CONTENTS

Overview of MGCP and Related Protocols 1
  Finding Feature Information 1
  Prerequisites for MGCP and Related Protocols 1
  Information About MGCP and Related Protocols 2
    Supported Gateways 3
      Residential Gateway 3
      Trunking Gateway 4
  Toll Fraud Prevention 5
  Additional References 6

Basic MGCP Configuration 9
  Finding Feature Information 9
  How to Configure MGCP and Related Protocols 9
    Configuring a TGW for MGCP 10
    Configuring a TGW for SGCP 12
    Configuring an RGW 13
    Configuring a SDP Aware NSE Mode 15
    Verifying NSE Mode Configuration 16
    Verifying the TGW or RGW Configuration 17
    Blocking New Calls 17

Configuration Examples for MGCP and Related Protocols 18
  Configuring a Cisco AS5300 as a TGW with MGCP Example 18
  Configuring a Cisco AS5300 as a TGW with SGCP Example 19
  Configuring a Cisco 3660 as a TGW with MGCP Example 21
  Configuring a Cisco uBR924 as an RGW Example 22
  Configuring a Cisco 2620 as an RGW Example 23

Additional References 24
  Feature Information for Basic MGCP Configuration 24

Configuring MGCP 1.0 27
  Finding Feature Information 28
Stutter Dial Tone  70
Off-Hook Warning Tone  70
911 Calls  70
Three-Way Calling  71
  Considerations for Three-way Calling  71
  Examples of Service-Provider Solutions  71
Troubleshooting MGCP Basic CLASS and Operator Services  74
Configuration Examples for MGCP Basic CLASS and Operator Services  75
Configuring NAS Package for MGCP  77
  Finding Feature Information  78
  Prerequisites for NAS Package for MGCP  78
  Information About NAS Package for MGCP  78
  How to Configure NAS Package for MGCP  79
    Configuring the NAS for MGCP  80
    Configuring Controllers  80
    Configuring Dialer Interfaces and Routing  83
  Verifying the NAS Package for MGCP  88
  Troubleshooting Tips  89
  MGCP Troubleshooting  90
    Example Output for show mgcp nas info Command  91
    Example Output for show mgcp nas dump Command  91
    Example Output for show mgcp connection Command  92
    Example Output for show xcsp slot Command  92
    Example Output for show xcsp port Command  92
    Example Output for show cdapi Command  93
  MGCP Debugging  93
    Example Output for debug mgcp all Command  94
    Example Output for debug mgcp events Command  94
    Example Output for debug mgcp packets Command  94
    Example Output for debug mgcp parser Command  95
    Example Output for debug mgcp nas Command  95
    Example Output for debug xcsp Command  95
    Example Output for debug cdapi Command  97
Controller Troubleshooting  97
  Example Output for show controllers e1 or t1 Command  99
Restrictions for MGCP CAS MD Package 130
Information About MGCP CAS MD Package 130

MD Package 130

How to Configure the MGCP CAS MD Package 130
Configuring the Incoming Called Number in the MGCP Dial Peer 130
Modifying ANI and DNIS Order when Using CAS MD Package 132

Configuration Examples for MGCP CAS MD Package 133
CAS MD Package Configuration Example 133
Cisco AS5850 Configuration Example 134

Configuring MGCP CAS PBX and AAL2 PVC 137
Finding Feature Information 138
Prerequisites for MGCP CAS PBX and AAL2 PVC 138
Restrictions for MGCP CAS PBX and AAL2 PVC 138
Information About MGCP CAS PBX and AAL2 PVC 139
How to Configure MGCP CAS PBX and AAL2 PVC 142
Configuring the Gateway 143
Configuring Subcell Multiplexing for AAL2 Voice 149
Configuring the Cable Access Router for SGCP and MGCP 149
Verifying the MGCP CAS PBX and AAL2 PVC Configurations 150

Configuration Examples for MGCP CAS PBX and AAL2 PVC 150
Example 1 MGCP Residential Gateway 151
Example 2 MGCP Gateway using Voice over ATM AAL2 151
Example 3 MGCP and SGCP EM Wink-Start 153
Example 4 SGCP 1.5 CAS PBX using Voice over ATM AAL2 154
Example 5 SGCP 1.5 CAS PBX using Voice over IP over ATM AAL5 158
Example 6 SGCP 1.5 Analog EM PBX using Voice over ATM AAL2 163
Example 7 SGCP 1.5 Analog EM PBX using Voice over IP over ATM AAL5 166
Example 8 SGCP 1.5 RGW using Voice over ATM AAL2 170
Example 9 SGCP 1.5 RGW using Voice over IP over ATM AAL5 174

Glossary 179
Overview of MGCP and Related Protocols

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

Note

Finding Feature Information
Your software release may not support all the features documented in this module. For the latest feature information and caveats, see the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the Feature Information Table at the end of this document.

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Prerequisites for 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.
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 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, page 3
- Trunking Gateway, page 4

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. The figure below shows an RGW configuration.
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 the table below.

### Table 1  RGW Functionality

<table>
<thead>
<tr>
<th>Functionality</th>
<th>Platform</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco 2600 Series</td>
<td>Cisco uBR924</td>
</tr>
<tr>
<td>Call waiting</td>
<td>Yes</td>
</tr>
<tr>
<td>Default call-agent address specifiable for each foreign exchange station (FXS) port</td>
<td>--</td>
</tr>
<tr>
<td>Distinctive ringing</td>
<td>--</td>
</tr>
<tr>
<td>Fax and modem calls</td>
<td>Yes</td>
</tr>
<tr>
<td>On-hook caller identification (ID)</td>
<td>--</td>
</tr>
<tr>
<td>Ring splash</td>
<td>--</td>
</tr>
<tr>
<td>Stutter dial tone</td>
<td>Yes</td>
</tr>
</tbody>
</table>

**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. The figure below shows a TGW configuration.

### Figure 2  Trunking Gateways

![Diagram of Trunking Gateways](image.png)
TGW functionality supports SGCP and MGCP as shown in the table below.

### Table 2   TGW Functionality

<table>
<thead>
<tr>
<th>Functionality</th>
<th>Platform</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco AS5300</td>
<td>Cisco 3660</td>
</tr>
<tr>
<td>911 outgoing calls on T1 lines</td>
<td>Yes</td>
</tr>
<tr>
<td>Fax and modem calls</td>
<td>Yes</td>
</tr>
<tr>
<td>PRI/ISDN signaling (calls are backhauled to the call agent)</td>
<td>Yes</td>
</tr>
<tr>
<td>SS7</td>
<td>Yes</td>
</tr>
<tr>
<td>T1 and E1 interfaces</td>
<td>Yes</td>
</tr>
</tbody>
</table>

---

**Toll Fraud Prevention**

When a Cisco router platform is installed with a voice-capable Cisco IOS software image, appropriate features must be enabled on the platform to prevent potential toll fraud exploitation by unauthorized users. Deploy these features on all Cisco router Unified Communications applications that process voice calls, such as Cisco Unified Communications Manager Express (CME), Cisco Survivable Remote Site Telephony (SRST), Cisco Unified Border Element (UBE), Cisco IOS-based router and standalone analog and digital PBX and public-switched telephone network (PSTN) gateways, and Cisco contact-center VoiceXML gateways. These features include, but are not limited to, the following:

- Disable secondary dial tone on voice ports--By default, secondary dial tone is presented on voice ports on Cisco router gateways. Use private line automatic ringdown (PLAR) for foreign exchange office (FXO) ports and direct-inward-dial (DID) for T1/E1 ports to prevent secondary dial tone from being presented to inbound callers.
- Cisco router access control lists (ACLs)--Define ACLs to allow only explicitly valid sources of calls to the router or gateway, and therefore to prevent unauthorized Session Initiation Protocol (SIP) or H.323 calls from unknown parties to be processed and connected by the router or gateway.
- Close unused SIP and H.323 ports--If either the SIP or H.323 protocol is not used in your deployment, close the associated protocol ports. If a Cisco voice gateway has dial peers configured to route calls outbound to the PSTN using either time division multiplex (TDM) trunks or IP, close the unused H. 323 or SIP ports so that calls from unauthorized endpoints cannot connect calls. If the protocols are used and the ports must remain open, use ACLs to limit access to legitimate sources.
- Change SIP port 5060--If SIP is actively used, consider changing the port to something other than well-known port 5060.
- SIP registration--If SIP registration is available on SIP trunks, turn on this feature because it provides an extra level of authentication and validation that only legitimate sources can connect calls. If it is not available, ensure that the appropriate ACLs are in place.
- SIP Digest Authentication--If the SIP Digest Authentication feature is available for either registrations or invites, turn this feature on because it provides an extra level of authentication and validation that only legitimate sources can connect calls.

---

1 Server must have SGCP 1.1+ protocol for Feature Group D Operator Services (FGD-OS)
• Explicit incoming and outgoing dial peers--Use explicit dial peers to control the types and parameters of calls allowed by the router, especially in IP-to-IP connections used on CME, SRST, and Cisco UBE. Incoming dial peers offer additional control on the sources of calls, and outgoing dial peers on the destinations. Incoming dial peers are always used for calls. If a dial peer is not explicitly defined, the implicit dial peer 0 is used to allow all calls.

• Explicit destination patterns--Use dial peers with more granularity than .T for destination patterns to block disallowed off-net call destinations. Use class of restriction (COR) on dial peers with specific destination patterns to allow even more granular control of calls to different destinations on the PSTN.

• Translation rules--Use translation rules to manipulate dialed digits before calls connect to the PSTN to provide better control over who may dial PSTN destinations. Legitimate users dial an access code and an augmented number for PSTN for certain PSTN (for example, international) locations.

• Tcl and VoiceXML scripts--Attach a Tcl/VoiceXML script to dial peers to do database lookups or additional off-router authorization checks to allow or deny call flows based on origination or destination numbers. Tcl/VoiceXML scripts can also be used to add a prefix to inbound DID calls. If the prefix plus DID matches internal extensions, then the call is completed. Otherwise, a prompt can be played to the caller that an invalid number has been dialed.

• Host name validation--Use the "permit hostname" feature to validate initial SIP Invites that contain a fully qualified domain name (FQDN) host name in the Request Uniform Resource Identifier (Request URI) against a configured list of legitimate source hostnames.

• Dynamic Domain Name Service (DNS)--If you are using DNS as the "session target" on dial peers, the actual IP address destination of call connections can vary from one call to the next. Use voice source groups and ACLs to restrict the valid address ranges expected in DNS responses (which are used subsequently for call setup destinations).

For more configuration guidance, see the "Cisco IOS Unified Communications Toll Fraud Prevention" paper.

### Additional References

The following sections provide references related to MGCP.

<table>
<thead>
<tr>
<th>Related Topic</th>
<th>Document Title</th>
</tr>
</thead>
</table>
### Related Topic

<table>
<thead>
<tr>
<th>Related Topic</th>
<th>Document Title</th>
</tr>
</thead>
</table>

### MIBs

<table>
<thead>
<tr>
<th>MIBs</th>
<th>MIBs Link</th>
</tr>
</thead>
<tbody>
<tr>
<td>•</td>
<td>To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: <a href="http://www.cisco.com/go/mibs">http://www.cisco.com/go/mibs</a></td>
</tr>
</tbody>
</table>

### Technical Assistance

<table>
<thead>
<tr>
<th>Description</th>
<th>Link</th>
</tr>
</thead>
<tbody>
<tr>
<td>The Cisco Support website provides extensive online resources, including documentation and tools for troubleshooting and resolving technical issues with Cisco products and technologies. To receive security and technical information about your products, you can subscribe to various services, such as the Product Alert Tool (accessed from Field Notices), the Cisco Technical Services Newsletter, and Really Simple Syndication (RSS) Feeds. Access to most tools on the Cisco Support website requires a Cisco.com user ID and password.</td>
<td><a href="http://www.cisco.com/techsupport">http://www.cisco.com/techsupport</a></td>
</tr>
</tbody>
</table>

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Any Internet Protocol (IP) addresses and phone numbers used in this document are not intended to be actual addresses and phone numbers. Any examples, command display output, network topology diagrams, and other figures included in the document are shown for illustrative purposes only. Any use of actual IP addresses or phone numbers in illustrative content is unintentional and coincidental.
Basic MGCP Configuration

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

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

- "Overview of MGCP and Related Protocols" on page 3
  - Finding Feature Information, page 9
  - How to Configure MGCP and Related Protocols, page 9
  - Configuration Examples for MGCP and Related Protocols, page 18
  - Additional References, page 24
  - Feature Information for Basic MGCP Configuration, page 24

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest feature information and caveats, see the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the Feature Information Table at the end of this document.

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How to Configure MGCP and Related Protocols

Note

RGWs are configured only with MGCP.

- Configuring a TGW for MGCP, page 10
- Configuring a TGW for SGCP, page 12
- Configuring an RGW, page 13
- Configuring a SDP Aware NSE Mode, page 15
- Verifying NSE Mode Configuration, page 16
- Verifying the TGW or RGW Configuration, page 17
Configuring a TGW for MGCP

To configure a trunking gateway (TGW) for MGCP, perform this task:

**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`
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. `mgcp sdp simple`
12. `exit`

**DETAILED STEPS**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> mgcp</td>
<td>Initiates the MGCP application.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# mgcp</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> mgcp call-agent [ipaddr</td>
<td>hostname] [port] service-type mgcp</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# mgcp call-agent [ipaddr</td>
<td>hostname] [port] service-type mgcp</td>
</tr>
<tr>
<td><strong>Step 3</strong> controller t1 number</td>
<td>Specifies the channel number of the T1 trunk to be used for analog calls and enters controller configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# controller t1 number</td>
<td></td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>------------------</td>
<td>---------</td>
</tr>
<tr>
<td><strong>Step 4</strong> ds0-group channel-number timeslots range type none service mgcp</td>
<td>Configures the channelized T1 time slots to accept the analog calls.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-controller)# ds0-group channel-number timeslots range type none service mgcp</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> exit</td>
<td>Exits the current mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-controller)# exit</td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong> mgcp restart-delay value</td>
<td>(Optional) Specifies the delay value sent in the RSIP graceful teardown method, in seconds. Range is from 0 to 600. Default is 0.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# mgcp restart-delay value</td>
<td></td>
</tr>
<tr>
<td><strong>Step 7</strong> mgcp package-capability {s-package</td>
<td>dtmf-package</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# mgcp package-capability {trunk-package</td>
<td>dtmf-package</td>
</tr>
<tr>
<td><strong>Step 8</strong> mgcp default-package {as-package</td>
<td>dtmf-package</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# mgcp default-package {as-package</td>
<td>dtmf-package</td>
</tr>
</tbody>
</table>
### Command or Action

| Step 9 | mgcp dtmf-relay {codec | low-bit-rate} mode {cisco | out-of-band} |
|--------|---------------------------------------------------------------|
| Purpose| (Optional) Used for relaying digits through the IP network. Default is no mgcp dtmf-relay for all codecs. |

**Example:**

```
Router(config)# mgcp dtmf-relay {codec | low-bit-rate} mode {cisco | out-of-band}
```

| Step 10 | mgcp modem passthru {cisco | ca} |
|---------|----------------------------------|
| Purpose| (Optional) Configures the gateway for modem and fax data. |

**Example:**

```
Router(config)# mgcp modem passthru {cisco | ca}
```

<table>
<thead>
<tr>
<th>Step 11</th>
<th>mgcp sdp simple</th>
</tr>
</thead>
<tbody>
<tr>
<td>Purpose</td>
<td>(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.</td>
</tr>
</tbody>
</table>

**Example:**

```
Router(config)# mgcp sdp simple
```

<table>
<thead>
<tr>
<th>Step 12</th>
<th>exit</th>
</tr>
</thead>
<tbody>
<tr>
<td>Purpose</td>
<td>Exits the current mode.</td>
</tr>
</tbody>
</table>

**Example:**

```
Router(config)# exit
```

---

### Configuring a TGW for SGCP

Perform this task to configure a trunking gateway (TGW) for Simple Gateway Control Protocol (SGCP):

**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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> mgcp</td>
<td>Initiates the MGCP application.</td>
</tr>
<tr>
<td></td>
<td>Example:</td>
</tr>
<tr>
<td>Router(config)# mgcp</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> mgcp call-agent [ipaddr</td>
<td>hostname] [port] service-type sgcp</td>
</tr>
<tr>
<td></td>
<td>Example:</td>
</tr>
<tr>
<td>Router(config)# mgcp call-agent [ipaddr</td>
<td>hostname] [port] service-type sgcp</td>
</tr>
<tr>
<td><strong>Step 3</strong> controller t1 number</td>
<td>Specifies the channel number of the T1 trunk to be used for analog calls and enters controller configuration mode.</td>
</tr>
<tr>
<td></td>
<td>Example:</td>
</tr>
<tr>
<td>Router(config)# controller t1 number</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> ds0-group channel-number timeslots range type [none</td>
<td>fgdos] [tone_type] [addr_info] service {sgcp</td>
</tr>
<tr>
<td></td>
<td>Example:</td>
</tr>
<tr>
<td>Router(config-controller)# ds0-group channel-number timeslots range type [none</td>
<td>fgdos] [tone_type] [addr_info] service {sgcp</td>
</tr>
<tr>
<td><strong>Step 5</strong> exit</td>
<td>Exits the current mode.</td>
</tr>
<tr>
<td></td>
<td>Example:</td>
</tr>
<tr>
<td>Router(config-controller)# exit</td>
<td></td>
</tr>
</tbody>
</table>

### Configuring an RGW

To configure a residential gateway (RGW), perform this task:
SUMMARY STEPS

1. mgcp
2. mgcp call-agent [ipaddr | hostname] [port] service-type sgcp
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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 mgcp</td>
<td>Initiates the MGCP application.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# mgcp</td>
<td></td>
</tr>
</tbody>
</table>

| Step 2 mgcp call-agent [ipaddr | hostname] [port] service-type sgcp | Specifies the call-agent IP address or domain name, port, and gateway control service type. |
| Example:                    |                                              |
| Router(config)# mgcp call-agent [ipaddr | hostname] [port] service-type mgcp |

| Step 3 dial-peer voice number pots | Sets up the dial peer for a voice port. |
| Example:                           |                                              |
| Router(config)# dial-peer voice number pots |

| Step 4 application MGCPAPP        | Selects the MGCP application to run on the voice port. |
| Example:                          |                                              |
| Router(config-dial-peer)# application MGCPAPP |

| Step 5 exit                       | Exits the current mode.                     |
| Example:                          |                                              |
| Router(config-dial-peer)# exit    |                                              |
### Configuring a SDP Aware NSE Mode

The Cisco IOS MGCP gateway relies only on the local modem or fax configuration to determine whether Named Signaling Event (NSE) should be used or not for the current call. SDP-aware NSE mode enables the Cisco IOS MGCP gateway to negotiate NSE-based modem and fax features by considering both the local configuration and the remote support for NSE.

**Note**

Cisco Unified Call Manager (UCM) does not support modem or fax passthrough. This feature should not be enabled when Cisco UCM is the call agent.

**SUMMARY STEPS**

1. `mgcp`
2. `mgcp behavior negotiate-nse enable`
3. `exit`
### DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> mgcp</td>
<td>Initiates the MGCP application.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# mgcp</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> mgcp behavior negotiate-nse enable</td>
<td>Enables SDP-aware NSE mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# mgcp behavior negotiate-nse enable</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> exit</td>
<td>Exits global configuration mode and returns to privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# exit</td>
<td></td>
</tr>
</tbody>
</table>

### Verifying NSE Mode Configuration

#### SUMMARY STEPS

1. show mgcp

#### DETAILED STEPS

`show mgcp`

Use this command to display the state of the `mgcp behavior` command.

**Example:**

```
Router# show mgcp
MGCP Admin State ACTIVE, Oper State ACTIVE - Cause Code NONE
MGCP call-agent: 10.7.0.200 Initial protocol service is MGCP 0.1
```

The following lines show that the `mgcp behavior negotiate-nse enable` command is enabled:

**Example:**

```
mgcp modem passthrough voip mode nse
mgcp codec g723ar53 packetization-period 30
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp package-capability pre-package
mgcp package-capability mdste-package
```
Verifying the TGW or RGW Configuration

SUMMARY STEPS

1. show running-configuration

DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1  show running-configuration</td>
<td>Displays the current configuration settings.</td>
</tr>
</tbody>
</table>

Example:

Router(config)# show running-configuration

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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> mgcp block-newcalls</td>
<td>Prevents the gateway from accepting new calls.</td>
</tr>
</tbody>
</table>
| **Example:**  
  Router(config)# mgcp block-newcalls | |
| **Step 2** no mgcp block-newcalls | Restarts normal MGCP call operation. |
| **Example:**  
  Router(config)# no mgcp block-newcalls | |

**Configuration Examples for MGCP and Related Protocols**

- Configuring a Cisco AS5300 as a TGW with MGCP Example, page 18
- Configuring a Cisco AS5300 as a TGW with SGCP Example, page 19
- Configuring a Cisco 3660 as a TGW with MGCP Example, page 21
- Configuring a Cisco uBR924 as an RGW Example, page 22
- Configuring a Cisco 2620 as an RGW Example, page 23

**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.

```plaintext
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
```
Configuring a Cisco uBR924 as an RGW Example

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

```plaintext
! end

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
  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.

```bash
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
```
See the "Additional References for MGCP and SGCP" section on page x for related documents, standards, and MIBs.

- See the "Glossary" for definitions of terms in this guide.

### Additional References

#### Related Documents

<table>
<thead>
<tr>
<th>Related Topic</th>
<th>Document Title</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco IOS Voice commands</td>
<td>Cisco IOS Voice Command Reference</td>
</tr>
</tbody>
</table>

#### Technical Assistance

<table>
<thead>
<tr>
<th>Description</th>
<th>Link</th>
</tr>
</thead>
<tbody>
<tr>
<td>The Cisco Support and Documentation website provides online resources to download documentation, software, and tools. Use these resources to install and configure the software and to troubleshoot and resolve technical issues with Cisco products and technologies. Access to most tools on the Cisco Support and Documentation website requires a Cisco.com user ID and password.</td>
<td><a href="http://www.cisco.com/cisco/web/support/index.html">http://www.cisco.com/cisco/web/support/index.html</a></td>
</tr>
</tbody>
</table>

### Feature Information for Basic MGCP Configuration

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on Cisco.com is not required.
### Table 3: Feature Information for MGCP Basic Configuration

<table>
<thead>
<tr>
<th>Feature Name</th>
<th>Releases</th>
<th>Feature Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Configuring a TGW and RGW for MGCP</td>
<td>12.4(22)Y</td>
<td>Support was added for negotiating remote NSE support by configuring modem pass through on the gateway.</td>
</tr>
<tr>
<td>SDP Aware NSE Mode</td>
<td>15.1(3)T</td>
<td>Support was added for negotiating remote NSE support by configuring modem pass through on the gateway.</td>
</tr>
</tbody>
</table>

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Configuring MGCP 1.0

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.

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

• "Overview of MGCP and Related Protocols" on page 3

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

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(2)XA</td>
<td>This feature was introduced on the following platforms: Cisco CVA122, Cisco uBR924, and Cisco AS5300.</td>
</tr>
<tr>
<td>12.2(2)XA1</td>
<td>This feature was implemented on the following platforms: Cisco CVA122, Cisco uBR925, and Cisco AS5300</td>
</tr>
<tr>
<td>12.2(2)XB</td>
<td>This feature was implemented on the following platforms: Cisco AS5350 and Cisco AS5400.</td>
</tr>
</tbody>
</table>
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.

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.

The fax keyword was added to the mgcp playout command.

The maximum number of MGCP profiles that can be configured was increased from 13 (12 plus 1 default) to 29 (28 plus 1 default).

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest feature information and caveats, see the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the Feature Information Table at the end of this document.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on Cisco.com is not required.

Prerequisites for MGCP 1.0

Prerequisites are described in the “Prerequisites for Configuring MGCP and Related Protocols” section. 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). 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
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.

MGCP 1.0 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."

The figure below 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.

The figure below 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

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 32
- Configuring Global MGCP Parameters,  page 42
- Configuring an MGCP Profile and Profile-Related MGCP Parameters,  page 48
- Verifying the Configuration,  page 54
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 32
- Digital CAS Trunks, page 33
- ISUP Signaling Trunks, page 37
- FGD-OS Trunks, page 38
- Digital VoATM with AAL2 PVC, page 39

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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> dial-peer voice tag pots</td>
<td>Enters dial-peer configuration mode and specifies the method of voice encapsulation.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Router(config)# dial-peer voice tag pots</td>
</tr>
<tr>
<td><strong>Step 2</strong> application mgcpapp</td>
<td>Enables the MGCP application on this dial peer.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Router(config-dial-peer)# application mgcpapp</td>
</tr>
</tbody>
</table>
### 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:

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 3</strong> <code>port port-number</code></td>
<td>Associates a dial peer with a specific voice port.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td><code>Router(config-dial-peer)# port port-number</code></td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> <code>exit</code></td>
<td>Exits the current mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td><code>Router(config-dial-peer)# exit</code></td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> <code>mgcp [gw-port]</code></td>
<td>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.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td><code>Router(config)# mgcp [gw-port]</code></td>
<td></td>
</tr>
</tbody>
</table>
SUMMARY STEPS

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

DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 1**
controller `{t1 | e1} cntrl-number` | Configures a T1 or E1 controller and enters controller configuration mode for the digital CAS port. |
| **Example:**
`Router(config)# controller {t1 | e1} cntrl-number` | |
| **Step 2**
mode cas | (Required for Cisco MC3810 only) Configures the T1 or E1 controller to support CAS mode. |
| **Example:**
`Router(config-controller)# mode cas` | |
### Step 3
Do one of the following:

- **framing** `{sf | esf}`
- for T1 lines
- or for E1 lines
- **framing** `{crc4 | no-crc4} [australia]`

**Purpose:** Selects frame type for T1 or E1 line.

**Example:**

```plaintext
Router(config-controller)# framing {sf | esf}
```

### Step 4
Do one of the following:

- **linecode** `{ami | b8zs}`
- for T1 lines
- or for E1 lines
- **linecode** `{ami | hdb3}`

**Purpose:** Specifies the line encoding to use.

**Example:**

```plaintext
Router(config-controller)# linecode {ami | b8zs}
```

**Example:**

```plaintext
Router(config-controller)# linecode {ami | hdb3}
```
### Command or Action

<table>
<thead>
<tr>
<th>Step 5</th>
<th>ds0-group channel-number timeslots range type type</th>
</tr>
</thead>
</table>

**Example:**

```
Router(config-controller)# ds0-group channel-number timeslots range type type
```

**Purpose:** 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.

<table>
<thead>
<tr>
<th>Step 6</th>
<th>exit</th>
</tr>
</thead>
</table>

**Example:**

```
Router(config-controller)# exit
```

**Purpose:** Exits the current mode.

<table>
<thead>
<tr>
<th>Step 7</th>
<th>Do one of the following:</th>
</tr>
</thead>
</table>

- **voice-port slot/port:ds0-group-no**
- for Cisco 2600 and Cisco 3600 series
- or for Cisco MC3810
- **voice-port slot:ds0-group-no**

**Example:**

```
Router(config)# voice-port slot/port:ds0-group-no
```

**Example:**

```
Router(config)# voice-port slot:ds0-group-no
```

**Purpose:** Enters voice-port configuration mode.

| Step 8 | dial-type {dtmf | mf | pulse} |
|--------|--------------------------------|

**Example:**

```
Router(config-voiceport)# dial-type {dtmf | mf | pulse}
```

**Purpose:** (Required for MF trunks) Specifies the type of out-dialing for voice port interfaces. Default is `dtmf`. 

---

**MGCP and Related Protocols Configuration Guide, Cisco IOS Release 12.4T**

---

**Digital CAS Trunks**

**Configuring MGCP 1.0**
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 9 exit</td>
<td>Exits the current mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-voiceport)# exit</td>
</tr>
<tr>
<td>Step 10 dial peer voice <em>tag</em> pots</td>
<td>Enters dial-peer configuration mode and specifies the method of voice encapsulation.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config)# dial peer voice <em>tag</em> pots</td>
</tr>
<tr>
<td>Step 11 application mgcpapp</td>
<td>Enables the MGCP application on this dial peer.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-dial-peer)# application mgcpapp</td>
</tr>
<tr>
<td>Step 12 <strong>port</strong> port-number</td>
<td>Associates a dial peer with a specific voice port.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-dial-peer)# port port-number</td>
</tr>
<tr>
<td>Step 13 exit</td>
<td>Exits the current mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-dial-peer)# exit</td>
</tr>
<tr>
<td>Step 14 mgcp [gw-port]</td>
<td>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.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config)# mgcp [gw-port]</td>
</tr>
</tbody>
</table>

**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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> controller {t1</td>
<td>e1} cntlr-number</td>
</tr>
<tr>
<td>Example: Router(config)# controller {t1</td>
<td>e1} cntlr-number</td>
</tr>
<tr>
<td><strong>Step 2</strong> ds0-group channel-number timeslots range type none service mgcp</td>
<td>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.</td>
</tr>
<tr>
<td>Example: Router(config-controller)# ds0-group channel-number timeslots range type none service mgcp</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> exit</td>
<td>Exits the current mode.</td>
</tr>
<tr>
<td>Example: Router(config-controller)# exit</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> mgcp [gw-port]</td>
<td>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.</td>
</tr>
<tr>
<td>Example: Router(config)# mgcp [gw-port ]</td>
<td></td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>**controller {t1</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td><code>ds0-group channel-number timeslots range type fgd-os service mgcp</code>&lt;br&gt;Example: &lt;br&gt;<code>Router(config-controller)# ds0-group channel-number timeslots range type fgd-os service mgcp</code></td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td><code>exit</code>&lt;br&gt;Example: &lt;br&gt;<code>Router(config-controller)# exit</code></td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td><code>mgcp [gw-port]</code>&lt;br&gt;Example: &lt;br&gt;<code>Router(config)# mgcp [gw-port]</code></td>
</tr>
</tbody>
</table>

### 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. Do one of the following:
   - framing \{sf | esf\}
   - for T1 lines
   - or for E1 lines
   - framing \{crc4 | no-crc4\} [australia]
4. Do one of the following:
   - linecode \{ami | b8zs\}
   - for T1 lines
   - or for E1 lines
   - linecode \{ami | hdb3\}
5. **exit**
6. **dial peer voice tag pots**
7. **application mgcpapp**
8. **port port-number**
9. **exit**
10. **mgcp** \[gw-port\]

DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> controller {t1</td>
<td>e1} <em>cntlr-number</em></td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# controller {t1</td>
<td>e1} <em>cntlr-number</em></td>
</tr>
<tr>
<td><strong>Step 2</strong> mode atm</td>
<td>Specifies that the controller supports ATM encapsulation and create ATM interface 0. When the controller is set to ATM mode, the following takes place:</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-controller)# mode atm</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Controller framing is automatically set to Extended Superframe (ESF).</td>
</tr>
<tr>
<td></td>
<td>• The line code is automatically set to B8ZS.</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
</tr>
<tr>
<td><strong>Step 3</strong> Do one of the following:</td>
<td>Selects frame type for T1 or E1 line. T1 default is sf. E1 default is crc4.</td>
</tr>
<tr>
<td>• framing {sf</td>
<td>esf}</td>
</tr>
<tr>
<td>• for T1 lines</td>
<td></td>
</tr>
<tr>
<td>• or for E1 lines</td>
<td></td>
</tr>
<tr>
<td>• framing {crc4</td>
<td>no-crc4} [australia]</td>
</tr>
</tbody>
</table>

**Example:**

```
Router(config-controller)# framing {sf | esf}
```

**Example:**

```
Router(config-controller)# framing {crc4 | no-crc4} [australia]
```

<table>
<thead>
<tr>
<th><strong>Step 4</strong> Do one of the following:</th>
<th>Specifies the line encoding to use. T1 default is ami. E1 default is hdb3.</th>
</tr>
</thead>
<tbody>
<tr>
<td>• linecode {ami</td>
<td>b8zs}</td>
</tr>
<tr>
<td>• for T1 lines</td>
<td></td>
</tr>
<tr>
<td>• or for E1 lines</td>
<td></td>
</tr>
<tr>
<td>• linecode {ami</td>
<td>hdb3}</td>
</tr>
</tbody>
</table>

**Example:**

```
Router(config-controller)# linecode {ami | b8zs}
```

**Example:**

```
Router(config-controller)# linecode {ami | hdb3}
```

<table>
<thead>
<tr>
<th><strong>Step 5</strong> exit</th>
<th>Exits the current mode.</th>
</tr>
</thead>
</table>

**Example:**

```
Router(config-controller)# exit
```
### Command or Action

<table>
<thead>
<tr>
<th>Step 6</th>
<th>dial peer voice tag pots</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Purpose</strong></td>
<td>Enters dial-peer configuration mode and specifies the method of voice encapsulation.</td>
</tr>
</tbody>
</table>

**Example:**
```
Router(config)# dial peer voice tag pots
```

<table>
<thead>
<tr>
<th>Step 7</th>
<th>application mgcpapp</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Purpose</strong></td>
<td>Enables the MGCP application on this dial peer.</td>
</tr>
</tbody>
</table>

**Example:**
```
Router(config-dial-peer)# application mgcpapp
```

<table>
<thead>
<tr>
<th>Step 8</th>
<th>port port-number</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Purpose</strong></td>
<td>Associates a dial peer with a specific voice port.</td>
</tr>
</tbody>
</table>

**Example:**
```
Router(config-dial-peer)# port port-number
```

<table>
<thead>
<tr>
<th>Step 9</th>
<th>exit</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Purpose</strong></td>
<td>Exits the current mode.</td>
</tr>
</tbody>
</table>

**Example:**
```
Router(config-dial-peer)# exit
```

<table>
<thead>
<tr>
<th>Step 10</th>
<th>mgcp [gw-port]</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Purpose</strong></td>
<td>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.</td>
</tr>
</tbody>
</table>

**Example:**
```
Router(config)# mgcp [gw-port]
```

---

## 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, page 48.
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}`
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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> mgcp call-agent {dns-name</td>
<td>ip-address} [port] [service-type type] [version protocol-version]</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>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.</td>
</tr>
<tr>
<td>Router(config)# mgcp call-agent {dns-name</td>
<td>ip-address} [port] [service-type type] [version protocol-version]</td>
</tr>
<tr>
<td><strong>Step 2</strong> mgcp behavior {auep</td>
<td>signal} v0.1</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>• <strong>auep</strong> --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.</td>
</tr>
<tr>
<td>Router(config)# mgcp behavior {auep</td>
<td>signal} v0.1</td>
</tr>
<tr>
<td><strong>Step 3</strong> mgcp sdp simple</td>
<td>Specifies that a subset of the SDP fields should be used.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>• <strong>v0.1</strong> --Selects MGCP Version 0.1.</td>
</tr>
<tr>
<td>Router(config)# mgcp sdp simple</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> mgcp sdp xpc-codec</td>
<td>Enables codec negotiation in the SDP.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# mgcp sdp xpc-codec</td>
<td></td>
</tr>
</tbody>
</table>
### Command or Action

<table>
<thead>
<tr>
<th>Step 5</th>
<th><code>mgcp codec type [packetization-period value]</code></th>
<th>Selects the default codec type and its optional packetization period value.</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Example:</strong></td>
<td><code>Router(config)# mgcp codec type [packetization-period value]</code></td>
<td></td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
<td></td>
</tr>
<tr>
<td>------------------</td>
<td>---------</td>
<td></td>
</tr>
<tr>
<td><strong>Step 11</strong> mgcp dtmf-relay voip codec {all</td>
<td>low-bit-rate} mode {cisco</td>
<td>nse</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config)# mgcp dtmf-relay voip codec {all</td>
<td>low-bit-rate} mode {cisco</td>
<td>nse</td>
</tr>
<tr>
<td><strong>Step 12</strong> mgcp max-waiting-delay <em>value</em></td>
<td>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.</td>
<td></td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config)# mgcp max-waiting-delay <em>value</em></td>
<td>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.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 13</strong> mgcp restart-delay <em>value</em></td>
<td>Sets the delay value sent in the RestartInProgress (RSIP) graceful teardown, in seconds. Range is from 0 to 600. Default is 0.</td>
<td></td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config)# mgcp restart-delay <em>value</em></td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Step 14</strong> mgcp vad</td>
<td>Enables voice activity detection (VAD) as a default for MGCP calls. Default is disabled.</td>
<td></td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config)# mgcp vad</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Step 15</strong> mgcp ip-tos {high-reliability</td>
<td>high-throughput</td>
<td>low-cost</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config)# mgcp ip-tos {high-reliability</td>
<td>high-throughput</td>
<td>low-cost</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
<td></td>
</tr>
<tr>
<td>------------------</td>
<td>---------</td>
<td></td>
</tr>
<tr>
<td><strong>Step 16</strong> mgcp quality-threshold {hwm-cell-loss value</td>
<td>hwm-jitter-buffer value</td>
<td>hwm-latency value</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Router(config)# mgcp quality-threshold {hwm-cell-loss value</td>
<td>hwm-jitter-buffer value</td>
<td>hwm-latency value</td>
</tr>
<tr>
<td><strong>Step 17</strong> mgcp playout / adaptive init-value min-value max-value</td>
<td>fax value</td>
<td>fixed init-value</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Router(config)# mgcp playout adaptive init-value min-value max-value</td>
<td>fax value</td>
<td>fixed init-value</td>
</tr>
<tr>
<td><strong>Step 18</strong> mgcp package-capability [package-type]</td>
<td>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.</td>
<td></td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Router(config)# mgcp package-capability [package-type]</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Step 19</strong> mgcp default package [package-type]</td>
<td>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.</td>
<td></td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Router(config)# mgcp default package [package-type]</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
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, page 42.

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.

```
mhcp 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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong>\ m<strong>gc</strong>p profile (\textit{profile-name} \mid \textit{default})</td>
<td>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. Effective with Cisco IOS Release 12.4(24)T3, the maximum number of MGCP profiles that can be configured is increased from 13 (12 plus 1 default) to 29 (28 plus 1 default).</td>
</tr>
<tr>
<td><strong>Example:</strong> [\text{Router(config)# mgcp profile } {\textit{profile-name } \mid \textit{default}}]</td>
<td></td>
</tr>
<tr>
<td>Step</td>
<td>Command or Action</td>
</tr>
<tr>
<td>--------</td>
<td>-------------------------</td>
</tr>
<tr>
<td>7</td>
<td><code>timeout tmax tmax-value</code></td>
</tr>
<tr>
<td></td>
<td>Example:</td>
</tr>
<tr>
<td>8</td>
<td><code>timeout tdmax tdmax-value</code></td>
</tr>
<tr>
<td></td>
<td>Example:</td>
</tr>
<tr>
<td>9</td>
<td><code>timeout tdinit tdinit-value</code></td>
</tr>
<tr>
<td></td>
<td>Example:</td>
</tr>
<tr>
<td>10</td>
<td><code>timeout tcrit tcrit-value</code></td>
</tr>
<tr>
<td></td>
<td>Example:</td>
</tr>
<tr>
<td>11</td>
<td><code>timeout tpar tpar-value</code></td>
</tr>
<tr>
<td></td>
<td>Example:</td>
</tr>
<tr>
<td>12</td>
<td><code>timeout thist thist-value</code></td>
</tr>
<tr>
<td></td>
<td>Example:</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>-------------------------------------------</td>
<td>--------------------------------------------------------------------------</td>
</tr>
<tr>
<td><strong>Step 13</strong> timeout tone mwi mwitone-value</td>
<td>Configures the message waiting indicator timeout value, in seconds. Range is from 1 to 600. Default is 16.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-mgcp-profile)# timeout tone mwi mwitone-value</td>
<td></td>
</tr>
<tr>
<td><strong>Step 14</strong> timeout tone ringback ringbacktone-value</td>
<td>Configures the ringback tone timeout value, in seconds. Range is from 1 to 600. Default is 180.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-mgcp-profile)# timeout tone ringback ringbacktone-value</td>
<td></td>
</tr>
<tr>
<td><strong>Step 15</strong> timeout tone ringback connection connectiontone-value</td>
<td>Configures the timeout value for ringback tone on connection, in seconds. Range is from 1 to 600. Default is 180.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-mgcp-profile)# timeout tone ringback connection connectiontone-value</td>
<td></td>
</tr>
<tr>
<td><strong>Step 16</strong> timeout tone network congestion congestiontone-value</td>
<td>Configures the network congestion tone timeout value, in seconds. Range is from 1 to 600. Default is 180.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-mgcp-profile)# timeout tone network congestion congestiontone-value</td>
<td></td>
</tr>
<tr>
<td><strong>Step 17</strong> timeout tone busy busytone-value</td>
<td>Configures the busy tone timeout value, in seconds. Range is from 1 to 600. Default is 3.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-mgcp-profile)# timeout tone busy busytone-value</td>
<td></td>
</tr>
<tr>
<td><strong>Step 18</strong> timeout tone dial dialtone-value</td>
<td>Configures the dial tone timeout value, in seconds. Range is from 1 to 600. Default is 16.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-mgcp-profile)# timeout tone dial dialtone-value</td>
<td></td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>------------------</td>
<td>---------</td>
</tr>
<tr>
<td><strong>Step 19</strong> timeout tone dial stutter <em>stuttertone-value</em></td>
<td>Configures the stutter dial tone timeout value, in seconds. Range is from 1 to 600. Default is 16.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-mgcp-profile)# timeout tone dial stutter <em>stuttertone-value</em></td>
<td></td>
</tr>
<tr>
<td><strong>Step 20</strong> timeout tone ringing <em>ringingtone-value</em></td>
<td>Configures the ringing tone timeout value, in seconds. Range is from 1 to 600. Default is 180.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-mgcp-profile)# timeout tone ringing <em>ringingtone-value</em></td>
<td></td>
</tr>
<tr>
<td><strong>Step 21</strong> timeout tone ringing distinctive <em>distinctivetone-value</em></td>
<td>Configures the distinctive ringing tone timeout value, in seconds. Range is from 1 to 600. Default is 180.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-mgcp-profile)# timeout tone ringing distinctive <em>distinctivetone-value</em></td>
<td></td>
</tr>
<tr>
<td><strong>Step 22</strong> timeout tone reorder <em>reordertone-value</em></td>
<td>Configures the reorder tone timeout value, in seconds. Range is from 1 to 600. Default is 30.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-mgcp-profile)# timeout tone reorder <em>reordertone-value</em></td>
<td></td>
</tr>
<tr>
<td><strong>Step 23</strong> timeout tone cot1 <em>continuity1tone-value</em></td>
<td>Configures the continuity1 tone timeout value, in seconds. Range is from 1 to 600. Default is 3.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-mgcp-profile)# timeout tone cot1 <em>continuity1tone-value</em></td>
<td></td>
</tr>
<tr>
<td><strong>Step 24</strong> timeout tone cot2 <em>continuity2tone-value</em></td>
<td>Configures the continuity2 tone timeout value, in seconds. Range is from 1 to 600. Default is 3.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-mgcp-profile)# timeout tone cot2 <em>continuity2tone-value</em></td>
<td></td>
</tr>
</tbody>
</table>
### Command or Action | Purpose
--- | ---
**Step 25** max1 lookup | Enables the DNS lookup procedure after the suspicion threshold is reached. Default is enabled.
**Example:**
```
Router(config-mgcp-profile)# max1 lookup
```
**Step 26** max1 retries value | Sets the suspicion threshold number of retries. Range is from 3 to 30. Default is 5.
**Example:**
```
Router(config-mgcp-profile)# max1 retries value
```
**Step 27** max2 lookup | Enables the DNS lookup procedure after the disconnect threshold is reached. Default is enabled.
**Example:**
```
Router(config-mgcp-profile)# max2 lookup
```
**Step 28** max2 retries value | Sets the disconnect threshold number of retries. Range is from 3 to 30. Default is 7.
**Example:**
```
Router(config-mgcp-profile)# max2 retries value
```
**Step 29** exit | Exits the current mode.
**Example:**
```
Router(config-mgcp-profile)# exit
```

### Verifying the Configuration

**SUMMARY STEPS**

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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> show running-configuration</td>
<td>Displays the current configuration settings.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router# show running-configuration</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> show mgcp [connection</td>
<td>endpoint</td>
</tr>
<tr>
<td><strong>Example:</strong> Router# show mgcp [connection</td>
<td>endpoint</td>
</tr>
</tbody>
</table>

- **Troubleshooting Tips, page 55**

## 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 `mgcapp` 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 0x0A 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 (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 33 20 2A 20 4D 47 43 50 20 30 2E 31 A
send_mgcp_msg, MGCP Packet sent --->
250 49993
```

Configuration Examples for MGCP 1.0

- Cisco uBR925 Using Radio Frequency Interface Example, page 56
- Cisco uBR925 Using Ethernet0 Interface Example, page 58
- Cisco CVA122 Using Radio Frequency Interface Example, page 59
- Cisco 2600 Series as a Residential Gateway Example, page 60
- Cisco 3660 Platform as a Trunking Gateway Example, page 62
- Cisco MC3810 as a Residential Gateway Example, page 63
- Cisco MC3810 as a VoAAL2 Gateway using AAL2 PVCs Example, page 64

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
!
smtp-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
!
mgc 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
```
Cisco uBR925 Using Ethernet0 Interface Example

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

```bash
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
!
!
mccp
!
!
mccp call-agent 10.0.0.224 service-type ncs version 1.0
! Use this CLI with NCS 1.0
```

![Configuration Examples for MGCP 1.0](mgcp-related-protocols-configuration-guide-cisco-ios-release-12.4t)
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.

```plaintext
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.

```plaintext
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
!
```
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 sdtp 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 tmax 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
Cisco MC3810 as a Residential Gateway Example

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

```plaintext
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
```

Cisco MC3810 as a Residential Gateway Example

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

```conf
testing
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:

```conf
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 vty 0 4
 password lab
 login
!
end

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.
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actual addresses and phone numbers. Any examples, command display output, network topology diagrams,
and other figures included in the document are shown for illustrative purposes only. Any use of actual IP
addresses or phone numbers in illustrative content is unintentional and coincidental.
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.

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

- "Overview of MGCP and Related Protocols" on page 3

Feature History for MGCP Basic (CLASS) and Operation Services

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
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<tbody>
<tr>
<td>12.2(2)T</td>
<td>This feature was introduced.</td>
</tr>
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</table>

- Finding Feature Information, page 67
- Prerequisites for MGCP Basic CLASS and Operator Services, page 68
- Restrictions for MGCP Basic CLASS and Operator Services, page 68
- Information About MGCP Basic CLASS and Operator Services, page 68
- Troubleshooting MGCP Basic CLASS and Operator Services, page 74
- Configuration Examples for MGCP Basic CLASS and Operator Services, page 75

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest feature information and caveats, see the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the Feature Information Table at the end of this document.
Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on Cisco.com is not required.

Prerequisites for 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.

• ◦ 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):

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

• Distinctive Power Ring, page 69
• Visual Message Waiting Indicator, page 69
• Caller ID, page 69
• Caller ID with Call Waiting, page 69
• Call Forwarding, page 69
• Ring Splash, page 70
• Distinctive Call-Waiting Tone, page 70
• Message-Waiting Tone, page 70
• Stutter Dial Tone, page 70
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

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, page 71
- Examples of Service-Provider Solutions, page 71

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. The figure below illustrates a possible residential cable access solution.

**Figure 4  Residential Cable Access Solution**

![Figure 4 Residential Cable Access Solution](image)

Note that, in the figure above, 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. The figure below illustrates an IP Centrex solution:

**Figure 5  IP Centrex Solution**

![Figure 5 IP Centrex Solution](image)

In the figure above, 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. The figure below illustrates an integrated access solution.

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

• Telecommuter or small-office-home-office solution
The figure below illustrates a telecommuter or small-office-home-office solution:

**Figure 7** Telecommuter or Small Office-Home Office Solution

In the figure above, the residential gateway must support the CLASS features. Other solutions are possible using the MGCP open protocol.

**Troubleshooting 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

The table below summarizes the findings for the models tested.

**Table 4** Telephones and Feature Capabilities

<table>
<thead>
<tr>
<th>Telephone</th>
<th>CID</th>
<th>VMWI</th>
<th>CIDCW</th>
</tr>
</thead>
<tbody>
<tr>
<td>Casio TI-345</td>
<td>Y</td>
<td>--</td>
<td>N</td>
</tr>
<tr>
<td>Casio TI-360</td>
<td>Y</td>
<td>--</td>
<td>N</td>
</tr>
<tr>
<td>Dial Digital CP-2892C</td>
<td>Y</td>
<td>Y</td>
<td>Y</td>
</tr>
<tr>
<td>Telephone</td>
<td>CID</td>
<td>VMWI</td>
<td>CIDCW</td>
</tr>
<tr>
<td>--------------------</td>
<td>-----</td>
<td>------</td>
<td>-------</td>
</tr>
<tr>
<td>GE 29299GE1-A</td>
<td>Y</td>
<td>--</td>
<td>Y</td>
</tr>
<tr>
<td>Panasonic KX-TSC7</td>
<td>Y</td>
<td>N</td>
<td>N</td>
</tr>
<tr>
<td>Panasonic KX-TSC55-b</td>
<td>Y</td>
<td>Y</td>
<td>Y</td>
</tr>
<tr>
<td>Sony IT-ID80</td>
<td>Y</td>
<td>--</td>
<td>Y</td>
</tr>
</tbody>
</table>

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:

1. Attach the phone to the voice port.
2. Do a "shut" to the voice port.
3. Do a "no shut" to the voice port.
4. 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 x for related documents, standards, and MIBs and the "Glossary" for definitions of terms in this guide.

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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

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

- "Overview of MGCP and Related Protocols" on page 3

### Feature History for NAS Package for MGCP

<table>
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<tr>
<th>Release</th>
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<td>12.2(2)XB</td>
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<td>12.2(11)T</td>
<td>This feature was implemented on the Cisco AS5850.</td>
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- Finding Feature Information, page 78
- Prerequisites for NAS Package for MGCP, page 78
- Information About NAS Package for MGCP, page 78
- How to Configure NAS Package for MGCP, page 79
- Configuration Examples for NAC Package for MGCP, page 108
Finding Feature Information

Your software release may not support all the features documented in this module. For the latest feature information and caveats, see the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the Feature Information Table at the end of this document.

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Prerequisites for NAS 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 NAS Package for MGCP

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.

The figure below shows a typical network topology for universal port media gateways.

*Figure 8  Media Gateways Operating As Network Access Servers*

---

**How to Configure NAS Package for MGCP**

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 80
- Configuring Controllers, page 80
- Configuring Dialer Interfaces and Routing, page 83
- Verifying the NAS Package for MGCP, page 88
- Troubleshooting Tips, page 89
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 NAS Package for MGCP 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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> mgcp [gw-port]</td>
<td>Allocates resources for MGCP and starts the MGCP daemon. If no port is specified, the command defaults to port 2427.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# mgcp [gw-port]</td>
<td></td>
</tr>
</tbody>
</table>

| **Step 2** 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. |
| Example: | |
| Router(config)# mgcp call-agent {dns-name | ip-address} [ca-port] [service-type type] [version protocol-version] | |

| **Step 3** 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. |
| Example: | |
| Router(config)# mgcp default-package nas-package | |

**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. Do one of the following:
   - `framing {sf | esf}`
   - for TI lines
   - or for E1 lines
   - `framing {crc4 | no-crc4} [australia]`

3. `extsig mgcp`

4. `guard-timer milliseconds [on-expiry {accept | reject}]`

5. Do one of the following:
   - `linecode {ami | b8zs}`
   - for TI lines
   - or for E1 lines
   - `linecode {ami | hdb3}`

6. `ds0-group channel-number timeslots range type none service mgcp`

7. `exit`

**DETAILED STEPS**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>**Step 1 controller {t1</td>
<td>e1} slot/port**</td>
</tr>
</tbody>
</table>

**Example:**

```
Router(config)# controller {t1 | e1} slot/port
```
### Command or Action

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Do one of the following:</td>
<td>Selects the frame type for the T1 or E1 trunk. T1 default is <strong>sf</strong>, E1 default is <strong>crc4</strong>.</td>
</tr>
<tr>
<td></td>
<td>- framing {sf</td>
<td>esf}</td>
</tr>
<tr>
<td></td>
<td>- for T1 lines</td>
<td></td>
</tr>
<tr>
<td></td>
<td>- or for E1 lines</td>
<td></td>
</tr>
<tr>
<td></td>
<td>- framing {crc4</td>
<td>no-crc4} [australia]</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-controller)# framing {sf</td>
<td>esf}</td>
</tr>
<tr>
<td>Example:</td>
<td>Routere(config-controller)# framing {crc4</td>
<td>no-crc4} [australia]</td>
</tr>
<tr>
<td>Step 3</td>
<td><strong>extsig mgcp</strong></td>
<td>Configures external signaling control by MGCP for this controller. For T3 trunks, each logical T1 must be configured with the <strong>extsig mgcp</strong> command.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-controller)# extsig mgcp</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td>**guard-timer milliseconds [on-expiry {accept</td>
<td>reject}]**</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-controller)# guard-timer milliseconds [on-expiry {accept</td>
<td>reject}]</td>
</tr>
</tbody>
</table>
### 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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 5** Do one of the following:  
  • linecode \{ami \| b8zs\}  
  • for T1 lines  
  • or for E1 lines  
  • linecode \{ami \| hdb3\}  | Specifies the line encoding to use. T1 default is **ami**. E1 default is **hdb3**. |
| Example:  
  Router(config-controller)# linecode \{ami \| b8zs\}  |  |
| Example:  
  Router(config-controller)# linecode \{ami \| hdb3\}  |  |
| **Step 6** 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. |
| Example:  
  Router(config-controller)# ds0-group channel-number  |  |
| Example:  
  Router(config-controller)# ds0-group channel-number timeslots range type none service mgcp  |  |
| **Step 7** exit  | Exits the current mode. |
| Example:  
  Router(config-controller)# exit  |  |
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. Do one of the following:
   - `ip unnumbered interface-number`
   - `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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> <code>interface dialer-name</code></td>
<td>Enters interface mode for the dialer interface.</td>
</tr>
</tbody>
</table>

**Example:**

```
Router(config)# interface dialer-name
```
### Command or Action

<table>
<thead>
<tr>
<th>Step 2</th>
<th>Do one of the following:</th>
</tr>
</thead>
<tbody>
<tr>
<td>• ip unnumbered interface-number</td>
<td></td>
</tr>
<tr>
<td>• ip address ip-address subnet-mask [secondary]</td>
<td></td>
</tr>
</tbody>
</table>

**Example:**

```
Router(config-if)# ip unnumbered interface-number
```

**Example:**

```
Router(config-if)# ip address ip-address subnet-mask [secondary]
```

### Purpose

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

**encapsulation ppp**

**Example:**

```
Router(config-if)# encapsulation ppp
```

Sets encapsulation type for PPP.

### Step 4

**dialer in-band [no-parity | odd-parity]**

**Example:**

```
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

**dialer idle-timeout seconds [inbound | either]**

**Example:**

```
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.
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 6</strong> dialer map protocol next-hop-address [name host-name] [dial-string[: isdn-subaddress]]</td>
<td>Configures a serial interface to make digital calls or to accept incoming calls from a specified location and to authenticate if so configured.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-if)# dialer map protocol next-hop-address [name host-name] [dial-string[: isdn-subaddress]]</td>
<td></td>
</tr>
<tr>
<td><strong>Step 7</strong> dialer extsig</td>
<td>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.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-if)#</td>
<td></td>
</tr>
<tr>
<td>dialer extsig</td>
<td></td>
</tr>
<tr>
<td><strong>Step 8</strong> dialer-group number</td>
<td>Controls access by configuring an interface to belong to a specific dialing group.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-if)#</td>
<td></td>
</tr>
<tr>
<td>dialer-group number</td>
<td></td>
</tr>
<tr>
<td><strong>Step 9</strong> no cdp enable</td>
<td>Disables Cisco Discovery Protocol (CDP) on the interface.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-if)# no cdp enable</td>
<td></td>
</tr>
<tr>
<td><strong>Step 10</strong> ppp authentication chap</td>
<td>Enables Challenge Handshake Authentication Protocol (CHAP) authentication on the interface.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-if)# ppp authentication chap</td>
<td></td>
</tr>
<tr>
<td><strong>Step 11</strong> exit</td>
<td>Exits the current mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-if)# exit</td>
<td></td>
</tr>
</tbody>
</table>
### Command or Action

**Step 12** `dialer list number protocol protocol-name {permit | deny [list access-list-number | access-group]}`

**Example:**

```
Router(config)# dialer list number
protocol protocol-name
{permit | deny [list access-list-number
| access-group ]}
```

**Purpose**
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** `ip route prefix mask [ip-address | interface-type interface-number] [distance] [tag tag] [permanent]`

**Example:**

```
Router(config)# ip route prefix mask
{ip-address | interface-type interface-number
| [distance
] [tag tag
| [permanent]
```

**Purpose**
Establishes a static route. Because you do not want dynamic routing protocols running across the DDR links, you manually configure static routes.

### Verifying the NAS Package for MGCP

To verify configuration, use the following commands.

**SUMMARY STEPS**

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

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

**DETAILED STEPS**

**Step 1**

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

**Example:**

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

The following example shows the configuration for serial interface 1:

**Example:**

```
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:

Example:

Router# show mgcp nas
dump
slot
port
channel 
info

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

Example:

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:

Example:

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 ID ID ID 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 90
MGCP Troubleshooting

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

<table>
<thead>
<tr>
<th>Command</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>show mgcp nas info</td>
<td>Displays status of the MGCP data channels.</td>
</tr>
<tr>
<td>show mgcp nas dump</td>
<td>Displays status and details about the specified MGCP data slot, port, and channel.</td>
</tr>
<tr>
<td>slot port chan</td>
<td></td>
</tr>
<tr>
<td>show mgcp connection</td>
<td>Displays active MGCP connections on the router.</td>
</tr>
<tr>
<td>show xcsp slot slot-num</td>
<td>Displays the status of a router slot under the control of the External Call Service Provider (XCSP) subsystem.</td>
</tr>
<tr>
<td>show xcsp port slot</td>
<td>Displays the status of a port under the control of the External Call Service Provider (XCSP) subsystem.</td>
</tr>
</tbody>
</table>

Router# show mgcp nas info

Router# show mgcp nas dump slot port chan

Router# show mgcp connection

Router# show xcsp slot slot-num

Router# show xcsp port slot
## Command

<table>
<thead>
<tr>
<th>Command</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>show cdapi</td>
<td>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.</td>
</tr>
</tbody>
</table>

Router# show cdapi

See Example Output for show cdapi Command, page 93.
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
```

MGCP Debugging

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

<table>
<thead>
<tr>
<th>Command</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>debug mgcp all</code></td>
<td>Enables all MGCP debugs.</td>
</tr>
<tr>
<td></td>
<td>See Example Output for <code>debug mgcp all</code> Command, page 94.</td>
</tr>
<tr>
<td><code>debug mgcp events</code></td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>See Example Output for <code>debug mgcp events</code> Command, page 94.</td>
</tr>
<tr>
<td><code>debug mgcp packets</code></td>
<td>Enables debugging of MGCP packets. Useful for displaying contents of NTFY, CRCX, DLCX, and other packets.</td>
</tr>
<tr>
<td></td>
<td>See Example Output for <code>debug mgcp packets</code> Command, page 94.</td>
</tr>
<tr>
<td>Command</td>
<td>Purpose</td>
</tr>
<tr>
<td>-----------------</td>
<td>---------------------------------------------------------------------------------------------------</td>
</tr>
</tbody>
</table>
| 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.  
See Example Output for debug mgcp parser Command, page 95. |
| debug mgcp nas   | Enables debugging for MGCP data channels and events.  
See Example Output for debug mgcp nas Command, page 95. |
| 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.  
See Example Output for debug xcsp Command, page 95. |
| debug cdapi {detail | events} | Displays real-time information about the call distributor application programming interface (CDAPI).  
See Example Output for debug cdapi Command, page 97. |

**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#
lwd: MGCP stat - 172.19.184.65, total=44, succ=7, failed=21
lwd: MGCP msg 1
lwd: remove_old_under_specified_ack:
lwd: MGCP stat - 172.19.184.65, total=44, succ=8, failed=21
lwd: updating lport with 2427setup_ipsocket: laddr=172.29.248.193, lport=2427,
faddr=172.19.184.65, fport=2427
lwd: 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_chq_nas_pkg: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:cdn
   mgcp_chq_nas_pkg:string past colon:1000
   CHECK DATA CALL for S7/DS1-0/23
   mgcppapp_xcsp_get_chan_cb -Found - Channel state Idle
   CRCX Recv
   mgcppapp_endpt_is_data:endpt S7/DS1-0/23, slot 7, port 0 chan 23
   mgcppapp_data_call_hnd:mgcppapp_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:Received CONN_RESP, channel state = 8 for slot port channel 7 0 23
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 connect mgcpapp_xcsp_alert:
mgcpapp_xcsp_get_chan_cb -Found - Channel state Connection in progress
200 S8 Alert
I:630AED90
<---:Ack send SUCCESSFUL
01:49:14:xcsp fsm:slot 7 p
c5400#port 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: process_cdapi_msg,new slot/port/channel 7/0/23
01:49:14: Received CALL_CONN:callid=0x7016
01:49:14:process_cdapi:Event CALL_CONN, channel state = 7 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_xcsp_get_chan_cb -Found - Channel state In Use
01:49:14:STOP TIMER
c5400#process_cdapi_msg: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: 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 CALL_DISC_REQ, channel state = 7 for slot port channel 7 0 23
01:50:23:xcsp_process_sig_fsm:state/event Connection in progress / in call connect mgcpapp_xcsp_disconnect
mgcpapp_xcsp_get_chan_cb -Found - Channel state Out Release in progress
mgcpapp_xcsp_disconnect
mgcpapp_xcsp_get_chan_cb -Found - Channel state Out Release in progress
mgcpapp_xcsp_disconnect
```

Example Output for debug xcsp Command
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 = 0x99FAA068001008201008B0100A118020274C020108008F534341524C415454492D3530303733
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
```

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.
<table>
<thead>
<tr>
<th>Command</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| `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.  
See Example Output for show controllers e1 or t1 Command, page 99. |
| `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.  
See Example Output for show voice port Command, page 100. |
| `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. |
| `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, page 99
- Example Output for show voice port Command, page 100
- Dialer Interface and Routing Troubleshooting, page 102
- Example Output for show dialer map Command, page 103
- Example Output for show dialer Command, page 103
- Example Output for show interface Command, page 104
- Example Output for show ip route Command, page 105
- Example Output for debug dialer Command, page 106
- Example Output for clear interface Command, page 106
- Example Output for debug ppp negotiation Command, page 107
- Example Output for debug ppp authentication Command, page 107
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
  7 Slip Secs, 7 Fr Loss Secs, 7 Line Err Secs, 0 Degraded Mins
  0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 0 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
```
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 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 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
```
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 a-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.
<table>
<thead>
<tr>
<th>Command</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>show dialer map</strong></td>
<td>Displays configured dynamic and static dialer maps.</td>
</tr>
<tr>
<td></td>
<td>See Example Output for show dialer map Command, page 103.</td>
</tr>
<tr>
<td><strong>show dialer</strong></td>
<td>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 timer 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.</td>
</tr>
<tr>
<td></td>
<td>See Example Output for show dialer Command, page 103.</td>
</tr>
<tr>
<td><strong>show interface Dialer-num</strong></td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>See Example Output for show interface Command, page 104.</td>
</tr>
<tr>
<td><strong>show ip route</strong></td>
<td>Displays the routes known to the router, including static and dynamically learned routes.</td>
</tr>
<tr>
<td></td>
<td>See Example Output for show ip route Command, page 105.</td>
</tr>
</tbody>
</table>

**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
```
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)
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), Fast 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, BR10
 P 172.27.4.0/8 [1/0] via 103.1.1.1, BR10
 S 172.31.0.0/16 [1/0] via 172.21.114.65, Ethernet0
 S 10.0.0.0/8 is directly connected, BR10
 P 10.0.0.0/8 is directly connected, BR10
172.21.0.0/16 is variably subnetted, 5 subnets, 2 masks
 S 172.21.114.201/32 is directly connected, BR10
 S 172.21.114.209/32 is directly connected, BR10
 S 172.21.114.174/32 is directly connected, BR10
 S 172.21.114.12/32 is directly connected, BR10
 P 10.0.0.0/8 is directly connected, BR10
 P 10.1.0.0/8 is directly connected, BR10
 P 10.2.2.0/8 is directly connected, BR10
 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

<table>
<thead>
<tr>
<th>Command</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>debug dialer</strong></td>
<td>Displays the activity that triggers a dial attempt.</td>
</tr>
<tr>
<td></td>
<td>See Example Output for show dialer Command, page 103.</td>
</tr>
<tr>
<td><strong>clear interface</strong></td>
<td>Clears a call that is in progress. In a troubleshooting situation, it</td>
</tr>
<tr>
<td></td>
<td>is sometimes useful to clear historical statistics to track the</td>
</tr>
<tr>
<td></td>
<td>current number of successful calls relative to failures. Use this</td>
</tr>
<tr>
<td></td>
<td>command with care. It sometimes requires that you clear both the</td>
</tr>
<tr>
<td></td>
<td>local and remote routers.</td>
</tr>
<tr>
<td><strong>debug ppp negotiation</strong></td>
<td>Displays negotiation of PPP options and Network Control Protocol</td>
</tr>
<tr>
<td></td>
<td>(NCP) parameters.</td>
</tr>
<tr>
<td></td>
<td>See Example Output for debug ppp negotiation Command, page 107.</td>
</tr>
<tr>
<td><strong>debug ppp authentication</strong></td>
<td>Displays exchange of Challenge Handshake Authentication Protocol</td>
</tr>
<tr>
<td></td>
<td>(CHAP) and Password Authentication Protocol (PAP) packets.</td>
</tr>
<tr>
<td></td>
<td>See Example Output for debug ppp authentication Command, page 107.</td>
</tr>
</tbody>
</table>

**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
```
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) value = C025/3E8 acked (ok)
PPP Serial14: 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
```

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
```

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 NAC Package for MGCP

NAS Package for MGCP Example

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

```plaintext
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 user��sample
!
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-recipes 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
```

MGCP and Related Protocols Configuration Guide, Cisco IOS Release 12.4T
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
See the "Additional References for MGCP and SGCP" section on page x for related documents, standards, and MIBs and 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.

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

- "Overview of MGCP and Related Protocols" on page 3

Feature History for SGCP RSIP and AUEP Enhancements

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(11)T</td>
<td>This feature was introduced on the following platforms: Cisco IAD2420 series, Cisco 2600 series, and Cisco MC3810.</td>
</tr>
</tbody>
</table>

- Finding Feature Information, page 113
- Prerequisites for SGCP RSIP and AUEP Enhancements, page 114
- Restrictions for SGCP RSIP and AUEP Enhancements, page 114
- Information About SGCP RSIP and AUEP Enhancements, page 114
- How to Configure SGCP RSIP and AUEP Enhancements, page 115
- Configuration Examples for SGCP RSIP and AUEP Enhancements, page 116

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest feature information and caveats, see the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the Feature Information Table at the end of this document.
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To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on Cisco.com is not required.

Prerequisites for SGCP RSIP and AUEP Enhancements

- Configure SGCP 1.5 on the gateway.

Restrictions for SGCP RSIP and AUEP Enhancements

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

Information About 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, page 115.

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

- Configuring SGCP RSIP and AUEP Enhancements, page 115
- Verifying SGCP RSIP Configuration, page 115

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:

```
Router(config)# mgcp sgcp disconnect notify
```

<table>
<thead>
<tr>
<th>Command</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>Router(config)# mgcp sgcp disconnect notify</code></td>
<td>Enables enhanced endpoint synchronization with a call agent after a disconnected procedure. The command is disabled by default.</td>
</tr>
</tbody>
</table>

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 voa1l2 modem passthrough mode:NSE, codec:g711ulaw, redundancy:DISABLED,
MGCP voa1l2 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 rtrcab DISABLED
MGCP system resource check DISABLED
MGCP xpc-codc: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
```
Configuration Examples for SGCP RSIP and AUEP Enhancements

- Disconnected RSIP Messaging Example, page 116

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 10.0.0.0 255.255.255.0 172.16.193.1
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
termination mgcpapp
port 1/1
!
dial-peer voice 2 pots
termination mgcpapp
port 1/2
!
dial-peer voice 3 pots
termination 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

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actual addresses and phone numbers. Any examples, command display output, network topology diagrams,
and other figures included in the document are shown for illustrative purposes only. Any use of actual IP
addresses or phone numbers in illustrative content is unintentional and coincidental.
Configuring MGCP Gateway Support

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.

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

- "Overview of MGCP and Related Protocols" on page 3

### Feature History for MGCP Gateway Support for the `mgcp bind` Command

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(13)T</td>
<td>This feature was introduced.</td>
</tr>
</tbody>
</table>

- Finding Feature Information, page 120
- Prerequisites for Configuring MGCP Gateway Support, page 120
- Information About MGCP Gateway Support, page 120
Finding Feature Information

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Prerequisites for Configuring MGCP Gateway Support

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.

Information About MGCP Gateway Support

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.
The figure below shows a typical configuration flow using the `mgcp bind` command.

Figure 9  Bind Configuration Flowchart

Post SYS event to MGCP application to open new UDP socket.
The figure below 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**  \hspace{1cm} **Bind Configuration for Control Flowchart**

The time frame for execution of the `mgcp bind` command for media is different from that for control. The figure below 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**  \hspace{1cm} **Bind Configuration for Media Flowchart**

From XGCP call creation
# How to Configure MGCP Gateway Support

- Configuring the MGCP Application, page 123
- Configuring the bind Command, page 124
- Verifying MGCP Gateway Support, page 127

## Configuring the MGCP Application

**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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 enable</td>
<td>Enables higher privilege levels, such as privileged EXEC mode. Enter your password if prompted.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router&gt; enable</td>
</tr>
<tr>
<td>Step 2 configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router# configure terminal</td>
</tr>
<tr>
<td>Step 3 mgcp call-agent {dns-name</td>
<td>ip-address} [port] [service-type type] [version protocol-version]</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config)# mgcp call-agent 209.165.200.225 service-type mgcp version 1.0</td>
</tr>
<tr>
<td>Step 4 mgcp</td>
<td>Enables MGCP on the gateway.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config)# mgcp</td>
</tr>
</tbody>
</table>
### Configuring the bind Command

#### SUMMARY STEPS

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

#### DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> <code>enable</code></td>
<td>Enables higher privilege levels, such as privileged EXEC mode. Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td><code>Router&gt; enable</code></td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> <code>configure terminal</code></td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td><code>Router# configure terminal</code></td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> `mgcp bind {control</td>
<td>media} source-interface interface-id`</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td><code>Router(config)# mgcp bind {control} source-interface FastEthernet</code></td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> <code>mgcp</code></td>
<td>Enables MGCP on the gateway.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td><code>Router(config)# mgcp</code></td>
<td></td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
</tr>
<tr>
<td>Step 5 exit</td>
<td>Exits the current mode.</td>
</tr>
</tbody>
</table>

**Example:**

Router(config)# exit

- **Troubleshooting Tips**, page 125

**Troubleshooting Tips**

To troubleshoot the MGCP Gateway Support for the Bind Command feature, use the `debug mgcp` command to enable debug traces for MGCP errors, events, media, packets, parser, and call admission control (CAC).

The following example illustrates the output for the `debug mgcp` 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 -
  CRX 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_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_mandatoryParms()
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
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: No SDP connection info
20:54:13: selected codec=5, bw=16000, codec=5
20:54:13: codec index=0, bw=16000, codec=5
20:54:13: selected codec=5
20:54:13: mgcp_process_quarantine_mode- callp 61C28130, voice_if 61C281A4
20:54:13: mgcp_select_codec - LC option, num codec=1, 1st codec=5
20:54:13: selected codec=5, bw=16000, codec=5
20:54:13: selected codec=5
20:54:13: IP Precedence=60
20:54:13: MGCP msg qos value=0
20:54:13: mgcp_get_pkt_period: voip codec=2, pkt_period=0, call adjust_packetization_period
20:54:13: 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=0
20:54:13: mgcp_get_pkt_period: voip codec=2, pkt_period=0, call adjust_packetization_period
20:54:13: 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=0
20:54:13: mgcp_get_pkt_period: voip codec=2, pkt_period=0, call adjust_packetization_period
20:54:13: 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=0
20:54:13: mgcp_get_pkt_period: voip codec=2, pkt_period=0, call adjust_packetization_period
20:54:13: 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=0
20:54:13: mgcp_get_pkt_period: voip codec=2, pkt_period=0, call adjust_packetization_period
20:54:13: mgcp_get_pkt_period: voip codec=2, pkt_period=10, after calling adjust_packetization_period
20:54:13: selected codec=5

Verifying MGCP Gateway Support

SUMMARY STEPS
1. Router# show mgcp
2. Router# show ip socket
3. Router# show running-configuration

DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 Router# show mgcp</td>
<td>Checks your configuration.</td>
</tr>
<tr>
<td>Step 2 Router# show ip socket</td>
<td>Displays IP socket information.</td>
</tr>
<tr>
<td>Step 3 Router# show running-config</td>
<td>Verifies bind functionality.</td>
</tr>
</tbody>
</table>

Configuration Examples for MGCP Gateway Support

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
!
```

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Configuring MGCP CAS MD Package

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.

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

- "Overview of MGCP and Related Protocols" on page 3

Feature History for MGCP CAS MD Package

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)T</td>
<td>This feature was introduced on the Cisco AS5850.</td>
</tr>
<tr>
<td>12.4(15)T</td>
<td>Support was added for the Cisco AS5350, Cisco AS5350XM, Cisco AS5400XM, and Cisco AS5400HPX platforms.</td>
</tr>
</tbody>
</table>

- Finding Feature Information, page 129
- Prerequisites for MGCP CAS MD Package, page 130
- Restrictions for MGCP CAS MD Package, page 130
- Information About MGCP CAS MD Package, page 130
- How to Configure the MGCP CAS MD Package, page 130
- Configuration Examples for MGCP CAS MD Package, page 133

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest feature information and caveats, see the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the Feature Information Table at the end of this document.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on Cisco.com is not required.
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

FGD Exchange Access International (EAIN) signaling is not supported.

Information About MGCP CAS MD Package

• MD Package, page 130

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

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, page 130
• Modifying ANI and DNIS Order when Using CAS MD Package, page 132

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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 1** enable | Enables privileged EXEC mode.  
- Enter your password if prompted. |
| **Example:** | Router> enable |
| **Step 2** configure terminal | Enters global configuration mode. |
| **Example:** | Router# configure terminal |
| **Step 3** dial-peer voice tag pots | Defines a dial peer as a POTS device and enters dial-peer configuration mode. |
| **Example:** | Router(config)# dial-peer voice 1003 pots |
| **Step 4** service mgcpapp | Enables MGCP on the dial peer.  
**Note** Do not use this command in dial peers that support PRI backhaul or BRI backhaul. |
| **Example:** | Router(config-dial-peer)# service mgcpapp |
| **Step 5** incoming called number string | Specifies the digit string that is used to match incoming calls to the dial peer. |
| **Example:** | Router(config-dial-peer)# incoming called number . |
### Command or Action | Purpose
--- | ---
**Step 6** | **Purpose**
port | Binds the MGCP application to the specified voice port.

**Example:**

```
Router(config-dial-peer)# port 0/0:3:0
```

**Step 7** | **Purpose**
end | Exits to privileged EXEC mode.

**Example:**

```
Router(config-dial-peer)# end
```

---

## 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** | **Purpose**
enable | Enables privileged EXEC mode.

- Enter your password if prompted.

**Example:**

```
Router> enable
```

**Step 2** | **Purpose**
configure terminal | Enters global configuration mode.

**Example:**

```
Router# configure terminal
```
## Configuration Examples for MGCP CAS MD Package

- CAS MD Package Configuration Example, page 133
- Cisco AS5850 Configuration Example, page 134

### CAS MD Package Configuration Example

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

```plaintext
... 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 ani-dnis
!
dial-peer voice 1003 pots
   service mgcpapp
```

### Command or Action | Purpose
--- | ---
**Step 3** `mgcp profile {profile-name | default}` | Defines an MGCP profile to be associated with one or more MGCP endpoints.

**Example:**

```
Router(config)# mgcp profile default
```

**Step 4** `notify {ani-dnis | 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.

**Example:**

```
Router(config-mgcp-profile)# notify dnis-ani
```

**Step 5** `end` | Exits to privileged EXEC mode.

**Example:**

```
Router(config-mgcp-profile)# end
```

**Step 6** `show mgcp profile [profile-name]` | Displays configuration information for MGCP profiles including the setting of the `notify` command.

**Example:**

```
Router# show mgcp profile default
```
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.

```plaintext
incoming called-number .
    port 0/0:3:0
!
dial-peer voice 1004 pots
    service mgcpapp
    incoming called-number .
    port 0/0:4:0
...
```

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

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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.

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

- "Overview of MGCP and Related Protocols" on page 3

Feature History for MGCP CAS PBX and AAL2 PVC

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(5)XM</td>
<td>This feature was introduced on the following platforms: Cisco 1750, Cisco 2600 series, Cisco 3600 series, Cisco AS5300, Cisco MC3810, and Cisco uBR924.</td>
</tr>
<tr>
<td>12.2(2)T</td>
<td>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).</td>
</tr>
<tr>
<td>12.2(11)T</td>
<td>This feature was implemented on the Cisco AS5300 and Cisco AS5850.</td>
</tr>
</tbody>
</table>

Note: AAL2 PVC is not supported on the Cisco AS5850.

- Finding Feature Information, page 138
- Prerequisites for MGCP CAS PBX and AAL2 PVC, page 138
Finding Feature Information

Your software release may not support all the features documented in this module. For the latest feature information and caveats, see the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the Feature Information Table at the end of this document.

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Prerequisites for MGCP CAS PBX and AAL2 PVC

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

Restrictions for 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.

- 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 MGCP CAS PBX and AAL2 PVC

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. The figure below illustrates a possible residential cable access solution:

![Residential Cable Access Solution](image1)

Note that in the figure above, 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. The figure below illustrates an IP Centrex solution:

![IP Centrex Solution](image2)

In the figure above, 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. The figure below illustrates an integrated access solution:

Figure 14  Integrated Access Solution

In the figure above, 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

The figure below illustrates a telecommuter/small office-home office solution:

Figure 15  Telecommuter or Small Office-Home Office Solution

In the figure above, MGCP must control the calls over AAL2 PVCs, and an analog FXS interface is required.
How to Configure MGCP CAS PBX and AAL2 PVC

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, page 143
- Configuring Subcell Multiplexing for AAL2 Voice, page 149
- Configuring the Cable Access Router for SGCP and MGCP, page 149
- Verifying the MGCP CAS PBX and AAL2 PVC Configurations, page 150
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}`
5. `mgcp sgcp restart notify`
6. `mgcp modem passthrough [voip | voaal2] mode [cisco | nse]`
7. `mgcp tse payload type`
8. `mgcp rtp unreachable timeout timer-value`
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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> mgcp</td>
<td>Starts the MGCP daemon.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td><code>Router(config)# mgcp</code></td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> mgcp call-agent {ipaddr</td>
<td>hostname} [port] [service-type type] version version-number</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>`Router(config)# mgcp call-agent {ipaddr</td>
<td>hostname} [port] [service-type type] version version-number`</td>
</tr>
<tr>
<td><strong>Step 3</strong> mgcp dtmf-relay voip codec {all</td>
<td>low-bit-rate} mode {cisco</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>`Router(config)# mgcp dtmf-relay voip codec {all</td>
<td>low-bit-rate} mode {cisco</td>
</tr>
<tr>
<td><strong>Step 4</strong> mgcp package-capability {as-package</td>
<td>atm-package</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>`Router(config)# mgcp package-capability {as-package</td>
<td>atm-package</td>
</tr>
<tr>
<td><strong>Step 5</strong> mgcp sgcp restart notify</td>
<td>(Required only for SGCP mode with a call agent supporting RSIP. See Configuration Examples 4 through 9.) Causes MGCP to send SGCP RSIP messages.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td><code>Router(config-if)# mgcp sgcp restart notify</code></td>
<td></td>
</tr>
</tbody>
</table>
## Configuring MGCP CAS PBX and AAL2 PVC

### How to Configure MGCP CAS PBX and AAL2 PVC

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 6</strong> mgcp modem passthrough [voip</td>
<td>voaal2] mode [cisco</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-if)# mgcp modem passthrough [voip</td>
<td>voaal2] mode [cisco</td>
</tr>
<tr>
<td><strong>Step 7</strong> mgcp tse payload type</td>
<td>(Required for nse mode. See Step 6.) Enables the TSE payload for fax and modem messages.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# mgcp tse payload type</td>
<td></td>
</tr>
<tr>
<td><strong>Step 8</strong> mgcp rtp unreachable timeout timer-value</td>
<td>(Optional) Enables detection of unreachable remote VoIP endpoints.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# mgcp rtp unreachable timeout timer-value</td>
<td></td>
</tr>
<tr>
<td><strong>Step 9</strong> no mgcp timer receive-rtcp</td>
<td>(Required for non-RGWs. See Configuration Examples 2 through 9.) Turns off the RTP RTCP receive timeout interval at the gateway.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# no mgcp timer receive-rtcp</td>
<td></td>
</tr>
<tr>
<td><strong>Step 10</strong> mgcp timer net-cont-test timer</td>
<td>(Optional for non-RGWs. See Configuration Examples 2 through 9.) Turns on the continuity test timeout interval at the gateway.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# mgcp timer net-cont-test timer</td>
<td></td>
</tr>
<tr>
<td><strong>Step 11</strong> controller T1 0</td>
<td>(Required for ATM mode. See Configuration Examples 2 through 9.) Select s the T1 controller 0.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# controller T1 0</td>
<td></td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
</tr>
</tbody>
</table>
| **Step 12** 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.  

Example:  
Router(config-controller)# mode atm |
| **Step 13** no shutdown | (Optional for ATM mode. See Configuration Examples 2 through 9.) Ensures that the controller is activated. |
| Example:  
Router(config-controller)# no shutdown |
| **Step 14** exit | (Required for ATM mode. See Configuration Examples 2 through 9.) Exits the current mode. |
| Example:  
Router(config-controller)# exit |
| **Step 15** mgcp quarantine mode process | (Optional) Turns on processing for SGCP quarantine mode. |
| Example:  
Router(config)# mgcp quarantine mode process |
| **Step 16** controller T1 1 | (Required for CAS PBX. See Configuration Examples 3, 4, and 5.) Select the T1 controller 1. |
| Example:  
Router(config)# controller T1 1 |
| **Step 17** mode cas | (Required for CAS PBX. See Configuration Examples 3, 4, and 5.) Specify that the controller will support CAS. |
| Example:  
Router(config-controller)# mode cas |
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 18</strong> ds0-group channel-number timeslots range type signaling-type tone type addr info service service-type</td>
<td>(Required for CAS PBX. See Configuration Examples 3, 4, and 5.) Configure the T1 timeslots for CAS calls.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-controller)# ds0-group channel-number timeslots range type signaling-type tone type addr info service service-type</td>
<td></td>
</tr>
<tr>
<td><strong>Step 19</strong> exit</td>
<td>(Required for CAS PBX. See Configuration Examples 3, 4, and 5.) Exit controller configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-controller)# exit</td>
<td></td>
</tr>
<tr>
<td><strong>Step 20</strong> interface atm0 [subinterface-number [multipoint</td>
<td>point-to-point]]</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# interface atm0 [subinterface-number [multipoint</td>
<td>point-to-point]]</td>
</tr>
<tr>
<td><strong>Step 21</strong> pvc [name] vpi/vci</td>
<td>(Required for ATM mode. See Configuration Examples 2 through 9.) Create an ATM PVC for voice traffic and enter ATM virtual circuit configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td><strong>Note</strong> The <strong>ilmi</strong> and <strong>qsaal</strong> options are not supported for AAL2.</td>
</tr>
<tr>
<td>Router(config-if)# pvc [name] vpi/vci</td>
<td></td>
</tr>
<tr>
<td><strong>Step 22</strong> encapsulation aal-encap</td>
<td>(Required for ATM mode. See Configuration Examples 2 through 9.) Set the encapsulation of the PVC for voice traffic. <strong>aal2</strong> automatically creates channel identifiers (CIDs) 1 through 255. Some of the Scenarios use <strong>aal5snap for ATM0.1 and ATM0.3. Use aal2 for ATM0.2.</strong></td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-if-atm-vc)# encapsulation aal-encap</td>
<td></td>
</tr>
</tbody>
</table>
### Command or Action

<table>
<thead>
<tr>
<th>Step 23</th>
<th>vbr-rt peak-rate average-rate [burst]</th>
</tr>
</thead>
<tbody>
<tr>
<td>Example:</td>
<td>Router(config-if-atm-vc)# vbr-rt peak-rate average-rate [burst]</td>
</tr>
</tbody>
</table>

**Purpose**: (Required for ATM mode. See Configuration Examples 2 through 9.) Configures the PVC for the variable-bit-rate real-time (voice) traffic.

<table>
<thead>
<tr>
<th>Step 24</th>
<th>vcci pvc-identifier</th>
</tr>
</thead>
<tbody>
<tr>
<td>Example:</td>
<td>Router(config-if-atm-vc)# vcci pvc-identifier</td>
</tr>
</tbody>
</table>

**Purpose**: (Optional for ATM mode. See Configuration Examples 2 through 9.) Assigns a unique identifier to the PVC.

<table>
<thead>
<tr>
<th>Step 25</th>
<th>exit</th>
</tr>
</thead>
<tbody>
<tr>
<td>Example:</td>
<td>Router(config-if-atm-vc)# exit</td>
</tr>
</tbody>
</table>

**Purpose**: (Required for ATM mode. See Configuration Examples 2 through 9.) Exits the current mode.

<table>
<thead>
<tr>
<th>Step 26</th>
<th>exit</th>
</tr>
</thead>
<tbody>
<tr>
<td>Example:</td>
<td>Router(config-if)# exit</td>
</tr>
</tbody>
</table>

**Purpose**: (Required for ATM mode. See Configuration Examples 2 through 9.) Exits the current mode.

<table>
<thead>
<tr>
<th>Step 27</th>
<th>dial-peer voice number pots</th>
</tr>
</thead>
<tbody>
<tr>
<td>Example:</td>
<td>Router(config)# dial-peer voice number pots</td>
</tr>
</tbody>
</table>

**Purpose**: Enter dial peer configuration mode for the POTS dial peer.

<table>
<thead>
<tr>
<th>Step 28</th>
<th>application MGCPAPP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Example:</td>
<td>Router(config-dial-peer)# application MGCPAPP</td>
</tr>
</tbody>
</table>

**Purpose**: Initiates the MGCP protocol for the voice ports.

<table>
<thead>
<tr>
<th>Step 29</th>
<th>exit</th>
</tr>
</thead>
<tbody>
<tr>
<td>Example:</td>
<td>Router(config-dial-peer)# exit</td>
</tr>
</tbody>
</table>

**Purpose**: 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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> voice service voatm</td>
<td>(Required) Enters voice-service configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# voice service voatm</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> session protocol aal2</td>
<td>(Required) Enters voice-service-session configuration mode and specifies AAL2 trunking.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-voice-service)# session protocol aal2</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> subcell-mux</td>
<td>(Required) Enables subcell multiplexing. By default, subcell multiplexing is not enabled.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-voice-service-session)# subcell-mux</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> end</td>
<td>(Required) Exits the current mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-voice-service-session)# end</td>
<td></td>
</tr>
</tbody>
</table>

Configuring the Cable Access Router for SGCP and MGCP

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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> show dial-peer voice sum</td>
<td>Displays the status of the dial peer. The dial peer should be active. If it is not, use the <strong>no shut</strong> command to make it so.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router# show dial-peer voice sum</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> show running-configuration</td>
<td>Displays the current configuration settings.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router# show running-configuration</td>
<td></td>
</tr>
</tbody>
</table>

Configuration Examples for MGCP CAS PBX and AAL2 PVC

- Example 1 MGCP Residential Gateway, page 151
- Example 2 MGCP Gateway using Voice over ATM AAL2, page 151
- Example 3 MGCP and SGCP EM Wink-Start, page 153
- Example 4 SGCP 1.5 CAS PBX using Voice over ATM AAL2, page 154
- Example 5 SGCP 1.5 CAS PBX using Voice over IP over ATM AAL5, page 158
- Example 6 SGCP 1.5 Analog EM PBX using Voice over ATM AAL2, page 163
- Example 7 SGCP 1.5 Analog EM PBX using Voice over IP over ATM AAL5, page 166
- Example 8 SGCP 1.5 RGW using Voice over ATM AAL2, page 170
- Example 9 SGCP 1.5 RGW using Voice over IP over ATM AAL5, page 174
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
!ngcp
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 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
Example 3 MGCP and SGCP EM 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:

```plaintext
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
```

Example 3 MGCP and SGCP EM 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:
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**

```bash
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
iadm 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-group 3 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 mrroute-cache
  shutdown
!
interface Serial1
  no ip address
  no ip route-cache
  no ip mrroute-cache
  shutdown
  no cdp enable
!
interface ATM0
  no ip address
  ip mrroute-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
Example 6 SGCP 1.5 Analog EM 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**

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

cns event-service server
voice-card 0

timeout 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
shutdown

interface Serial1
no ip address
no ip route-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
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
vccl 2

interface ATM0.3 point-to-point
description data vc
ip address 10.0.0.5 255.0.0.0
vbr-rt 1536 170 8
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 mroutecache
  shutdown
!
interface Serial1
  no ip address
  no ip route-cache
  no ip mroutecache
  shutdown
  no cdp enable
!
interface ATM0
  no ip address
  ip mroutecache
  no atm ilmi-keepalive
Example 7 SGCP 1.5 Analog EM PBX using Voice over IP over ATM AAL5

The following figure 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**

![Diagram of 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 10.0.0.2 255.0.0.0.
```
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
!
mgc
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
iadn 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 mrouting-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 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

![Diagram of SGCP 1.5 RGW using Voice over ATM AAL2 Configuration]

**Router A Configuration**

```plaintext
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
iadn 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
  vccl 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
Configuring MGCP CAS PBX and AAL2 PVC

MGCP and Related Protocols Configuration Guide, Cisco IOS Release 12.4T

Configuring MGCP CAS PBX and AAL2 PVC

Configuration Examples for MGCP CAS PBX and AAL2 PVC

![hostname B]
! network-clock base-rate 56K
ip subnet-zero
!
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
vccl 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 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 1imi-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 10.0.0.2
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 for related documents, standards, and MIBs, and the "Glossary" for definitions of terms in this guide.

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Any Internet Protocol (IP) addresses and phone numbers used in this document are not intended to be actual addresses and phone numbers. Any examples, command display output, network topology diagrams, and other figures included in the document are shown for illustrative purposes only. Any use of actual IP addresses or phone numbers in illustrative content is unintentional and coincidental.
Glossary


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.

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.

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.

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&M--Ear and Mouth analog signaling.

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.

GW--Gateway.

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.

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.

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*.

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.

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.

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.

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.

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.

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