`Router(config)# mgcp package-capability {line-package | dtmf-package | gm-package | rtp-package}`	(Optional) Specifies event packages that are supported on the residential gateway. Default is **line-package**.

	Command	Purpose
Step 7	`Router(config)# mgcp default-package [line-package	dtmf-package	gm-package]`	(Optional) Specifies the default event package. Overrides the **mgcp package-capability** command.
Step 8	`Router(config)# exit`	Exits the current mode.

Verifying the TGW or RGW Configuration

To verify configuration, use the following command.

SUMMARY STEPS

1. **show running-configuration**

DETAILED STEPS

	Command	Purpose
Step 1	`Router(config)# show running-configuration`	Displays the current configuration settings.

Blocking New Calls

You can block all new MGCP calls to the router (Step 1) and terminate all existing active calls (Step 2), which means that an active call is not terminated until the caller hangs up.

To block all new calls, use the following commands in global configuration mode:

SUMMARY STEPS

1. **mgcp block-newcalls**
2. **no mgcp block-newcalls**

DETAILED STEPS

	Command	Purpose
Step 1	`Router(config)# mgcp block-newcalls`	Prevents the gateway from accepting new calls.
Step 2	`Router(config)# no mgcp block-newcalls`	Restarts normal MGCP call operation.

Configuration Examples for MGCP and Related Protocols

This section provides configuration examples for each of the supported platforms:

- Configuring a Cisco AS5300 as a TGW with MGCP Example, page 14
- Configuring a Cisco AS5300 as a TGW with SGCP Example, page 15
- Configuring a Cisco 3660 as a TGW with MGCP Example, page 16

- Configuring a Cisco uBR924 as an RGW Example, page 18
- Configuring a Cisco 2620 as an RGW Example, page 19

Configuring a Cisco AS5300 as a TGW with MGCP Example

The following example illustrates a configuration only for MGCP calls. FGD-OS calls are not supported.

```
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname A
!
resource-pool disable
!
ip subnet-zero
ip ftp username smith
ip host B 209.165.200.225
ip host C 209.165.200.226
ip domain-name cisco.com
ip name-server 209.165.202.129
!
mgcp
mgcp request timeout 10000
mgcp call-agent 192.168.10.10 2302
mgcp restart-delay 5
mgcp package-capability gm-package
mgcp package-capability dtmf-package
mgcp package-capability trunk-package
mgcp package-capability rtp-package
mgcp package-capability as-package
mgcp package-capability mf-package
mgcp package-capability script-package
mgcp default-package trunk-package
mta receive maximum-recipients 0
!
controller T1 0
 framing esf
 clock source line primary
 linecode b8zs
 ds0-group 0 timeslots 1-24 type none service mgcp
!
controller T1 1
 framing esf
 clock source line secondary 1
 linecode b8zs
 ds0-group 0 timeslots 1-24 type none service mgcp
!
controller T1 2
 framing esf
 linecode b8zs
 ds0-group 0 timeslots 1-24 type none service mgcp
!
controller T1 3
 framing esf
 linecode b8zs
 ds0-group 0 timeslots 1-24 type none service mgcp
!
voice-port 0:0
!
```

```
voice-port 1:0
!
voice-port 2:0
!
voice-port 3:0
!
interface Ethernet0
 ip address 192.168.10.9 255.255.255.0
 no ip directed-broadcast
!
interface FastEthernet0
 ip address 172.22.91.73 255.255.255.0
 no ip directed-broadcast
 shutdown
 duplex auto
 speed auto
!
no ip classless
ip route 0.0.0.0 0.0.0.0 172.22.91.1
ip route 209.165.200.225 255.255.255.255 192.168.0.1
no ip http server
!
line con 0
 exec-timeout 0 0
 transport input none
line aux 0
line vty 0 4
 login
!
end
```

Configuring a Cisco AS5300 as a TGW with SGCP Example

The following example illustrates a configuration that supports MGCP and FGD-OS calls:

```
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname A
!
resource-pool disable
!
ip subnet-zero
ip ftp username smith
ip host B 209.165.200.225
ip host C 209.165.200.226
ip domain-name cisco.com
ip name-server 209.165.202.129
!
mgcp
mgcp request timeout 10000
mgcp call-agent 192.168.10.10 2302 sgcp
mta receive maximum-recipients 0
!
controller T1 0
 framing esf
 clock source line primary
 linecode b8zs
 ds0-group 0 timeslots 1-24 type none service mgcp
!
```

```
controller T1 1
 framing esf
 clock source line secondary 1
 linecode b8zs
 ds0-group 0 timeslots 1-24 type fgd-os mf dnis-ani service voice
!
controller T1 2
 framing esf
 linecode b8zs
 ds0-group 0 timeslots 1-24 type none service mgcp
!
controller T1 3
 framing esf
 linecode b8zs
 ds0-group 0 timeslots 1-24 type none service mgcp
!
!voice-port 0:0
!
voice-port 1:0
!
voice-port 2:0
!
voice-port 3:0
!
interface Ethernet0
 ip address 192.168.10.9 255.255.255.0
 no ip directed-broadcast
!
interface FastEthernet0
 ip address 172.22.91.73 255.255.255.0
 no ip directed-broadcast
 shutdown
 duplex auto
 speed auto
!
no ip classless
ip route 0.0.0.0 0.0.0.0 172.22.91.1
ip route 209.165.200.225 255.255.255.255 192.168.0.1
no ip http server
!
line con 0
 exec-timeout 0 0
 transport input none
line aux 0
line vty 0 4
 login
!
end
```

Configuring a Cisco 3660 as a TGW with MGCP Example

The following example illustrates a platform that does not support FGD-OS calls.

```
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname A
!
memory-size iomem 40
voice-card 1
```

```
!
ip subnet-zero
!
mgcp 4000
mgcp call-agent 209.165.202.129 4000
mgcp package-capability gm-package
mgcp package-capability dtmf-package
mgcp package-capability rtp-package
mgcp package-capability as-package
isdn voice-call-failure 0
cns event-service server
!
controller T1 1/0
 framing esf
 clock source internal
 ds0-group 1 timeslots 1-24 type none service mgcp
!
controller T1 1/1
 framing esf
 clock source internal
 ds0-group 1 timeslots 1-24 type none service mgcp
!
voice-port 1/0:1
!
voice-port 1/1:1
!
interface FastEthernet0/0
 ip address 209.165.202.140 255.255.255.0
 no ip directed-broadcast
 load-interval 30
 duplex auto
 speed auto
!
interface FastEthernet0/1
 no ip address
 no ip directed-broadcast
 no ip mroute-cache
 load-interval 30
 shutdown
 duplex auto
 speed auto
!
ip default-gateway 209.165.202.130
ip classless
ip route 209.165.200.225 255.255.255.255 FastEthernet0/0
no ip http server
!
snmp-server engineID local 00000009020000107BD8CD80
snmp-server community public RO
!
line con 0
 exec-timeout 0 0
 transport input none
line aux 0
line vty 0 4
 login
!
end
```

Configuring a Cisco uBR924 as an RGW Example

The following example illustrates a platform that does not support FGD-OS calls.

```
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname A
!
logging buffered 200000 debugging
!
clock timezone - -8
ip subnet-zero
no ip routing
no ip domain-lookup
ip host A 192.168.147.91
ip host C 209.165.200.224
ip host D 209.165.200.225
!
mgcp
mgcp call-agent 192.168.10.10 2490
mgcp package-capability gm-package
mgcp package-capability dtmf-package
mgcp package-capability line-package
mgcp default-package line-package
!
voice-port 0
 input gain -3
!
voice-port 1
 input gain -3
!
dial-peer voice 1 pots
 application MGCPAPP
 port 1
!
dial-peer voice 2 pots
 application MGCPAPP
 port 0
!
interface Ethernet0
 ip address 192.168.147.91 255.255.255.0
 no ip directed-broadcast
 no ip route-cache
 no ip mroute-cache
!
interface cable-modem0
 ip address negotiated
 no ip directed-broadcast
 no ip route-cache
 no ip mroute-cache
 cable-modem downstream saved channel 459000000 20
cable-modem downstream saved channel 699000000 19 2
 cable-modem mac-timer t2 100000
 no cable-modem compliant bridge
 bridge-group 59
 bridge-group 59 spanning-disabled
!
ip default-gateway 10.1.1.1
ip classless
```

```
no ip http server
!
line con 0
 exec-timeout 0 0
 transport input none
line vty 0 4
 login
!
end
```

Configuring a Cisco 2620 as an RGW Example

The following example illustrates a platform that does not support FGD-OS calls.

```
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname D
!
memory-size iomem 10
ip subnet-zero
!
mgcp
mgcp call-agent 172.20.5.20
mgcp package-capability gm-package
mgcp package-capability dtmf-package
mgcp package-capability line-package
mgcp package-capability rtp-package
mgcp default-package line-package
cns event-service server
!
voice-port 1/0/0
!
voice-port 1/0/1
!
dial-peer voice 1 pots
 application MGCPAPP
 port 1/0/0
!
dial-peer voice 2 pots
 application MGCPAPP
 port 1/0/1
!
interface Ethernet0/0
 no ip address
 no ip directed-broadcast
 shutdown
!
interface Serial0/0
 no ip address
 no ip directed-broadcast
 no ip mroute-cache
 shutdown
 no fair-queue
!
interface Ethernet0/1
 ip address 172.20.5.25 255.255.255.0
 no ip directed-broadcast
!
interface Serial0/1
```

```
 no ip address
 no ip directed-broadcast
 shutdown
!
ip default-gateway 209.165.202.130
ip classless
ip route 209.165.200.225 255.255.255.224 Ethernet0/1
no ip http server
!
line con 0
 exec-timeout 0 0
 transport input none
line aux 0
line vty 0 4
 login
!
end
```

Tip

- See the "Additional References for MGCP and SGCP" section on page xi for related documents, standards, and MIBs.
- See the "Glossary" for definitions of terms in this guide.

Configuring MGCP 1.0 Including NCS 1.0 and TGCP 1.0 Profiles

This chapter provides configuration information on configuring the MGCP 1.0 Including Network-based Call Signaling (NCS) 1.0 and Trunking Gateway Control Protocol (TGCP) 1.0 Profiles feature. The feature implements MGCP 1.0, NCS 1.0, and TGCP 1.0 support in existing MGCP stacks.

Feature benefits include the following:

- MGCP 1.0 provides flexible interoperability with a wide variety of call agents, thus enabling a wide range of solutions.
- MGCP 1.0 contains many improvements over its previous release.
- NCS 1.0 and TGCP 1.0 allow participation in packet cable solutions.
- The ability to interoperate with H.323 and Session Initiation Protocol (SIP) control agents allows leverage of the feature sets available in the different protocols and provides the ability to migrate smoothly from one protocol to another.

Note For more information about this and related Cisco IOS voice features, see the following:

- "Overview of MGCP and Related Protocols" on page 3
- Entire Cisco IOS Voice Configuration Library—including library preface and glossary, other feature documents, and troubleshooting documentation—at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vcl.htm.

Feature History for MGCP 1.0 Including NCS 1.0 and TGCP 1.0 Profiles

Release	Modification
12.2(2)XA	This feature was introduced on the following platforms: Cisco CVA122, Cisco uBR924, and Cisco AS5300.
12.2(2)XA1	This feature was implemented on the following platforms: Cisco CVA122, Cisco uBR925, and Cisco AS5300
12.2(2)XB	This feature was implemented on the following platforms: Cisco AS5350 and Cisco AS5400.

12.2(4)T	This feature was implemented on the following platforms: Cisco CVA122, Cisco CVA122E, Cisco uBR925, Cisco 2600 series, Cisco 2650, Cisco 3660, and Cisco MC3810. AAL2 PVC support was introduced for MGCP 1.0 on the Cisco MC3810. Certain gateway features were integrated into MGCP 1.0. **Note** The Cisco AS5300 is not supported in this release.
12.2(8)T	The voice-port (MGCP profile) command was changed to port (MGCP profile) for all platforms supported in this release. **Note** The Cisco AS5300 is not supported in this release.
12.2(13)T	The **fax** keyword was added to the **mgcp playout** command.

Contents

Prerequisites for Configuring MGCP 1.0 Including NCS 1.0 and TGCP 1.0 Profiles

Prerequisites are described in the “Prerequisites for Configuring MGCP and Related Protocols” section on page 3. In addition, the following apply:

- Ensure that the minimum software requirements are met.Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at http://www.cisco.com/go/fn.
- Configure Voice over ATM AAL2 PVC (optional step that applies to Cisco MC3810 only). Refer to Cisco IOS Release 12.3 Configuration Guides and Command References. The router that is intending to use the VoAAL2 features must have hardware support for VoAAL2.
- Set up the cable modems, if any. See the documentation for the cable product as listed in the “Preface”.

Note IP addresses and host names in these examples are fictitious.

Information About MGCP 1.0 Including NCS 1.0 and TGCP 1.0 Profiles

This feature implements the following MGCP protocols on supported Cisco media gateways:

- MGCP 1.0 (RFC 2705)
- Network-based Call Signaling (NCS) 1.0, the MGCP 1.0 profile for residential gateways (RGWs)
- Trunking Gateway Control Protocol (TGCP) 1.0, the MGCP 1.0 profile for trunking gateways (TGWs)
- VoIP—Includes signaling methods under VoIP.
- AAL2 PVC—Includes signaling methods under ATM adaptation layer 2 (AAL2) permanent virtual circuit (PVC).
- Basic/Extended RGW—Includes a collection of residential gateway features supporting channel-associated signaling (CAS). Digital CAS (recEive and transMit, or E&M) interfaces and analog (Foreign Exchange Office [FXO], Foreign Exchange Station [FXS], and E&M) interfaces are supported on platforms with the appropriate voice hardware.
- ISUP—Supports ISDN user part signaling for SS7 trunks.
- FGD-OS—Supports Feature Group D Operator Services signaling over T1 or E1 trunks.
- Incoming CAS—Supports digital CAS interfaces for digital incoming multifrequency tones (MF) CAS wink-start trunks in which an operator at an Operator Services Console can initiate the Operator Interrupt and Busy Line Verify (OI and BLV) functions.
- CAS PBX—Includes CAS private branch exchange (PBX) trunks, digit maps, CAS events, and quarantine buffer software. These features are supported on digital CAS interfaces.

MGCP1.0 is a protocol for the control of VoIP calls by external call-control elements known as media gateway controllers (MGCs) or call agents (CAs). It is described in the informational RFC 2705, published by the Internet Society.

PacketCable is an industry-wide initiative for developing interoperability standards for multimedia services over cable facilities using packet technology. PacketCable developed the NCS and TGCP protocols, which contain extensions and modifications to MGCP while preserving basic MGCP architecture and constructs. NCS is designed for use with analog, single-line user equipment on residential gateways, while TGCP is intended for use in VoIP-to-PSTN trunking gateways in a cable environment. To meet European cable requirements and equipment characteristics, the EuroPacketCable working group has adapted PacketCable standards under the name *IP Cablecom*.

MGCP Model

MGCP bases its call control and intelligence in centralized *call agents*, also called media gateway controllers. The call agents issue commands to simple, low-cost endpoints, which are housed in media gateways (MGs), and the call agents also receive event reports from the gateways. MGCP messages between call agents and media gateways are sent with Internet Protocol over User Datagram Protocol (IP/UDP).

The MGCP 1.0 Including NCS 1.0 and TGCP 1.0 Profiles feature provides protocols for RGWs and TGWs, which sit at the border of the packet network to provide an interface between traditional, circuit-based voice services and the packet network. Residential gateways offer a small number of analog line interfaces, while trunking gateways generally manage a large number of digital trunk circuits.

Two basic MGCP constructs are *endpoints* and *connections*. An endpoint is a source or sink for call data (RTP/IP) that is flowing through the gateway. A common type of endpoint is found at the physical interface between the POTS (plain old telephone service) or Public Switched Telephone Network (PSTN) service and the gateway; this type of endpoint might be an analog voice port or a digital DS0 group. There are other types of endpoints as well, and some are logical rather than physical. An endpoint is identified by a two-part endpoint name that contains the name of the entity on which it exists (for example, an access server or router) and the local name by which it is known (for example, a port identifier).

A connection is a temporary allocation of resources that enables a call to be completed. One or more connections is necessary to complete a call. Connections have names that identify them with the call to which they belong. Connections can be one-to-one or multipoint. Calls and connections are initiated, modified, and deleted on instructions from call agents.

Call agents manage call flow through standard MGCP *commands* that are sent to the endpoints under their control. The commands are delivered in standard ASCII text, and may contain session descriptions transmitted in Session Description Protocol (SDP), a text-based protocol. These messages are sent over IP/UDP.

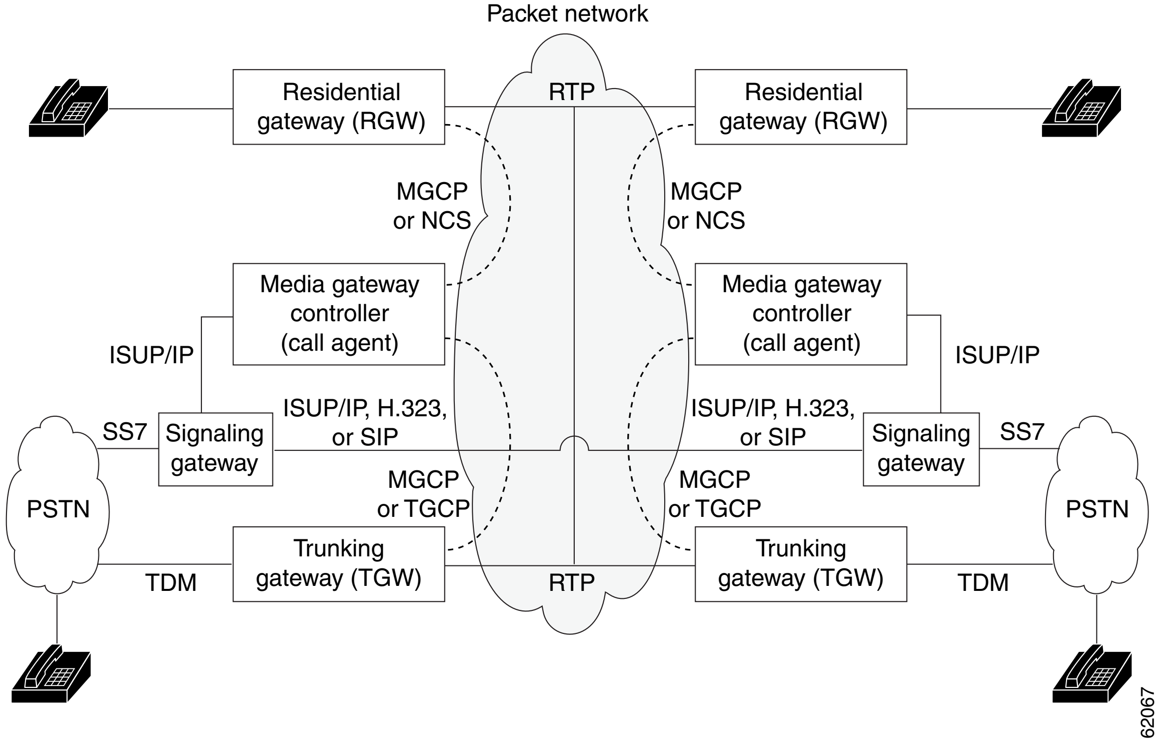
Call agents keep track of endpoint and connection status through the gateway's reporting of standard *events* that are detected from endpoints and connections. Call agents also direct gateways to apply certain standard *signals* when a POTS or PSTN connection expects them. For example, when someone picks up a telephone handset, an off-hook event is detected on an endpoint on the residential gateway to which the telephone is connected. The gateway reports the event to a call agent, which orders the gateway to apply the dial-tone signal to the endpoint reporting the off-hook event. The person picking up the handset hears dial tone.

Related events and signals are grouped into standard *packages* that apply to particular types of endpoints. For instance, the off-hook event is found in the line package, which is associated with analog-line endpoints, which in turn are associated with residential gateways. Standard events, signals, and packages are defined in the NCS, TGCP, and MGCP standards and RFCs listed in the "Preface."

Figure 3 on page 25 shows a hypothetical MGCP network with both residential and trunking gateways. The residential gateway has telephone sets connected to the gateway's FXS voice ports. MGCP or NCS over IP/UDP is used for call control and reporting to the call agent, while Real-Time Transport Protocol (RTP) is used to transmit the actual voice data.

Figure 3 on page 25 also shows two trunking gateways with T1 (or E1) connections to the PSTN. Incoming time-division multiplexing (TDM) data is sent through the gateway into the packet network using RTP. MGCP or TGCP over IP/UDP is used for call control and reporting to the call agent. Signaling System 7 (SS7) data travels a different route, however, bypassing the trunking gateway entirely in favor of a specialized signaling gateway, where the signaling data is transformed to ISUP/IP format and relayed to the call agent. Communication between two signaling gateways in the same packet network can be done with Integrated Services Digital Networks User Part over Internet Protocol (ISUP/IP), H.323, or Session Initiation Protocol (SIP).

Figure 3 MGCP Network Model



How to Configure MGCP 1.0 Including NCS 1.0 and TGCP 1.0 Profiles

The three tasks listed below configure the MGCP 1.0 Including NCS 1.0 and TGCP 1.0 Profiles feature on a media gateway. The first task names the voice ports or DS1 groups that are serving as MGCP endpoints. This task also associates the ports with an MGCP service type or application and starts the MGCP daemon.

The last two tasks allow you to configure MGCP parameters to meet your requirements. Each MGCP parameter is either a global parameter or a profile-related parameter. When you configure a global MGCP parameter value, it applies to all the MGCP endpoints on the gateway. When you configure a profile-related MGCP parameter value, it applies only to the endpoints associated with the MGCP profile that you are configuring at that moment (an MGCP profile is a user-defined subset of all the MGCP endpoints on the gateway). There is also a predefined MGCP profile named *default* that you can use to configure profile-related parameters for endpoints that do not belong to a user-defined MGCP profile.

See the following sections for configuration tasks for the MGCP 1.0 including NCS 1.0 and TGCP 1.0 Profiles feature. Each task in the list is identified as either required or optional:

- Identifying Endpoints and Configuring the MGCP Application, page 26 (required)
- Configuring Global MGCP Parameters, page 31 (optional)
- Configuring an MGCP Profile and Profile-Related MGCP Parameters, page 35 (optional)

Identifying Endpoints and Configuring the MGCP Application

This task is required. Voice ports or DS0 groups that are acting as MGCP endpoints must be identified and associated with the MGCP application. The commands to identify MGCP endpoints depend on the type of endpoint that you are configuring.

To identify endpoints and configure the MGCP application, use the commands in the appropriate table, beginning in global configuration mode:

- Analog CAS and POTS Lines, page 26
- Digital CAS Trunks, page 26
- ISUP Signaling Trunks, page 29
- FGD-OS Trunks, page 29
- Digital VoATM with AAL2 PVC, page 30

Analog CAS and POTS Lines

To identify endpoints and configure the MGCP application for use with analog CAS and POTS lines, use these commands, beginning in global configuration mode:

SUMMARY STEPS

1. **dial-peer voice** *tag* **pots**
2. **application mgcpapp**
3. **port** *port-number*
4. **exit**
5. **mgcp** [*gw-port*]

DETAILED STEPS

	Command	Purpose
Step 1	`Router(config)# dial-peer voice tag pots`	Enters dial-peer configuration mode and specifies the method of voice encapsulation.
Step 2	`Router(config-dial-peer)# application mgcpapp`	Enables the MGCP application on this dial peer.
Step 3	`Router(config-dial-peer)# port port-number`	Associates a dial peer with a specific voice port.
Step 4	`Router(config-dial-peer)# exit`	Exits the current mode.
Step 5	`Router(config)# mgcp [gw-port]`	Initiates the MGCP daemon. The optional argument is the UDP port over which the gateway receives messages from the call agent (the gateway MGCP port number). Default is 2427.

Digital CAS Trunks

To identify endpoints and configure the MGCP application for use with digital CAS trunks, use these commands, beginning in global configuration mode:

SUMMARY STEPS

1. **controller** {**t1** | **e1**} *cntlr-number*
2. **mode cas**
3. **framing** {**sf** | **esf**} (T1 lines) or **framing** {**crc4** | **no-crc4**} [**australia**] (E1 lines)
4. **linecode** {**ami** | **b8zs**} (T1 lines only) or **linecode** {**ami** | **hdb3**} (E1 lines only)
5. **ds0-group** *channel-number* **timeslots** *range* **type** *type*
6. **exit**
7. **voice-port** *slot/port:ds0-group-no* (Cisco 2600 and Cisco 3600 series) or **voice-port** *slot:ds0-group-no* (Cisco MC3810)
8. **dial-type** {**dtmf** | **mf** | **pulse**}
9. **exit**
10. **dial peer voice** *tag* **pots**
11. **application mgcpapp**
12. **port** *port-number*
13. **exit**
14. **mgcp** [*gw-port*]

DETAILED STEPS

	Command	Purpose
Step 1	`Router(config)# controller {t1 | e1} cntlr-number`	Configures a T1 or E1 controller and enters controller configuration mode for the digital CAS port.
Step 2	`Router(config-controller)# mode cas`	(Required for Cisco MC3810 only) Configures the T1 or E1 controller to support CAS mode.

Step 3	**T1 Lines**
	`Router(config-controller)# framing {sf | esf}`
	E1 Lines
	`Router(config-controller)# framing {crc4 | no-crc4} [australia]`
		Selects frame type for T1 or E1 line.
		T1 default is **sf**. E1 default is **crc4**.

Step 4	**T1 Lines**
	`Router(config-controller)# linecode {ami | b8zs}`
	E1 Lines
	`Router(config-controller)# linecode {ami | hdb3}`
		Specifies the line encoding to use.
		T1 default is **ami**. E1 default is **hdb3**.

Step 5	`Router(config-controller)# ds0-group channel-number timeslots range type type`	Specifies the DS0 time slots that make up a logical voice port on a T1 or E1 controller and specifies the signaling type by which the router connects to the PBX or PSTN. Use command-line interface (CLI) help (enter **?** after **type**) for valid signaling types.
Step 6	`Router(config-controller)# exit`	Exits the current mode.

	Command	Purpose
Step 7	**Cisco 2600 and Cisco 3600 Series** `Router(config)#` **`voice-port`** *`slot/port:ds0-group-no`* **Cisco MC3810** `Router(config)#` **`voice-port`** *`slot:ds0-group-no`*	Enters voice-port configuration mode.
Step 8	`Router(config-voiceport)#` **`dial-type`** `{`**`dtmf`** `|` **`mf`** `|` **`pulse`**`}`	(Required for MF trunks) Specifies the type of out-dialing for voice port interfaces. Default is **dtmf**.
Step 9	`Router(config-voiceport)#` **`exit`**	Exits the current mode.
Step 10	`Router(config)#` **`dial peer voice`** *`tag`* **`pots`**	Enters dial-peer configuration mode and specifies the method of voice encapsulation.
Step 11	`Router(config-dial-peer)#` **`application mgcpapp`**	Enables the MGCP application on this dial peer.
Step 12	`Router(config-dial-peer)#` **`port`** *`port-number`*	Associates a dial peer with a specific voice port.
Step 13	`Router(config-dial-peer)#` **`exit`**	Exits the current mode.
Step 14	`Router(config)#` **`mgcp`** `[`*`gw-port`*`]`	Initiates the MGCP daemon. The optional port-number argument is the UDP port over which the gateway receives messages from the call agent (the gateway MGCP port number). Default is 2427.

ISUP Signaling Trunks

To identify endpoints and configure the MGCP application for use with Integrated Services Digital Network Upper Part (ISUP) signaling trunks, use these commands, beginning in global configuration mode:

SUMMARY STEPS

1. **controller** {**t1** | **e1**} *cntlr-number*
2. **ds0-group** *channel-number* **timeslots** *range* **type none service mgcp**
3. **exit**
4. **mgcp** [*gw-port*]

DETAILED STEPS

	Command	Purpose
Step 1	`Router(config)#` **`controller`** `{`**`t1`** `|` **`e1`**`}` *`cntlr-number`*	Configures a T1 or E1 controller and enters controller configuration mode for the ISUP trunk port.
Step 2	`Router(config-controller)#` **`ds0-group`** *`channel-number`* **`timeslots`** *`range`* **`type none service mgcp`**	Specifies the DS0 time slots that make up a logical voice port on a T1 or E1 controller and specifies the signaling type by which the router connects to the PBX or PSTN. Specify the **type none** and **service mgcp** options to identify this voice port as an MGCP endpoint.
Step 3	`Router(config-controller)#` **`exit`**	Exits the current mode.
Step 4	`Router(config)#` **`mgcp`** `[`*`gw-port`*`]`	Initiates the MGCP daemon. The optional port number argument allows you to specify the UDP port over which the gateway receives messages from the call agent (the gateway MGCP port number). Default UDP port number for gateways is 2427.

FGD-OS Trunks

To identify endpoints and configure the MGCP application for use with Feature Group D Operator Services (FGD-OS) signaling over T1 or E1 trunks, use these commands, beginning in global configuration mode:

SUMMARY STEPS

1. **controller {t1** | **e1**} *cntlr-number*
2. **ds0-group** *channel-number* **timeslots** *range* **type fgd-os service mgcp**
3. **exit**
4. **mgcp** [*gw-port*]

DETAILED STEPS

	Command	Purpose
Step 1	`Router(config)#` **`controller`** `{`**`t1`** `|` **`e1`**`}` *`cntlr-number`*	Configures a T1 or E1 controller and enters controller configuration mode for the FGD-OS trunk port.
Step 2	`Router(config-controller)#` **`ds0-group`** *`channel-number`* **`timeslots`** *`range`* **`type fgd-os service mgcp`**	Specifies the DS0 time slots that make up a logical voice port on a T1 or E1 controller and specifies the signaling type by which the router connects to the PBX or PSTN. Specify the **type fgd-os** option for FGD-OS signaling, and the **service mgcp** option to identify this voice port as an MGCP endpoint.
Step 3	`Router(config-controller)#` **`exit`**	Exits the current mode.
Step 4	`Router(config)#` **`mgcp`** `[`*`gw-port`*`]`	Initiates the MGCP daemon. The optional argument is the UDP port over which the gateway receives messages from the call agent (the gateway MGCP port number). Default is 2427.

Digital VoATM with AAL2 PVC

To identify endpoints and configure the MGCP application for use with digital Voice over Asynchronous Transfer Mode (VoATM) with ATM Adaptation Layer 2 (AAL2) Permanent Virtual Circuit (PVC), use these commands, beginning in global configuration mode:

SUMMARY STEPS

1. **controller** {**t1** | **e1**} *cntlr-number*
2. **mode atm**
3. **framing** {**sf** | **esf**} (T1 lines) or **framing** {**crc4** | **no-crc4**} [**australia**] (E1 lines)
4. **linecode** {**ami** | **b8zs**} (T1 lines) or **linecode** {**ami** | **hdb3**} (E1 lines)
5. **exit**
6. **dial peer voice** *tag* **pots**
7. **application mgcpapp**
8. **port** *port-number*
9. **exit**
10. **mgcp** [*gw-port*]

DETAILED STEPS

	Command	Purpose
Step 1	`Router(config)#` **`controller`** `{`**`t1`** `|` **`e1`**`}` *`cntlr-number`*	Enters dial-peer configuration mode and specifies the method of voice encapsulation.
Step 2	`Router(config-controller)#` **`mode atm`**	Specifies that the controller supports ATM encapsulation and create ATM interface 0. When the controller is set to ATM mode, the following takes place: • Controller framing is automatically set to Extended Superframe (ESF). • The line code is automatically set to B8ZS.

Step 3	**T1 Lines**
	`Router(config-controller)#` **`framing`** `{`**`sf`** `|` **`esf`**`}`
	E1 Lines
	`Router(config-controller)#` **`framing`** `{`**`crc4`** `|` **`no-crc4`**`} [`**`australia`**`]`
		Selects frame type for T1 or E1 line.
		T1 default is **sf**. E1 default is **crc4**.

Step 4	**T1 Lines**
	`Router(config-controller)#` **`linecode`** `{`**`ami`** `|` **`b8zs`**`}`
	E1 Lines
	`Router(config-controller)#` **`linecode`** `{`**`ami`** `|` **`hdb3`**`}`
		Specifies the line encoding to use.
		T1 default is **ami**. E1 default is **hdb3**.

Step 5	`Router(config-controller)#` **`exit`**	Exits the current mode.
Step 6	`Router(config)#` **`dial peer voice`** *`tag`* **`pots`**	Enters dial-peer configuration mode and specifies the method of voice encapsulation.
Step 7	`Router(config-dial-peer)#` **`application mgcpapp`**	Enables the MGCP application on this dial peer.
Step 8	`Router(config-dial-peer)#` **`port`** *`port-number`*	Associates a dial peer with a specific voice port.
Step 9	`Router(config-dial-peer)#` **`exit`**	Exits the current mode.
Step 10	`Router(config)#` **`mgcp`** `[`*`gw-port`*`]`	Initiates the MGCP daemon. The optional argument is the UDP port over which the gateway receives messages from the call agent (the gateway MGCP port number). Default is 2427.

Configuring Global MGCP Parameters

This optional task configures global MGCP parameters on the gateway so that you can set these values to conform to the requirements of the call agent, trunks, or lines that are being used with this gateway. The global parameter values that you configure are associated with every MGCP endpoint that you have identified on this gateway.

In addition to the global MGCP parameters, there are other MGCP parameters that apply only to MGCP profiles on the gateway. For configuration of profile-related parameters, see the "Configuring an MGCP Profile and Profile-Related MGCP Parameters" section on page 35.

Note The only parameter that is common to both profile and global configurations is the call-agent parameter, which is configured with the **call-agent** command for MGCP profile configuration and with the **mgcp call-agent** command for the global configuration. These commands are mutually exclusive; whichever command you configure first blocks configuration of the other. For example, if the MGCP profile **call-agent** command is configured on an endpoint, then you are not allowed to configure the global **mgcp call-agent** command.

To configure global MGCP parameters, complete these steps as needed, beginning in global configuration mode:

SUMMARY STEPS

1. **mgcp call-agent** {*dns-name* | *ip-address*} [*port*] [**service-type** *type*] [**version** *protocol-version*]
2. **mgcp behavior** {**auep** | **signal**} **v0.1**
3. **mgcp sdp simple**
4. **mgcp sdp xpc-codec**
5. **mgcp codec** *type* [**packetization-period** *value*]
6. **no mgcp timer receive-rtcp**
7. **no mgcp piggyback message**
8. **mgcp endpoint offset**
9. **mgcp persistent** {**hookflash** | **offhook** | **onhook**}
10. **mgcp request timeout** {*timeout-value* | **max** *maxtimeout-value*}
11. **mgcp dtmf-relay voip codec** {**all** | **low-bit-rate**} **mode** {**cisco** | **nse** | **out-of-band**}
12. **mgcp max-waiting-delay** *value*
13. **mgcp restart-delay** *value*
14. **mgcp vad**
15. **mgcp ip-tos** {**high-reliability** | **high-throughput** | **low-cost** | **low-delay** | **rtp precedence** *value* | **signaling precedence** *value*}
16. **mgcp quality-threshold** {**hwm-cell-loss** *value* | **hwm-jitter-buffer** *value* | **hwm-latency** *value* | **hwm-packet-loss** *value* | **lwm-cell-loss** *value* | **lwm-jitter-buffer** *value* | **lwm-latency** *value* | **lwm-packet-loss** *value*}
17. **mgcp playout** {**adaptive** *init-value min-value max-value* | **fax** *value* | **fixed** *init-value*}
18. **mgcp package-capability** [*package-type*]
19. **mgcp default package** [*package-type*]

DETAILED STEPS

	Command	Purpose
Step 1	`Router(config)#` **`mgcp call-agent`** `{`*`dns-name`*` | `*`ip-address`*`} [`*`port`*`] [`**`service-type`** *`type`*`] [`**`version`** *`protocol-version`*`]`	Configures parameters for communicating with the call agent (media gateway controller). **Note** You can define a call agent globally with the **mgcp call-agent** command, or locally for each MGCP profile with the **call-agent** command, but not both. Whichever command you configure first blocks configuration of the other.
Step 2	`Router(config)#` **`mgcp behavior`** `{`**`auep`**` | `**`signal`**`}` **`v0.1`**	(Optional) Forces a gateway to follow the MGCP Version 0.1 protocol for a specified behavior. All other MGCP functionality continues to behave according to the version of MGCP that is specified in the **mgcp call-agent** command. • **auep**—Forces the gateway to reply to an Audit Endpoint (AUEP) command according to the MGCP Version 0.1 specification. If this keyword is used, an AUEP command on an out-of-service endpoint results in a return code of 501. Use this keyword with Cisco IOS Release 12.3(2)T1 or a later release. • **signal**—Forces the gateway to handle signaling tones according to the MGCP Version 0.1 specification. The MGCP 0.1 specification treats call signaling tones as on-off tones, which terminate only after a specific MGCP message has been sent to stop the signal. The specifications for MGCP 1.0 and later versions treat call signaling tones as timeout tones, which terminate when the appropriate timeout timer expires. Use this keyword with Cisco IOS Release 12.3(4)T or a later release. • **v0.1**—Selects MGCP Version 0.1.
Step 3	`Router(config)#` **`mgcp sdp simple`**	Specifies that a subset of the SDP fields should be used.
Step 4	`Router(config)#` **`mgcp sdp xpc-codec`**	Enables codec negotiation in the SDP.
Step 5	`Router(config)#` **`mgcp codec`** *`type`* `[`**`packetization-period`** *`value`*`]`	Selects the default codec type and its optional packetization period value.
Step 6	`Router(config)#` **`no mgcp timer receive-rtcp`**	Disables the timer used by a gateway to disconnect a VoIP call when the IP connectivity is lost with the remote gateway. The timer is known as the RTP Control Protocol (RTCP) transmission interval timer.
Step 7	`Router(config)#` **`no mgcp piggyback message`**	Disables piggyback messages.
Step 8	`Router(config)#` **`mgcp endpoint offset`**	Increments the voice-port or DS0-group portion of the endpoint name for NCS 1.0.
Step 9	`Router(config)#` **`mgcp persistent`** `{`**`hookflash`**` | `**`offhook`**` | `**`onhook`**`}`	Enables call-agent notification of the specified type of event.

	Command	Purpose
Step 10	`Router(config)#` **`mgcp request timeout`** {*`timeout-value`* | **`max`** *`maxtimeout-value`*}	Specifies how long the gateway waits for a call-agent response to a request before retransmitting the request.
Step 11	`Router(config)#` **`mgcp dtmf-relay voip codec`** {**`all`** | **`low-bit-rate`**} **`mode`** {**`cisco`** | **`nse`** | **`out-of-band`**}	Ensures accurate forwarding of digits with a compressed codec.
Step 12	`Router(config)#` **`mgcp max-waiting-delay`** *`value`*	Specifies the number of milliseconds to wait after a restart before connecting with the call agent. Range is from 0 to 600,000 (600 seconds). Default is 3000. If used, these values should be staggered among gateways to avoid having large numbers of gateways connecting with the call agent at the same time after a mass restart.
Step 13	`Router(config)#` **`mgcp restart-delay`** *`value`*	Sets the delay value sent in the RestartInProgress (RSIP) graceful teardown, in seconds. Range is from 0 to 600. Default is 0.
Step 14	`Router(config)#` **`mgcp vad`**	Enables voice activity detection (VAD) as a default for MGCP calls. Default is disabled.
Step 15	`Router(config)#` **`mgcp ip-tos`** {**`high-reliability`** | **`high-throughput`** | **`low-cost`** | **`low-delay`** | **`rtp precedence`** *`value`* | **`signaling precedence`** *`value`*}	Enables the IP type of service (ToS) for MGCP-controlled connections.
Step 16	`Router(config)#` **`mgcp quality-threshold`** {**`hwm-cell-loss`** *`value`* | **`hwm-jitter-buffer`** *`value`* | **`hwm-latency`** *`value`* | **`hwm-packet-loss`** *`value`* | **`lwm-cell-loss`** *`value`* | **`lwm-jitter-buffer`** *`value`* | **`lwm-latency`** *`value`* | **`lwm-packet-loss`** *`value`*}	Sets the jitter buffer size threshold, latency threshold, and packet-loss threshold parameters.
Step 17	`Router(config)#` **`mgcp playout`** {**`adaptive`** *`init-value min-value max-value`* | **`fax`** *`value`* | **`fixed`** *`init-value`*}	Configures the jitter buffer packet size in milliseconds for MGCP calls. The default is **adaptive** 60 4 200 • **adaptive** *init-value min-value max-value*—Defines the range for the jitter-buffer packet size. The range for each value is 4 to 250. Default is 60 4 200. Note that *init-value* must be between *min-value* and *max-value*. • **fax** *value*—Defines the fax playout buffer size. The range is 0 to 700. The default value is 300. The range and default value might vary with different platforms. See the platform digital signal processor (DSP) specifications before setting this value. • **fixed** *init-value*—Defines the fixed size for the jitter-buffer packet size. The range is 4 to 250. There is no default value.
Step 18	`Router(config)#` **`mgcp package-capability`** [*`package-type`*]	Specifies an MGCP package to be supported on this gateway. Configure one package at a time and repeat this command to configure support for more than one package. Available package types vary with the type of gateway.
Step 19	`Router(config)#` **`mgcp default package`** [*`package-type`*]	Defines the package to be used as the default when no package is named with an event. Available package types vary with the type of gateway.

Configuring an MGCP Profile and Profile-Related MGCP Parameters

This optional task creates a named, user-defined MGCP *profile* that consists of a subset of all the MGCP endpoints on this gateway. More than one MGCP profile can be configured on a gateway. Each MGCP profile is associated with a call agent and one or more endpoints. When multiple MGCP profiles are configured, endpoints on a single media gateway can be controlled by different call agents. When each endpoint comes on line, an RSIP (RestartInProgress) message notifies the appropriate call agent of the endpoint's presence.

Note When partitioning a gateway for multiple call-agent control, the call agents must be coordinated so that there are no overlapping transaction identification numbers.

In addition, this task allows you to configure profile-related MGCP parameters to conform to the requirements of the call agent, trunks, or lines that are being used with the profile's endpoints. These parameters are called *profile-related* MGCP parameters because they are associated with a particular MGCP profile, or subset of endpoints, and they are configured in MGCP profile configuration mode. Other parameters are considered *global* MGCP parameters; when they are configured, they apply to all the endpoints on a gateway. Global MGCP parameters are discussed in the "Configuring Global MGCP Parameters" section on page 31.

The parameters for an MGCP profile are configured in a special MGCP profile configuration mode that you enter with the **mgcp profile** command. One or more endpoints are associated with the profile by using the **voice-port** command in MGCP profile configuration mode.

Note The only parameter that can be configured in both profile configuration mode and in global configuration mode is call agent, which is configured with the **call-agent** command for MGCP profiles, and with the **mgcp call-agent** command for global configurations. These commands are mutually exclusive; whichever command you configure first blocks configuration of the other. For example, if the MGCP profile **call-agent** command is configured on an endpoint, then you are not allowed to configure the global **mgcp call-agent** command.

You do not have to define MGCP profiles to configure profile-related parameters. For endpoints that are not associated with a user-defined MGCP profile, the values for profile-related parameters are provided by a predefined profile with the name *default*. The default profile is configured in the same way that a user-defined MGCP profile is configured, except that the keyword **default** is used in place of a profile name in the **mgcp profile** command. The default profile has no association with voice ports or a call agent (the call agent for these endpoints is defined by the global **mgcp call-agent** command).

In the excerpt below from a **show running-config** command output, two MGCP profiles are defined: MAX1 and MAX2. Each profile is associated with a different call agent and a different voice port. The MAX1 profile is configured with a value of 3 for the max1 retries parameter and 5 for max2 retries. The MAX2 profile uses the values in the default profile for those parameters. In the MAX2 profile, the MT package is configured as a persistent package. The max1 retries parameter for the default profile is configured with a value of 2. The max2 retries parameter is not configured, so the value used is the default value, which is 7. The MAX2 profile has a value of 2 for the max1 retries parameter and 7 for max2 retries.

```
!
mgcp profile MAX1
 call agent ca1.example.com 4022 service-type mgcp version 1.0
 max1 retries 3
 max2 retries 5
 voice-port 2/1:1
!
mgcp profile MAX2
 call-agent ca2.example.com 50031 service-type mgcp version 0.1
 package persistent mt-package
 voice-port 2/0:1
!
mgcp profile default
 max1 retries 2
```

To configure parameters for a user-defined MGCP profile or for the default profile, use the following commands as appropriate, beginning in global configuration mode:

SUMMARY STEPS

1. **mgcp profile** {*profile-name* | **default**}
2. **description** {*text*}
3. **call-agent** {*dns-name* | *ip-address*} [*port*] [**service-type** *type*] [**version** *protocol-version*]
4. **voice-port** *port-number*
5. **default** {*command*}
6. **package persistent** *package-name*
7. **timeout tsmax** *tsmax-value*
8. **timeout tdmax** *tdmax-value*
9. **timeout tdinit** *tdinit-value*
10. **timeout tcrit** *tcrit-value*
11. **timeout tpar** *tpar-value*
12. **timeout thist** *thist-value*
13. **timeout tone mwi** *mwitone-value*
14. **timeout tone ringback** *ringbacktone-value*
15. **timeout tone ringback connection** *connectiontone-value*
16. **timeout tone network congestion** *congestiontone-value*
17. **timeout tone busy** *busytone-value*
18. **timeout tone dial** *dialtone-value*
19. **timeout tone dial stutter** *stuttertone-value*
20. **timeout tone ringing** *ringingtone-value*
21. **timeout tone ringing distinctive** *distinctivetone-value*
22. **timeout tone reorder** *reordertone-value*
23. **timeout tone cot1** *continuity1tone-value*

24. **timeout tone cot2** *continuity2tone-value*
25. **max1 lookup**
26. **max1 retries** *value*
27. **max2 lookup**
28. **max2 retries** *value*
29. **exit**

DETAILED STEPS

	Command	Purpose
Step 1	`Router(config)# mgcp profile {profile-name | default}`	Initiates MGCP profile mode, in which you create and configure a named MGCP profile associated with one or more endpoints, or configure the default profile.
Step 2	`Router(config-mgcp-profile)# description {text}`	Provides a description for the profile.
Step 3	`Router(config-mgcp-profile)# call-agent {dns-name | ip-address} [port] [service-type type] [version protocol-version]`	Defines the call agent's DNS name or IP address, UDP port number, service type, and protocol version. (Not used when configuring the default profile.) **Note** You can define a call agent globally with the **mgcp call-agent** command, or locally for each MGCP profile with the **call-agent** command, but not both. Whichever command you configure first blocks configuration of the other.
Step 4	`Router(config-mgcp-profile)# voice-port port-number`	Provides the voice port number or DS0 group number for the endpoint to be associated with this MGCP profile. Repeat this command to add more than one endpoint to the profile. (Not used when configuring the default profile.)
Step 5	`Router(config-mgcp-profile)# default {command}`	Restores the parameter represented by *command* to its default value.
Step 6	`Router(config-mgcp-profile)# package persistent package-name`	Configures the package type used when reporting persistent events for an MF CAS endpoint type. Valid types are **ms-package** and **mt-package.** Default is **ms-package**.
Step 7	`Router(config-mgcp-profile)# timeout tsmax tsmax-value`	Configures the maximum timeout value after which MGCP messages are removed from the retransmission queue, in seconds. Range is from 1 to 1000. Default is 20.
Step 8	`Router(config-mgcp-profile)# timeout tdmax tdmax-value`	Configures the maximum timeout value for the disconnected procedure (Tdmax), in seconds. Range is from 300 to 600. Default is 600.
Step 9	`Router(config-mgcp-profile)# timeout tdinit tdinit-value`	Configures the initial waiting delay value (Tdinit) used as the timer for the disconnect procedure, in seconds. Range is from 1 to 30. Default is 15.
Step 10	`Router(config-mgcp-profile)# timeout tcrit tcrit-value`	Configures the critical timeout value (Tcritical) for the interdigit timer used in digit map matching, in seconds. Range is from 1 to 600. Default is 4.

	Command	Purpose
Step 11	`Router(config-mgcp-profile)#` **`timeout tpar`** *`tpar-value`*	Configures the partial timeout value (Tpartial) for the interdigit timer used in digit map matching, in seconds. Range is from 1 to 60. Default is 16.
Step 12	`Router(config-mgcp-profile)#` **`timeout thist`** *`thist-value`*	Configures the packet storage timeout value, in seconds. Range is from 1 to 1100. Default is 30.
Step 13	`Router(config-mgcp-profile)#` **`timeout tone mwi`** *`mwitone-value`*	Configures the message waiting indicator timeout value, in seconds. Range is from 1 to 600. Default is 16.
Step 14	`Router(config-mgcp-profile)#` **`timeout tone ringback`** *`ringbacktone-value`*	Configures the ringback tone timeout value, in seconds. Range is from 1 to 600. Default is 180.
Step 15	`Router(config-mgcp-profile)#` **`timeout tone ringback connection`** *`connectiontone-value`*	Configures the timeout value for ringback tone on connection, in seconds. Range is from 1 to 600. Default is 180.
Step 16	`Router(config-mgcp-profile)#` **`timeout tone network congestion`** *`congestiontone-value`*	Configures the network congestion tone timeout value, in seconds. Range is from 1 to 600. Default is 180.
Step 17	`Router(config-mgcp-profile)#` **`timeout tone busy`** *`busytone-value`*	Configures the busy tone timeout value, in seconds. Range is from 1 to 600. Default is 3.
Step 18	`Router(config-mgcp-profile)#` **`timeout tone dial`** *`dialtone-value`*	Configures the dial tone timeout value, in seconds. Range is from 1 to 600. Default is 16.
Step 19	`Router(config-mgcp-profile)#` **`timeout tone dial stutter`** *`stuttertone-value`*	Configures the stutter dial tone timeout value, in seconds. Range is from 1 to 600. Default is 16.
Step 20	`Router(config-mgcp-profile)#` **`timeout tone ringing`** *`ringingtone-value`*	Configures the ringing tone timeout value, in seconds. Range is from 1 to 600. Default is 180.
Step 21	`Router(config-mgcp-profile)#` **`timeout tone ringing distinctive`** *`distinctivetone-value`*	Configures the distinctive ringing tone timeout value, in seconds. Range is from 1 to 600. Default is 180.
Step 22	`Router(config-mgcp-profile)#` **`timeout tone reorder`** *`reordertone-value`*	Configures the reorder tone timeout value, in seconds. Range is from 1 to 600. Default is 30.
Step 23	`Router(config-mgcp-profile)#` **`timeout tone cot1`** *`continuity1tone-value`*	Configures the continuity1 tone timeout value, in seconds. Range is from 1 to 600. Default is 3.
Step 24	`Router(config-mgcp-profile)#` **`timeout tone cot2`** *`continuity2tone-value`*	Configures the continuity2 tone timeout value, in seconds. Range is from 1 to 600. Default is 3.
Step 25	`Router(config-mgcp-profile)#` **`max1 lookup`**	Enables the DNS lookup procedure after the suspicion threshold is reached. Default is enabled.
Step 26	`Router(config-mgcp-profile)#` **`max1 retries`** *`value`*	Sets the suspicion threshold number of retries. Range is from 3 to 30. Default is 5.
Step 27	`Router(config-mgcp-profile)#` **`max2 lookup`**	Enables the DNS lookup procedure after the disconnect threshold is reached. Default is enabled.
Step 28	`Router(config-mgcp-profile)#` **`max2 retries`** *`value`*	Sets the disconnect threshold number of retries. Range is from 3 to 30. Default is 7.
Step 29	`Router(config-mgcp-profile)#` **`exit`**	Exits the current mode.

Verifying the Configuration

To verify configuration, use the following commands.

SUMMARY STEPS

1. **show running-configuration**
2. **show mgcp** [**connection** | **endpoint** | **profile** [*profile-name*] | **statistics**]

DETAILED STEPS

	Command	Purpose
Step 1	`Router# show running-configuration`	Displays the current configuration settings.
Step 2	`Router# show mgcp [connection	endpoint	profile [profile-name]	statistics]`	Displays the current MGCP settings.

Troubleshooting Tips

The following suggestions help with troubleshooting:

- Use the **show running-config** command to verify that the following are properly configured:
  - For CAS and POTS endpoints, POTS dial peers are configured with the **mgcpapp** application.
  - The correct packages are enabled in the **mgcp package-capability** command.
  - The **mgcp call-agent** or **call-agent** command defines the call agent and service type correctly.
- Reset the MGCP statistical counters with the **clear mgcp statistics** command.
- If RTP traffic is not getting through, make sure that IP routing is enabled. Use the **show rtp statistics** command, then use the **debug ip udp** command and track down the MGCP RTP packets.

```
Router# show rtp statistics

RTP Statistics info:
No. CallId Xmit-pkts Xmit-bytes Rcvd-pkts  Rcvd-bytes Lost pkts  Jitter Latenc
1   17492  0x8A      0x5640     0x8A       0x5640     0x0        0x0      0x0

Router# show rtp statistics

RTP Statistics info:
No. CallId Xmit-pkts Xmit-bytes Rcvd-pkts  Rcvd-bytes Lost pkts  Jitter Latenc
1   17492  0xDA      0x8840     0xDB       0x88E0     0x0        0x160    0x0
```

- If an RSIP message is not received by the call agent, make sure that the **mgcp call-agent** command or the MGCP profile **call-agent** command is configured with the correct call agent name or IP address and UDP port. Use the **show mgcp** command or the **show mgcp profile** command to display this information:

```
Router# show mgcp

MGCP Admin State ACTIVE, Oper State ACTIVE - Cause Code NONE
MGCP call-agent: 172.29.248.51  Initial protocol service is MGCP, v. 1.0
...
MGCP gateway port: 2727, MGCP maximum waiting delay 3000
...

Router# show mgcp profile

MGCP Profile nycprofile
Description: NY branch office configuration
Call-agent: 10.14.2.200 Initial protocol service is MGCP, v. 1.0
```

- To verify connections and endpoints, use the **show mgcp** command:

```
Router# 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/DS1-1/5      C=F123AB,5,6  I=0x3  P=16506,16602  M=3  S=4  C=1 E=2,0,0,2  R=0,0
2. S0/DS1-1/6      C=F123AB,7,8  I=0x4  P=16602,16506  M=3  S=4  C=1 E=0,0,0,0  R=0,0

Router# show mgcp endpoint

T1/0 ds0-group 0 timeslots 1-24
T1/1 ds0-group 0 timeslots 1-24
T1/2 ds0-group 0 timeslots 1-24
T1/3 ds0-group 0 timeslots 1-24
```

- If an MGCP message is rejected, it may be because the remote media gateway does not support SDP mandatory parameters (the *o*=, *s*=, and *t*= lines). If this is the case, configure the **mgcp sdp simple** command to send SDP messages without those parameters.
- If you notice problems with voice quality, make sure that the **cptone** (voice-port configuration) command is set for the correct country code. Capturing RTP packets from the sniffer may help to debug the problem, such as whether the payload type or timestamps are set correctly, and so forth.
- To check operation of interfaces, use the **show interface** command.
- To view information about activity on the T1 or E1 line, use the **show controllers** command. Alarms, line conditions, and other errors are displayed. The data is updated every 10 seconds; and every 15 minutes, the cumulative data is stored and retained for 24 hours.
- When necessary, you can enable debug traces for errors, events, media, packets, and parser. The command **debug mgcp packets** can be used to verify that your packets are arriving at the gateway and to monitor message flow in general. Note that there is always a performance penalty when using debug commands. The sample output below shows the use of the optional **input-hex** keyword to enable display of hexadecimal values.

```
Router# debug mgcp packets input-hex

Media Gateway Control Protocol input packets in hex value debugging is on
MGCP Packet received -
DLCX 49993 * MGCP 0.1
MGCP Packet received in hex -
44 4C 43 58 20 34 39 39 39 33 20 2A 20 4D 47 43 50 20 30 2E 31 A
send_mgcp_msg, MGCP Packet sent --->
250 49993
```

Configuration Examples for Configuring MGCP 1.0 Including NCS 1.0 and TGCP 1.0 Profiles

This section provides the following configuration examples:

Cisco uBR925 Using Radio Frequency Interface Example

This example shows how to set up a Cisco uBR925 as an MGCP residential gateway. The call agent is specified to the cable router (Cisco uBR925, Cisco CVA122, or Cisco CVA122E) by a Dynamic Host Configuration Protocol (DHCP) offer on a cable radio frequency (RF) network. On completion of the DHCP offer, the call agent is set in the MGCP profile on the cable modem. This setting is displayed with the **show mgcp profile** command. The router does not show the call agent in the CLI.

```
version 12.2
no service single-slot-reload-enable
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname hydepark
!
logging rate-limit console 10 except errors
!
clock timezone - 0 6
ip subnet-zero
no ip routing
ip domain-name example.com
ip name-server 10.0.0.229
!
ip ssh time-out 120
ip ssh authentication-retries 3
no ip dhcp-client network-discovery
!
interface Ethernet0
 ip address 192.168.0.11 255.255.0.0
 no ip route-cache
 no ip mroute-cache
 bridge-group 59
 bridge-group 59 spanning-disabled
!
interface cable-modem0
 no ip route-cache
 no ip mroute-cache
 cable-modem boot admin 2
 cable-modem boot oper 5
```

```
 bridge-group 59
 bridge-group 59 spanning-disabled
!
ip classless
no ip http server
no ip http cable-monitor
!
snmp-server manager
!
voice-port 0
 input gain -2
 output attenuation 0
!
voice-port 1
 input gain -2
 output attenuation 0
!
mgcp
! Use this CLI with NCS 1.0
mgcp endpoint offset
!
mgcp profile default
!
dial-peer voice 100 pots
 application MGCPAPP
 port 0
!
dial-peer voice 101 pots
 application MGCPAPP
 port 1
!
line con 0
line vty 0 4
 login
!
end
```

Cisco uBR925 Using Ethernet0 Interface Example

This example shows how to set up a Cisco uBR925 as a residential gateway:

```
version 12.2
no service single-slot-reload-enable
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname monticello
!
logging rate-limit console 10 except errors
!
clock timezone - 0 6
ip subnet-zero
ip domain-name example.com
ip name-server 10.0.0.229
!
ip ssh time-out 120
ip ssh authentication-retries 3
no ip dhcp-client network-discovery
!
interface Ethernet0
```

```
 ip address 192.168.0.11 255.255.0.0
 no ip route-cache
 no ip mroute-cache
 bridge-group 59
 bridge-group 59 spanning-disabled
!
interface cable-modem0
 no ip route-cache
 no ip mroute-cache
 shutdown
 cable-modem boot admin 2
 cable-modem boot oper 5
 no cable-modem compliant bridge
 cable-modem voip clock-internal
 bridge-group 59
 bridge-group 59 spanning-disabled
!
ip classless
no ip http server
no ip http cable-monitor
!
ip default-gateway 172.16.1.1
!
! We are using the cable modem without its RF interface. So
! route IP traffic out the Ethernet0 interface.
!
ip route 0.0.0.0 0.0.0.0 Ethernet0
!
snmp-server manager
!
voice-port 0
 input gain -2
 output attenuation 0
!
voice-port 1
 input gain -2
 output attenuation 0
!
mgcp
!
! The ip address of call agent below can be a FQDN as well.
mgcp call-agent 10.0.0.224 service-type ncs version 1.0
! Use this CLI with NCS 1.0
mgcp endpoint offset
!
mgcp profile default
!
dial-peer voice 100 pots
 application MGCPAPP
 port 0
!
dial-peer voice 101 pots
 application MGCPAPP
 port 1
!
line con 0
line vty 0 4
 login
!
end
```

Cisco CVA122 Using Radio Frequency Interface Example

The call agent is specified to the cable router (Cisco uBR925, Cisco CVA122, or Cisco CVA122E) by a DHCP offer on a cable RF network. On completion of the DHCP offer, the call agent is set in the MGCP profile on the cable modem. This setting is displayed with the **show mgcp profile** command. The router does not show the call agent in the CLI.

```
version 12.2
no service single-slot-reload-enable
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
!
hostname mtvernon
!
no logging buffered
logging rate-limit console 10 except errors
!
clock timezone - -5
ip subnet-zero
no ip routing
ip domain-name example.com
ip name-server 10.0.0.229
!
no ip dhcp-client network-discovery
!
interface Ethernet0
 ip address 10.20.0.59 255.255.0.0
 no ip route-cache
 no ip mroute-cache
 shutdown
 bridge-group 59
 bridge-group 59 spanning-disabled
!
interface cable-modem0
 no ip route-cache
 no ip mroute-cache
 cable-modem boot admin 2
 cable-modem boot oper 5
 bridge-group 59
 bridge-group 59 spanning-disabled
!
interface usb0
 ip address 10.20.0.59 255.255.0.0
 no ip route-cache
 no ip mroute-cache
 arp timeout 0
 bridge-group 59
 bridge-group 59 spanning-disabled
!
ip classless
no ip http server
no ip http cable-monitor
!
access-list 1 deny   10.0.0.254
access-list 1 permit any
snmp-server packetsize 4096
snmp-server manager
```

```
call rsvp-sync
!
voice-port 0
 input gain -2
 output attenuation 0
 timeouts interdigit 2
!
voice-port 1
 input gain -2
 output attenuation 0
 timeouts interdigit 2
!
mgcp
!
mgcp profile default
!
mgcp profile test
 call-agent test service-type ncs version 1.0
!
dial-peer voice 100 pots
 application MGCPAPP
 port 0
!
dial-peer voice 101 pots
 application MGCPAPP
 port 1
!
line con 0
 exec-timeout 0 0
line vty 0 4
 exec-timeout 0 0
 login
!
end
```

Cisco 2600 Series as a Residential Gateway Example

This example shows a Cisco 2620 router being configured as an analog residential gateway:

```
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname betty-2620
!
voice-port 1/0/0
!
voice-port 1/0/1
!
dial-peer voice 1 pots
application mgcpapp
destination-pattern 100
port 1/0/0
!
dial-peer voice 2 pots
application mgcpapp
destination-pattern 101
port 1/0/1
!
process-max-time 200
!
mgcp 4000
mgcp call-agent 10.14.2.200 4000 service-type mgcp version 1.0
mgcp sdp simple
no mgcp timer receive-rtcp
mgcp sdp xpc-codec
no mgcp piggyback message
mgcp endpoint offset
no mgcp persistent hook on
no mgcp persistent hook flash
mgcp request timeout 1000
mgcp dtmf-relay codec all mode cisco
mgcp max-waiting-delay 600000
mgcp restart-delay 500
mgcp codec g711ulaw packetization-period 10
mgcp ip-tos rtp precedence 7
mgcp quality-threshold lwm-jitter-buffer 59
mgcp quality-threshold lwm-latency 199
mgcp quality-threshold lwm-packet-loss 2
mgcp playout adaptive 100 50 150
mgcp package-capability dtmf-package
mgcp package-capability mf-package
mgcp package-capability rtp-package
mgcp package-capability as-package
isdn voice-call-failure 0
srcp 2428
cns event-service server
!
mgcp profile cisco
 call-agent 10.14.2.200 4000 service-type mgcp version 1.0
 voice-port 0:1
 package persistent mt-package
 timeout tsmax 100
 timeout tdinit 30
 timeout tcrit 600
 timeout tpar 600
 timeout thist 60
```

```
 timeout tone mwi 600
 timeout tone ringback 600
 timeout tone ringback connection 600
 timeout tone network congestion 600
 timeout tone busy 600
 timeout tone dial 600
 timeout tone dial stutter 600
 timeout tone ringing 600
 timeout tone ringing distinctive 600
 timeout tone reorder 600
 timeout tone cot1 600
 timeout tone cot2 600
 max1 retries 10
 no max2 lookup
 max2 retries 10
!
interface Ethernet0/0
 ip address 10.14.12.9 255.0.0.0
!
interface Ethernet0/1
 no ip address
 shutdown
!
ip classless
ip route 0.0.0.0 0.0.0.0 10.14.0.1
no cdp run
!
line con 0
exec-timeout 0 0
line aux 0
line vty 0 4
exec-timeout 0 0
 password test
 login
!
end
```

Cisco 3660 Platform as a Trunking Gateway Example

This example shows a Cisco 3660 that is being configured for CAS trunks. The association of endpoints with the MGCP application is made in the dial-peer configuration.

```
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname charley-3660
!
controller t1 1/0
 linecode b8zs
 clock source line secondary 1
 ds0-group 0 timeslots 1-24 type e&m-winkstart
!
controller t1 1/1
 linecode b8zs
 clock source line secondary 1
 ds0-group 0 timeslots 1-24 type e&m-winkstart
!
ip subnet-zero
!
voice-port 1/0:0
 dial-type mf
!
voice-port 1/1:0
 dial-type mf
!
dial-peer voice 1 pots
 application mgcpapp
 destination-pattern 100
 port 1/0:0
!
dial-peer voice 2 pots
 application mgcpapp
 destination-pattern 101
 port 1/1:0
!
mgcp 4000
mgcp call-agent 10.14.2.200 4000 service-type mgcp version 1.0
mgcp sdp simple
no mgcp timer receive-rtcp
mgcp sdp xpc-codec
no mgcp piggyback message
mgcp endpoint offset
mgcp persistent hook on
mgcp persistent hook flash
mgcp request timeout 1000
mgcp dtmf-relay codec all mode cisco
mgcp max-waiting-delay 600000
mgcp restart-delay 500
mgcp codec g711ulaw packetization-period 10
mgcp ip-tos rtp precedence 7
mgcp quality-threshold lwm-jitter-buffer 59
mgcp quality-threshold lwm-latency 199
mgcp quality-threshold lwm-packet-loss 2
mgcp playout adaptive 100 50 150
mgcp package-capability dtmf-package
mgcp package-capability mf-package
mgcp package-capability rtp-package
mgcp package-capability as-package
```

```
isdn voice-call-failure 0
srcp 2428
cns event-service server
!
mgcp profile cisco
 call-agent 10.14.2.200 4000 service-type mgcp version 1.0
 voice-port 1/0:0
 package persistent mt-package
 timeout tsmax 100
 timeout tdinit 30
 timeout tcrit 600
 timeout tpar 600
 timeout thist 60
 timeout tone mwi 600
 timeout tone ringback 600
 timeout tone ringback connection 600
 timeout tone network congestion 600
 timeout tone busy 600
 timeout tone dial 600
 timeout tone dial stutter 600
 timeout tone ringing 600
 timeout tone ringing distinctive 600
 timeout tone reorder 600
 timeout tone cot1 600
 timeout tone cot2 600
 max1 retries 10
 no max2 lookup
 max2 retries 10
!
interface FastEthernet0/0
 ip address 10.14.12.12 255.0.0.0
 speed auto
 duplex auto
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
ip classless
ip route 0.0.0.0 0.0.0.0 10.14.0.1
no ip http server
!
line con 0
 exec-timeout 0 0
 transport input none
line aux 0
line vty 0 4
 exec-timeout 0 0
 password trial
 login
!
end
```

Cisco MC3810 as a Residential Gateway Example

The following example shows a Cisco MC3810 being configured as a residential gateway:

```
version 12.2
no service pad
service timestamps debug datetime msec
service timestamps log uptime
!
hostname harry
!
logging buffered
!
ip subnet-zero
ip host buffalo 192.168.254.254
!
mgcp
mgcp call-agent 10.14.90.1
!
voice-card 0
 codec complexity high
!
controller T1 0
 framing esf
 linecode b8zs
!
interface Ethernet0
 ip address 10.14.92.3 255.255.0.0
!
interface Serial0
 shutdown
!
interface Serial1
 no ip address
 no ip route-cache
 no ip mroute-cache
 shutdown
!
interface FR-ATM20
 no ip address
 shutdown
!
ip default-gateway 10.14.0.1
ip route 192.168.254.0 255.255.255.0 10.14.0.1
!
voice-port 1/1
!
dial-peer voice 1 pots
 application mgcpapp
 port 1/1
!
line con 0
 exec-timeout 0 0
 transport input none
line aux 0
line 2 3
line vty 0 4
login
!
end
```

Cisco MC3810 as a VoAAL2 Gateway using AAL2 PVCs Example

This example shows a Cisco MC3810 being configured as a VoAAL2 gateway using AAL2 PVCs:

```
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname stella-mc3810
!
network-clock base-rate 56k
ip subnet-zero
no ip domain-lookup
ip host camel 192.168.254.254
ip host buffalo 192.168.254.253
!
mgcp
mgcp call-agent 10.14.117.4 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode nse
mgcp dtmf-relay voaal2 codec all
mgcp modem passthrough nse
mgcp package-capability rtp-package
mgcp tse payload 100
mgcp timer receive-rtcp 100
mgcp timer net-cont-test 3000
isdn voice-call-failure 0
!
voice-card 0
!
controller T1 0
    mode atm
    framing esf
    linecode b8zs
!
interface Ethernet0
    ip address 10.14.121.1 255.255.0.0
!
interface Serial0
    no ip address
    no ip mroute-cache
    shutdown
    no fair-queue
!
interface Serial1
    no ip address
    shutdown
!
interface ATM0
    no ip address
    ip mroute-cache
    no atm ilmi-keepalive
!
interface ATM0.2 point-to-point
   pvc 2/200
      vbr-rt 760 760 100
      encapsulation aal2
      vcci 2
!
interface FR-ATM20
     no ip address
     shutdown
!
```

```
router igrp 1
    redistribute connected
    network 1.0.0.0
!
ip default-gateway 10.14.0.1
no ip http server
ip classless
ip route 192.168.254.0 255.255.255.0 10.14.0.1
!
dialer-list 1 protocol ip permit
dialer-list 1 protocol ipx permit
voice-port 1/1
!
voice-port 1/2
     shutdown
!
voice-port 1/6
     shutdown
!
dial-peer voice 1 pots
    application mgcpapp
    port 1/1
!
line con 0
    transport input none
line aux 0
   line 2 3
 line vty 0 4
    password lab
    login
!
end
```

Tip

- See the "Additional References for MGCP and SGCP" section in the Preface for related documents, standards, and MIBs.
- See "Glossary" for definitions of terms in this guide.

Configuring MGCP Basic CLASS and Operator Services

This chapter provides information on configuring and troubleshooting the MGCP Basic (CLASS) and Operation Services feature. The feature provides xGCP support for three-way calling on residential and trunking gateways.

Feature benefits include the following:

- The merged SGCP/MGCP software for RGWs, BGWs, and TGWs enables easier development and growth of Cisco and customer solutions.
- MGCP BCOS satisfies the requirements for providing basic CLASS services on Cisco IOS gateways that enable multiple xGCP solutions, particularly residential gateway and IP Centrex.

Note For more information about this and related Cisco IOS voice features, see the following:

- "Overview of MGCP and Related Protocols" on page 3
- Entire Cisco IOS Voice Configuration Library—including library preface and glossary, other feature documents, and troubleshooting documentation—at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vcl.htm.

Feature History for MGCP Basic (CLASS) and Operation Services

Release	Modification
12.2(2)T	This feature was introduced.

Contents

Prerequisites for Configuring MGCP Basic CLASS and Operator Services

Prerequisites are described in the "Prerequisites for Configuring MGCP and Related Protocols" section on page 3.

Restrictions for MGCP Basic CLASS and Operator Services

- For the Cisco MC3810 series platform, the MGCP BCOS software is supported on an HCM version of an DSP card; it is not supported on an VCM version.

  To check the type of DSP card in your Cisco MC3810 series platform, enter a **show version** command at the EXEC prompt. For example `Router#` **`show version`**:

  - If you have an HCM card, the following line appears as part of the **show version** information:

    1 6-DSP (slot 2) High Performance Compression Module(v01.A0)

  - If you have an VCM card, the following line appears as part of the **show version** information:

    1 6-DSP (slot 2) Voice Compression Module(v255.V7)

  If you have an HCM card, the MGCP BCOS features will function properly. If you have an VCM card, the feature is not supported.

- The G.728 and G.723 codecs do not support three-way calling.

Information About MGCP Basic CLASS and Operator Services

The MGCP BCOS are a set of calling features, sometimes called "custom calling" features, that use MGCP to transmit voice, video, and data over the IP network. These features are usually found in circuit-based networks. MGCP BCOS brings them to the Cisco IOS gateways on packet-based networks.

The MGCP BCOS software is built on the MGCP CAS PBX and AAL2 software package, and supports MGCP 0.1 and the earlier protocol version Simple Gateway Control Protocol (SGCP) 1.1 and 1.5.

The following MGCP BCOS features are available on residential gateways (RGWs) and business gateways (BGWs):

- Distinctive Power Ring, page 55
- Visual Message Waiting Indicator, page 55
- Caller ID, page 55
- Caller ID with Call Waiting, page 55
- Call Forwarding, page 55
- Ring Splash, page 56
- Distinctive Call-Waiting Tone, page 56
- Message-Waiting Tone, page 56
- Stutter Dial Tone, page 56
- Off-Hook Warning Tone, page 56

The following two features can be run as residential gateway (RGW) or trunking gateway (TGW) features:

- 911 Calls, page 57
- Three-Way Calling (TWC), page 57

Distinctive Power Ring

A telephone rings in a distinctive pattern when a call comes in from a predefined telephone number. The following patterns are available:

- R1: One long ring
- R2: Long ring -long ring
- R3: Short ring-short ring-long ring
- R4: Short ring - long ring - short ring
- R5: One short ring

Visual Message Waiting Indicator

A light goes on when a message is waiting for the line.

Caller ID

The calling party's telephone number, date, and time of the call appear on the receiving telephone's display between the first and second rings. A maximum of 18 digits are shown., and private and unlisted numbers are displayed. If the called party answers the phone on the first ring, the calling party's number does not appear.

If the called party has an appropriate name display unit, the calling party's name and telephone number appear on the display. The name and number appear between the first and second rings.

If the calling party has blocked Caller ID from displaying the telephone number, the called party sees "P" for private or "Anonymous" on the display unit.

Caller ID with Call Waiting

If the called party has Caller ID and has enabled the Call Waiting feature, then the calling party's name (if an appropriate display unit is available) and telephone number appear while the called party is on the line with another call.

If the calling party has blocked Caller ID from displaying the name and telephone number, the called party will see "P" for private or "Anonymous" on the display unit.

Call Forwarding

The following scenarios are available:

- The call agent transfers all incoming calls to a designated telephone number when the called number does not answer after a predetermined interval.
- The call agent transfers all incoming calls to a designated telephone number when the called number is busy.
- The call agent transfers all incoming calls to a specific destination when the user enters a code and a destination telephone number that receives the calls. The user is responsible for all charges between the original called number and the receiving telephone number.
- A user can activate Call Forwarding remotely using a touch-tone telephone and a user-defined personal identification number (PIN), which, by default, is the last four digits of the user's telephone number. The original destination telephone emits a Ring Splash when a call is forwarded.

Ring Splash

Also known as Reminder Ring, Ring Splash is activated when the user enables Call Forwarding on the telephone. The user hears Distinctive Power Ring R5 when the line is idle and a call has been forwarded. This reminds the user that Call Forwarding is active.

Distinctive Call-Waiting Tone

The called party hears four audible tone patterns (*waiting tones*, or WTs) when a call is waiting on the called party's line. The call agent provides the following tone patterns in sequence as the incoming call continues to wait:

- WT1: One short tone
- WT2: Short tone–short tone
- WT3: Short tone–short tone–short tone
- WT4: Short tone–long tone–short tone

Message-Waiting Tone

For users with an active voice mail system, a special dial tone is heard when the user goes off-hook and a message is waiting. The dial tone is a sequence of 10 short tones followed by a steady tone. If the user has a telephone with a visual message indicator, the indicator light goes on when a message is waiting.

Stutter Dial Tone

This tone is used in place of the dial tone to indicate that a message is waiting. When the user goes off-hook, a sequence of three short tones followed by a steady tone is heard.

Off-Hook Warning Tone

The user hears this tone when the telephone is off-hook. The tone is repeated bursts of sound of rising pitch.

911 Calls

The user can make a 911 call to an Emergency Service Bureau (ESB), and the call is maintained as long as the ESB does not hang up. If the user hangs up, the call is maintained. If the user hangs up and picks up the phone again, the call resumes. If the user hangs up and does not pick up the phone again, the ESB can ring the user and resume the call.

This feature is available in SGCP mode on the Cisco 3660 platforms and in MGCP mode on all supported platforms.

Three-Way Calling (TWC)

The user can create a 3-way call by pressing the switchhook quickly to put the first call on hold, dial a third party, and press the switchhook again quickly to join all parties to the call. This feature is supported on all five platforms.

Considerations for Three-way Calling

- The user who sets up the 3-way call must be connected to a residential gateway, which handles the call setup. TWC is transparent to a trunking gateway.
- Only the G.711u and G.711a codecs support TWC. If any part of a 3-way call is made on a codec other than the G.711u, that codec must be switched to G.711u mode before the second switchhook flash in order for the 3-way call to be set up.
- TWC supports calls originating as Voice over IP or Voice over AAL2 calls, not Voice over ATM or Voice over Frame Relay calls. However, if the network has ATM or Frame Relay as a transport protocol, the VoIP call is completed.
- The user originating the 3-way call is the *controller*. Each of the two other users on the call can add another person onto the call, which is referred to as *call chaining*. Those new users can also add another person to the call. However, when five people in total are on the call, adding more users causes voice quality to degrade.
- If the controller of the call hangs up, all the users are disconnected from the call. If one of the non-controller users hangs up, the remaining users are still connected to the call.
- If the controller presses the switchhook quickly for a third time, the last user connected to the call is dropped from the call.
- If two users are on a call and a third user calls one of them, that third user cannot be joined (bridged) into the two-party call.

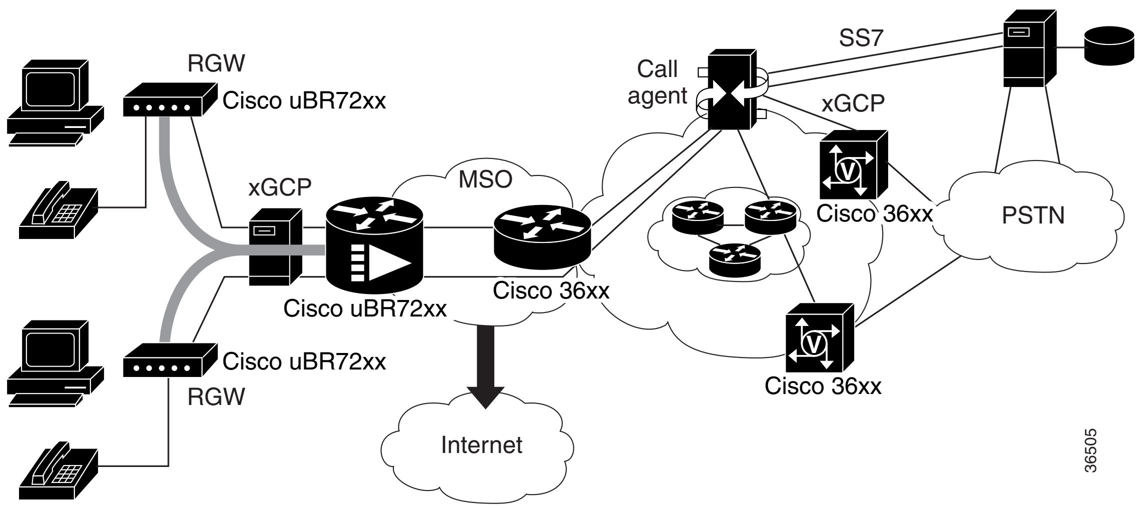
Examples of Service-Provider Solutions

The Basic CLASS and Operator Services features support MGCP solutions in the following areas:

- Residential cable access

  A CLEC can use residential cable access to provide residential customers with basic telephony and data services. CLASS features and Three-way calling, Caller ID with Call Waiting, and Distinctive Call Waiting Tone are features that support these customers. Figure 4 illustrates a possible residential cable access solution.

Figure 4 Residential Cable Access Solution

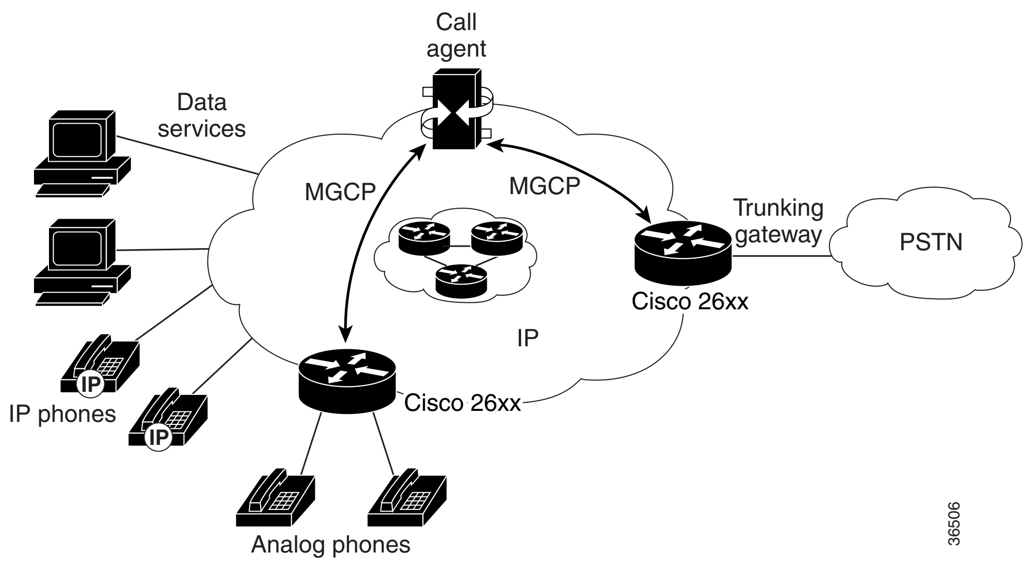


Note that, in Figure 4, the residential gateway must support the CLASS features and 911 capability.

- IP Centrex and IP PBX

  In these solutions, a call agent provides business voice services that are traditionally offered by a circuit-based PBX. CLASS features and Three-way calling, Caller ID with Call Waiting, Distinctive Call Waiting Tone, and Visual Message Waiting Indicator are features suitable for these customers. Figure 5 illustrates an IP Centrex solution:

Figure 5 IP Centrex Solution

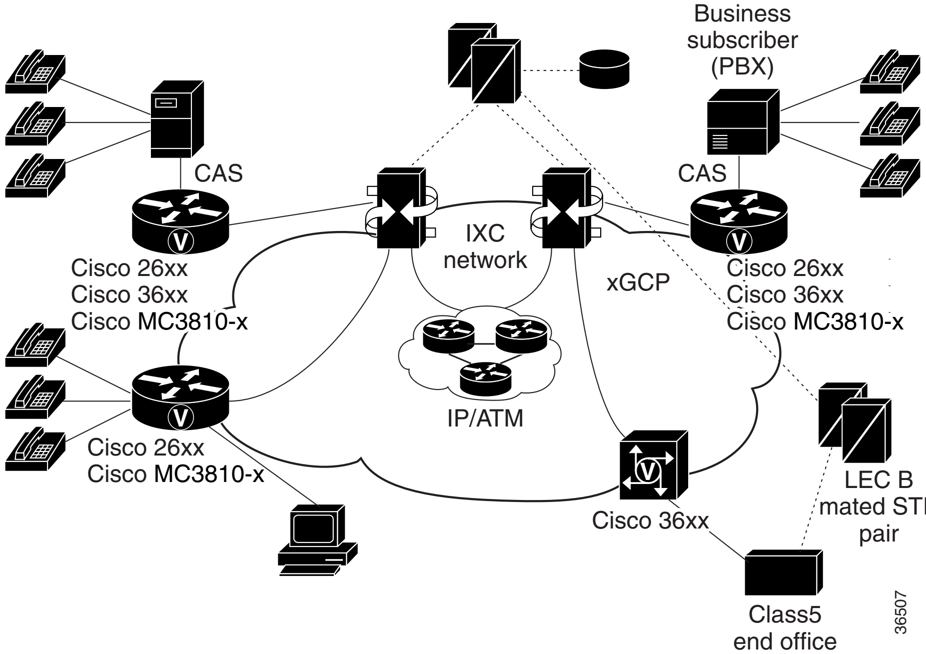


In Figure 5, the residential gateway (the Cisco 2600 series platforms) must support the CLASS features.

- Integrated access

  A CLEC or IXC can provide small, medium, and large businesses with integrated voice and data access services. The integrated access device can be located at the central office or on the customer's premises. Access to the subscriber can be analog or digital, and transport of voice and data can be over IP, Frame Relay, or ATM. CLASS features and Three-way calling, Caller ID with Call Waiting, Distinctive Call Waiting Tone, and Visual Message Waiting Indicator are features suitable for these customers. Figure 6 illustrates an integrated access solution.

Figure 6 Integrated Access Solution

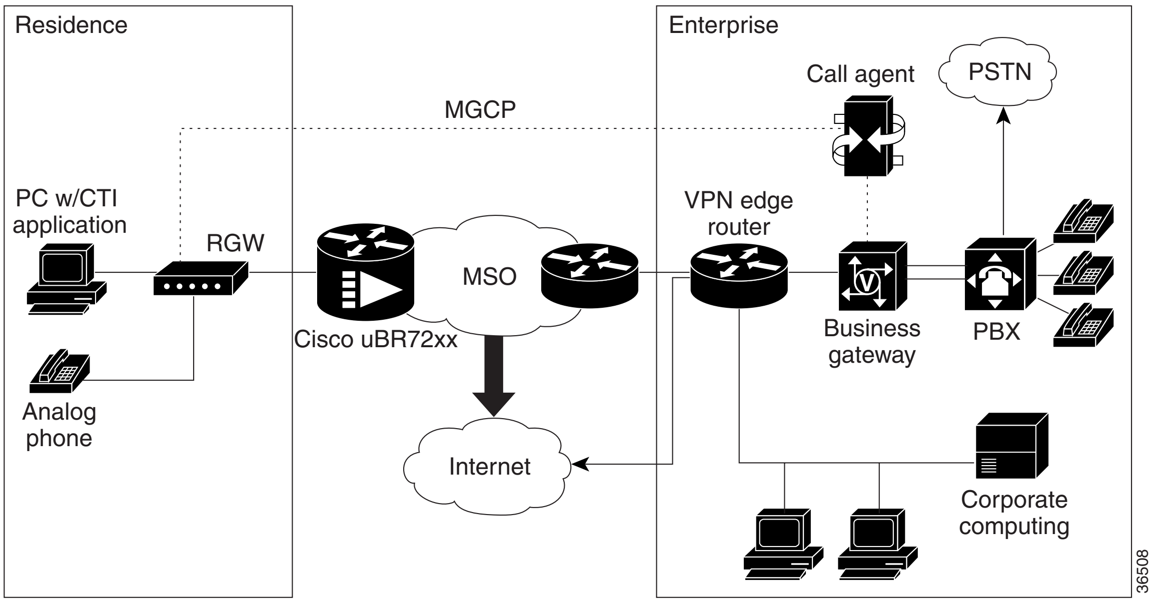


In Figure 6, the residential gateway (the Cisco 2600 series and Cisco MC3810 series platforms) must support the CLASS features.

- Telecommuter or small-office-home-office solution

  Figure 7 illustrates a telecommuter or small-office-home-office solution:

Figure 7 Telecommuter or Small Office-Home Office Solution



In Figure 7, the residential gateway must support the CLASS features.

Other solutions are possible using the MGCP open protocol.

How to Troubleshoot MGCP Basic CLASS and Operator Services

No new or modified configuration tasks are required to initiate MGCP Basic CLASS and Operator Services. MGCP BCOS co resides with MGCP CAS PBX and AAL2 PVC software, for which configuration activities are required. These are discussed in "Appendix A: Configuring MGCP CAS PBX and AAL2 PVC.

The following MGCP BCOS features do not work on telephones from all manufacturers when the telephones are connected to a Cisco MC3810 series platform:

- CID – Caller ID
- VMWI – Visual Message Waiting Indicator
- CIDCW – Caller ID with Call Waiting

Table 3 summarizes the findings for the models tested.

Table 3 Telephones and Feature Capabilities

Telephone	CID	VMWI	CIDCW
Casio TI-345	Y	—	N
Casio TI-360	Y	—	N
Dial Digital CP-2892C	Y	Y	Y

Table 3 Telephones and Feature Capabilities (continued)

Telephone	CID	VMWI	CIDCW
GE 29299GE1-A	Y	—	Y
Panasonic KX-TSC7	Y	N	N
Panasonic KX-TSC55-b	Y	Y	Y
Sony IT-ID80	Y	—	Y

To correct this operation, change the idle voltage in the voice port from low to high.

To change the voice port idle voltage, perform these additional steps:

- If the phone is already connected to the voice port, lift the phone's handset.
- If the phone is not connected to the voice port, do the following:
  - **a.** Attach the phone to the voice port.
  - **b.** Do a "shut" to the voice port.
  - **c.** Do a "no shut" to the voice port.
  - **d.** Lift the phone's handset.

Configuration Examples for MGCP Basic CLASS and Operator Services

No new or modified configuration settings are needed to implement MGCP Basic CLASS and Operator Services. See the MGCP CAS PBX and AAl2 PVC setup in "Appendix A: Configuring MGCP CAS PBX and AAL2 PVC" for sample configurations.

Tip

- See the "Additional References for MGCP and SGCP" section on page xi for related documents, standards, and MIBs.
- See the "Glossary" for definitions of terms in this guide.

Configuring NAS Package for MGCP

This chapter provides information on configuring the Network Access Server (NAS) Package for MGCP feature. The feature adds support for the MGCP NAS package on universal gateways. Data calls can be terminated on a trunking media gateway that is serving as a NAS. Trunks on the NAS are controlled and managed by a call agent supporting MGCP for both voice and data calls. The call agent must support the MGCP NAS package.

Key feature benefits derive from the presence of universal ports that are able to terminate both voice and data calls under control of the MGCP call agent. These benefits include the following:

- Cost savings
  - Sharing of trunks (T1 or E1) for dial and voice services
  - Collapsed IP backbone infrastructure
  - Simplified operations and management
- Increased revenue
  - Optimized utilization of trunk (T1 or E1) resources
- Flexibility in deploying new services
- Flexibility in access network engineering

Note For more information about this and related Cisco IOS voice features, see the following:

- "Overview of MGCP and Related Protocols" on page 3
- Entire Cisco IOS Voice Configuration Library—including library preface and glossary, other feature documents, and troubleshooting documentation—at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vcl.htm.

Feature History for NAS Package for MGCP

Release	Modification
12.2(2)XB	This feature was introduced on the Cisco AS5350 and Cisco AS5400.
12.2(11)T	This feature was implemented on the Cisco AS5850.

Contents

Prerequisites for Configuring Network Access Server Package for MGCP

Prerequisites are described in the "Prerequisites for Configuring MGCP and Related Protocols" section on page 3. In addition, the following apply:

- Configure a data network.
- Configure MGCP.

Information About Configuring Network Access Server Package for Media Gateway Control Protocol

This feature adds support for the Network Access Server Package for Media Gateway Control Protocol package on the Cisco AS5350, Cisco AS5400, and Cisco AS5850 universal gateways. With this implementation, data calls can be terminated on a trunking media gateway that is serving as a network access server (NAS). Trunks on the NAS are controlled and managed by a call agent that supports Media Gateway Control Protocol (MGCP) for both voice and data calls. The call agent must support the MGCP NAS package.

These capabilities are enabled by the universal port functionality of the Cisco AS5350, Cisco AS5400, and Cisco AS5850, which allows these platforms to operate simultaneously as network access servers and voice gateways to deliver universal services on any port at any time. These universal services include dial access, real-time voice and fax, wireless data access, and unified communications.

The MGCP NAS package implements signals and events to create, modify, and tear down data calls. The events include signaling the arrival of an outbound call (IP to Public Switched Telephone Network [PSTN]) to the media gateway controller (call agent), reporting carrier loss and call authorization status, and receiving callback requests. The following types of calls can be terminated as data calls:

- Data within the voice band (analog modem)
- ISDN data (digital modem)
- Data over voice when using a call agent that recognizes this call type and delivers these calls as digital data to the NAS

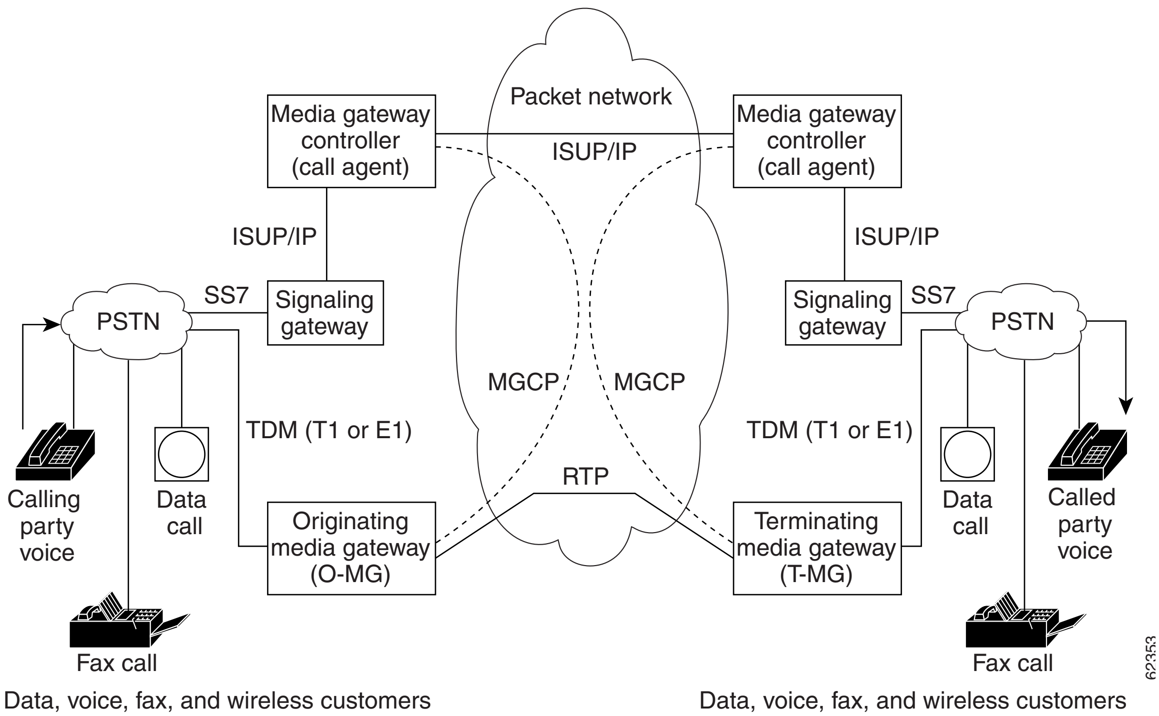
The NAS package provides MGCP capabilities for data calls on the Cisco AS5350, Cisco AS5400, and Cisco AS5850 that support all the dial-in and dial-out services, including the following:

- Virtual Private Network (VPN) with Layer 2 Tunneling Protocol (L2TP)
- Scalable Multichassis Multilink PPP (MMP) across multiple channels
- MGCP 1.0 and MGCP 0.1
- Call preauthentication with MGCP dial calls

Resource pool management can be used to manage dial ports when dialed number identification service (DNIS) preauthentication is enabled. The NAS returns an error with a preauthentication failure code to the call agent, which releases the call gracefully with a busy cause. Refer to the Cisco IOS Release 12.3 Configuration Guides and Command References, for more information about dial-pool management, and for more information about authentication, authorization, and accounting (AAA) preauthentication services.

Figure 8 on page 65 shows a typical network topology for universal port media gateways.

Figure 8 Media Gateways Operating As Network Access Servers



How to Configure Network Access Server Package for Media Gateway Control Protocol

With the Network Access Server Package for Media Gateway Control Protocol feature, the NAS supports both data and voice calls, which can be managed from a single call agent that supports MGCP with the NAS package. The NAS package provides the interface to a call agent (media gateway controller) for handling modem calls that terminate on the NAS and that originate from the PSTN, including callback requests. Results of AAA authorization and preauthorization requests from the NAS are reported to the call agent as notifications.

See the following sections for configuration tasks for the Network Access Server Package for Media Gateway Control Protocol feature. Each task in the list is identified as either required or optional.

- Configuring the NAS for MGCP, page 66 (required)
- Configuring Controllers, page 66 (required)
- Configuring Dialer Interfaces and Routing, page 67 (required)

- Verifying the Network Access Server Package for Media Gateway Control Protocol Feature, page 70 (optional)

Configuring the NAS for MGCP

In this task, MGCP is configured on the trunking gateway (NAS), and the NAS package is set as the default package. The steps that are listed are the minimum needed to configure MGCP on the NAS. For more commands and optional settings for MGCP, see the documents listed in the "Related Documents" section on page xi.

To configure the Network Access Server Package for Media Gateway Control Protocol feature, use the following commands in global configuration mode:

SUMMARY STEPS

1. **mgcp** [*gw-port*]
2. **mgcp call-agent** {*dns-name* | *ip-address*} [*ca-port*] [**service-type** *type*] [**version** *protocol-version*]
3. **mgcp default-package nas-package**

DETAILED STEPS

	Command	Purpose
Step 1	`Router(config)# mgcp [gw-port]`	Allocates resources for MGCP and starts the MGCP daemon. If no port is specified, the command defaults to port 2427.
Step 2	`Router(config)# mgcp call-agent {dns-name	ip-address} [ca-port] [service-type type] [version protocol-version]`	Configures the gateway with the address and protocol of the call agent (media gateway controller). Make sure to specify a call agent that supports the NAS package.
Step 3	`Router(config)# mgcp default-package nas-package`	(Optional) Defines the default package to be used for MGCP signaling. For this feature, specify the NAS-Package. Default generally used on trunking gateways is Trunk-Package and can be left unchanged.

Configuring Controllers

In this task, in addition to the standard controller commands, you configure a T1 or E1 controller for external signaling control by MGCP. You can also set the AAA preauthentication timer to expire after a certain number of milliseconds have elapsed without a response from the AAA server and indicate whether the call should be accepted or rejected if no response occurs before the timer expires.

To configure a controller to use the Network Access Server Package for Media Gateway Control Protocol feature, use the following commands beginning in global configuration mode:

SUMMARY STEPS

1. **controller** {**t1** | **e1**} *slot/port*
2. **framing** {**sf** | **esf**} (T1 lines) or **framing** {**crc4** | **no-crc4**} [**australia**] (E1 lines)

3. **extsig mgcp**
4. **guard-timer** *milliseconds* [**on-expiry** {**accept** | **reject**}]
5. **linecode** {**ami** | **b8zs**} (T1 lines) or **linecode** {**ami** | **hdb3**} (E1 lines)
6. **ds0-group** *channel-number* **timeslots** *range* **type none service mgcp**
7. **exit**

DETAILED STEPS

	Command	Purpose
Step 1	`Router(config)#` **`controller`** `{`**`t1`** `|` **`e1`**`}` *`slot/port`*	Configures a T1 or E1 controller and enters controller configuration mode.

Step 2	**T1**
	`Router(config-controller)#` **`framing`** `{`**`sf`** `|` **`esf`**`}`
	E1
	`Router(config-controller)#` **`framing`** `{`**`crc4`** `|` **`no-crc4`**`}` `[`**`australia`**`]`
		Selects the frame type for the T1 or E1 trunk.
		T1 default is **sf**. E1 default is **crc4**.

Step 3	`Router(config-controller)#` **`extsig mgcp`**	Configures external signaling control by MGCP for this controller. For T3 trunks, each logical T1 must be configured with the **extsig mgcp** command.
Step 4	`Router(config-controller)#` **`guard-timer`** *`milliseconds`* `[`**`on-expiry`** `{`**`accept`** `|` **`reject`**`}]`	(Optional) Sets a guard timer for the number of milliseconds to wait for a AAA server to respond to a preauthentication request before expiring. Also specifies the default action to take when the timer expires without a response from AAA.

Step 5	**T1**
	`Router(config-controller)#` **`linecode`** `{`**`ami`** `|` **`b8zs`**`}`
	E1
	`Router(config-controller)#` **`linecode`** `{`**`ami`** `|` **`hdb3`**`}`
		Specifies the line encoding to use.
		T1 default is **ami**. E1 default is **hdb3**.

Step 6	`Router(config-controller)#` **`ds0-group`** *`channel-number`* **`timeslots`** *`range`* **`type none service mgcp`**	Specifies the DS0 time slots that make up a logical voice port on a T1 or E1 controller and specifies the signaling type by which the router connects to the PBX or PSTN.
Step 7	`Router(config-controller)#` **`exit`**	Exits the current mode.

Configuring Dialer Interfaces and Routing

This set of tasks configures dial-on-demand routing (DDR) on a dialer interface that is under external call control by MGCP.

DDR refers to a collection of Cisco features that allows two or more Cisco routers to establish a dynamic connection over simple dial-up facilities to route packets and exchange routing updates on an as-needed basis. DDR is used for low-volume, periodic network connections over the PSTN or an ISDN. A connection is automatically established whenever *interesting traffic* is detected; during configuration you define what constitutes interesting traffic.

ISDN B channels, synchronous serial interfaces, and asynchronous interfaces can all be converted to dialer interfaces using dialer interface configuration commands.

DDR provides several functions. First, DDR *spoofs*, or pretends, that there are established configured routes to provide the image of full-time connectivity using the dialer interfaces. When the routing table forwards a packet to a dialer interface, DDR filters out the interesting packets for establishing, maintaining, and releasing switched connections. Internetworking is achieved over the DDR-maintained connection using PPP or other WAN encapsulation techniques.

The encapsulation methods available depend on the physical interface being used. Cisco supports PPP, High-Level Data Link Control (HDLC), Serial Line Internet Protocol (SLIP), and X.25 data-link encapsulations for DDR. PPP is the recommended encapsulation method because it supports multiple protocols and is used for synchronous, asynchronous, or ISDN connections. In addition, PPP performs address negotiation and authentication, and it is interoperable with different vendors.

There are two ways of setting up addressing on dialer interfaces:

- Applying a subnet to the dialer interfaces—Each site with a dialer interface is given a unique node address on a shared subnet for use on its dialer interface. This method is similar to numbering a LAN or multipoint WAN, and it simplifies the addressing scheme and creation of static routes.
- Using unnumbered interfaces—Similar to using unnumbered addressing on leased-line point-to-point interfaces, the address of another interface on the router is borrowed for use on the dialer interface. Unnumbered addressing takes advantage of the fact that there are only two devices on the point-to-point link.

DDR uses manually entered static network protocol routes. This eliminates the use of a routing protocol that broadcasts routing updates across the DDR connection, causing unnecessary connections.

Similar to the function provided by an Address Resolution Protocol (ARP) table, dialer map statements translate next-hop protocol addresses to telephone numbers. Without statically configured dialer maps, DDR call initiation cannot occur. When the routing table points at a dialer interface, and the next-hop address is not found in a dialer map, the packet is dropped.

Authentication in DDR network design provides two functions: security and dialer state. As most DDR networks connect to the PSTN, it is imperative that a strong security model be implemented to prevent unauthorized access to sensitive resources. Authentication also allows the DDR code to keep track of what sites are currently connected and provides for building of Multilink PPP bundles.

In summary, the following main tasks are involved in configuring the dialer interface and routing:

- Specification of interesting traffic—What traffic type should enable the link?
- Definition of static routes—What route do you take to get to the destination?
- Configuration of dialer information—What number do you call to get to the next-hop router, and what service parameters do you use for the call?

For MGCP NAS, configuration of dialer interfaces entails the use of the **dialer extsig** command in interface configuration mode, which enables the External Call Service Provider (XCSP) subsystem to provide an interface between the Cisco IOS software and the MGCP protocol. The XCSP subsystem enables services such as modem call setup and teardown for the dialer interface.

To configure the dialer interface and routing, use the following commands beginning in global configuration mode:

SUMMARY STEPS

1. **interface** *dialer-name*
2. **ip unnumbered** *interface-number* or **ip address** *ip-address subnet-mask* [**secondary**]

3. **encapsulation ppp**
4. **dialer in-band** [**no-parity** | **odd-parity**]
5. **dialer idle-timeout** *seconds* [**inbound** | **either**]
6. **dialer map** *protocol next-hop-address* [**name** *host-name*] [*dial-string*[**:***isdn-subaddress*]]
7. **dialer extsig**
8. **dialer-group** *number*
9. **no cdp enable**
10. **ppp authentication chap**
11. **exit**
12. **dialer list** *number* **protocol** *protocol-name* {**permit** | **deny** [**list** *access-list-number* | *access-group*]}
13. **ip route** *prefix mask* {*ip-address* | *interface-type interface-number*} [*distance*] [**tag** *tag*] [**permanent**]

DETAILED STEPS

	Command	Purpose
Step 1	`Router(config)# interface dialer-name`	Enters interface mode for the dialer interface.

Step 2	`Router(config-if)# ip unnumbered interface-number`
	or
	`Router(config-if)# ip address ip-address subnet-mask [secondary]`
		Enables IP processing on the dialer interface, configures the dialer interface not to have an explicit IP address, and assigns the IP address of the loopback interface instead. This command helps conserve IP addresses.

Step 3	`Router(config-if)# encapsulation ppp`	Sets encapsulation type for PPP.
Step 4	`Router(config-if)# dialer in-band [no-parity | odd-parity]`	Specifies that dial-on-demand routing (DDR) is to be supported. The **in-band** keyword specifies that the same interface that sends the data performs call setup and teardown operations between the router and an external dialing device such as a modem. By default, no parity is applied to the dialer string.
Step 5	`Router(config-if)# dialer idle-timeout seconds [inbound | either]`	Specifies the duration of idle time before a line is disconnected. Default direction is outbound. Default idle time is 120 seconds.
Step 6	`Router(config-if)# dialer map protocol next-hop-address [name host-name] [dial-string[:isdn-subaddress]]`	Configures a serial interface to make digital calls or to accept incoming calls from a specified location and to authenticate if so configured.
Step 7	`Router(config-if)# dialer extsig`	Specifies an interface for the initiation and termination of digital calls for external signaling protocols. Only one dialer with external signaling per NAS is permitted.
Step 8	`Router(config-if)# dialer-group number`	Controls access by configuring an interface to belong to a specific dialing group.
Step 9	`Router(config-if)# no cdp enable`	Disables Cisco Discovery Protocol (CDP) on the interface.

	Command	Purpose
Step 10	`Router(config-if)# ppp authentication chap`	Enables Challenge Handshake Authentication Protocol (CHAP) authentication on the interface.
Step 11	`Router(config-if)# exit`	Exits the current mode.
Step 12	`Router(config)# dialer list number protocol protocol-name {permit	deny [list access-list-number	access-group]}`	Defines a DDR dialer list for dialing by protocol or by a combination of a protocol and a previously defined access list. Each dialer interface can have only one dialer group, but the same dialer list can be assigned to multiple interfaces (using the **dialer-group** command).
Step 13	`Router(config)# ip route prefix mask {ip-address	interface-type interface-number} [distance] [tag tag] [permanent]`	Establishes a static route. Because you do not want dynamic routing protocols running across the DDR links, you manually configure static routes.

Verifying the Network Access Server Package for Media Gateway Control Protocol Feature

To verify configuration, use the following commands.

SUMMARY STEPS

1. **show running-configuration**
2. **show mgcp nas** {**dump** *slot port channel* | **info**}

DETAILED STEPS

Step 1 Use the following command to display the running configuration to verify configured parameters for MGCP, controllers, dialer interfaces, and routing:

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

The following example shows the configuration for serial interface 1:

```
Router# show running-configuration interface serial 1

Building configuration...
Current configuration:
!
interface Serial1
no ip address
no ip directed-broadcast
no ip route-cache
no ip mroute-cache
shutdown
!
end
```

Step 2 Use the following command to display MGCP configurations for NAS:

```
Router# show mgcp nas {dump slot port channel | info}
```

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

```
Router# show mgcp nas dump 7 0 23
```

```
Slot 7 state= Up
Port 0 state= Up
State  Idle  PortCb=0x630DE864 ss_id=0x0 handle=0x0
bearer cap=Modem call_id= conn_id=
Events req-
4d21h:
 callp=0x62D137D4 - state=MGCP_CALL_IDLE - data_call No
Endpt name=S7/DS1-0/23
```

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

```
Router# show mgcp nas info

Slot 7 state= Up
Port 0 state= Up
ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID XX XX XX
XX XX XX XX XX
Channel State Legend
NP=Not Present, OO=Out Of Service, ID=Idle, US=In Use
CI=Connection in progress, RI=In Release in progress
RO=Out Release in progress, DN=Down, SH=Shutdown
XX=Unconfigurable
```

Troubleshooting Tips

In addition, a number of **show** and **debug** commands are useful for troubleshooting the Network Access Server Package for Media Gateway Control Protocol feature. These commands are listed in the following sections:

- MGCP Troubleshooting, page 71
- Controller Troubleshooting, page 79
- Dialer Interface and Routing Troubleshooting, page 85

MGCP Troubleshooting

To display detailed information on the MGCP application and operations, use the following commands in privileged EXEC mode:

Command	Purpose
`Router# show mgcp nas info`	Displays status of the MGCP data channels. **Note** See "Example Output for show mgcp nas info Command" section on page 72.
`Router# show mgcp nas dump slot port chan`	Displays status and details about the specified MGCP data slot, port, and channel. **Note** See "Example Output for show mgcp nas dump Command" section on page 73.
`Router# show mgcp connection`	Displays active MGCP connections on the router. **Note** See "Example Output for show mgcp connection Command" section on page 73.

Command	Purpose
`Router# show xcsp slot slot-num`	Displays the status of a router slot under the control of the External Call Service Provider (XCSP) subsystem. **Note** See "Example Output for show xcsp slot Command" section on page 73.
`Router# show xcsp port slot port`	Displays the status of a port under the control of the External Call Service Provider (XCSP) subsystem. **Note** See "Example Output for show xcsp port Command" section on page 74.
`Router# show cdapi`	Displays information about the call distributor application programming interface (CDAPI), which is the internal API that provides an interface between the MGCP signaling stacks and applications. **Note** See "Example Output for show cdapi Command" section on page 74.

Example Output for show mgcp nas info Command

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

```
Router# show mgcp nas info

Slot 7 state= Up
Port 0 state= Up
ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID ID XX XX XX
XX XX XX XX XX
Channel State Legend
NP=Not Present, OO=Out Of Service, ID=Idle, US=In Use
CI=Connection in progress, RI=In Release in progress
RO=Out Release in progress, DN=Down, SH=Shutdown
XX=Unconfigurable
```

Example Output for show mgcp nas dump Command

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

```
Router# show mgcp nas dump 7 0 23

Slot 7 state= Up
Port 0 state= Up
State  Idle  PortCb=0x630DE864 ss_id=0x0 handle=0x0
bearer cap=Modem call_id= conn_id=
Events req-
4d21h:
 callp=0x62D137D4 - state=MGCP_CALL_IDLE - data_call No
Endpt name=S7/DS1-0/23
```

Example Output for show mgcp connection Command

The following is sample output from the **show mgcp connection** command for Voice over IP (VoIP) connections:

```
Router# 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
```

The following is sample output from the **show mgcp connection** command for VoAAL2 connections:

```
Router# show mgcp connection

Endpoint  Call_ID(C) Conn_ID(I)(V)cci/cid (M)ode (S)tate (C)odec (E)vent[SIFL]
(R)esult[EA]
1.aaln/S1/1   C=1,11,12    I=0x2    V=2/10     M=3    S=4,4    C=1    E=3,0,0,3     R=0,0

Total number of active calls 1
```

Example Output for show xcsp slot Command

The following is sample output from the **show xcsp slot** command:

```
Router# show xcsp slot 1

Slot 1 configured
Number of ports configured=1 slot state= Up
```

Example Output for show xcsp port Command

The following is sample output for the **show xcsp port** command:

```
Router# show xcsp port 1 0

Slot 1 configured
Number of ports configured=1 slot state= Up
===================================================
Port 0 State= Up type = 5850 24 port T1
Channel states
  0  Idle
  1  Idle
  2  Idle
  3  Idle
  4  Idle
  5  Idle
  6  Idle
  7  Idle
  8  Idle
  9  Idle
  10  Idle
  11  Idle
  12  Idle
  13  Idle
  14  Idle
  15  Idle
  16  Idle
  17  Idle
  18  Idle
  19  Idle
  20  Idle
  21  Idle
  22  Idle
  23  Idle
```

Example Output for show cdapi Command

The following is output for the **show cdapi** command:

```
Router# show cdapi

Registered CDAPI Applications/Stacks
====================================
Application TSP CDAPI Application
Application Type(s) Voice Facility Signaling
Application Level Tunnel
Application Mode Enbloc
Signaling Stack ISDN
Interface Se023
Signaling Stack ISDN
Interface Se123
Active CDAPI Calls
==================
Interface Se023
No active calls.
Interface Se123
Call ID = 0x39, Call Type = VOICE, Application = TSP CDAPI Application
CDAPI Message Buffers
=====================
Used Msg Buffers 0, Free Msg Buffers 1600
Used Raw Buffers 1, Free Raw Buffers 799
Used Large-Raw Buffers 0, Free Large-Raw Buffers 80
scarlatti1#
```

MGCP Debugging

To debug MGCP calls, events, and operations, use the following commands in privileged EXEC mode:

Command	Purpose
`Router# debug mgcp all`	Enables all MGCP debugs. **Note** See "Example Output for debug mgcp all Command" section on page 75.
`Router# debug mgcp events`	Enables MGCP events debugging, which shows information such as the following: whether the router is detected, the MGCP event that initiates a call, and the reset of an controller that is being serviced by MGCP. **Note** See "Example Output for debug mgcp events Command" section on page 75.
`Router# debug mgcp packets`	Enables debugging of MGCP packets. Useful for displaying contents of NTFY, CRCX, DLCX, and other packets. **Note** See "Example Output for debug mgcp packets Command" section on page 76.
`Router# debug mgcp parser`	Enables debugging of MGCP parser and builder. Useful to determine whether NTFY, CRCX, and other packets have the format that the router expects. **Note** See "Example Output for debug mgcp parser Command" section on page 76.
`Router# debug mgcp nas`	Enables debugging for MGCP data channels and events. **Note** See "Example Output for debug mgcp nas Command" section on page 76.
`Router# debug xcsp {all | cot | event}`	Enables reporting of the exchange of signaling information between the MGCP protocol stack and end applications, such as call switching module (CSM) and dialer. **Note** See "Example Output for debug xcsp Command" section on page 77.
`Router# debug cdapi {detail | events}`	Displays real-time information about the call distributor application programming interface (CDAPI). **Note** See "Example Output for debug cdapi Command" section on page 78.

Example Output for debug mgcp all Command

The **debug mgcp all** command and keyword would show a compilation of all this output, including the **debug mgcp voipcac** command and keyword output. Note that using the **debug mgcp all** command and keyword may severely impact network performance.

Example Output for debug mgcp events Command

The following example illustrates the output from the **debug mgcp events** command and keyword:

```
Router# debug mgcp events
```

```
Media Gateway Control Protocol events debugging is on
Router#
1w1d: MGC stat - 172.19.184.65, total=44, succ=7, failed=21
1w1d: MGCP msg 1
1w1d: remove_old_under_specified_ack:
1w1d: MGC stat - 172.19.184.65, total=44, succ=8, failed=21
1w1d: updating lport with 2427setup_ipsocket: laddr=172.29.248.193, lport=2427,
faddr=172.19.184.65, fport=2427
1w1d: enqueue_ack: ackqhead=0, ackqtail=0, ackp=1DC1D38, msg=21A037C
```

Example Output for debug mgcp packets Command

The following example illustrates the output from the **debug mgcp packets** command and keyword:

```
Router# debug mgcp packets

Media Gateway Control Protocol packets debugging is on
Router#
1w1d: MGCP Packet received -
DLCX 408631346 * MGCP 0.1
1w1d: send_mgcp_msg, MGCP Packet sent --->
1w1d: 250 408631346
<---
```

Example Output for debug mgcp parser Command

The following example illustrates the output from the **debug mgcp parser** command and keyword:

```
Router# debug mgcp parser

Media Gateway Control Protocol parser debugging is on
Router#
1w1d: -- 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_val_mandatory_parms()
- SUCCESS: mgcp_parse_packet()- END of Parsing
1w1d: -- mgcp_build_packet()-
1w1d: - mgcp_estimate_msg_buf_length() - 87 bytes needed for header
- mgcp_estimate_msg_buf_length() - 87 bytes needed after checking parameter lines
- mgcp_estimate_msg_buf_length() - 87 bytes needed after checking SDP lines
- SUCCESS: MGCP message building OK
- SUCCESS: END of building
```

Example Output for debug mgcp nas Command

The following example displays output for the **debug mgcp nas** command and keyword, with the **debug mgcp packets** command and keyword enabled as well:

```
Router# debug mgcp nas

Media Gateway Control Protocol nas pkg events debugging is on
Router# debug mgcp packets

Media Gateway Control Protocol packets debugging is on

Router#
01:49:14:MGCP Packet received -
CRCX 58 S7/DS1-0/23 MGCP 1.0
X:57
 M:nas/data
```

```
 C:3

 L:b:64, nas/bt:modem, nas/cdn:3000, nas/cgn:1000

mgcp_parse_conn_mode :string past nas = data
mgcp_chq_nas_pkg:Full string:nas/bt:modem
mgcp_chq_nas_pkg:string past slash:bt
mgcp_chq_nas_pkg:string past colon:modem
mgcp_chq_nas_pkg:Full string:nas/cdn:3000
mgcp_chq_nas_pkg:string past slash:cdn
mgcp_chq_nas_pkg:string past colon:3000
mgcp_chq_nas_pkg:Full string:nas/cgn:1000
c5400#
mgcp_chq_nas_pkg:string past slash:cgn
mgcp_chq_nas_pkg:string past colon:1000
CHECK DATA CALL for S7/DS1-0/23
 mgcpapp_xcsp_get_chan_cb -Found - Channel state  Idle

 CRCX Recv
mgcpapp_endpt_is_data:endpt S7/DS1-0/23, slot 7, port 0 chan 23
 mgcpapp_data_call_hnd:mgcpapp_xcsp_get_chan_cb -Found - Channel state  Idle
bw=64, bearer=E1,cdn=3000,cgn=1000
```

Example Output for debug xcsp Command

The following examples show output for the **debug xcsp all** command and keyword and the **debug xcsp event** command and keyword:

```
Router# debug xcsp all

xcsp all debugging is on

Router# debug xcsp event

xcsp events debugging is on

01:49:14:xcsp_call_msg:Event  Call Indication , channel state =  Idle  for slot port
channel 7
c5400# 0 23
01:49:14:xcsp_process_sig_fsm:state/event  Idle / Call Indication
01:49:14:xcsp_incall:
01:49:14:xcsp_incall CONNECT_IND:cdn=3000 cgn=1000
01:49:14:xcsp:START guard TIMER
01:49:14:xcsp_fsm:slot 7 port 0 chan 23 oldstate =  Idle  newstate= Connection
in progress mgcpapp_process_mgcp_msg PROCESSED NAS PACKAGE EVENT

01:49:14:Received message on XCSP_CDAPI
01:49:14:process_cdapi_msg :slot/port/channel 7/0/23
01:49:14: process_cdapi_msg:new  slot/port/channel 7/0/23
01:49:14:
c5400#Received CONN_RESP:callid=0x7016
01:49:14:process_cdapi:Event CONN_RESP, channel state = 8 for slot port channel 7 0 23
01:49:14:xcsp_process_sig_fsm:state/event  Connection in progress / In Call accept
 mgcpapp_xcsp_alert:
 mgcpapp_xcsp_get_chan_cb -Found - Channel state  Connection in progress

200 58  Alert
I:630AED90
<---:Ack send SUCCESSFUL

01:49:14:xcsp_fsm:slot 7 p
c5400#ort 0 chan 23 oldstate =  Connection in progress  newstate= Connection in progress
01:49:14:Received message on XCSP_CDAPI
```

```
01:49:14:process_cdapi_msg :slot/port/channel 7/0/23
01:49:14: process_cdapi_msg:new  slot/port/channel 7/0/23
01:49:14: Received CALL_CONN:callid=0x7016
01:49:14:process_cdapi:Event CONN_, channel state = 8 for slot port channel 7
0 23
01:49:14:xcsp_process_sig_fsm:state/event  Connection in progress / in call connect
 mgcpapp_xcsp_connect:
 mgcpapp_xc
c5400#sp_get_chan_cb -Found - Channel state In Use

01:49:14:STOP TIMER
01:49:14:xcsp_fsm:slot 7 port 0 chan 23 oldstate =  Connection in progress
newstate=In Use
c5400#
01:50:23:Received message on XCSP_CDAPI
01:50:23:process_cdapi_msg :slot/port/channel 7/0/23
01:50:23: process_cdapi_msg:new  slot/port/channel 7/0/23
01:50:23: Received CALL_DISC_REQ:callid=0x7016
01:50:23:process_cdapi:Event DISC_CONN_REQ, channel state = 7 for slot port
channel 7 0 23
01:50:23:xcsp_process_sig_fsm:state/event In Use / release Request
 mgcpapp_xcsp_disconnect
 mgcpapp_xcsp_get_chan_cb -Fou
c5400#nd - Channel state In Use
01:50:23:send_mgcp_msg, MGCP Packet sent --->

01:50:23:RSIP 1 *@c5400 MGCP 1.0
RM:restart
.
DLCX 4 S7/DS1-0/23 MGCP 1.0
C:3
I:630AED90
E:801 /NAS User request
<---
01:50:23:xcsp_fsm:slot 7 port 0 chan 23 oldstate = In Use  newstate=Out
Release in progress
 xcsp_restart Serial7/0:22 vc = 22
 xcsp_restart Put idb Serial7/0:22 in down state
01:50:23:MGCP Packet received -
200 4 bye

 Data call ack received callp=0x62AEEA70mgcpapp_xcsp
c5400#_ack_recv:mgcpapp_xcsp_get_chan_cb -Found - Channel state Out Release in progress

mgcpapp_xcsp_ack_recv ACK 200 rcvd:transaction id = 4 endpt=S7/DS1-0/23
01:50:23:xcsp_call_msg:Event  Release confirm , channel state = Out Release in progress
for slot port channel 7 0 23
01:50:23:xcsp_process_sig_fsm:state/event Out Release in progress/ Release confirm
01:50:23:STOP TIMER
01:50:23:xcsp_fsm:slot 7 port 0 chan 23 oldstate = Out Release in progress
newstate= Idle
```

Example Output for debug cdapi Command

The following example shows output for the **debug cdapi** command:

```
003909 ISDN Se123 RX <- SETUP pd = 8 callref = 0x06BB
003909 Bearer Capability i = 0x9090A2
003909 Channel ID i = 0xA18381
003909 Facility i =
0x9FAA068001008201008B0100A1180202274C020100800F534341524C415454492D3530303733
003909 Progress Ind i = 0x8183 - Origination address is non-ISDN
003909 Calling Party Number i = 0xA1, '50073'
```

```
003909 Called Party Number i = 0xC1, '3450070'
003909 CDAPI Se123 TX -> CDAPI_MSG_CONNECT_IND to TSP CDAPI Application call = 0x24
003909 From Appl/Stack = ISDN
003909 Call Type = VOICE
003909 B Channel = 0
003909 Cause = 0
003909 Calling Party Number = 50073
003909 Called Party Number = 3450070
003909 CDAPI Se123 TX -> CDAPI_MSG_CONNECT_RESP to ISDN call = 0x24
003909 From Appl/Stack = TSP CDAPI Application
003909 Call Type = VOICE
003909 B Channel = 0
003909 Cause = 0
003909 CDAPI-ISDN Se123 RX <- CDAPI_MSG_CONNECT_RESP from TSP CDAPI Application call =
0x24
003909 Call Type = VOICE
003909 B Channel = 0
003909 Cause = 0
003909 CDAPI Se123 TX -> CDAPI_MSG_SUBTYPE_CALL_PROC_REQ to ISDN call = 0x24
003909 From Appl/Stack = TSP CDAPI Application
003909 Call Type = VOICE
003909 B Channel = 0
003909 Cause = 0
003909 CDAPI-ISDN Se123 RX <- CDAPI_MSG_SUBTYPE_CALL_PROC_REQ from TSP CDAPI Application
call = 0x24
003909 Call Type = VOICE
003909 B Channel = 0
003909 Cause = 0
003909 ISDN Se123 TX -> CALL_PROC pd = 8 callref = 0x86BB
003909 Channel ID i = 0xA98381
```

Controller Troubleshooting

The commands in this section can be helpful in finding sources of problems with call connections and switching. The call switching module (CSM) associated with a controller contains digit collection logic that processes incoming calls for automatic number information (ANI) and dialed number identification service (DNIS) digits.

To display information on controller and CSM configuration and operation, use the following commands in privileged EXEC mode.

Command	Purpose
`Router# show controllers t1	e1 [slot/port]`	Displays whether the T1 or E1 connection between the router and switch (central office [CO] or PBX) is up or down and whether the connection is functioning properly. **Note** See "Example Output for show controllers e1 or t1 Command" section on page 80.
`Router# show voice port [slot/port]`	Displays the port state and the parameters configured on the voice ports of Cisco voice interface cards. Voice-port defaults, like all command-line interface default parameters, do not display in the output for the **show running-config** command, but they can be seen with the **show voice port** command. **Note** See "Example Output for show voice port Command" section on page 81.

Command	Purpose
`Router# show csm modem [slot/port	modem-group-number]`	Displays the CSM call statistics for a specific modem, for a group of modems, or for all modems.
`Router# debug csm modem [slot/port	group modem-group-number]`	Traces the complete sequence of switching of incoming and outgoing modem call.

Example Output for show controllers e1 or t1 Command

The following is an output example from the **show controllers e1** command on the Cisco 7500 series:

```
Router# show controllers e1

e1 0/0 is up.
Applique type is Channelized E1 - unbalanced
Framing is CRC4, Line Code is HDB3
No alarms detected.
Data in current interval (725 seconds elapsed):
0 Line Code Violations, 0 Path Code Violations
0 Slip Secs, 0 Fr Loss Secs, 0 Line Err Secs, 0 Degraded Mins
0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 0 Unavail Secs
Total Data (last 24 hours)
0 Line Code Violations, 0 Path Code Violations,
0 Slip Secs, 0 Fr Loss Secs, 0 Line Err Secs, 0 Degraded Mins,
0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 0 Unavail Secs
```

The following is an example of the **show controllers e1** display including the board identifier type:

```
Router# show controllers e1

E1 4/1 is up.
No alarms detected.
  Framing is CRC4, Line Code is hdb3
Data in current interval (0 seconds elapsed):
0 Line Code Violations, 0 Path Code Violations 0 Slip Secs, 0 Fr Loss Secs,
0 Line Err Secs, 0 Degraded Mins 0 Errored Secs, 0 Bursty Err Secs,
0 Severely Err Secs, 0 Unavail Secs
Total Data (last 79 15 minute intervals):
0 Line Code Violations, 0 Path Code Violations, 0 Slip Secs, 0 Fr Loss Secs,
0 Line Err Secs, 0 Degraded Mins, 0 Errored Secs, 0 Bursty Err Secs,
0 Severely Err Secs, 0 Unavail Secs
```

The following is an example from the **show controllers t1** command on the Cisco 7500 series routers:

```
Router# show controllers t1

T1 4/1 is up.
No alarms detected.
  Framing is ESF, Line Code is AMI, Clock Source is line
Data in current interval (0 seconds elapsed):
0 Line Code Violations, 0 Path Code Violations 0 Slip Secs, 0 Fr Loss Secs,
0 Line Err Secs, 0 Degraded Mins 0 Errored Secs, 0 Bursty Err Secs,
0 Severely Err Secs, 0 Unavail Secs
Total Data (last 79 15 minute intervals):
0 Line Code Violations, 0 Path Code Violations, 0 Slip Secs, 0 Fr Loss Secs,
0 Line Err Secs, 0 Degraded Mins, 0 Errored Secs, 0 Bursty Err Secs,
0 Severely Err Secs, 0 Unavail Secs
```

The following example shows the status of the T1 controllers connected to the Cisco AS5800 access servers:

```
Router# show controller T1
```

```
T1 1/0/0:1 is up.
No alarms detected.
Framing is ESF, Line Code is AMI, Clock Source is Line.
Data in current interval (770 seconds elapsed):
5 Line Code Violations, 8 Path Code Violations
0 Slip Secs, 0 Fr Loss Secs, 7 Line Err Secs, 0 Degraded Mins
0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 7 Unavail Secs
Total Data (last 81 15 minute intervals):
7 Line Code Violations, 4 Path Code Violations,
6 Slip Secs, 20 Fr Loss Secs, 2 Line Err Secs, 0 Degraded Mins,
0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 2 Unavail Secs
T1 1/0/1:5 is down.
Transmitter is sending remote alarm.
Receiver has loss of frame.
Framing is SF, Line Code is AMI, Clock Source is Line.
Data in current interval (770 seconds elapsed):
50 Line Code Violations, 5 Path Code Violations
0 Slip Secs, 7 Fr Loss Secs, 7 Line Err Secs, 0 Degraded Mins
0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 7 Unavail Secs
Total Data (last 81 15 minute intervals):
27 Line Code Violations, 22 Path Code Violations,
0 Slip Secs, 13 Fr Loss Secs, 13 Line Err Secs, 0 Degraded Mins,
0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 13 Unavail Secs
Router#
```

Example Output for show voice port Command

The following is sample output from the Cisco AS5800 for the **show voice port** command:

```
ISDN 1/0/0:D
Type of VoicePort is ISDN
Operation State is DORMANT
Administrative State is UP
No Interface Down Failure
Description is ""
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 16 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
```

The following example displays voice port configuration information for the digital voice port 0 located in slot 1, DS0 group 1:

```
receEive 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 DBMS
In Gain is Set to 0 dBm
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
```

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

```
receEive 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
```

The following is sample output from the show voice port command for an FXS analog voice port on a Cisco MC3810 multiservice concentrator:

```
Voice port 1/2 Slot is 1, Port is 2
Type of VoicePort is FXS
Operation State is UP
Administrative State is UP
No Interface Down Failure
Description is not set
Noise Regeneration is enabled
Non Linear Processing is enabled
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
Coder Type is g729ar8
Companding Type is u-law
Voice Activity Detection is disabled
Ringing Time Out is 180 s
Wait Release Time Out is 30 s
Nominal Playout Delay is 80 milliseconds
Maximum Playout Delay is 160 milliseconds
Analog Info Follows:
Region Tone is set for northamerica
Currently processing Voice
Maintenance Mode Set to None (not in mtc mode)
Number of signaling protocol errors are 0
Impedance is set to 600r Ohm
Analog interface A-D gain offset = -3 dB
Analog interface D-A gain offset = -3 dB
Voice card specific Info Follows:
Signal Type is loopStart
Ring Frequency is 20 Hz
Hook Status is On Hook
Ring Active Status is inactive
Ring Ground Status is inactive
Tip Ground Status is active
Digit Duration Timing is set to 100 ms
InterDigit Duration Timing is set to 100 ms
Ring Cadence are [20 40] * 100 msec
InterDigit Pulse Duration Timing is set to 500 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:

```
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
Hook Flash Duration Timing is set to 600 ms
```

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

```
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
```

Dialer Interface and Routing Troubleshooting

To obtain information on dialer interfaces, routing configuration, and routing operations, use the following commands in privileged EXEC mode.

Command	Purpose
`Router# show dialer map`	Displays configured dynamic and static dialer maps. **Note** See "Example Output for show dialer map Command" section on page 85.
`Router# show dialer`	Displays general diagnostic information about an interface configured for DDR, such as the number of times the dialer string has been successfully reached, and the idle timer and the fast idle timer values for each B channel. Current call-specific information is also provided, such as the length of a call and the number and name of the device to which the interface is currently connected. When external signaling is configured, the output also displays the CDAPI state. **Note** See "Example Output for show dialer Command" section on page 85.
`Router# show interface Dialer-num`	Shows whether the interface and protocol are *up (spoofing)*, a state in which the dialer interface pretends to be *up/up* so that associated routes remain in force and packets can be routed to the interface. **Note** See "Example Output for show interface Command" section on page 86.
`Router# show ip route`	Displays the routes known to the router, including static and dynamically learned routes. **Note** See "Example Output for show ip route Command" section on page 87.

Example Output for show dialer map Command

The following is sample output from the **show dialer map** command.

```
Router# show dialer map

Static dialer map ip 10.1.1.1 name peer_1 on Dialer1
Static dialer map ip 10.1.1.2 name peer_2 on Dialer1
BAP dialer map ip 10.1.1.2 name peer_2 on Dialer1
Dynamic dialer map ip 10.1.1.3 name peer_3 on Dialer1
BAP dialer map ip 10.1.1.3 name peer_3 on Dialer1
```

Example Output for show dialer Command

The following is sample output from the **show dialer** command for a BRI interface when dialer profiles are configured:

```
Router# show dialer interface bri 0

BRI0 - dialer type = ISDN
Dial String Successes Failures Last called Last status
0 incoming call(s) have been screened.
BRI0: B-Channel 1
Idle timer (120 secs), Fast idle timer (20 secs)
Wait for carrier (30 secs), Re-enable (15 secs)
Dialer state is data link layer up
```

```
Dial reason: ip (s=10.1.1.8, d=10.1.1.1)
Interface bound to profile Dialer0
Time until disconnect 102 secs
Current call connected 00:00:19
Connected to 5773872 (wolfman)
BRI0: B-Channel 2
Idle timer (120 secs), Fast idle timer (20 secs)
Wait for carrier (30 secs), Re-enable (15 secs)
Dialer state is idle
```

The following is sample output from the **show dialer** command for a dialer under external signaling control:

```
Router# show dialer

Se7/0:0 - dialer type = IN-BAND SYNC NO-PARITY
Rotary group 1, priority 0
Idle timer (222222 secs), Fast idle timer (20 secs)
Wait for carrier (30 secs), Re-enable (15 secs)
Dialer state is idle
Dialer cdapi state is idle <<<<<<<<<==================

Se7/0:1 - dialer type = IN-BAND SYNC NO-PARITY
Rotary group 1, priority 0
Idle timer (222222 secs), Fast idle timer (20 secs)
Wait for carrier (30 secs), Re-enable (15 secs)
Dialer state is idle
Dialer cdapi state is idle <<<<<<<<<<=================

The following is sample output from the show dialer command for an asynchronous interface:
Router# show dialer interface async 1
Async1 - dialer type = IN-BAND NO-PARITY
Idle timer (900 secs), Fast idle timer (20 secs)
Wait for carrier (30 secs), Re-enable (15 secs)
Time until disconnect 838 secs
Current call connected 0:02:16
Connected to 8986
Dial String Successes Failures Last called Last status
8986 0 0 never Defaults
8986 8 3 0:02:16 Success Defaults
```

When the show dialer EXEC command is issued for a synchronous serial interface configured for DTR dialing, output similar to the following is displayed:

```
Serial 0 - dialer type = DTR SYNC
Idle timer (120 secs), Fst idle timer (20 secs)
Wait for carrier (30 secs), Re-enable (15 secs)
Dial String Successes Failures Last called Last status
---- 1 0 1:04:47 Success DTR dialer
8986 0 0 never Defaults
```

Example Output for show interface Command

The following is sample output from the **show interface Dialer0** command:

```
Router# show interface Dialer0

Dialer0 is up (spoofing), line protocol is up (spoofing)
  Hardware is Unknown
  Internet address is 60.0.0.2/24
  MTU 1500 bytes, BW 56 Kbit, DLY 20000 usec,
     reliability 255/255, txload 1/255, rxload 1/255
  Encapsulation PPP, loopback not set
  DTR is pulsed for 1 seconds on reset
```

```
  Last input never, output never, output hang never
  Last clearing of "show interface" counters 1d17h
  Input queue: 0/75/0/0 (size/max/drops/flushes); Total output drops: 0
  Queueing strategy: weighted fair
  Output queue: 0/1000/64/0 (size/max total/threshold/drops)
     Conversations  0/0/16 (active/max active/max total)
     Reserved Conversations 0/0 (allocated/max allocated)
     Available Bandwidth 42 kilobits/sec
  5 minute input rate 0 bits/sec, 0 packets/sec
  5 minute output rate 0 bits/sec, 0 packets/sec
     0 packets input, 0 bytes
     0 packets output, 0 bytes
```

Example Output for show ip route Command

The following examples display all downloaded static routes. A P designates which route was installed using AAA route download.

```
Router# show ip route

Codes: C - connected, S - static, I - IGRP, R - RIP, M - mobile, B - BGP
D - EIGRP, EX - EIGRP external, O - OSPF, IA - OSPF inter area
N1 - OSPF NSSA external type 1, N2 - OSPF NSSA external type 2
E1 - OSPF external type 1, E2 - OSPF external type 2, E - EGP
i - IS-IS, L1 - IS-IS level-1, L2 - IS-IS level-2, * - candidate default
U - per-user static route, o - ODR, P - periodic downloaded static route
T - traffic engineered route
Gateway of last resort is 172.21.17.1 to network 0.0.0.0
172.31.0.0/32 is subnetted, 1 subnets
P 172.31.229.41 is directly connected, Dialer1 20.0.0.0/24 is subnetted, 3 subnets
P 10.1.1.0 [200/0] via 172.31.229.41, Dialer1
P 10.1.3.0 [200/0] via 172.31.229.41, Dialer1
P 10.1.2.0 [200/0] via 172.31.229.41, Dialer1
Router# show ip route static
172.27.4.0/8 is variably subnetted, 2 subnets, 2 masks
P 172.1.1.1/32 is directly connected, BRI0
P 172.27.4.0/8 [1/0] via 103.1.1.1, BRI0
S 172.31.0.0/16 [1/0] via 172.21.114.65, Ethernet0
S 10.0.0.0/8 is directly connected, BRI0
P 10.0.0.0/8 is directly connected, BRI0
172.21.0.0/16 is variably subnetted, 5 subnets, 2 masks
S 172.21.114.201/32 is directly connected, BRI0
S 172.21.114.205/32 is directly connected, BRI0
S 172.21.114.174/32 is directly connected, BRI0
S 172.21.114.12/32 is directly connected, BRI0
P 10.0.0.0/8 is directly connected, BRI0
P 10.1.0.0/8 is directly connected, BRI0
P 10.2.2.0/8 is directly connected, BRI0
S* 0.0.0.0/0 [1/0] via 172.21.114.65, Ethernet0
S 172.29.0.0/16 [1/0] via 172.21.114.65, Ethernet0
```

To debug dialer and authorization or to clear in-progress calls, use the following commands in privileged EXEC mode.

Command	Purpose
Router# **debug dialer**	Displays the activity that triggers a dial attempt. **Note** See "Example Output for debug dialer Command" section on page 88.
Router# **clear interface**	Clears a call that is in progress. In a troubleshooting situation, it is sometimes useful to clear historical statistics to track the current number of successful calls relative to failures. Use this command with care. It sometimes requires that you clear both the local and remote routers. **Note** See "Example Output for clear interface Command" section on page 88.
Router# **debug ppp negotiation**	Displays negotiation of PPP options and Network Control Protocol (NCP) parameters. **Note** See "Example Output for debug ppp negotiation Command" section on page 88.
Router# **debug ppp authentication**	Displays exchange of Challenge Handshake Authentication Protocol (CHAP) and Password Authentication Protocol (PAP) packets. **Note** See "Example Output for debug ppp authentication Command" section on page 89.

Example Output for debug dialer Command

Displays the activity that triggers a dial attempt.

```
Dialing cause: Async1: ip (s=172.16.1.111 d=172.16.2.22)
```

Example Output for clear interface Command

The following example demonstrates the use of the **clear interface** command with the RLM feature:

```
Router# clear interface loopback 1

02:48:52: rlm 1: [State_Up, rx ACTIVE_LINK_BROKEN] over link [10.1.1.1(Loopback1),
10.1.4.1]
02:48:52: rlm 1: link [10.1.1.2(Loopback2), 10.1.4.2] requests activation
02:48:52: rlm 1: link [10.1.1.1(Loopback1), 10.1.4.1] is deactivated
02:48:52: rlm 1: link [10.1.1.1(Loopback1), 10.1.4.1] = socket[10.1.1.1, 10.1.4.1]
02:48:52: rlm 1: [State_Recover, rx USER_SOCKET_OPENED] over link [10.1.1.1(Loopback1),
10.1.4.1] for user RLM_MGR
02:48:52: rlm 1: link [10.1.1.1(Loopback1), 10.1.4.1] is opened
02:48:52: rlm 1: link [10.1.1.1(Loopback1), 10.1.5.1] = socket[10.1.1.1, 10.1.5.1]
02:48:52: rlm 1: [State_Recover, rx USER_SOCKET_OPENED] over link [10.1.1.1(Loopback1),
10.1.5.1] for user RLM_MGR
02:48:52: rlm 1: link [10.1.1.1(Loopback1), 10.1.5.1] is opened
02:48:52: rlm 1: [State_Recover, rx START_ACK] over link [10.1.1.2(Loopback2), 10.1.4.2]
02:48:52: rlm 1: link [10.1.1.2(Loopback2), 10.1.4.2] is activated
02:48:52: rlm 1: [State_Up, rx LINK_OPENED] over link [10.1.1.1(Loopback1), 10.1.4.1]
```

Example Output for debug ppp negotiation Command

The following is sample output from the **debug ppp negotiation** command. This is a normal negotiation, where both sides agree on Network Control Program (NCP) parameters. In this case, protocol type IP is proposed and acknowledged.

```
Router# debug ppp negotiation
```

```
ppp: sending CONFREQ, type = 4 (CI_QUALITYTYPE), value = C025/3E8
ppp: sending CONFREQ, type = 5 (CI_MAGICNUMBER), value = 3D56CAC
ppp: received config for type = 4 (QUALITYTYPE) acked
ppp: received config for type = 5 (MAGICNUMBER) value = 3D567F8 acked (ok)
PPP Serial4: state = ACKSENT fsm_rconfack(C021): rcvd id 5
ppp: config ACK received, type = 4 (CI_QUALITYTYPE), value = C025
ppp: config ACK received, type = 5 (CI_MAGICNUMBER), value = 3D56CAC
ppp: ipcp_reqci: returning CONFACK.
(ok)
PPP Serial4: state = ACKSENT fsm_rconfack(8021): rcvd id 4
```

The following is sample output from the debug ppp negotiation command when the remote side of the connection is unable to respond to LQM requests:

```
Router# debug ppp negotiation

ppp: sending CONFREQ, type = 4 (CI_QUALITYTYPE), value = C025/3E8
ppp: sending CONFREQ, type = 5 (CI_MAGICNUMBER), value = 44B7010
ppp: sending CONFREQ, type = 4 (CI_QUALITYTYPE), value = C025/3E8
ppp: sending CONFREQ, type = 5 (CI_MAGICNUMBER), value = 44B7010
ppp: sending CONFREQ, type = 4 (CI_QUALITYTYPE), value = C025/3E8
ppp: sending CONFREQ, type = 5 (CI_MAGICNUMBER), value = 44B7010
ppp: sending CONFREQ, type = 4 (CI_QUALITYTYPE), value = C025/3E8
ppp: sending CONFREQ, type = 5 (CI_MAGICNUMBER), value = 44B7010
ppp: sending CONFREQ, type = 4 (CI_QUALITYTYPE), value = C025/3E8
ppp: sending CONFREQ, type = 5 (CI_MAGICNUMBER), value = 44B7010
ppp: sending CONFREQ, type = 4 (CI_QUALITYTYPE), value = C025/3E8
ppp: sending CONFREQ, type = 5 (CI_MAGICNUMBER), value = 44B7010
ppp: sending CONFREQ, type = 4 (CI_QUALITYTYPE), value = C025/3E8
ppp: sending CONFREQ, type = 5 (CI_MAGICNUMBER), value = 44B7010
ppp: sending CONFREQ, type = 4 (CI_QUALITYTYPE), value = C025/3E8
ppp: sending CONFREQ, type = 5 (CI_MAGICNUMBER), value = 44B7010
ppp: sending CONFREQ, type = 4 (CI_QUALITYTYPE), value = C025/3E8
ppp: sending CONFREQ, type = 5 (CI_MAGICNUMBER), value = 44B7010
ppp: sending CONFREQ, type = 4 (CI_QUALITYTYPE), value = C025/3E8
ppp: sending CONFREQ, type = 5 (CI_MAGICNUMBER), value = 44B7010
ppp: sending CONFREQ, type = 4 (CI_QUALITYTYPE), value = C025/3E8
ppp: sending CONFREQ, type = 5 (CI_MAGICNUMBER), value = 44B7010
ppp: sending CONFREQ, type = 4 (CI_QUALITYTYPE), value = C025/3E8
ppp: sending CONFREQ, type = 5 (CI_MAGICNUMBER), value = 44C1488
```

Example Output for debug ppp authentication Command

The following is sample output from the **debug ppp authentication** command. Use this debug command to determine why an authentication fails.

```
Router# debug ppp authentication

Serial0: Unable to authenticate. No name received from peer
Serial0: Unable to validate CHAP response. USERNAME pioneer not found.
Serial0: Unable to validate CHAP response. No password defined for USERNAME pioneer
Serial0: Failed CHAP authentication with remote.
Remote message is Unknown name
Serial0: remote passed CHAP authentication.
Serial0: Passed CHAP authentication with remote.
Serial0: CHAP input code = 4 id = 3 len = 48
```

Configuration Examples for Configuring Network Access Server for Media Gateway Control Protocol

This section provides the following configuration example:

- Network Access Server Package for MGCP Feature Example, page 90

Network Access Server Package for MGCP Feature Example

This example configures the Network Access Server Package for Media Gateway Control Protocol Feature on a Cisco AS5400:

```
version 12.2
no service single-slot-reload-enable
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 54iwo
!
no boot startup-test
logging rate-limit console 10 except errors
!
resource-pool disable
!
resource-pool profile service user1sample
!
voice-fastpath enable
ip subnet-zero
ip host 54ccxv 172.18.16.25
!
no ip dhcp-client network-discovery
isdn switch-type primary-ni
!
fax interface-type modem
mta receive maximum-recipients 0
!
controller T1 7/0
 framing esf
 extsig mgcp
 guard-timer 10 on-expiry reject
 linecode b8zs
 ds0-group 1 timeslots 1-24 type none service mgcp
!
controller T1 7/1
 framing esf
 linecode ami
 pri-group timeslots 1-24
!
controller T1 7/2
 framing sf
 linecode ami
!
controller T1 7/3
 framing sf
 linecode ami
!
controller T1 7/4
 framing sf
```

```
 linecode ami
!
controller T1 7/5
 framing sf
 linecode ami
!
controller T1 7/6
 framing sf
 linecode ami
!
controller T1 7/7
 framing sf
 linecode ami
!
interface Loopback0
 ip address 172.16.0.3 255.255.255.0
!
interface FastEthernet0/0
 ip address 172.18.184.183 255.255.255.0
 duplex auto
 speed auto
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface Serial0/0
 no ip address
 shutdown
 clockrate 2000000
!
interface Serial0/1
 no ip address
 shutdown
 clockrate 2000000
!
interface Serial7/1:23
 no ip address
 encapsulation ppp
 dialer rotary-group 9
 dialer-group 1
 isdn switch-type primary-ni
 isdn incoming-voice modem
 no cdp enable
!
interface Async1/00
 ip unnumbered Loopback0
 dialer in-band
 dialer map ip 172.23.0.1 234567
 dialer-group 1
!
interface Async1/01
 ip address 10.17.1.1 255.255.255.0
 encapsulation ppp
 dialer in-band
 dialer map ip 10.17.1.2 22222
 dialer-group 1
!
interface Async1/02
 no ip address
!
interface Async1/03
```

```
 no ip address
!
interface Async1/04
 no ip address
!
interface Async1/05
 no ip address
!
interface Async3/102
 no ip address
!
interface Async3/103
 no ip address
!
interface Async3/104
 no ip address
!
interface Async3/105
 no ip address
!
interface Async3/106
 no ip address
!
interface Async3/107
 no ip address
!
interface Group-Async0
 no ip address
 no group-range
!
interface Dialer1
 ip unnumbered Loopback0
 encapsulation ppp
 dialer in-band
 dialer idle-timeout 222222
 dialer map ip 172.16.0.1 name 53bxbv 1000
 dialer extsig
 dialer-group 1
 no cdp enable
 ppp authentication chap
 ppp direction dedicated
!
interface Dialer9
 ip address 10.1.1.1 255.255.255.0
 encapsulation ppp
 dialer in-band
 dialer map ip 10.1.1.2 23456
 dialer-group 1
 no cdp enable
!
ip classless
ip route 0.0.0.0 0.0.0.0 172.18.184.1
ip route 172.16.0.1 255.255.255.255 Dialer1
ip route 172.23.0.1 255.255.255.255 Async1/00
no ip http server
!
dialer-list 1 protocol ip permit
!
call rsvp-sync
!
voice-port 7/0:1
!
voice-port 7/1:D
!
```

```
mgcp
mgcp call-agent 172.18.64.242 service-type mgcp version 1.0
no mgcp timer receive-rtcp
!
mgcp profile default
 max2 retries 5
!
line con 0
 exec-timeout 0 0
 logging synchronous
line aux 0
 logging synchronous
line vty 0 4
 password mango
 login
line 1/00 1/107
 no flush-at-activation
 modem InOut
line 3/00 3/107
 no flush-at-activation
 modem InOut
!
scheduler allocate 10000 400
end
```

Tip

- See the "Additional References for MGCP and SGCP" section on page xi for related documents, standards, and MIBs.
- See the "Glossary" for definitions of terms in this guide.

Configuring SGCP RSIP and AUEP Enhancements

This section provides information on configuring the Simple Gateway Control Protocol (SGCP) Restart In Progress (RSIP) and Audit Endpoint (AUEP) Enhancements feature. The feature provides enhancements to SGCP for disconnected RSIP and audit endpoints requested by call agents.

Feature benefits include the following:

- Provides SGCP 1.5 gateways with the ability to synchronize endpoints with call agents after the disconnected procedure has occurred.

Note For more information about this and related Cisco IOS voice features, see the following:

- "Overview of MGCP and Related Protocols" on page 3
- Entire Cisco IOS Voice Configuration Library—including library preface and glossary, other feature documents, and troubleshooting documentation—at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vcl.htm.

Feature History for SGCP RSIP and AUEP Enhancements

Release	Modification
12.2(11)T	This feature was introduced on the following platforms: Cisco IAD2420 series, Cisco 2600 series, and Cisco MC3810.

Contents

Prerequisites for Configuring SGCP RSIP and AUEP Enhancements

- Configure SGCP 1.5 on the gateway.

Restrictions for Configuring SGCP RSIP and AUEP Enhancements

- This feature applies only to SGCP 1.5 gateways.
- This feature does not apply to MGCP gateways.

Information About Configuring SGCP RSIP and AUEP Enhancements

The SGCP RSIP and AUEP Enhancements feature provides additional messaging capabilities that allow an endpoint on a Simple Gateway Control Protocol (SGCP) 1.5 gateway to synchronize with a call agent after the endpoint returns to service from the disconnected procedure. The additional messaging capabilities provide the following:

- A special disconnected Restart In Progress (RSIP) message that the gateway sends to the call agent as a result of the disconnected procedure.
- Additional fields in the Audit Endpoint (AUEP) command that the call agent uses to query the endpoint's status when contact is reestablished.

Media Gateway Control Protocol (MGCP) provides this ability automatically, but it must be explicitly configured for SGCP networks, as described in the "How to Configure SGCP RSIP and AUEP Enhancements" section on page 97.

An endpoint may lose contact with its call agent because the call agent is temporarily off line or because of faults in the network. When a gateway recognizes that an endpoint has lost its communication with the call agent, it initiates the disconnected procedure. The disconnected procedure requires the endpoint to send RSIPs to the call agent and also to guarantee that the first message that the call agent sees from the endpoint is an RSIP command. The endpoint continues to attempt to send RSIPs at the intervals prescribed by the disconnected procedure until an attempt is successful. The RSIP identifies itself as an RSIP that was generated from a disconnected procedure rather than from a restart. The following output is seen on the gateway:

```
Disconnected RSIP sent from gateway

00:04:27:RSIP 7 ds1-3/2@RouterA SGCP 1.5
RM:disconnected
```

On receipt of a disconnected RSIP message, the call agent may decide to send an AUEP command to query the status of endpoints and synchronize endpoints. The SGCP RSIP and AUEP Enhancements feature provides the following additional fields of information in the AUEP:

- I—List of connection identifiers for current connections on the endpoint
- ES—Event state of the endpoint (off-hook or on-hook)

- RM—Restart method for the endpoint, which is one of the following:
  - Graceful—Endpoints are being taken out of service after a delay; the call agent should not make new connections.
  - Forced—Endpoints were abruptly taken out of service; connections were lost.
  - Restart—Endpoints with no connections will be returned to service after a delay.
  - Disconnected—Endpoints are being returned to service after the disconnected procedure.

How to Configure SGCP RSIP and AUEP Enhancements

See the following sections for configuration tasks for the SGCP RSIP and AUEP Enhancements feature. Each task in the list is identified as either required or optional.

- Configuring SGCP RSIP and AUEP Enhancements, page 97 (required)
- Verifying SGCP RSIP Configuration, page 98 (required)

Configuring SGCP RSIP and AUEP Enhancements

To configure enhanced restart and endpoint audit messaging capabilities on an SGCP gateway, use the following command in global configuration mode:

Command	Purpose
`Router(config)# mgcp sgcp disconnect notify`	Enables enhanced endpoint synchronization with a call agent after a disconnected procedure. The command is disabled by default.

Verifying SGCP RSIP Configuration

To verify your configuration, enter the **show mgcp** command. The following example shows that disconnected RSIP is enabled.

```
Router# show mgcp

MGCP Admin State ACTIVE, Oper State ACTIVE - Cause Code NONE
MGCP call-agent:172.16.193.148 Initial protocol service is SGCP 1.5
MGCP block-newcalls DISABLED
MGCP send SGCP RSIP:forced/restart/graceful DISABLED, disconnected ENABLED
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:0
MGCP Named Signalling Event (NSE) response timer:200
MGCP Network (IP/AAL2) Continuity Test timer:200
MGCP 'RTP stream loss' timer:5
MGCP request timeout 500
MGCP maximum exponential request timeout 4000
MGCP gateway port:2427, MGCP maximum waiting delay 3000
MGCP restart delay 0, MGCP vad DISABLED
MGCP rtrcac DISABLED
MGCP system resource check DISABLED
MGCP xpc-codec:DISABLED, MGCP persistent hookflash:DISABLED
MGCP persistent offhook:ENABLED, MGCP persistent onhook:DISABLED
MGCP piggyback msg ENABLED, MGCP endpoint offset DISABLED
MGCP simple-sdp DISABLED
MGCP undotted-notation DISABLED
MGCP codec type g711ulaw, MGCP packetization period 20
MGCP JB threshold lwm 30, MGCP JB threshold hwm 150
MGCP LAT threshold lmw 150, MGCP LAT threshold hwm 300
MGCP PL threshold lwm 1000, MGCP PL threshold hwm 10000
MGCP CL threshold lwm 1000, MGCP CL threshold hwm 10000
MGCP playout mode is adaptive 60, 4, 200 in msec
MGCP IP ToS low delay disabled, MGCP IP ToS high throughput disabled
MGCP IP ToS high reliability disabled, MGCP IP ToS low cost disabled
MGCP IP RTP precedence 5, MGCP signaling precedence:3
MGCP default package:line-package
MGCP supported packages:gm-package dtmf-package trunk-package line-package
        hs-package atm-package ms-package dt-package res-package
        mt-package
MGCP Digit Map matching order:shortest match
SGCP Digit Map matching order:always left-to-right
MGCP VoAAL2 ignore-lco-codec DISABLED
```

Configuration Examples for Configuring SGCP RSIP and AUEP Enhancements

This section contains the following example:

- Disconnected RSIP Messaging Example, page 99

Disconnected RSIP Messaging Example

The following example shows the configuration of disconnected RSIP messaging on a Cisco MC3810.

```
version 12.2
no parser cache
no service pad
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Router2
!
boot system tftp smithj/mc3810-r3jjl 172.16.206.10
logging buffered 2000000 debugging
no logging console
enable password lab
!
network-clock base-rate 56k
ip subnet-zero
!
no ip domain-lookup
ip host corona 172.16.206.10
ip host redlands 172.31.140.33
ip host rialto 172.16.193.147
!
voice service voip
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
no voice confirmation-tone
voice-card 0
!
controller T1 0
 mode cas
 framing esf
 clock source internal
 linecode ami
 ds0-group 0 timeslots 1-24 type fxs-ground-start
!
interface Ethernet0
 ip address 172.16.193.162 255.255.255.0
 no ip mroute-cache
!
interface Serial0
 no ip address
 no ip route-cache
 no ip mroute-cache
 shutdown
!
interface Serial1
 no ip address
 no ip route-cache
 no ip mroute-cache
 shutdown
!
interface FR-ATM20
 no ip address
 shutdown
ip classless
ip route 10.0.0.0 10.0.0.0 172.16.193.1
ip route 172.16.0.0 255.255.0.0 172.16.193.1
no ip http server
!
```

```
!
call rsvp-sync
!
voice-port 0:0
!
voice-port 1/1
!
voice-port 1/2
 description package
!
mgcp
mgcp call-agent 172.16.193.148 service-type sgcp version 1.5
mgcp sgcp disconnect notify
!
mgcp profile default
!
dial-peer cor custom
!
dial-peer voice 1 pots
 application mgcpapp
 port 1/1
!
dial-peer voice 2 pots
 application mgcpapp
 port 1/2
!
dial-peer voice 3 pots
 application mgcpapp
 port 0:0
!
gatekeeper
 shutdown
!
line con 0
 exec-timeout 0 0
line aux 0
line 2 3
line vty 0 4
 exec-timeout 0 0
 password hemet
 login
!
end
```

Tip

- See the Additional References for MGCP and SGCP, page xi for related documents, standards, and MIBs.
- See "Glossary" for definitions of terms in this guide.

Configuring MGCP Gateway Support for the mgcp bind Command

This section provides information on configuring the MGCP Gateway Support for the **mgcp bind** Command feature.

Feature benefits include the following:

- Media gateway controller-to-media gateway (MGC-to-MG) signaling and identification

  The command allows you to use a loopback interface IP address for sourcing MGCP packets, which is transparent to any interface failure.
- Security of the media gateway

  The command allows you to obtain a predefined interface for both MGCP and media control, which can be used for security configuration.
- Possible clash of voice and dial addressing

  This feature allows you to assign a media bind interface other than loopback 0, which allows dial calls to conserve IP addresses.
- No interface diversity using routing and reduced MGCP voice diversity

  You can use routing capability more efficiently if you configure the loopback interface for control. Using the command to configure the loopback interface helps in creating redundant MGCP control or media interface.
- MGCP backward compatibility

  This feature is backward compatible with earlier MGCP features.

Note For more information about this and related Cisco IOS voice features, see the following:

- "Overview of MGCP and Related Protocols" on page 3
- Entire Cisco IOS Voice Configuration Library—including library preface and glossary, other feature documents, and troubleshooting documentation—at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vcl.htm.

Feature History for MGCP Gateway Support for the mgcp bind Command

Release	Modification
12.2(13)T	This feature was introduced.

Contents

Prerequisites for Configuring MGCP Gateway Support for the mgcp bind Command

The current Media Gateway Control Protocol (MGCP) implementation does not allow the assignment of particular IP addresses for sourcing MGCP commands and media packets, which can cause firewall and security problems. This feature allows you to configure interfaces on which control and media packets can be exchanged. This new functionality allows you to separate signaling from voice by binding control (MGCP signaling) and media (Real-Time Transport Protocol, or RTP voice, fax, and modem) to specific gateway interfaces.

This feature includes new commands that can be used to configure the required interface for MGCP control and control of the required media packets.

- Configure MGCP and the interface that will be bound. See the flow charts in the "Information About MGCP Gateway Support for the mgcp bind Command" section on page 102 to understand the behavior of the configuration command depending on interface status.
- Set the bind address before using the **mgcp bind** command. For more information about using this command, see the "Configuring the MGCP Application" section on page 107.

Information About MGCP Gateway Support for the mgcp bind Command

If the media gateway (MG) uses an IP address, which is the outgoing interface of the MG, the media gateway controller (MGC) identifies the MG entity with that address. If that interface fails, MG sources MGCP from another interface, which is not known to the MGC. Some form of name lookup (host or Domain Name System, or DNS) needs to occur on the MGC at this time. Using the **mgcp bind** command, a loopback interface IP address can be used for sourcing MGCP packets, which is transparent to any interface failure.

Present implementation of MGCP media uses the "loopback 0" or best available IP address in the order indicated for media. A fixed default loopback 0 address for media streams breaks the dial address pool convention used for most configurations, where dial IP addresses are assigned from the loopback 0 address range. With this feature, it is possible to assign a media bind interface other than loopback 0, which helps dial calls conserve IP addresses.

If you configure the loopback interface for control, you can use routing capability more efficiently. Using the **mgcp bind** command to configure the loopback interface helps in creating redundant MGCP control or media interface.

In the current implementation of MGCP, the source address of MGCP and media control is given by the IP layer. Because of this inconsistency, it is not possible to include a reliable access list or firewall configuration. Using the **mgcp bind** command for both MGCP and media control, you can get a predefined interface or IP address that can be used for security configuration.

Figure 9 on page 104 shows a typical configuration flow using the **mgcp bind** command.

Figure 9 Bind Configuration Flowchart

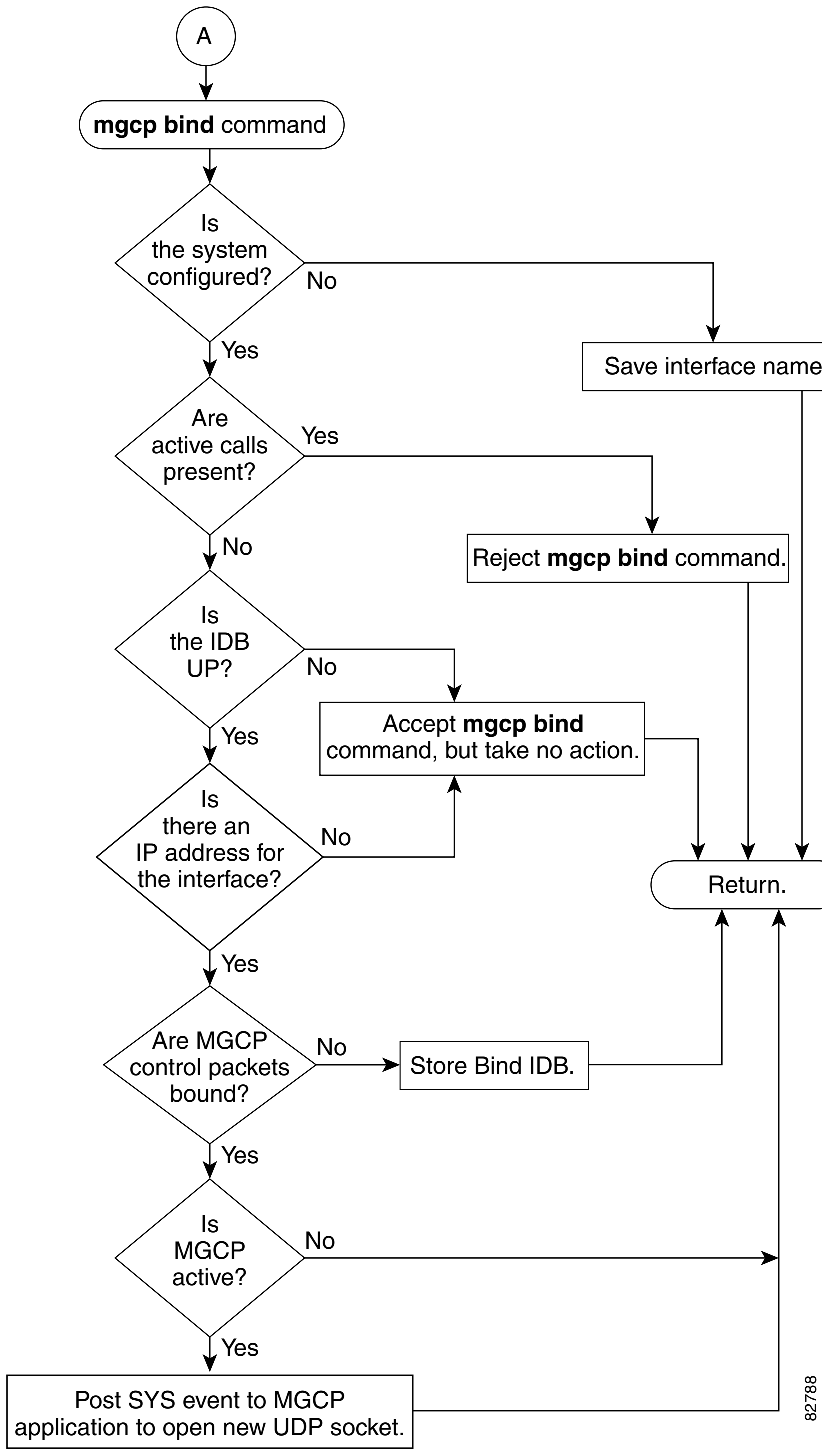
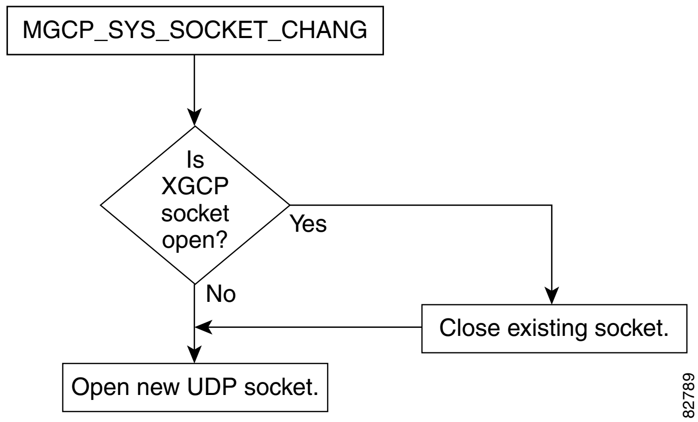


Figure 10 on page 105 shows how the **mgcp bind** command takes effect for MGCP control. When the **mgcp bind** command is configured for MGCP control, the MGCP_SYS_SOCKET_CHANG system event is posted to MGCPAPP. This event is processed by opening a new socket based on the configured interface.

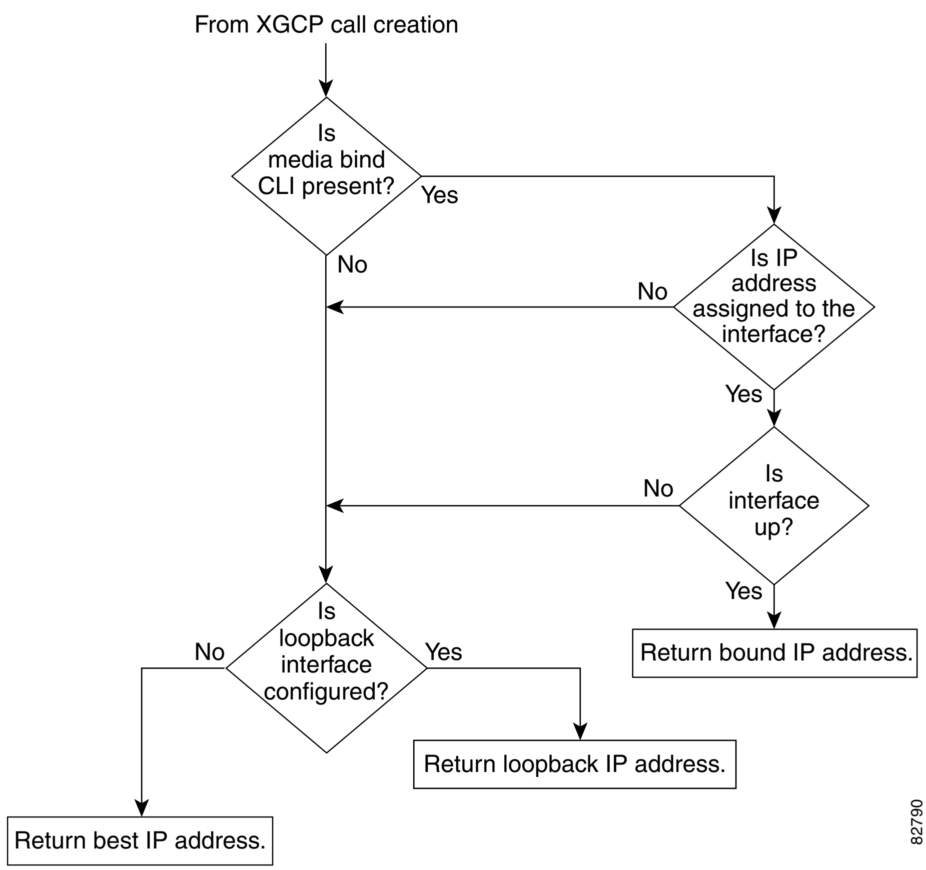
Figure 10 Bind Configuration for Control Flowchart



The time frame for execution of the **mgcp bind** command for media is different from that for control. Figure 11 on page 106 shows how the **mgcp bind** command is used for media. In this case, the IP address used for media Session Description Protocol (SDP) negotiation is taken from the configured interface. This flow is not active until an MGCP call is created.

The function call to get an IP address for the media returns a configured interface IP address, a loopback interface IP address, or a best available IP address in the order specified in the figure.

Figure 11 Bind Configuration for Media Flowchart



How to Configure MGCP Gateway Support for the mgcp bind Command

This section contains the following procedures. Each procedure is identified as either required or optional.

- Configuring the MGCP Application, page 107 (required)
- Configuring the bind Command, page 108 (required)
- Verifying MGCP Gateway Support for the Bind Command, page 111 (optional)

Configuring the MGCP Application

To configure the MGCP application, use the following commands.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **mgcp call-agent** {*dns-name* | *ip-address*} [*port*] [**service-type** *type*] [**version** *protocol-version*]
4. **mgcp**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**mgcp call-agent** {*dns-name* | *ip-address*} [*port*] [**service-type** *type*] [**version** *protocol-version*] **Example:** `Router(config)# mgcp call-agent 209.165.200.225 service-type mgcp version 1.0`	Configures the MGCP protocol and corresponding call agent.
Step 4	**mgcp** **Example:** `Router(config)# mgcp`	Enables MGCP on the gateway.
Step 5	**exit** **Example:** `Router(config)# exit`	Exits the current mode.

Configuring the bind Command

To configure the **mgcp bind** command, use the following commands.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **mgcp bind** {**control** | **media**} **source interface** *interface-id*
4. **mgcp**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`mgcp bind`** {**`control`** | **`media`**} **`source-interface`** *`interface-id`* **Example:** `Router(config)# mgcp bind {control} source-interface FastEthernet`	Sets a source interface for signaling and media packets.
Step 4	**`mgcp`** **Example:** `Router(config)# mgcp`	Enables MGCP on the gateway.
Step 5	**`exit`** **Example:** `Router(config)# exit`	Exits the current mode.

Troubleshooting Tips

To troubleshoot the MGCP Gateway Support for the Bind Command feature, perform the following:

- Use the **debug mgcp** command to enable debug traces for MGCP errors, events, media, packets, parser, and call admission control (CAC). See "Example Output for debug mgcp Command" section on page 109.

Example Output for debug mgcp Command

The following example illustrates the output for the **debug mgcp all** command with the all keyword:

```
Router# debug mgcp all

20:54:13: MGC stat - 192.168.10.10, total=37, succ=28, failed=8
20:54:13: MGCP Packet received -
CRCX 55560 s0/ds1-0/1 SGCP 1.1
C: 78980
M: sendrecv
L: a:G.726-16
20:54:13: -- mgcp_parse_packet() - call mgcp_parse_header
- mgcp_parse_header()- Request Verb FOUND CRCX
- mgcp_parse_packet() - out mgcp_parse_header
- SUCCESS: mgcp_parse_packet()-MGCP Header parsing was OK
- mgcp_parse_parameter_lines(), code_str:: 78980, code_len:2, str:1640150312
- mgcp_parse_parameter_lines(str:C: 78980) -num_toks: 19
- mgcp_parse_parameter_lines() check NULL str(78980), in_ptr(C: 78980)
- mgcp_parse_parameter_lines() return Parse function in
mgcp_parm_rules_array[1]
- mgcp_parse_call_id(in_ptr: 78980)
- SUCCESS: mgcp_parse_call_id()-Call ID string(78980) parsing is OK
- mgcp_parse_parameter_lines(), code_str:: sendrecv, code_len:2, str:1640150312
- mgcp_parse_parameter_lines(str:M: sendrecv) -num_toks: 19
- mgcp_parse_parameter_lines() check NULL str(sendrecv), in_ptr(M: sendrecv)
- mgcp_parse_parameter_lines() return Parse function in
mgcp_parm_rules_array[6]
- mgcp_parse_conn_mode(in_ptr: sendrecv)
- mgcp_parse_conn_mode()- tmp_ptr:(sendrecv)
- mgcp_parse_conn_mode(match sendrecv sendrecv
- mgcp_parse_conn_mode(case MODE_SENDRECV)
- SUCCESS: Connection Mode parsing is OK
- mgcp_parse_parameter_lines(), code_str:: a:G.726-16, code_len:2,
str:1640150312
- mgcp_parse_parameter_lines(str:L: a:G.726-16) -num_toks: 19
- mgcp_parse_parameter_lines() check NULL str(a:G.726-16), in_ptr(L:
a:G.726-16)
- mgcp_parse_parameter_lines() return Parse function in mgcp_parm_rules_array[5]
- mgcp_parse_con_opts()
- mgcp_parse_codecs()
- SUCCESS: CODEC strings parsing is OK- SUCCESS: Local Connection option
parsing is OK- mgcp_val_mandatory_parms()
20:54:13: - SUCCESS: mgcp_parse_packet()- END of Parsing
20:54:13: MGCP msg 1
20:54:13: mgcp_search_call_by_endpt: endpt = s0/ds1-0/1, new_call = 1
20:54:13: slot=0,ds1=0,ds0=1
20:54:13: search endpoint - New call=1, callp 61C28130
20:54:13: callp: 61C28130, vdbptr: 0, state: 0
20:54:13: mgcp_remove_old_ack:
20:54:13: mgcp_idle_crcx: get capability
passthru is 3
20:54:13: process_request_ev- callp 61C28130, voice_if 61C281A4
20:54:13: process_detect_ev- callp 61C28130, voice_if 61C281A4
process_signal_ev- callp 61C28130, voice_ifp 61C281A4
20:54:13: mgcp_process_quarantine_mode- callp 61C28130, voice_if 61C281A4
20:54:13: mgcp_process_quarantine_mode- new q mode: process=0, loop=0
20:54:13: mgcp_xlat_ccapi_error_code - ack_code_tab_index = 0,
20:54:13: No SDP connection info
20:54:13: mgcp_select_codec - LC option, num codec=1, 1st codec=5
20:54:13: mgcp_select_codec - num supprt codec=11
20:54:13: mgcp_select_codec - LC codec list only
20:54:13: codec index=0, bw=16000, codec=5
20:54:13: selected codec=5mgcp_get_pkt_period: voip_codec=2, pkt_period=0, call
```

```
adjust_packetization_period
mgcp_get_pkt_period: voip_codec=2, pkt_period=10, after calling
adjust_packetization_period
20:54:13: selected codec 5
20:54:13: IP Precedence=60
20:54:13: MGCP msg qos value=0mgcp_get_pkt_period: voip_codec=2, pkt_period=0,
call adjust_packetization_period
mgcp_get_pkt_period: voip_codec=2, pkt_period=10, after calling
adjust_packetization_period
mgcp_new_codec_bytes: voip_codec=2, pkt_period=10, codec_bytes=20
20:54:13: callp : 61C28AE8, state : 2, call ID : 40, event : 5, minor evt:
1640137008
20:54:13: MGCPAPP state machine: state = 2, event = 5
20:54:13: mgcp_call_connect: call_id=40, ack will be sent later.
20:54:13: callp : 61C28AE8, new state : 3, call ID : 40
20:54:14: xlate_ccapi_ev - Protocol is SGCP, change pkg=2
20:54:14: MGCP Session Appl: ignore CCAPI event 22, callp 61C28130
20:54:14: xlate_ccapi_ev - Protocol is SGCP, change pkg=2
20:54:14: callp : 61C28130, state : 2, call ID : 39, event : 5, minor evt: 20
20:54:14: MGCPAPP state machine: state = 2, event = 5
20:54:14: callp : 61C28130, new state : 3, call ID : 39
20:54:14: xlate_ccapi_ev - Protocol is SGCP, change pkg=2
20:54:14: callp : 61C28130, state : 3, call ID : 39, event : 6, minor evt: 20
20:54:14: MGCPAPP state machine: state = 3, event = 6
20:54:14: call_id=39, mgcp_ignore_ccapi_ev: ignore 6 for state 3
20:54:14: callp : 61C28130, new state : 3, call ID : 39
20:54:14: MGCP voice mode event
20:54:14: xlate_ccapi_ev - Protocol is SGCP, change pkg=2
20:54:14: callp : 61C28130, state : 3, call ID : 39, event : 17, minor evt: 0
20:54:14: MGCPAPP state machine: state = 3, event = 17
20:54:14: mgcp_voice_mode_done(): callp 61C28130, major ev 17,
minor ev 0mgcp_start_ld_timer: timer already initialized
20:54:14: send_mgcp_create_ack
20:54:14: map_mgcp_error_code_to_string error_tab_index = 0, protocol version:
2
20:54:14: MGC stat - 1.13.89.3, total=37, succ=29, failed=8
20:54:14: Codec Cnt, 1, first codec 5
20:54:14: First Audio codec, 5, local encoding, 96
20:54:14: -- mgcp_build_packet()-
20:54:14: - mgcp_estimate_msg_buf_length() - 87 bytes needed for header
- mgcp_estimate_msg_buf_length() - 125 bytes needed after checking parameter lines
- mgcp_estimate_msg_buf_length() - 505 bytes needed after cheking SDP lines
20:54:14: --- mgcp_build_parameter_lines() ---
- mgcp_build_conn_id()
- SUCCESS: Conn ID string building is OK
- SUCCESS: Building MGCP Parameter lines is OK
- SUCCESS: building sdp owner id (o=) line
- SUCCESS: building sdp session name (s=) line
- SUCCESS: MGCP message building OK
- SUCCESS: END of building
updating lport with 2427
20:54:14: send_mgcp_msg, MGCP Packet sent --->
200 55560
I: 10
v=0
o=- 78980 0 IN IP4 192.168.10.9
s=Cisco SDP 0
c=IN IP4 192.168.10.9
t=0 0
m=audio 16444 RTP/AVP 96
a=rtpmap:96 G.726-16/8000/1
<---
20:54:14: enqueue_ack: voice_if=61C281A4, ackqhead=0, ackqtail=0,
ackp=61D753E8, msg=61D00010
```

```
20:54:14:
mgcp_process_quarantine_after_ack:ack_code=200mgcp_delete_qb_evt_q:cleanup QB
evt q
20:54:14: callp : 61C28130, new state : 4, call ID : 39
```

Verifying MGCP Gateway Support for the Bind Command

To verify configuration, use the following commands.

SUMMARY STEPS

1. **show mgcp**
2. **show ip socket**
3. **show running-configuration**

DETAILED STEPS

	Command	Purpose
Step 1	`Router# show mgcp`	Checks your configuration.
Step 2	`Router# show ip socket`	Displays IP socket information.
Step 3	`Router# show running-configuration`	Verifies bind functionality.

Configuration Examples for MGCP Gateway Support for the Bind Command

The following is partial output from the **show running-configuration** command indicating that bind is functional on receiving router 172.18.192.204. Updated output for MGCP binding is highlighted under the voice service VoIP indicator.

```
ip subnet-zero
ip ftp source-interface Ethernet0
!
voice service voip
mgcp bind control source-interface FastEthernet0
mgcp bind media source-interface FastEthernet0
!
interface FastEthernet0
 ip address 172.18.192.204 255.255.255.0
 duplex auto
 speed auto
 fair-queue 64 256 1000
 ip rsvp bandwidth 75000 100
!
```

Tip

- See the "Additional References for MGCP and SGCP" section on page xi for related documents, standards, and MIBs.
- See "Glossary" for definitions of terms in this guide.

Configuring MGCP CAS MD Package

Revised: October 31, 2005

This chapter provides information on configuring the MGCP channel-associated signaling (CAS) MD Package feature. This feature introduces support for Feature Group D (FGD) Exchange Access North American (EANA) protocol signaling. The CAS MD package adds support for the reporting of automatic number identification (ANI) and dialed number identification service (DNIS) digits to enable the MGCP call agent to better handle customer billing.

Note For more information about this and related Cisco IOS voice features, see the following:

- "Overview of MGCP and Related Protocols" on page 3
- Entire Cisco IOS Voice Configuration Library—including library preface and glossary, other feature documents, and troubleshooting documentation—at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/voice_c/vcl.htm.

Feature History for MGCP CAS MD Package

Release	Modification
12.4(4)T	This feature was introduced on the Cisco AS5850.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at http://www.cisco.com/go/fn. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

Contents

Prerequisites for MGCP CAS MD Package

Prerequisites are described in "Prerequisites for Configuring MGCP and Related Protocols" on page 3.

Restrictions for MGCP CAS MD Package

- The CAS MD package is supported only on the Cisco AS5850 universal gateway.
- FGD Exchange Access International (EAIN) signaling is not supported.

Information About MGCP CAS MD Package

To configure the MGCP CAS MD Package feature, you should understand the following concept:

- MD Package, page 114

MD Package

The MD package supports the FGD EANA protocol for T1 CAS interfaces as defined in RFC 3064. It includes support for ANI and DNIS reporting that enables the MGCP call agent to improve its handling of customer billing. The MD package is enabled automatically when a T1 interface is configured using the **ds0-group** command with the **fgd-eana** keyword. The order in which the voice gateway sends the ANI and DNIS digits can be controlled by using the **notify** command in the MGCP profile.

How to Configure the MGCP CAS MD Package

This section contains the following tasks for configuring the MGCP CAS MD package:

- Configuring the Incoming Called Number in the MGCP Dial Peer, page 115 (required)
- Modifying ANI and DNIS Order when Using CAS MD Package, page 116 (optional)

Note You do not have to enable the CAS MD package with the **mgcp package-capability** command. The CAS MD package is enabled automatically when a T1 controller is configured for FGD EANA signaling using the **ds0-group** command.

Configuring the Incoming Called Number in the MGCP Dial Peer

Perform this procedure to specify the dial string to use for matching incoming calls to the MGCP dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **pots**
4. **service mgcpapp**
5. **incoming called-number** *string*
6. **port** *port*
7. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`dial-peer voice`** *`tag`* **`pots`** **Example:** `Router(config)# dial-peer voice 1003 pots`	Defines a dial peer as a POTS device and enters dial-peer configuration mode.
Step 4	**`service mgcpapp`** **Example:** `Router(config-dial-peer)# service mgcpapp`	Enables MGCP on the dial peer. **Note** Do not use this command in dial peers that support PRI backhaul or BRI backhaul.
Step 5	**`incoming called number`** *`string`* **Example:** `Router(config-dial-peer)# incoming called number .`	Specifies the digit string that is used to match incoming calls to the dial peer.

	Command or Action	Purpose
Step 6	**port** *port* **Example:** `Router(config-dial-peer)# port 0/0:3:0`	Binds the MGCP application to the specified voice port.
Step 7	**end** **Example:** `Router(config-dial-peer)# end`	Exits to privileged EXEC mode.

Modifying ANI and DNIS Order when Using CAS MD Package

Perform this procedure to specify the order in which ANI and DNIS digits are sent in notify messages to the call agent when using the CAS MD package.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **mgcp profile** {*profile-name* | **default**}
4. **notify** {**ani-dnis** | **dnis-ani**}
5. **end**
6. **show mgcp profile** [*profile-name*]

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**mgcp profile** {*profile-name* | **default**} **Example:** `Router(config)# mgcp profile default`	Defines an MGCP profile to be associated with one or more MGCP endpoints.

	Command or Action	Purpose
Step 4	**notify** {**ani-dnis** | **dnis-ani**} **Example:** `Router(config-mgcp-profile)# notify dnis-ani`	Specifies the order in which ANI and DNIS digits are reported to the MGCP call agent. • **ani-dnis**—ANI digits are sent in the first notify message. This is the default order. • **dnis-ani**—DNIS digits are sent in the first notify message.
Step 5	**end** **Example:** `Router(config-mgcp-profile)# end`	Exits to privileged EXEC mode.
Step 6	**show mgcp profile** [*profile-name*] **Example:** `Router# show mgcp profile default`	Displays configuration information for MGCP profiles including the setting of the **notify** command.

Configuration Examples for MGCP CAS MD Package

This section contains the following configuration examples:

- CAS MD Package Configuration: Example, page 117
- Cisco AS5850 Configuration: Example, page 118

CAS MD Package Configuration: Example

The following example shows the significant portions of a configuration for the CAS MD package.

```
...
controller T1 0/0:3
 framing esf
 ds0-group 0 timeslots 1 type fgd-eana mf ani-dnis
!
controller T1 0/0:4
 framing esf
 ds0-group 0 timeslots 1 type fgd-eana mf ani-dnis
...
mgcp profile default
 notify dnis-ani
!
!
dial-peer voice 1003 pots
 service mgcpapp
 incoming called-number .
 port 0/0:3:0
!
dial-peer voice 1004 pots
 service mgcpapp
 incoming called-number .
 port 0/0:4:0
...
```

Cisco AS5850 Configuration: Example

The following example shows a complete running configuration for a Cisco AS5850 universal gateway that is using the CAS MD package.

```
Current configuration : 2636 bytes
!
version 12.4
no service pad
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
service internal
!
hostname Sample
!
boot-start-marker
boot system flash:c5850-p9-mz
boot-end-marker
!
!
redundancy
 mode classic-split
logging buffered 20000000 debugging
no logging console
enable password temp
!
no aaa new-model
!
resource policy
!
!
!
resource-pool disable
dial-tdm-clock priority 1 external t1 120ohm
spe default-firmware spe-firmware-1
!
!
!
!
ip subnet-zero
ip cef distributed
!
!
!
isdn switch-type primary-5ess
!
!
!
!
!
!
controller T3 0/0
 framing c-bit
 cablelength 224
 t1 1-7 controller
!
controller T1 0/0:1
 shutdown
 framing sf
 ds0-group 0 timeslots 1 type fgd-eana mf ani-dnis
!
```

```
controller T1 0/0:2
 shutdown
 framing sf
 ds0-group 0 timeslots 1 type fgd-eana mf ani-dnis
!
controller T1 0/0:3
 framing esf
 ds0-group 0 timeslots 1 type fgd-eana mf ani-dnis
!
controller T1 0/0:4
 framing esf
 ds0-group 0 timeslots 1 type fgd-eana mf ani-dnis
!
controller T1 0/0:5
 framing esf
!
controller T1 0/0:6
 shutdown
 framing esf
!
controller T1 0/0:7
 framing esf
!
!
!
interface Loopback0
 no ip address
 no ip route-cache cef
 no ip route-cache distributed
 no ip route-cache
!
interface FastEthernet6/0
 ip address 172.16.0.46 255.255.255.0
 no ip proxy-arp
 logging event link-status
 speed 100
 full-duplex
 no keepalive
!
interface GigabitEthernet6/0
 no ip address
 logging event link-status
 shutdown
 negotiation auto
!
interface GigabitEthernet6/1
 no ip address
 logging event link-status
 shutdown
 negotiation auto
!
interface Group-Async0
 no ip address
 encapsulation ppp
 group-range 0/00 3/323
!
!
ip classless
ip route 0.0.0.0 0.0.0.0 172.16.0.200
no ip http server
!
!
!
!
```

```
!
!
voice-port 0/0:1:0
!
voice-port 0/0:2:0
!
voice-port 0/0:3:0
!
voice-port 0/0:4:0
!
mgcp
mgcp call-agent 172.16.0.200 18384 service-type mgcp version 0.1
mgcp package-capability dtmf-package
mgcp package-capability mf-package
mgcp package-capability rtp-package
no mgcp piggyback message
mgcp persistent onhook
mgcp fax t38 inhibit
!
mgcp profile default
!
!
dial-peer voice 1003 pots
 service mgcpapp
 incoming called-number .
 port 0/0:3:0
!
dial-peer voice 1004 pots
 service mgcpapp
 incoming called-number .
 port 0/0:4:0
!
!
!
line con 0
 exec-timeout 0 0
 transport output all
line aux 0
 exec-timeout 0 0
 transport output all
line vty 0 4
 exec-timeout 0 0
 privilege level 15
 no login
 transport input all
 transport output all
line 0/00 0/215
 modem InOut
 transport input all
line 3/00 3/323
 modem InOut
 transport input all
!
end
```

Appendix A: Configuring MGCP CAS PBX and AAL2 PVC

This section provides information on configuring the MGCP Channel-Associated Signaling (CAS) Private-Branch-Exchange (PBX) and ATM Adaptation Layer 2 (AAL2) Permanent Virtual Circuit (PVC) feature.

Feature benefits include the following:

- The merged Simple Gateway Control Protocol/Media Gateway Control Protocol (SGCP/MGCP) software for residential gateways (RGWs), business gateways (BGWs), and trunking gateways (TGWs) enables easier development and growth of Cisco and customer solutions.
- MGCP CAS PBX and AAL2 PVC software meets customer requirements for CAS connectivity to traditional PBXs and regulatory requirements for support of 911, Barge In, and Busy Line Verify features.

Note For more information about this and related Cisco IOS voice features, see the following:

- "Overview of MGCP and Related Protocols" on page 3
- Entire Cisco IOS Voice Configuration Library—including library preface and glossary, other feature documents, and troubleshooting documentation—at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vcl.htm.

Feature History for MGCP CAS PBX and AAL2 PVC

Release	Modification
12.1(5)XM	This feature was introduced on the following platforms: Cisco 1750, Cisco 2600 series, Cisco 3600 series, Cisco AS5300, Cisco MC3810, and Cisco uBR924.
12.2(2)T	This feature was integrated into this release on all previously supported platforms except the Cisco AS5300. A new command was added (mgcp rtp unreachable timeout) and an existing command was modified (mgcp sdp).
12.2(11)T	This feature was implemented on the Cisco AS5300 and Cisco AS5850. **Note** AAL2 PVC is not supported on the Cisco AS5850.

Contents

Prerequisites for Configuring MGCP CAS PBX and AAL2 PVC

Prerequisites are described in the "Prerequisites for Configuring MGCP and Related Protocols" section on page 3.

Restrictions for Configuring MGCP CAS PBX and AAL2 PVC

MGCP CAS PBX and AAL2 PVC Software Caveats

- Only the Cisco MC3810 series platform supports MGCP 0.1 control of AAL2 voice transport in this Cisco IOS release.
- For the Cisco MC3810 series platform, the AAL2 PVC functionality is supported on an high-performance compression module (HCM) version of an digital signal processor (DSP) card; it is not supported on an voice compression module (VCM) version.

  To check the type of DSP card, enter a **show version** command at the EXEC prompt. For example:

  ```
  Router# show version
  ```

  - If you have an HCM card, the following line appears as part of the **show version** information:

    ```
    1 6-DSP (slot 2) High Performance Compression Module(v01.A0)
    ```

  - If you have an VCM card, the following line appears as part of the **show version** information:

    ```
    1 6-DSP (slot 2) Voice Compression Module(v255.V7)
    ```

  If you have an HCM card, the MGCP Basic CLASS and Operator Services (BCOS) features will function correctly. If you have an VCM card, the AAL2 PVC feature is not supported.

- The Cisco AS5300 multiservice platform supports only the Feature Group-D Operator Services (FGD-OS) Barge-In/Busy Line Verify and 911 features of the MGCP CAS PBX and AAL2 PVC software.

Features Not Supported

Basic CLASS and Operator features are covered in the MGCP Basic CLASS and Operator Services software. For more information on these capabilities, see *Configuring MGCP Basic CLASS and Operator Services*.

- The MGCP CAS PBX and AAL2 PVC software has not implemented DSP clock slotting changes, Comfort Noise Indication, ATM SVCs, TGCP, AAL1, FXO support in SGCP, ATM on the Cisco 3660 platform, and VoIP Call Admission Control (CAC). These capabilities are part of other Cisco development efforts.

Information About Configuring MGCP CAS PBX and AAL2 PVC Feature

The MGCP CAS PBX and AAL2 PVC features extend the earlier Simple Gateway Control Protocol (SGCP) Channel Associated Signaling (CAS) and AAL2 support onto the merged SGCP/MGCP software base to enable various service provider solutions.

MGCP CAS PBX and AAL2 PVC Features

- CAS termination and translation to MGCP on Business Gateways (BGWs) and Trunking Gateways (TGWs).

  Digital CAS (E&M) interfaces are supported in addition to the analog (FXO, FXS, and E&M) interfaces.

  For this feature release, the BGWs are the Cisco 3810 series and Cisco 2600 series routers. The TGWs are the Cisco 3600 series multiservice platforms.
- Support for CAS PBX and Feature Group D CAS Functions.

  MGCP 0.1 has been expanded to support CAS packages that handle CAS PBX and Feature Group D CAS functions, including Barge-In/Busy Line Verify, and 911 capabilities on the TGW.
- Expanding MGCP 0.1 to control AAL2 voice transport.

  The earlier version of the merged SGCP/MGCP stack supported only Voice over IP. The merged stack will now support both VoIP and VoAAL2.

  Only the Cisco MC3810 series platforms supports this feature in this release.
- Addition of SGCP CAS PBX support to the existing merged SGCP/MGCP software stack.

  The CAS PBX gateway features include CAS PBX trunks, digit maps, CAS events, and quarantine buffer software. These features were available in the existing standalone SGCP software; now they are supported in the merged stack.
- Consolidation of various SGCP and MGCP feature sets onto one software image for Residential Gateways (RGWs), BGWs, and TGWs.

  For this feature release, the RGWs are the Cisco uBR924 cable router and Cisco 1750 access router.

The essential difference for current SGCP users is that support for the SGCP application has been replaced with the MGCP application. The MGCP application supports both SGCP commands and MGCP commands, permitting access to a larger feature set than with the SGCP application alone. The MGCP CAS PBX and AAL2 PVC software assumes the MGCP mode as the default environment. This allows the gateway to recognize both MGCP and SGCP messages sent by the call agent. However, the user can specify SGCP mode for certain messages that will be sent by the gateway (by using the **sgcp** option as the service type in the **mgcp call-agent** command).

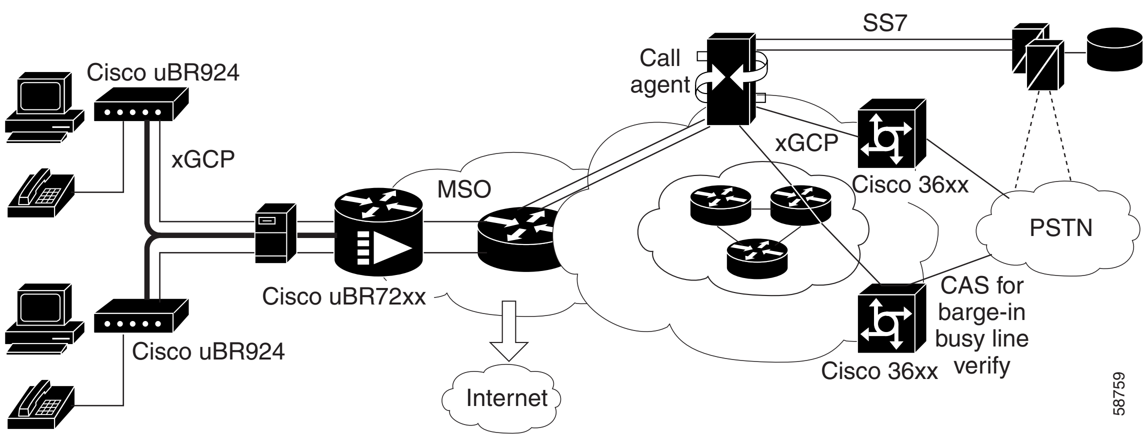
Examples of Service Provider Solutions

MGCP CAS PBX and AAL2 PVC features support several types of service provider solutions:

- Residential cable access

  A CLEC can use residential cable access to provide residential customers with basic telephony and data services. Figure 12 on page 124 illustrates a possible residential cable access solution:

Figure 12 Residential Cable Access Solution

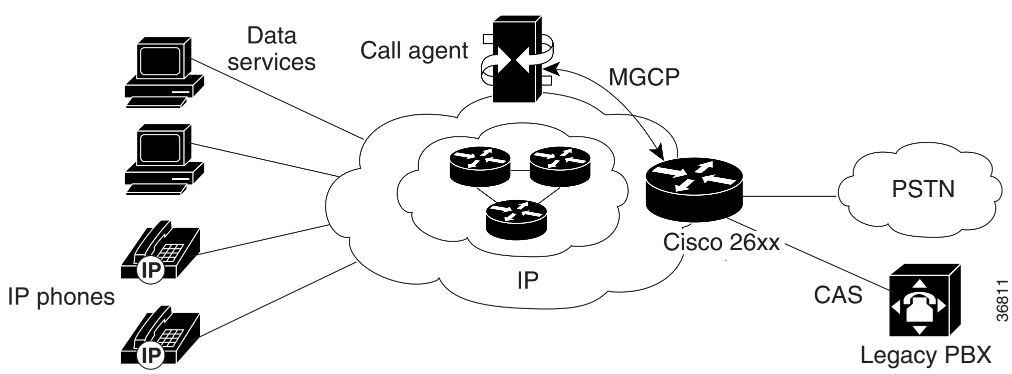


  Note that in Figure 12 on page 124, the trunking gateway (the Cisco 3660 platform) requires support of incoming and outgoing MF signaling for the Barge-In and Busy-Line Verify features. The residential gateway (the Cisco uBR924 cable access router) must support the CLASS features and 911 capability.

- IP Centrex and IP PBX

  In these solutions, a call agent provides business voice services traditionally offered by a circuit-based PBX. Figure 13 on page 124 illustrates an IP Centrex solution:

Figure 13 IP Centrex Solution

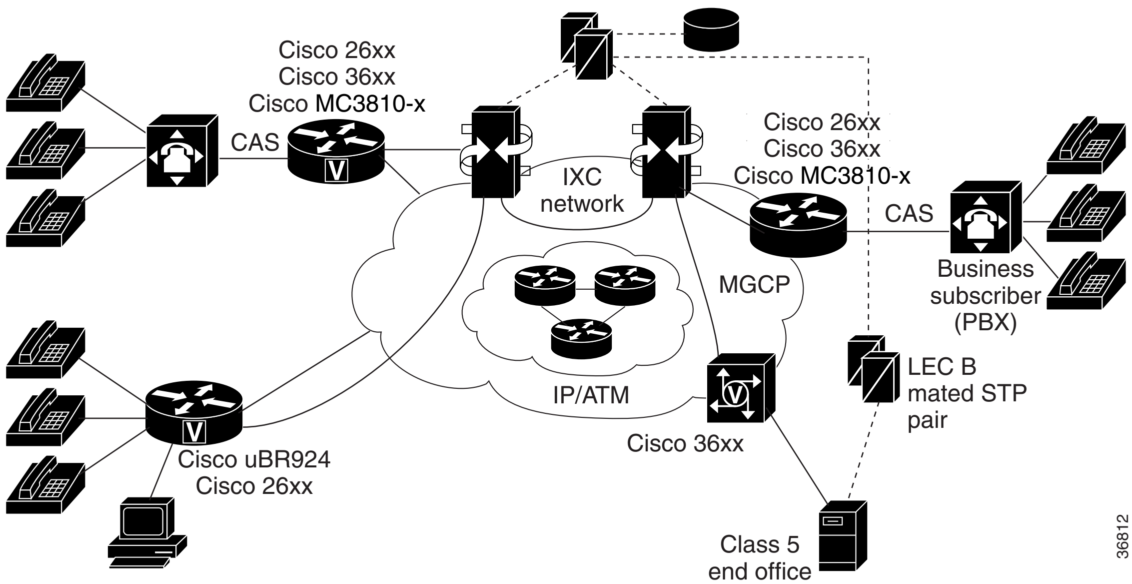


  In Figure 13 on page 124, the BGW (the Cisco 2600 platform) requires PBX connectivity to interface with the legacy PBX.

- Integrated Access

  A CLEC or IXC can provide small, medium, and large businesses with integrated voice and data access services. The integrated access device can be located at the central office or on the customer's premises. Access to the subscriber can be analog or digital T1 interfaces in addition to DSL. Transport of voice and data can be over IP, Frame Relay, or ATM. Figure 14 on page 125 illustrates an integrated access solution:

Figure 14 Integrated Access Solution

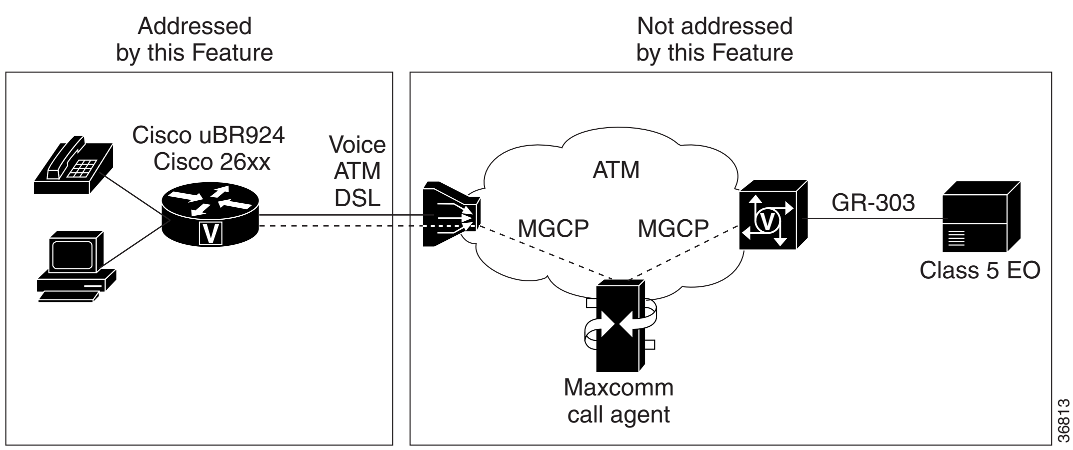


In Figure 14 on page 125, MGCP control of calls over the AAL2 PVCs is required on the BGWs (the Cisco 2600 series, Cisco 3600, and Cisco 3810 series platforms) to connect into the ATM network for VToA.

- Telecommuter or Small Office-Home Office

  Figure 15 on page 125 illustrates a telecommuter/small office-home office solution:

Figure 15 Telecommuter or Small Office-Home Office Solution



In Figure 15 on page 125, MGCP must control the calls over AAL2 PVCs, and an analog FXS interface is required.

How to Configure MGCP CAS PBX and AAL2 PVC Feature

See the following sections for configuration tasks for the MGCP CAS PBX and AAL2 PVC feature. Each task in the list indicates if the task is optional or required.

- Configuring the Gateway, page 126 (required)
- Configuring Subcell Multiplexing for AAL2 Voice, page 129 (optional)
- Configuring the Cisco uBR924 Cable Access Router for SGCP and MGCP Functionality, page 129 (optional)
- Verifying the MGCP CAS PBX and AAL2 PVC Configurations, page 130 (optional)

Some tasks indicate one or more configuration examples affected by the command. See the specific configuration example listing for the parameter values.

Configuring the Gateway

SUMMARY STEPS

1. **mgcp**
2. **mgcp call-agent** {*ipaddr* | *hostname*} [*port*] [**service-type** *type*] **version** *version-number*
3. **mgcp dtmf-relay voip codec** {**all** | **low-bit-rate**} **mode** {**cisco** | **nse** | **out-of-band**}
4. **mgcp package-capability** {**as-package** | **atm-package** | **dtmf-package** | **gm-package** | **hs-package** | **nas-package** | **rtp-package** | **script-package** | **trunk-package**}
5. **mgcp sgcp restart notify**
6. **mgcp modem passthrough** [**voip** | **voaal2**] **mode** [**cisco** | **nse**]
7. **mgcp tse payload** *type*
8. **mgcp rtp unreachable timeout**
9. **no mgcp timer receive-rtcp**
10. **mgcp timer net-cont-test** *timer*
11. **controller T1 0**
12. **mode atm**
13. **no shutdown**
14. **exit**
15. **mgcp quarantine mode process**
16. **controller T1 1**
17. **mode cas**
18. **ds0-group** *channel-number* **timeslots** *range* **type** *signaling-type* **tone** *type addr info* **service** *service-type*
19. **exit**
20. **interface atm0** [*subinterface-number* [**multipoint** | **point-to-point**]]

21. **pvc** [*name*] *vpi/vci*
22. **encapsulation** *aal-encap*
23. **vbr-rt** *peak-rate average-rate* [*burst*]
24. **vcci** *pvc-identifier*
25. **exit**
26. **exit**
27. **dial-peer voice** *number* **pots**
28. **application MGCPAPP**
29. **exit**

DETAILED STEPS

	Command	Purpose
Step 1	`Router(config)# mgcp`	Starts the MGCP daemon.
Step 2	`Router(config)# mgcp call-agent {ipaddr	hostname} [port] [service-type type] version version-number`	Configures the MGCP call agent and service type. If you want SGCP mode, use sgcp as the service type.
Step 3	`Router(config)# mgcp dtmf-relay voip codec {all	low-bit-rate} mode {cisco	nse	out-of-band}`	(Optional. See Configuration Example 2.) Specifies compressed codecs for digit forwarding.
Step 4	`Router(config)# mgcp package-capability {as-package	atm-package	dtmf-package	gm-package	hs-package	nas-package	rtp-package	script-package	trunk-package}`	(Optional. See Configuration Example 2.) Assigns packages to the gateway. Also refer to the **mgcp default-package** command.
Step 5	`Router(config-if)# mgcp sgcp restart notify`	(Required only for SGCP mode with a call agent supporting RSIP. See Configuration Examples 4 through 9.) Causes MGCP to send SGCP RSIP messages.
Step 6	`Router(config-if)# mgcp modem passthrough [voip	voaal2] mode [cisco	nse]`	(Optional for **nse** mode) Enables the gateway to process fax or modem messages. VoAAL2 does not support **cisco**.
Step 7	`Router(config)# mgcp tse payload type`	(Required for **nse** mode. See Step 6.) Enables the TSE payload for fax and modem messages.
Step 8	`Router(config)# mgcp rtp unreachable timeout timer-value`	(Optional) Enables detection of unreachable remote VoIP endpoints.
Step 9	`Router(config)# no mgcp timer receive-rtcp`	(Required for non-RGWs. See Configuration Examples 2 through 9.) Turns off the RTP RTCP receive timeout interval at the gateway.
Step 10	`Router(config)# mgcp timer net-cont-test timer`	(Optional for non-RGWs. See Configuration Examples 2 through 9.) Turns on the continuity test timeout interval at the gateway.
Step 11	`Router(config)# controller T1 0`	(Required for ATM mode. See Configuration Examples 2 through 9.) Select s the T1 controller 0.

	Command	Purpose
Step 12	`Router(config-controller)#` **`mode atm`**	(Required for ATM mode. See Configuration Examples 2 through 9.) Specifies that the controller will support ATM encapsulation and create ATM interface 0. When the controller is set to ATM mode, the following takes place: • Controller framing is automatically set to Extended SuperFrame (ESF). • The linecode is automatically set to B8ZS.
Step 13	`Router(config-controller)#` **`no shutdown`**	(Optional for ATM mode. See Configuration Examples 2 through 9.) Ensures that the controller is activated.
Step 14	`Router(config-controller)#` **`exit`**	(Required for ATM mode. See Configuration Examples 2 through 9.) Exits the current mode.
Step 15	`Router(config)#` **`mgcp quarantine mode process`**	(Optional) Turns on processing for SGCP quarantine mode.
Step 16	`Router(config)#` **`controller T1 1`**	(Required for CAS PBX. See Configuration Examples 3, 4, and 5.) Select the T1 controller 1.
Step 17	`Router(config-controller)#` **`mode cas`**	(Required for CAS PBX. See Configuration Examples 3, 4, and 5.) Specify that the controller will support CAS.
Step 18	`Router(config-controller)#` **`ds0-group`** *`channel-number`* **`timeslots`** *`range`* **`type`** *`signaling-type`* **`tone`** *`type addr info`* **`service`** *`service-type`*	(Required for CAS PBX. See Configuration Examples 3, 4, and 5.) Configure the T1 timeslots for CAS calls.
Step 19	`Router(config-controller)#` **`exit`**	(Required for CAS PBX. See Configuration Examples 3, 4, and 5.) Exit controller configuration mode.
Step 20	`Router(config)#` **`interface atm0`** `[`*`subinterface-number`* `[`**`multipoint`** `|` **`point-to-point`**`]]`	(Required for ATM mode. See Configuration Examples 2 through 9.) Enter interface configuration mode to configure ATM interface 0 or an ATM subinterface. Default for subinterfaces is **multipoint.** *For all scenarios*: Set up three subinterfaces for point-to-point.
Step 21	`Router(config-if)#` **`pvc`** `[`*`name`*`]` *`vpi/vci`*	(Required for ATM mode. See Configuration Examples 2 through 9.) Create an ATM PVC for voice traffic and enter ATM virtual circuit configuration mode. **Note** The **ilmi** and **qsaal** options are not supported for AAL2.
Step 22	`Router(config-if-atm-vc)#` **`encapsulation`** *`aal-encap`*	(Required for ATM mode. See Configuration Examples 2 through 9.) Set the encapsulation of the PVC for voice traffic. **aal2** automatically creates channel identifiers (CIDs) 1 through 255. *Some of the Scenarios* use **aal5snap** for ATM0.1 and ATM0.3. Use **aal2** for ATM0.2.
Step 23	`Router(config-if-atm-vc)#` **`vbr-rt`** *`peak-rate average-rate`* `[`*`burst`*`]`	(Required for ATM mode. See Configuration Examples 2 through 9.) Configures the PVC for the variable-bit-rate real-time (voice) traffic.

	Command	Purpose
Step 24	`Router(config-if-atm-vc)#` **vcci** *pvc-identifier*	(Optional for ATM mode. See Configuration Examples 2 through 9.) Assigns a unique identifier to the PVC.
Step 25	`Router(config-if-atm-vc)#` **exit**	(Required for ATM mode. See Configuration Examples 2 through 9.) Exits the current mode.
Step 26	`Router(config-if)#` **exit**	(Required for ATM mode. See Configuration Examples 2 through 9.) Exits the current mode.
Step 27	`Router(config)#` **dial-peer voice** *number* **pots**	Enter dial peer configuration mode for the POTS dial peer.
Step 28	`Router(config-dial-peer)#` **application MGCPAPP**	Initiates the MGCP protocol for the voice ports.
Step 29	`Router(config-dial-peer)#` **exit**	Exits the current mode.

Configuring Subcell Multiplexing for AAL2 Voice

This section describes the configuration tasks necessary to enable AAL2 common part sublayer (CPS) subcell multiplexing when the Cisco MC3810 series platform interoperates with a voice interface service module (VISM) in an MGX switch.

SUMMARY STEPS

1. **voice service voatm**
2. **session protocol aal2**
3. **subcell-mux**
4. **end**

DETAILED STEPS

	Command	Purpose
Step 1	`Router(config)#` **voice service voatm**	(Required) Enters voice-service configuration mode.
Step 2	`Router(config-voice-service)#` **session protocol aal2**	(Required) Enters voice-service-session configuration mode and specifies AAL2 trunking.
Step 3	`Router(config-voice-service-session)#` **subcell-mux**	(Required) Enables subcell multiplexing. By default, subcell multiplexing is not enabled.
Step 4	`Router(config-voice-service-session)#` **end**	(Required) Exits the current mode.

Configuring the Cisco uBR924 Cable Access Router for SGCP and MGCP Functionality

The Cisco uBR924 cable access router requires standard per-port provisioning to work with MGCP CAS PBX and AAL2 PVC:

To access SGCP functionality, use the command:

```
S|0|ca1@call-agent.abc.com:2427|S|1|ca2@call-agent.abc.com:2427
```

To access MGCP functionality, use the command:

```
M|0|ca1@call-agent.abc.com:2427|M|1|ca2@call-agent.abc.com:2427
```

For either functionality type, port 0 points to call agent 1 and port 1 points to call agent 2. If needed, both ports can point to the same call agent.

Verifying the MGCP CAS PBX and AAL2 PVC Configurations

To verify configuration, use the following commands.

SUMMARY STEPS

1. **show dial-peer voice sum**
2. **show running-configuration**

DETAILED STEPS

	Command	Purpose
Step 1	`Router#` **`show dial-peer voice sum`**	Displays the status of the dial peer. The dial peer should be active. If it is not, use the **no shut** command to make it so.
Step 2	`Router#` **`show running-configuration`**	Displays the current configuration settings.

Configuration Examples for Configuring MGCP CAS PBX and AAL2 PVC Feature

This section provides the following configuration diagrams and software examples:

Example 1: MGCP Residential Gateway

The following example illustrates the configuration for a Cisco MC3810 series platform with CAS running the MGCP application:

```
version 12.2
no service pad
service timestamps debug datetime msec
service timestamps log uptime
!
hostname Router
!
logging buffered
!
ip subnet-zero
ip host first 192.168.254.254
!
mgcp
mgcp call-agent 172.16.90.1
!
voice-card 0
  codec complexity high
!
controller T1 0
  framing esf
  linecode b8zs
!
interface Ethernet0
  ip address 172.16.92.3 255.255.0.0
!
interface Serial0
  shutdown
!
interface Serial1
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
!
interface FR-ATM20
  no ip address
  shutdown
!
ip default-gateway 172.16.0.1
ip route 198.168.254.0 255.255.255.0 172.16.0.1
!
voice-port 1/1
!
dial-peer voice 1 pots
  application MGCPAPP
  port 1/1
!
line con 0
  exec-timeout 0 0
  transport input none
line aux 0
line 2 3
line vty 0 4
  login
!
end
```

Example 2: MGCP Gateway using Voice over ATM AAL2

The following configuration illustrates a Cisco MC3810 series platform running the MGCP application using ATM AAL2 to carry voice traffic:

```
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname main.office
!
network-clock base-rate 56k
ip subnet-zero
no ip domain-lookup
ip host second 192.168.254.254
ip host first 192.168.254.253
!
mgcp
mgcp call-agent 172.16.117.4 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode nse
mgcp dtmf-relay voaal2 codec all
mgcp package-capability rtp-package
mgcp tse payload 100
mgcp timer receive-rtcp 100
mgcp timer net-cont-test 3000
isdn voice-call-failure 0
!
voice-card 0
!
controller T1 0
  mode atm
  framing esf
  linecode b8zs
!
interface Ethernet0
  ip address 171.16.121.1 255.255.0.0
!
interface Serial0
  no ip address
  no ip mroute-cache
  shutdown
  no fair-queue
!
interface Serial1
  no ip address
  shutdown
!
interface ATM0
  no ip address
  ip mroute-cache
  no atm ilmi-keepalive

interface ATM0.2 point-to-point
  pvc 2/200
   vbr-rt 760 760 100
   encapsulation aal2
vcci 2
!
interface FR-ATM20
  no ip address
  shutdown
!
router group1 1
  redistribute connected
  network 172.0.0.0
!
ip default-gateway 172.16.0.1
no ip http server
```

```
ip classless
ip route 192.168.254.0 255.255.255.0 172.16.0.1
!
dialer-list 1 protocol ip permit
dialer-list 1 protocol ipx permit
voice-port 1/1
  codec g711ulaw
!
voice-port 1/2
  shutdown
!
voice-port 1/6
  shutdown
!
dial-peer voice 1 pots
  application MGCPAPP
  destination-pattern 2220001
  port 1/1
!
line con 0
  transport input none
line aux 0
  line 2 3
line vty 0 4
login
!
end
```

Example 3: MGCP/SGCP E&M Wink-Start

The following example illustrates an E&M wink-start configuration on the Cisco MC3810 series platform that can be defined for either the SGCP or MGCP modes:

```
version 12.2
no service pad
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname sales
!
network-clock base-rate 56k
ip subnet-zero
!
mgcp
no mgcp timer receive-rtcp
call rsvp-sync
!
voice service voatm
!
 session protocol aal2
  subcell-mux
!
voice-card 0
!
controller T1 0
 mode atm
 framing esf
 clock source internal
 linecode b8zs
!
controller T1 1
```

```
 mode cas
 framing esf
 linecode b8zs
 ds0-group 1 timeslots 1-24 type e&m-wink-start
!
interface Ethernet0
 ip address 172.29.248.199 255.255.255.0
 no ip route-cache
 no ip mroute-cache
!
interface Serial0
 no ip address
 no ip route-cache
 no ip mroute-cache
!
interface Serial1
 no ip address
 no ip route-cache
 no ip mroute-cache
 shutdown
!
interface ATM0
 no ip address
 ip mroute-cache
 no atm ilmi-keepalive
!
interface ATM0.2 point-to-point
 pvc 2/200
  vbr-rt 1536 1536 100
  encapsulation aal2
  vcci 10
!
interface FR-ATM20
 no ip address
 no ip route-cache
 shutdown
!
ip classless
ip route 0.0.0.0 0.0.0.0 172.29.248.1
no ip http server
!
voice-port 1:1
 dial-type mf
!
dial-peer cor custom
!
dial-peer voice 1 pots
 application mgcpapp
 destination-pattern 1
 port 1:1
!
gatekeeper
 shutdown
!
line con 0
 exec-timeout 0 0
 transport input none
line aux 0
line 2 3
line vty 0 4
login
 length 0
!
ntp clock-period 17248569
```

```
ntp server 172.29.1.129
end
```

Example 4: SGCP 1.5 CAS PBX using Voice over ATM AAL2

The following figure and configuration illustrate the network connections for a Cisco MC3810 series platform with CAS running the MGCP application in SGCP 1.5 mode. ATM AAL2 carries voice traffic.

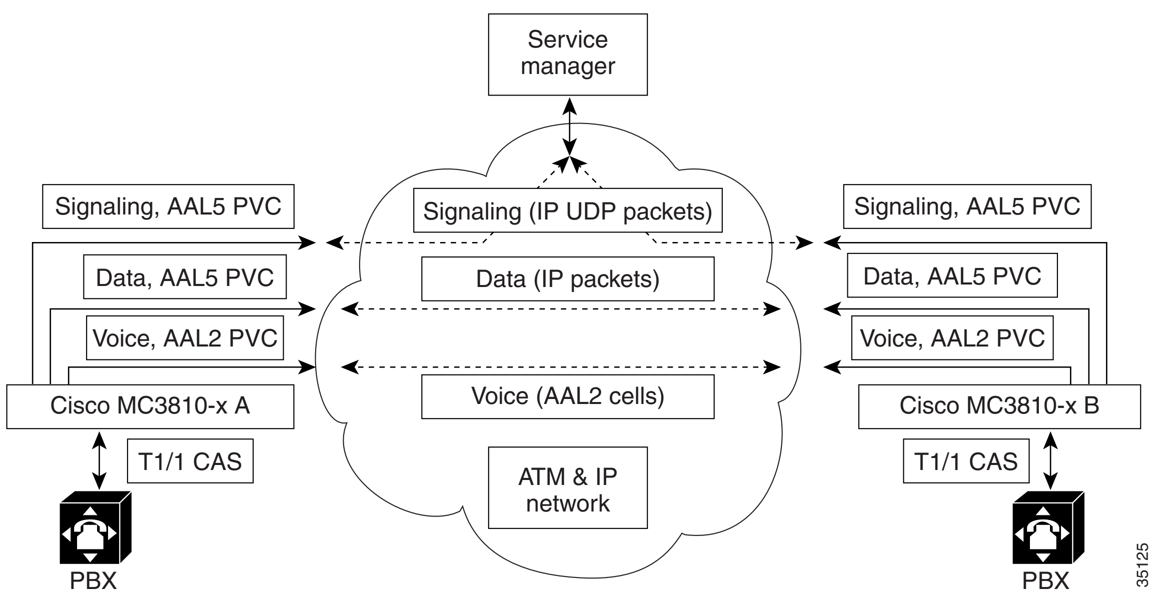
- T1/0 is configured to run ATM with three permanent virtual circuits (PVCs):
  - 1 PVC with encapsulation AAL5 carries SGCP messages (signaling VC)
  - 1 PVC with encapsulation AAL5 carries data traffic (data VC)
  - 1 PVC with encapsulation AAL2 carries voice traffic (bearer VC)

    This bearer VC has a **vcci** of 2 assigned to it. The service manager uses this **vcci** value and a selected channel identifier (CID) value for a voice call on this router.

    For AAL2, allocate 200 ATM cells/sec (84.8K bits/sec) for each G711u no vad call, 100 ATM cells/sec (42.4K bits/sec) for each G726-32 no vad or G729a no vad call.
- In this configuration, T1/1 is configured as three DS-0 groups:
  - 1 FXS ground start group
  - 1 E&M immediate start group
  - 1 E&M wink start group

    For these voice ports, the dial type is set to **mf** to support mf dialing.
- **mgcp sdp** is configured to enable SGCP RSIP messages notification.
- **mgcp modem passthrough mode** is configured to allow **nse** processing of fax or modem calls.

Figure 16 SGCP 1.5 CAS PBX using Voice over ATM AAL2 Configuration



Router A Configuration

```
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname A
!
network-clock base-rate 56K
ip subnet-zero
!
mgcp
mgcp call-agent 10.0.0.1 service-type sgcp version 1.5
mgcp sgcp restart notify
mgcp tse payload 100
no mgcp timer receive-rtcp
mgcp timer net-cont-test 3000
isdn voice-call-failure 0
!
cns event-service server
voice-card 0
!
controller T1 0
  mode atm
  framing esf
  clock source line
  linecode b8zs
!
controller T1 1
  mode cas
  framing esf
  clock source line
  linecode b8zs
  ds0-group 1 timeslots 1-8 type e&m-immediate-start
  ds0-group 2 timeslots 9-16 type e&m-wink-start
  ds0-group 3 timeslots 17-24 type fxs-ground-start
!
interface Ethernet0
  ip address 172.16.24.103 255.255.0.0
!
interface Serial0
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
!
interface Serial1
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
  no cdp enable
!
interface ATM0
  no ip address
  ip mroute-cache
  no atm ilmi-keepalive
!
interface ATM0.1 point-to-point
  description signaling vc
  ip address 10.0.0.2 255.0.0.0
  pvc 1/1
```

```
   vbr-rt 1536 64
   encapsulation aal5snap
!
interface ATM0.2 point-to-point
  description bearer vc
  pvc 2/200
   vbr-rt 1536 1400 100
   encapsulation aal2
   vcci 2
!
interface ATM0.3 point-to-point
  description data vc
  ip address 10.0.0.5 255.0.0.0
  pvc 1/100
   encapsulation aal5snap
!
interface FR-ATM20
  no ip address
  no ip route-cache
  shutdown
!
ip classless
no ip http server
!
voice-port 1:1
!
voice-port 1:2
  dial-type mf
!
voice-port 1:3
!
dial-peer voice 1 pots
  application MGCPAPP
  port 1:1
!
dial-peer voice 2 pots
  application MGCPAPP
  port 1:2
!
dial-peer voice 3 pots
  application MGCPAPP
  port 1:3
!
line con 0
  exec-timeout 0 0
  privilege level 15
  transport input none
line aux 0
line 2 3
line vty 0 4
  login
!
end
```

Router B Configuration

```
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname B
!
```

```
network-clock base-rate 56K
ip subnet-zero
!
mgcp
mgcp call-agent 10.0.0.1 service-type sgcp version 1.5
mgcp sgcp restart notify
mgcp tse payload 100
no mgcp timer receive-rtcp
mgcp timer net-cont-test 3000
isdn voice-call-failure 0
!
cns event-service server
voice-card 0
!
controller T1 0
  mode atm
  framing esf
  clock source line
  linecode b8zs
!
controller T1 1
  mode cas
  framing esf
  clock source line
  linecode b8zs
  ds0-group 1 timeslots 1-8 type e&m-immediate-start
  ds0-group 2 timeslots 9-16 type e&m-wink-start
  ds0-group3 timeslots 17-24 type fxs-ground-start
!
interface Ethernet0
  ip address 172.17.24.103 255.255.0.0
!
interface Serial0
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
!
interface Serial1
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
  no cdp enable
!
interface ATM0
  no ip address
  ip mroute-cache
  no atm ilmi-keepalive
!
interface ATM0.1 point-to-point
  description signaling vc
  ip address 10.0.0.3 255.0.0.0
  pvc 1/1
   vbr-rt 1536 64
   encapsulation aal5snap
!
interface ATM0.2 point-to-point
  description bearer vc
  pvc 2/200
   vbr-rt 1536 1400 100
   encapsulation aal2
   vcci 2
!
```

```
interface ATM0.3 point-to-point
  description data vc
  ip address 10.0.0.8 255.0.0.0
  pvc 1/100
   encapsulation aal5snap
!
interface FR-ATM20
  no ip address
  no ip route-cache
  shutdown
!
ip classless
no ip http server
!
voice-port 1:1
!
voice-port 1:2
  dial-type mf
!
voice-port 1:3
!
dial-peer voice 1 pots
  application MGCPAPP
  port 1:1
!
dial-peer voice 2 pots
  application MGCPAPP
  port 1:2
!
dial-peer voice 3 pots
  application MGCPAPP
  port 1:3
!
line con 0
  exec-timeout 0 0
  privilege level 15
  transport input none
line aux 0
line 2 3
line vty 0 4
  login
!
end
```

Example 5: SGCP 1.5 CAS PBX using Voice over IP over ATM AAL5

The following figure and configuration illustrate the network connections for a Cisco MC3810 series platform with CAS running the MGCP application in SGCP 1.5 mode. Voice over IP over ATM AAL5 carries voice traffic.

This configuration is very similar to the AAL2 example in the previous section except that an AAL5 PVC is the bearer PVC for voice traffic.

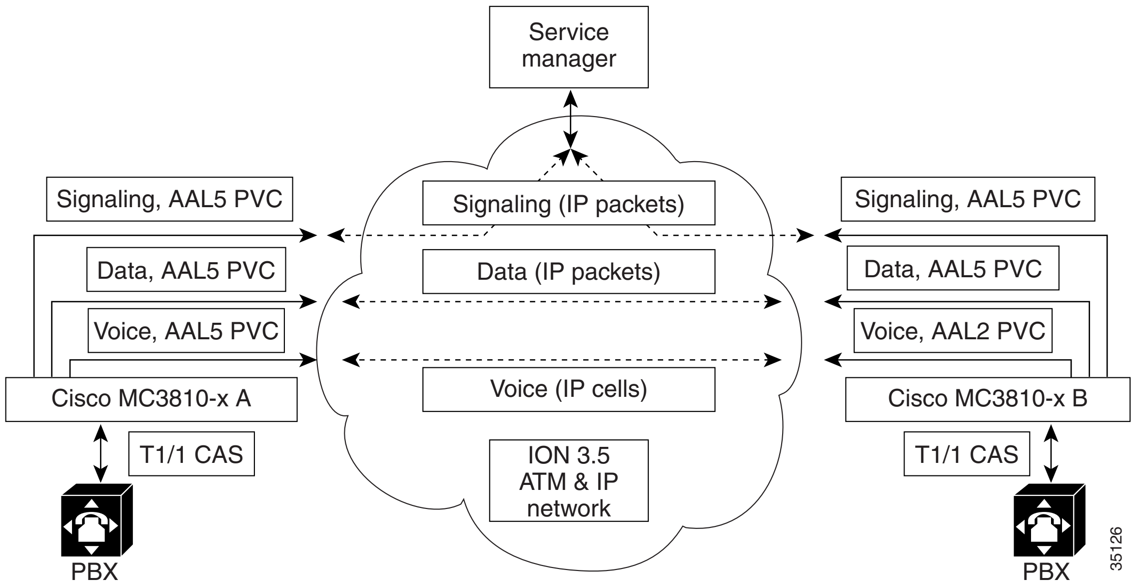
This configuration has a loopback interface with an IP address assigned to it. During voice calls, the gateway gives this IP address to the service manager as the address for the other gateway of the voice connection to use as the destination IP address.

In the example below, Router A's loopback address is 192.168.1.0 and Router B's address is 192.168.5.0. If Router A originated a call to Router B, A would give 192.168.1.0 to the Service Manager and B would give 192.168.5.0. The IP route configuration commands of both routers direct the IP traffic into the voice bearer PVC since the loopback addresses are on different IP subnets.

For Voice over IP, allocate 300 ATM cells/sec (127.2K bits/sec) for each G711u no vad call, and 200 ATM cells/sec (84.8K bits/sec) for each G726-32 no vad or G729a no vad call.

Note For G711u no vad calls, a T1 running ATM does not have enough bandwidth to carry 24 voice calls.

Figure 17 SGCP 1.5 CAS PBX using Voice over IP over ATM AAL5 Configuration



Router A Configuration

```
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname A
!
network-clock base-rate 56K
ip subnet-zero
!
mgcp
mgcp call-agent 10.0.0.1 service-type sgcp version 1.5
mgcp modem passthrough nse
mgcp sgcp restart notify
mgcp tse payload 100
no mgcp timer receive-rtcp
mgcp timer net-cont-test 3000
isdn voice-call-failure 0
!
cns event-service server
voice-card 0
!
controller T1 0
  mode atm
  framing esf
  linecode b8zs
!
```

```
controller T1 1
  mode cas
  framing esf
  clock source line
  linecode b8zs
  ds0-group 1 timeslots 1-8 type e&m-immediate-start
  ds0-group 2 timeslots 9-16 type e&m-wink-start
  ds0-group 3 timeslots 17-24 type fxs-ground-start
  framing esf
  linecode b8zs
!
interface Loopback0
  ip address 192.168.1.0 255.255.255.0
!
interface Ethernet0
  ip address 172.16.24.103 255.255.0.0
!
interface Serial0
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
!
interface Serial1
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
  no cdp enable
!
interface ATM0
  no ip address
  ip mroute-cache
  no atm ilmi-keepalive
!
interface ATM0.1 point-to-point
  description signaling vc
  ip address 10.0.0.2 255.0.0.0
  pvc 1/1
   vbr-rt 1536 64
   encapsulation aal5snap
!
interface ATM0.2 point-to-point
  description bearer vc
  ip address 10.0.0.5 255.0.0.0
  pvc 1/2
   vbr-rt 1536 1400 100
   encapsulation aal5mux ip
!
interface ATM0.3 point-to-point
  description data vc
  ip address 10.0.0.8 255.0.0.0
  pvc 1/100
   encapsulation aal5snap
!
interface FR-ATM20
  no ip address
  no ip route-cache
  shutdown
!
ip classless
ip route 10.0.0.15 255.0.0.0 ATM0.2
no ip http server
!
```

```
!
voice-port 1:1
!
voice-port 1:2
  dial-type mf
!
voice-port 1:3
!
dial-peer voice 1 pots
  application MGCPAPP
  port 1:1
!
dial-peer voice 2 pots
  application MGCPAPP
  port 1:2
!
dial-peer voice 3 pots
  application MGCPAPP
  port 1:3
!
line con 0
  exec-timeout 0 0
  privilege level 15
  transport input none
line aux 0
line 2 3
line vty 0 4
  login
!
end
```

Router B Configuration

```
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname B
!
network-clock base-rate 56K
ip subnet-zero
!
mgcp
mgcp call-agent 10.0.0.1 service-type sgcp version 1.5
mgcp modem passthrough nse
mgcp sgcp restart notify
mgcp tse payload 100
no mgcp timer receive-rtcp
mgcp timer net-cont-test 3000
isdn voice-call-failure 0
!
cns event-service server
voice-card 0
!
controller T1 0
  mode atm
  framing esf
  linecode b8zs
!
controller T1 1
  mode cas
  ds0-group 1 timeslots 1-8 type e&m-immediate-start
```

```
  ds0-group 2 timeslots 9-16 type e&m-wink-start
  ds0-group3 timeslots 17-24 type fxs-ground-start
  framing esf
  linecode b8zs
!
interface Loopback 0
  ip address 192.168.5.0 255.255.255.0
!
interface Ethernet0
  ip address 172.17.24.103 255.255.0.0
!
interface Serial0
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
!
interface Serial1
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
  no cdp enable
!
interface ATM0
  no ip address
  ip mroute-cache
  no atm ilmi-keepalive
!
interface ATM0.1 point-to-point
  description signaling vc
  ip address 10.0.0.3 255.0.0.0
  pvc 1/1
   vbr-rt 1536 64
   encapsulation aal5snap
!
interface ATM0.2 point-to-point
  description bearer vc
  ip address 10.0.0.6 255.0.0.0
  pvc 1/2
   vbr-rt 1536 1400 100
   encapsulation aal5mux ip
!
interface ATM0.3 point-to-point
  description data vc
  ip address 10.0.0.9 255.0.0.0
  pvc 1/100
   encapsulation aal5snap
!
interface FR-ATM20
  no ip address
  no ip route-cache
  shutdown
!
ip classless
ip route 10.0.0.16 255.0.0.0 ATM0.2
no ip http server
!
!
voice-port 1:1
!
voice-port 1:2
  dial-type mf
!
```

```
voice-port 1:3
!
dial-peer voice 1 pots
  application MGCPAPP
  port 1:1
!
dial-peer voice 2 pots
  application MGCPAPP
  port 1:2
!
dial-peer voice 3 pots
  application MGCPAPP
  port 1:3
!
!
line con 0
  exec-timeout 0 0
  privilege level 15
  transport input none
line aux 0
line 2 3
line vty 0 4
  login
!
end
```

Example 6: SGCP 1.5 Analog E&M PBX using Voice over ATM AAL2

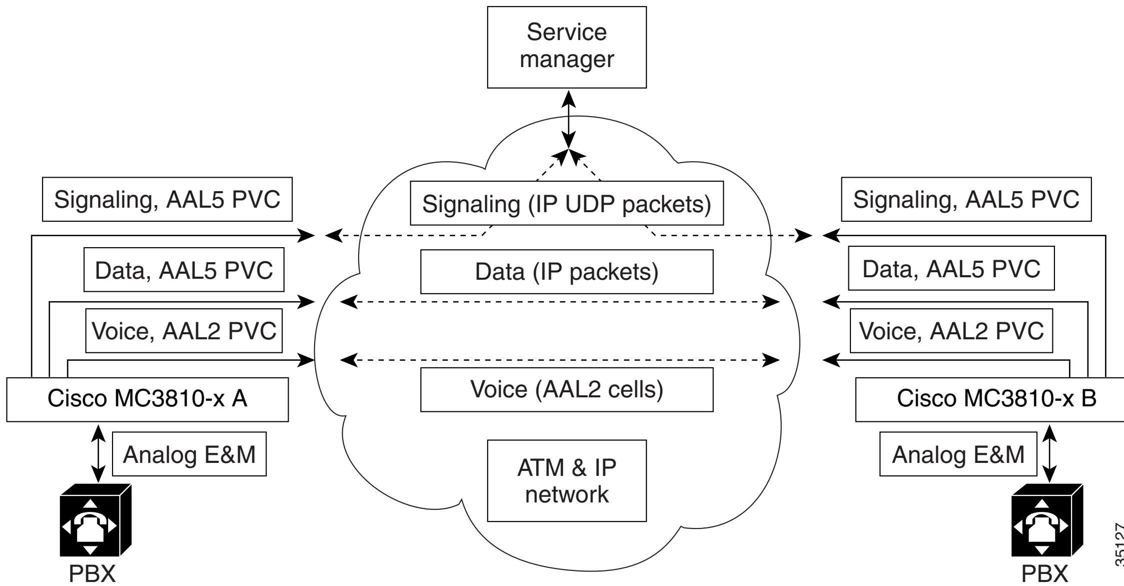
The following figure and configuration illustrate the network connections for a Cisco MC3810 series platform with Analog E&M running the MGCP application in SGCP 1.5 mode. ATM AAL2 carries voice traffic.

This configuration is similar to the SGCP 1.5 CAS PBX using Voice over ATM AAL2 configuration, with these exceptions:

- No DS-0 groups are configured for T1/1 because the slot is used by analog voice.
- The E&M port must be configured to match the type of analog PBX to which the port is connected.
- E&M protocol is set to either E&M immediate or wink start. For wink start, set the dial-type to **mf**.
- Operation must be set to 2-w (for 2-wire) or 4-w (for 4-wire).
- Type is set to I, II, IV, or V.

In this example, the bearer PVC has enough bandwidth for two G711u no vad calls because the router has only two voice ports.

Figure 18 SGCP 1.5 Analog E&M PBX using Voice over ATM AAL2 Configuration



Router A Configuration

```
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname A
!
network-clock base-rate 56K
ip subnet-zero
!
mgcp
mgcp call-agent 10.0.0.1 service-type sgcp version 1.5
mgcp tse payload 100
no mgcp timer receive-rtcp
mgcp timer net-cont-test 3000
mgcp sgcp restart notify
isdn voice-call-failure 0
!
!
cns event-service server
voice-card 0
!
controller T1 0
  mode atm
  framing esf
  linecode b8zs
!
interface Ethernet0
  ip address 172.16.24.101 255.255.0.0
!
interface Serial0
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
```

```
!
interface Serial1
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
  no cdp enable
!
interface ATM0
  no ip address
  ip mroute-cache
  no atm ilmi-keepalive
!
interface ATM0.1 point-to-point
  description signaling vc
  ip address 10.0.0.2 255.0.0.0
  pvc 1/1
   vbr-rt 1536 64
   encapsulation aal5snap
!
interface ATM0.2 point-to-point
  description bearer vc
  pvc 1/2
   vbr-rt 1536 170 8
   encapsulation aal2
   vcci 2
!
interface ATM0.3 point-to-point
  description data vc
  ip address 10.0.0.5 255.0.0.0
  pvc 1/100
   encapsulation aal5snap
!
interface FR-ATM20
  no ip address
  no ip route-cache
  shutdown
!
ip classless
no ip http server
!
voice-port 1/3
  operation 4-wire
  type 2
  signal immediate
!
voice-port 1/4
  operation 4-wire
  type 2
  dial-type mf
!
!
dial-peer voice 3 pots
  application MGCPAPP
  port 1/3
!
dial-peer voice 4 pots
  application MGCPAPP
  port 1/4
!
line con 0
  exec-timeout 0 0
  privilege level 15
  transport input none
```

```
line aux 0
line 2 3
line vty 0 4
  login
!
end
```

Router B Configuration

```
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname B
!
network-clock base-rate 56K
ip subnet-zero
!
mgcp
mgcp call-agent 10.0.0.1 service-type sgcp version 1.5
mgcp tse payload 100
no mgcp timer receive-rtcp
mgcp timer net-cont-test 3000
mgcp sgcp restart notify
isdn voice-call-failure 0
!
cns event-service server
voice-card 0
!
controller T1 0
  mode atm
  framing esf
  linecode b8zs
!
interface Ethernet0
  ip address 172.17.24.101 255.255.0.0
!
interface Serial0
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
!
interface Serial1
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
  no cdp enable
!
interface ATM0
  no ip address
  ip mroute-cache
  no atm ilmi-keepalive
!
interface ATM0.1 point-to-point
  description signaling vc
  ip address 10.0.0.3 255.0.0.0
  pvc 1/1
   vbr-rt 1536 64
   encapsulation aal5snap
!
```

```
interface ATM0.2 point-to-point
  description bearer vc
  pvc 1/2
   vbr-rt 1536 170 8
   encapsulation aal2
   vcci 2
!
interface ATM0.3 point-to-point
  description data vc
  ip address 10.0.0.6 255.0.0.0
  pvc 1/100
   encapsulation aal5snap
!
interface FR-ATM20
  no ip address
  no ip route-cache
  shutdown
!
ip classless
no ip http server
!
voice-port 1/3
  operation 2-wire
  type 1
  signal immediate
!
voice-port 1/4
  operation 4-wire
  type 2
  dial-type mf
!
dial-peer voice 3 pots
  application MGCPAPP
  port 1/3
!
dial-peer voice 4 pots
  application MGCPAPP
  port 1/4
!
!
line con 0
  exec-timeout 0 0
  privilege level 15
  transport input none
line aux 0
line 2 3
line vty 0 4
  login
!
end
```

Example 7: SGCP 1.5 Analog E&M PBX using Voice over IP over ATM AAL5

The following Figure 19 on page 149 and configuration illustrate the network connections for a Cisco MC3810 series platform RGW with analog FXS loopstart ports running the MGCP application in SGCP 1.5 mode. Voice over IP over ATM AAL5 carries voice traffic.

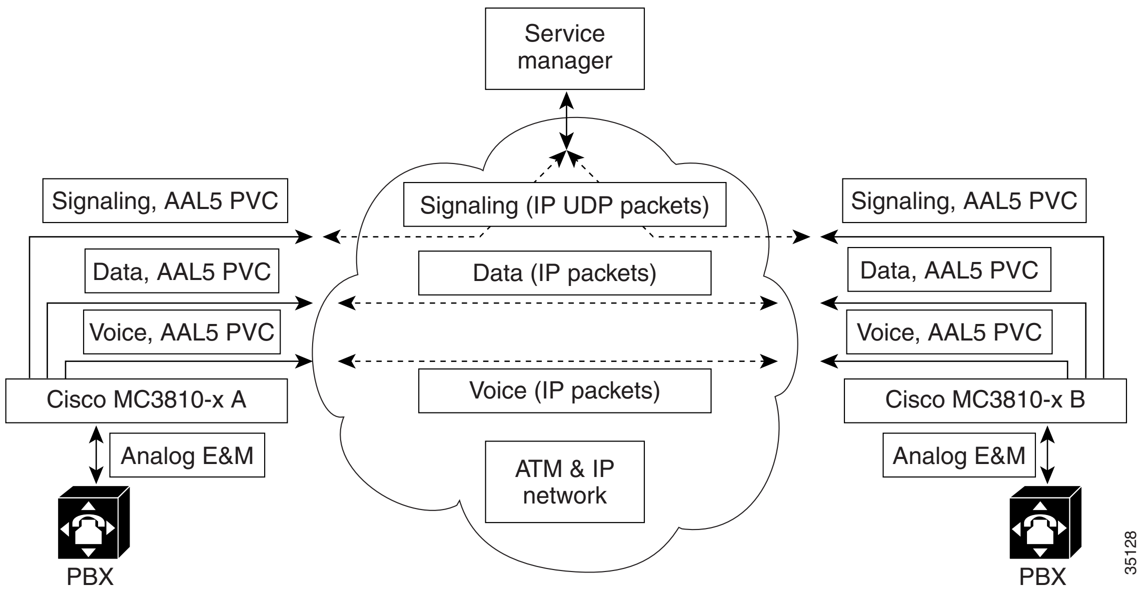
This configuration is similar to the SGCP 1.5 CAS PBX using Voice over IP over ATM AAL5 configuration, with these exceptions:

- No DS-0 groups are configured for T1/1 because the slot is used by analog voice.

- The E&M port must be configured to match the type of analog PBX to which the port is connected.
- E&M protocol is set to either E&M immediate or wink start. For wink start, set the dial-type to **mf**.
- Operation must be set to 2-w (for 2-wire) or 4-w (for 4-wire).
- Type is set to I, II, IV, or V.

In this example, the bearer PVC has enough bandwidth for two G711u no vad calls because the router has only two voice ports.

Figure 19 SGCP 1.5 Analog E&M PBX using Voice over IP over ATM AAL5 Configuration



Router A Configuration

```
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname A
!
network-clock base-rate 56K
ip subnet-zero
!
mgcp
mgcp call-agent 10.0.0.1 service-type sgcp version 1.5
mgcp tse payload 100
no mgcp timer receive-rtcp
mgcp timer net-cont-test 3000
mgcp sgcp restart notify
isdn voice-call-failure 0
!
cns event-service server
voice-card 0
!
controller T1 0
  mode atm
  framing esf
```

```
  linecode b8zs
!
interface Loopback0
  ip address 10.0.0.2 255.0.0.
!
interface Ethernet0
  ip address 172.16.24.101 255.255.0.0
!
interface Serial0
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
!
interface Serial1
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
!
interface ATM0
  no ip address
  ip mroute-cache
  no atm ilmi-keepalive
!
interface ATM0.1 point-to-point
  description signaling vc
  ip address 10.0.0.5 255.0.0.0
  pvc 1/1
   vbr-rt 1536 64
   encapsulation aal5snap
!
interface ATM0.2 point-to-point
  description bearer vc
  ip address 10.0.0.6 255.0.0.0
  pvc 1/2
   vbr-rt 1536 260 8
   encapsulation aal5mux ip
!
interface ATM0.3 point-to-point
  description data vc
  ip address 10.0.0.8 255.0.0.0
  pvc 1/100
   encapsulation aal5snap
!
interface FR-ATM20
  no ip address
  no ip route-cache
  shutdown
!
ip classless
ip route 10.0.0.0 255.0.0.0 ATM0.2
no ip http server
!
voice-port 1/3
  operation 4-wire
  type 2
  signal immediate
!
voice-port 1/4
  operation 4-wire
  type 2
  dial-type mf
!
```

```
dial-peer voice 3 pots
  application MGCPAPP
  port 1/3
!
dial-peer voice 4 pots
  application MGCPAPP
  port 1/4
!
line con 0
  exec-timeout 0 0
  privilege level 15
  transport input none
line aux 0
line 2 3
line vty 0 4
  login
!
end
```

Router B Configuration

```
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname B
!
network-clock base-rate 56K
ip subnet-zero
!
mgcp
mgcp call-agent 10.0.0.1 service-type sgcp version 1.5
mgcp tse payload 100
no mgcp timer receive-rtcp
mgcp timer net-cont-test 3000
mgcp sgcp restart notify
isdn voice-call-failure 0
!
cns event-service server
voice-card 0
!
controller T1 0
  mode atm
  framing esf
  linecode b8zs
!
interface Loopback0
  ip address 10.0.0.3 255.0.0.0

interface Ethernet0
  ip address 172.17.24.101 255.255.0.0
!
interface Serial0
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
!
interface Serial1
  no ip address
  no ip route-cache
  no ip mroute-cache
```

```
  shutdown
!
interface ATM0
  no ip address
  ip mroute-cache
  no atm ilmi-keepalive
!
interface ATM0.1 point-to-point
  description signaling vc
  ip address 10.0.0.7 255.0.0.0
  pvc 1/1
   vbr-rt 1536 64
   encapsulation aal5snap
!
interface ATM0.2 point-to-point
  description bearer vc
  ip address 10.0.0.9 255.0.0.0
  pvc 1/2
   vbr-rt 1536 170 8
   encapsulation aal5mux ip
!
interface ATM0.3 point-to-point
  description data vc
  ip address 10.0.0.10 255.0.0.0
  pvc 1/100
   encapsulation aal5snap
!
interface FR-ATM20
  no ip address
  no ip route-cache
  shutdown
!
ip classless
ip route 10.0.0.20 255.0.0.0 ATM0.2
no ip http server
!
voice-port 1/3
  operation 4-wire
  type 2
  signal immediate
!
voice-port 1/4
  operation 4-wire
  type 2
  dial-type mf
!
dial-peer voice 3 pots
  application MGCPAPP
  port 1/3
!
dial-peer voice 4 pots
  application MGCPAPP
  port 1/4
!
line con 0
  exec-timeout 0 0
  privilege level 15
  transport input none
line aux 0
line 2 3
line vty 0 4
  login
!
end
```

Example 8: SGCP 1.5 RGW using Voice over ATM AAL2

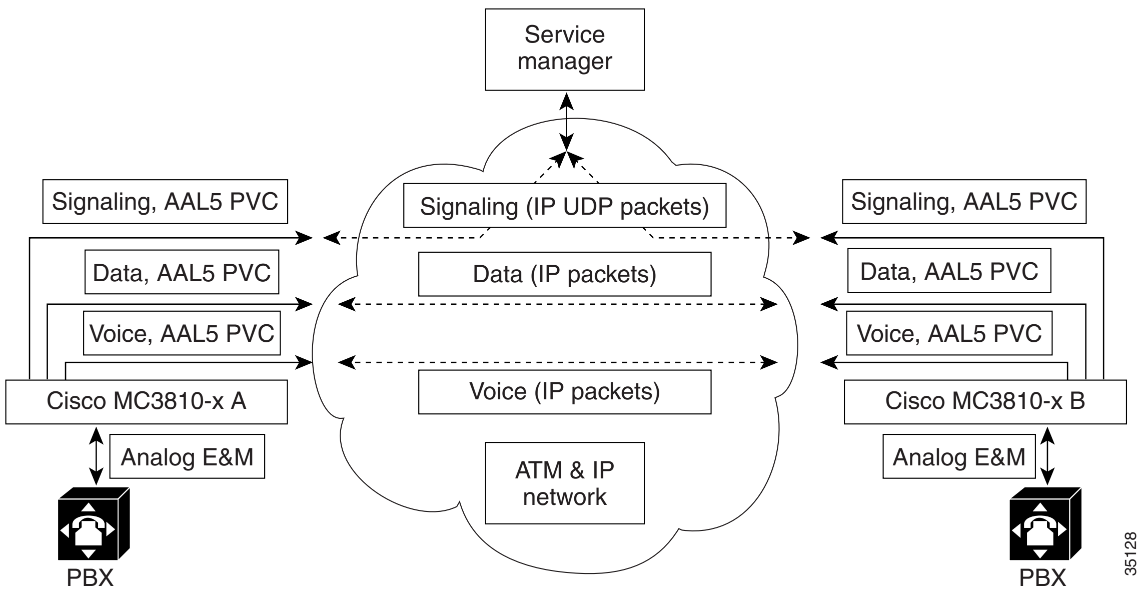
The following Figure 20 on page 153 and configuration illustrate the network connections for a Cisco MC3810 series platform RGW with analog FXS port running the MGCP application in SGCP 1.5 mode. ATM AAL2 carries voice traffic.

This configuration is similar to the SGCP 1.5 CAS PBX using Voice over ATM AAL2 configuration, with these exceptions:

- No DS-0 groups are configured for T1/1 because the slot is used by analog voice.
- For RGW, the FXS ports' signaling are set to loop start, which is the default.

In this example, the bearer PVC has enough bandwidth for two G711u no vad calls because the router has only two voice ports.

Figure 20 SGCP 1.5 RGW using Voice over ATM AAL2 Configuration



Router A Configuration

```
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname A
!
network-clock base-rate 56K
ip subnet-zero
!
mgcp
mgcp call-agent 10.0.0.1 service-type sgcp version 1.5
mgcp sgcp restart notify
mgcp tse payload 100
no mgcp timer receive-rtcp
mgcp timer net-cont-test 3000
isdn voice-call-failure 0
!
```

```
!
cns event-service server
voice-card 0
!
controller T1 0
  mode atm
  framing esf
  linecode b8zs
!
interface Ethernet0
  ip address 172.16.24.101 255.255.0.0
!
interface Serial0
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
!
interface Serial1
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
  no cdp enable
!
interface ATM0
  no ip address
  ip mroute-cache
  no atm ilmi-keepalive
!
interface ATM0.1 point-to-point
  description signaling vc
  ip address 10.0.0.2 255.0.0.0
  pvc 1/1
   vbr-rt 1536 64
   encapsulation aal5snap
!
interface ATM0.2 point-to-point
  description bearer vc
  pvc 1/2
   vbr-rt 1536 170 8
   encapsulation aal2
   vcci 2
!
interface ATM0.3 point-to-point
  description data vc
  ip address 10.0.0.5 255.0.0.0
  pvc 1/100
   encapsulation aal5snap
!
interface FR-ATM20
  no ip address
  no ip route-cache
  shutdown
!
ip classless
no ip http server
!
!
voice-port 1/1
!
voice-port 1/2
!
dial-peer voice 1 pots
```

```
  application MGCPAPP
  port 1/1
!
dial-peer voice 2 pots
  application MGCPAPP
  port 1/2
!
line con 0
  exec-timeout 0 0
  privilege level 15
  transport input none
line aux 0
line 2 3
line vty 0 4
  login
!
end
```

Router B Configuration

```
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname B
!
network-clock base-rate 56K
ip subnet-zero
!
mgcp
mgcp call-agent 10.0.0.1 service-type sgcp version 1.5
mgcp sgcp restart notify
mgcp tse payload 100
no mgcp timer receive-rtcp
mgcp timer net-cont-test 3000
isdn voice-call-failure 0
!
cns event-service server
voice-card 0
!
controller T1 0
  mode atm
  framing esf
  linecode b8zs
!
interface Ethernet0
  ip address 172.17.24.101 255.255.0.0
!
interface Serial0
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
!
interface Serial1
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
  no cdp enable
!
interface ATM0
```

```
  no ip address
  ip mroute-cache
  no atm ilmi-keepalive
!
interface ATM0.1 point-to-point
  description signaling vc
  ip address 10.0.0.3 255.0.0.0
  pvc 1/1
   vbr-rt 1536 64
   encapsulation aal5snap
!
interface ATM0.2 point-to-point
  description bearer vc
  pvc 1/2
   vbr-rt 1536 170 8
   encapsulation aal2
   vcci 2
!
interface ATM0.3 point-to-point
  description data vc
  ip address 10.0.0.6 255.0.0.0
  pvc 1/100
   encapsulation aal5snap
!
interface FR-ATM20
  no ip address
  no ip route-cache
  shutdown
!
ip classless
no ip http server
!
voice-port 1/1
!
voice-port 1/2
!
dial-peer voice 1 pots
  application MGCPAPP
  port 1/1
!
dial-peer voice 2 pots
  application MGCPAPP
  port 1/2
!
line con 0
  exec-timeout 0 0
  privilege level 15
  transport input none
line aux 0
line 2 3
line vty 0 4
  login
!
end
```

Example 9: SGCP 1.5 RGW using Voice over IP over ATM AAL5

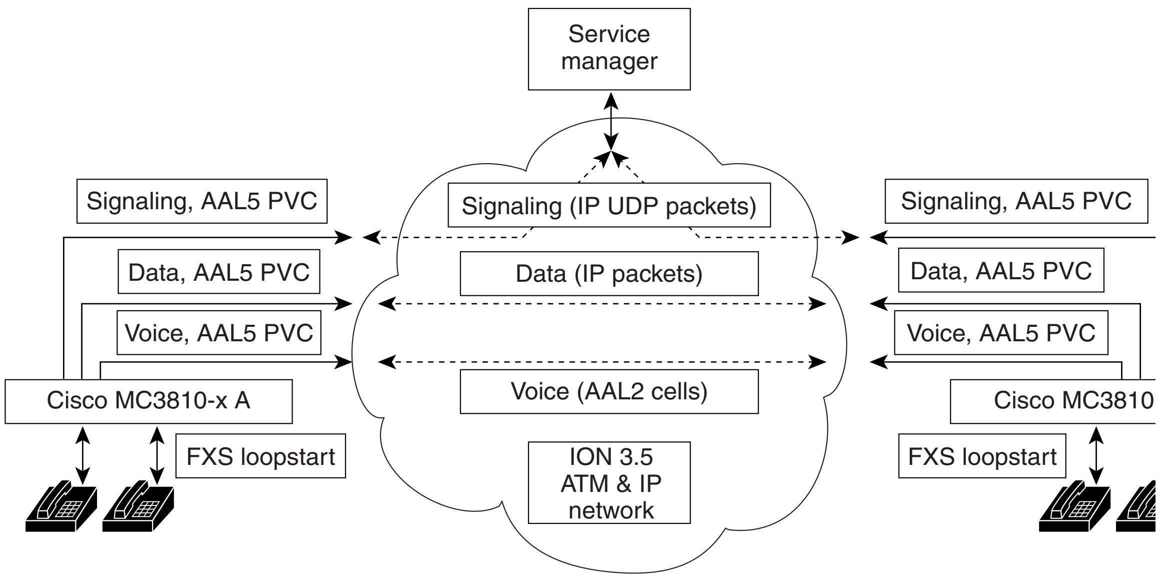
The following Figure 21 on page 157 and configuration illustrate the network connections for a Cisco MC3810 series platform RGW with analog FXS port running the MGCP application in SGCP 1.5 mode. Voice over IP over ATM AAL5 carries voice traffic.

This configuration is similar to the SGCP 1.5 CAS PBX Voice Over ATM AAL5 configuration, with these exceptions:

- No DS-0 groups are configured for T1/1 because the slot is used by analog voice.
- For RGW, the FXS ports' signaling are set to loop start, which is the default.

In this example, the bearer PVC has enough bandwidth for two G711u no vad calls because the router has only two voice ports.

Figure 21 SGCP 1.5 RGW using Voice over IP over ATM AAL5 Configuration



Router A Configuration

```
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname A
!
network-clock base-rate 56K
ip subnet-zero
!
mgcp
mgcp call-agent 10.0.0.1 service-type sgcp version 1.5
mgcp sgcp restart notify
mgcp tse payload 100
no mgcp timer receive-rtcp
mgcp timer net-cont-test 3000
isdn voice-call-failure 0
!
cns event-service server
voice-card 0
!
controller T1 0
  mode atm
  framing esf
  linecode b8zs
```

```
!
interface Ethernet0
  ip address 172.16.24.101 255.255.0.0
!
interface Serial0
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
!
interface Serial1
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
!
interface ATM0
  no ip address
  ip mroute-cache
  no atm ilmi-keepalive
!
interface ATM0.1 point-to-point
  description signaling vc
  ip address 10.0.0.2 255.0.0.0
  pvc 1/1
   vbr-rt 1536 64
   encapsulation aal5snap
!
interface ATM0.2 point-to-point
  description bearer vc
  ip address 10.0.0.5 255.0.0.0
  pvc 1/2
   vbr-rt 1536 260 8
   encapsulation aal5mux ip
!
interface ATM0.3 point-to-point
  description data vc
  ip address 10.0.0.8 255.0.0.0
  pvc 1/100
   encapsulation aal5snap
!
interface FR-ATM20
  no ip address
  no ip route-cache
  shutdown
!
ip classless
ip route 10.0.0.10 255.0.0.0 ATM0.2
no ip http server
!
voice-port 1/1
!
voice-port 1/2
!
dial-peer voice 1 pots
  application MGCPAPP
  port 1/1
!
dial-peer voice 2 pots
  application MGCPAPP
  port 1/2
!
line con 0
  exec-timeout 0 0
```

```
  privilege level 15
  transport input none
line aux 0
line 2 3
line vty 0 4
  login
!
end
```

Router B Configuration

```
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname B
!
network-clock base-rate 56K
ip subnet-zero
!
mgcp
mgcp call-agent 10.0.0.1 service-type sgcp version 1.5
mgcp sgcp restart notify
mgcp tse payload 100
no mgcp timer receive-rtcp
mgcp timer net-cont-test 3000
isdn voice-call-failure 0
!
!
cns event-service server
voice-card 0
!
controller T1 0
  mode atm
  framing esf
  linecode b8zs
!
interface Ethernet0
  ip address 172.17.24.101 255.255.0.0
!
interface Serial0
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
!
interface Serial1
  no ip address
  no ip route-cache
  no ip mroute-cache
  shutdown
!
interface ATM0
  no ip address
  ip mroute-cache
  no atm ilmi-keepalive
!
interface ATM0.1 point-to-point
  description signaling vc
  ip address 10.0.0.3 255.0.0.0
  pvc 1/1
   vbr-rt 1536 64
```

```
   encapsulation aal5snap
!
interface ATM0.2 point-to-point
  description bearer vc
  ip address 10.0.0.6 255.0.0.0
  pvc 1/2
   vbr-rt 1536 260 8
   encapsulation aal5mux ip
!
interface ATM0.3 point-to-point
  description data vc
  ip address 10.0.0.7 255.0.0.0
  pvc 1/100
   encapsulation aal5snap
!
interface FR-ATM20
  no ip address
  no ip route-cache
  shutdown
!
ip classless
ip route 10.0.0.12 255.0.0.0 ATM0.2
no ip http server
!
voice-port 1/1
!
voice-port 1/2
!
dial-peer voice 1 pots
  application MGCPAPP
  port 1/1
!
dial-peer voice 2 pots
  application MGCPAPP
  port 1/2
!
line con 0
  exec-timeout 0 0
  privilege level 15
  transport input none
line aux 0
line 2 3
line vty 0 4
  login
!
end
```

Tip

- See the "Additional References for MGCP and SGCP" section on page xi for related documents, standards, and MIBs.
- See "Glossary" for definitions of terms in this guide.

Glossary

> **Note** Refer to the *Internetworking Terms and Acronyms* for terms not included in this glossary.

A

AAA Authentication, authorization, and accounting. Security services for packet networks.

AAL2 ATM Adaptation Layer 2. Supports traffic needing precise timing, such as compressed voice and video.

AAL5 ATM Adaptation Layer 5

AUEP Audit Endpoint command. An MGCP or SGCP command that is used by the call agent to determine the status of a given endpoint on a gateway.

B

BGW Business gateway.

bind In MGCP, configuring the source address for signaling and media packets to the IP address of a specific interface.

business gateway An xGCP media gateway which is a business customer premises equipment that has connection(s) to the VoIP network as well as connection(s) to the user's telephony equipment (typically a PBX, a corporate LAN or WAN). Such gateways are used to eliminate or reduce the need for individual medium (voice, data, and so forth) connectivity.

C

CA Call agent.

CAC Call admission control. Controls whether a call can be established, according to availability of local or network resources.

call agent An intelligent entity in an IP telephony network which handles call control in an MGCP model Voice over IP network.

An intelligent entity in an IP telephony network that handles call control in an xGCP model Voice-over-IP network. A Call Agent is also known as a media gateway controller (MGC).

CAS Channel Associated Signaling**.** A signaling technique that uses the same facility path for both voice and signaling traffic.

CCS Common Channel Signaling

CID AAL2 Channel Identifier

CLASS Custom Local Area Subscriber Services, usually referred to as “Custom Calling” features

CLI command-line interface.

CRCX Create Connection request. Used in an MGCP call sequence by gateways to exchange SDP.

CSM Call switching module. Card that contains digit collection logic to process incoming calls for automatic number information (ANI) and dialed number identification service (DNIS) digits.

D

DDR Dial-on-demand routing.

disconnected procedure When an endpoint attempts to communicate with its call agent and fails, the gateway may initiate the disconnected procedure, in which a timer (the disconnected timer) is started and RSIP messages are sent to the call agent at prescribed intervals until contact is established. This procedure ensures that an RSIP message is the first message to reach the call agent after communications are reestablished. The call agent may request an AUEP after the RSIP is received.

DNS Domain Name System. Used to translate H.323 IDs, URLs, or e-mail IDs into IP addresses. DNS is also used to assist in locating remote gatekeepers and to reverse-map raw IP addresses to host names of administrative domains.

DS-0 64-kbps channel in a T1/E1 line.

DTMF Dual-tone multifrequency. Tones made by pushing buttons on a telephone.

Tones used to send phone number digits to and from a switch. DTMF tones identify the numbers 0 through 9 and the * and # symbols.

E

E&M Ear and Mouth analog signaling.

F

fax passthrough The GW sends fax data encoded using a codec such as G.711 ulaw or G.711 alaw, resulting in a more reliable transmission. Fax passthrough provides repression of compression, echo cancellation, and other functions; issues redundant packets to ensure complete transmission; and provides a buffer to protect against clock skew.

FGD Feature Group-D. Identifies a standardized service available to carriers delivered on a channelized T1 line.

FGD-OS Feature Group-D Operator Services protocol. OS is a telephony signaling protocol for calls that originate from the Bell Operating Company (BOC) and are sent towards the carrier switch.

fx: extension An extension of the local connection option used by the CA to instruct the GW to be in CA-controlled mode or GW-controlled mode.

FXO Foreign Exchange Operator—An interface from a telephone to a PSTN central office or a station interface on a PBX.

FXS Foreign Exchange Station—An interface that connects to a telephone, key set, or PBX to supply ring, voltage, and dial tone.

G

GW Gateway.

I

IETF Internet Engineering Task Force. Task force that consists of over 80 working groups responsible for developing Internet standards. The IETF operates under the auspices of the Internet Society.

IPDC Internet Protocol Device Control. A device control specification.

ISP Internet service provider.

M

MDCX Modify connection request. Used in an MGCP call sequence by gateways to exchange SDP information.

media gateway The emerging industry standard generic term for a gateway. Equipment that connects the PSTN or a PBX with the Voice-over-IP network. It is controlled by a call agent using MGCP.

MGC Media gateway controller. The emerging industry standard generic term for the VSC. Another name for call agent.

MGCP Media Gateway Control Protocol. A merging of the IPDC and SGCP protocols.

N

NAS network access server. Communications processor that connects asynchronous devices to a LAN or WAN through network and terminal emulation software. Performs both synchronous and asynchronous routing of supported protocols.

NCS Network-based Call Signaling. PacketCable protocol, profile of MGCP 1.0 for residential gateways.

NSE Named Signaling Event. Format of RTP packets used for applications such as modem relay and fax relay. NSEs have different payload values than NTEs.

NTE Named Telephony Event. Format of RTP packets used to transport DTMF digits as defined in RFC 2833, *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*.

P

package A set of signals and events that define a type of voice endpoint or connection. Examples include line-package, trunk-package, dtmf-package, and atm-package

payload type Payload types are defined in RFC 2833, *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*. A payload type is a number from 96 through 127 that identifies the type of payload carried in the packet (for example, a payload type of 121 denotes a Cisco DTMF relay payload; a payload type of 122 denotes a fax payload). The payload type should be identical on the GW and call agent.

PDU protocol data units. Used by bridges to transfer connectivity information.

PRI ISDN primary rate interface

PSTN Public Switched Telephone Network

PVC Permanent Virtual Circuit

R

residential gateway An xGCP media gateway that is customer premises equipment and has one or more connections to the VoIP network as well as to user telephony equipment.

RGW Residential gateway.

RSC Router switch controller.

RSIP Restart In Progress command. An MGCP or SGCP command that is used by the gateway to signal that an endpoint, or a group of endpoints, is being taken in or out of service.

RTCP RTP Control Protocol. The protocol monitors an RTP connection and conveys information about the on-going session.

RTP Real-Time Transport Protocol. The protocol provides end-to-end network transport functions for applications transmitting real-time data and services such as payload type identification, sequence numbering, timestamping, and delivery monitoring.

S

SDP Session Description Protocol. Messages containing capabilities information that are exchanged between gateways.

SGCP Simple Gateway Control Protocol. Controls Voice-over-IP gateways using an external call control element (called a call agent). SGCP is used to establish, maintain, and disconnect calls across an IP network.

SIP Session Initiation Protocol.

SPVC Soft Permanent Virtual Circuit

SVC Switched Virtual Circuit

T

T1 **T1**—24 64-Kbps time slots on a 1.544-Mbps serial interface.

TGW see Trunking Gateway. Trunking gateway, also called the trunk side gateway. An xGCP media gateway that provides PSTN/IP gateway functionality.

trunking gateway An xGCP media gateway that provides connectivity between the PSTN and VoIP networks.

An external gateway control protocol (xGCP) media gateway that provides connectivity between the PSTN and VoIP networks.

TSE Inband Telephony Signaling Events

V

VAD Voice Activity Detection

VCC Virtual Channel Connection (used where it may be a PVC, SPVC, or SVC)

VoIP Voice over IP. The ability to carry normal telephone-style voice over an IP-based Internet with POTS-like functionality, reliability, and voice quality. VoIP is a blanket term that generally refers to the Cisco standards-based approach (for example, H.323) to IP voice traffic.

VToA Voice Trunking on ATM.

X

XCSP External Call Service Provider. Subsystem that interoperates with external call protocols to provide services such as modem call setup and teardown.


INDEX