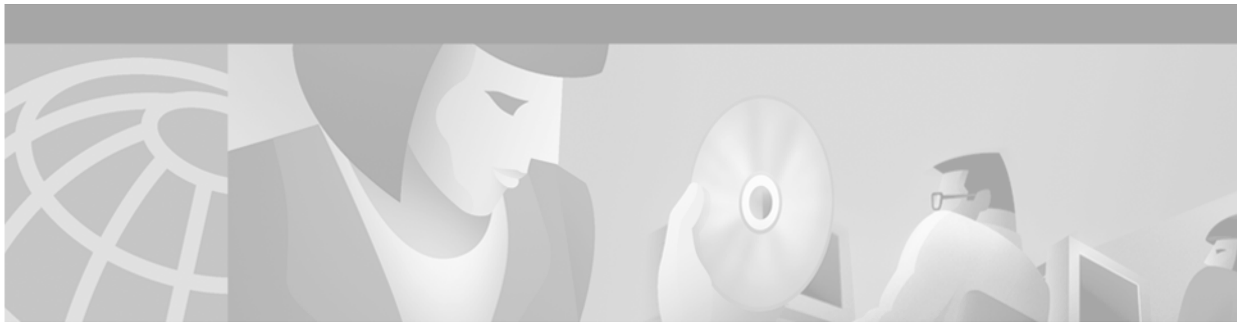


Cisco IOS Voice, Video, and Fax Commands: G Through P

This chapter presents the commands to configure and maintain Cisco IOS voice, video, and fax applications. The commands are presented in alphabetical order beginning with G. Some commands required for configuring voice, video, and fax may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

For detailed information on how to configure these applications and features, refer to the *Cisco IOS Voice, Video, and Fax Configuration Guide*.

gatekeeper

To enter gatekeeper configuration mode, use the **gatekeeper** command in global configuration mode.

gatekeeper

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced on the Cisco 2500 and 3600 series routers.
12.0(3)T	This command was integrated into the Cisco IOS Release 12.0(3)T and, support for the Cisco MC3810 multiservice concentrator was added.

Usage Guidelines

Press Ctrl-Z or use the **exit** command to exit gatekeeper configuration mode.

Examples

The following example brings the gatekeeper online:

```
gatekeeper
 no shutdown
```

gateway

To enable the H.323 Voice over IP (VoIP) gateway, use the **gateway** command in global configuration mode. To disable the gateway, use the **no** form of this command.

gateway

no gateway

Syntax Description

This command has no arguments or keywords.

Defaults

The gateway is unregistered.

Command Modes

Global configuration

Command History

Release	Modification
11.3(6)NA2	This command was introduced on the Cisco 3600 series routers and Cisco AS5300 and Cisco AS5800 universal access servers.

Usage Guidelines

Use the **gateway** command to enable H.323 VoIP gateway functionality. After you enable the gateway, it will attempt to discover a gatekeeper by using the H.323 RAS GRQ message. If you enter **no gateway voip**, the VoIP gateway will unregister with the gatekeeper via the H.323 RAS URQ message.

Examples

The following example enables the gateway:

```
gateway
```

group

To create a session-group and associate it with a specified session-set, use the **group** command in backhaul session manager configuration mode. To delete the group, use the **no** form of this command.

group *group-name* **set** *set-name*

no group *group-name* **set** *set-name*

Syntax Description

group-name	Session-group name.
set *set-name*	Session-set name.

Defaults

No default behavior or values.

Command Modes

Backhaul session manager configuration

Command History

Release	Modification
12.1(1)T	This command was introduced.

Examples

To associate the group named Group5 with the set named Set1, see the following example:

```
Router(config-bsm)# group group5 set set1
```

Related Commands

Command	Description
group auto-reset	Configures the maximum auto-reset value.
group cumulative-ack	Configures maximum cumulative acknowledgments.
group out-of-sequence	Configures maximum out-of-sequence segments that are received before an EACK is sent.
group receive	Configures maximum receive segments.
group retransmit	Configures maximum retransmits.
group timer cumulative-ack	Configures cumulative acknowledgment timeout.
group timer keepalive	Configures keepalive (or null segment) timeout.
group timer retransmit	Configures retransmission timeout.
group timer transfer	Configures state transfer timeout.

group auto-reset

To configure the maximum auto-reset value, use the **group auto-reset** command in backhaul session manager configuration mode. To set the value to the default value, use the **no** form of this command.

group *group-name* **auto-reset** *count*

no group *group-name* **auto-reset** *count*

Caution Do not change this parameter unless instructed to do so by Cisco technical support. There are relationships between group parameters that can cause sessions to fail if not set correctly.

Syntax Description

group-name	Session-group name.
auto-reset *count*	Maximum number of auto-resets. Range is 0 through 255.

Defaults

5

Command Modes

Backhaul session manager configuration

Command History

Release	Modification
12.1(1)T	This command was introduced.

Examples

To configure the maximum auto-reset value for the group named Group5 to 6, see the following example:

```
Router(config-bsm)# group group5 auto-reset 6
```

Related Commands

Command	Description
group cumulative-ack	Configures maximum cumulative acknowledgments.
group out-of-sequence	Configures maximum out-of-sequence segments that are received before an EACK is sent.
group receive	Configures maximum receive segments.
group retransmit	Configures maximum retransmits.

group cumulative-ack

To configure maximum cumulative acknowledgments, use the **group cumulative-ack** command in backhaul session manager configuration mode. Maximum cumulative acknowledgments are the maximum number of segments that are received before an acknowledgment is sent. To set the value to the default value, use the **no** form of this command.

group *group-name* **cumulative ack** *count*

no group *group-name* **cumulative ack** *count*

Caution Do not change this parameter unless instructed to do so by Cisco technical support. There are relationships between group parameters that can cause sessions to fail if not set correctly.

Syntax Description

group-name	Session-group name.
cumulative ack *count*	Maximum number of segments received before acknowledgment. Range is 0 through 255.

Defaults

3

Command Modes

Backhaul session manager configuration

Command History

Release	Modification
12.1(1)T	This command was introduced.

Examples

To set the cumulative acknowledgment maximum for Group5 to 4, see the following example:

```
Router(config-bsm)# group group5 cumulative-ack 4
```

Related Commands

Command	Description
group auto-reset	Configures the maximum auto-reset value.
group out-of-sequence	Configures maximum out-of-sequence segments that are received before an EACK is sent.
group receive	Configures maximum receive segments.
group retransmit	Configures maximum retransmits.

group out-of-sequence

To configure maximum out-of-sequence segments that are received before an EACK is sent, use the **group out-of-sequence** command in backhaul session manager configuration mode. To set the value to the default value, use the **no** form of this command.

group *group-name* **out-of-sequence** *count*

no group *group-name* **out-of-sequence** *count*

Caution Do not change this parameter unless instructed to do so by Cisco technical support. There are relationships between group parameters that can cause sessions to fail if not set correctly.

Syntax Description

group-name	Session-group name.
out-of-sequence *count*	Maximum number of out-of-sequence segments. Range is 0 through 255.

Defaults

3

Command Modes

Backhaul session manager configuration

Command History

Release	Modification
12.1(1)T	This command was introduced.

Examples

To set the out-of-sequence maximum for Group5 to 4, see the following example:

```
Router(config-bsm)# group group5 out-of-sequence 4
```

Related Commands

Command	Description
group auto-reset	Configures the maximum auto-reset value.
group cumulative-ack	Configures maximum cumulative acknowledgments.
group receive	Configures maximum receive segments.
group retransmit	Configures maximum retransmits.

group receive

To configure maximum receive segments, use the **group receive** command in backhaul session manager configuration mode. To set the value to the default value, use the **no** form of this command.

group *group-name* **receive** *count*

no group *group-name* **receive** *count*

Caution Do not change this parameter unless instructed to do so by Cisco technical support. There are relationships between group parameters that can cause sessions to fail if not set correctly.

Syntax Description

group-name	Session-group name.
receive *count*	Maximum number of segments in our receive window. The other side should send no more than this number of segments before receiving an acknowledgment for the oldest outstanding segment. Range is 1 through 64.

Defaults

32

Command Modes

Backhaul session manager configuration

Command History

Release	Modification
12.1(1)T	This command was introduced.

Examples

To set the receive maximum to 10 for Group5, see the following example:

```
Router(config-bsm)# group group5 receive 10
```

Related Commands

Command	Description
group auto-reset	Configures the maximum auto-reset value.
group cumulative-ack	Configures maximum cumulative acknowledgments.
group out-of-sequence	Configures maximum out-of-sequence segments that are received before an EACK is sent.
group retransmit	Configures maximum retransmits.

group retransmit

To configure maximum retransmits, use the **group retransmit** command in backhaul session manager configuration mode. To set the value to the default value, use the **no** form of this command.

group *group-name* **retransmit** *count*

no group *group-name* **retransmit** *count*

Caution Do not change this parameter unless instructed to do so by Cisco technical support. There are relationships between group parameters that can cause sessions to fail if not set correctly.

Syntax Description

group-name	Session-group name.
retransmit *count*	Maximum number of retransmits. Range is 0 through 255.

Defaults

2

Command Modes

Backhaul session manager configuration

Command History

Release	Modification
12.1(1)T	This command was introduced.

Examples

To set the retransmit maximum for Group5 to 3, see the following example:

```
Router(config-bsm)# group group5 retrans 3
```

Related Commands

Command	Description
group auto-reset	Configures the maximum auto-reset value.
group cumulative-ack	Configures maximum cumulative acknowledgments.
group out-of-sequence	Configures maximum out-of-sequence segments that are received before an EACK is sent.
group receive	Configures maximum receive segments.

group timer cumulative-ack

To configure cumulative acknowledgment timeout, use the **group timer cumulative ack** command in backhaul session manager configuration mode. Cumulative acknowledgment timeout is the maximum number of milliseconds RUDP will delay before sending an acknowledgment for a received segment. To set the value to the default value, use the **no** form of this command.

group *group-name* **timer cumulative ack** *time*

no group *group-name* **timer cumulative ack** *time*

Caution Do not change this parameter unless instructed to do so by Cisco technical support. There are relationships between group parameters that can cause sessions to fail if not set correctly.

Syntax Description

group-name	Session-group name.
timer cumulative ack *time*	Number of milliseconds RUDP will delay. Range is 100 through 65535.

Defaults

100

Command Modes

Backhaul session manager configuration

Command History

Release	Modification
12.1(1)T	This command was introduced.

Examples

To set the cumulative acknowledgment timer for Group5 to 325, see the following example:

```
Router(config-bsm)# group group5 timer cumulative-ack 325
```

Related Commands

Command	Description
group timer keepalive	Configures keepalive (or null segment) timeout.
group timer retransmit	Configures retransmission timeout.
group timer transfer	Configures state transfer timeout.

group timer keepalive

To configure keepalive (or null segment) timeout, use the **group timer keepalive** command in backhaul session manager configuration mode. Keepalive timeout is the number of milliseconds RUDP will wait before sending a keepalive segment. To set the value to the default value, use the **no** form of this command.

group *group-name* **timer keepalive** *time*

no group *group-name* **timer keepalive** *time*

Caution Do not change this parameter unless instructed to do so by Cisco technical support. There are relationships between group parameters that can cause sessions to fail if not set correctly.

Syntax Description

group-name	Session-group name.
timer keepalive *time*	Number of milliseconds before RUDP sends a keepalive segment. Range is 100 through 65535.

Defaults

1000

Command Modes

Backhaul session manager configuration

Command History

Release	Modification
12.1(1)T	This command was introduced.

Examples

To configure the keepalive timer for Group5 to 2050 milliseconds, see the following example:

```
Router(config-bsm)# group group5 timer keepalive 2050
```

Related Commands

Command	Description
group timer cumulative-ack	Configures cumulative acknowledgment timeout.
group timer retransmit	Configures retransmission timeout.
group timer transfer	Configures state transfer timeout.

group timer retransmit

To configure retransmission timeout, use the **group timer retransmit** command in backhaul session manager configuration mode. Retransmission timeout is the number of milliseconds RUDP will wait to receive an acknowledgment for a segment. To set the value to the default value, use the **no** form of this command.

group *group-name* **timer retransmit** *time*

no group *group-name* **timer retransmit** *time*

Caution Do not change this parameter unless instructed to do so by Cisco technical support. There are relationships between group parameters that can cause sessions to fail if not set correctly.

Syntax Description

group-name	Session-group name.
timer retransmit *time*	Number of milliseconds RUDP will delay. Range is 100 through 65535.

Defaults

300

Command Modes

Backhaul session manager configuration

Command History

Release	Modification
12.1(1)T	This command was introduced.

Usage Guidelines

The retransmit timer must be greater than the cumulative-ack timer.

Examples

To set the retransmit timer for Group5 to 650, see the following example:

```
Router(config-bsm)# group group5 timer retransmit 650
```

Related Commands

Command	Description
group timer cumulative-ack	Configures cumulative acknowledgment timeout.
group timer keepalive	Configures keepalive (or null segment) timeout.
group timer transfer	Configures state transfer timeout.

group timer transfer

To configure state transfer timeout, use the **group timer transfer** command in backhaul session manager configuration mode. To set the value to the default value, use the **no** form of this command.

group *group-name* **timer transfer** *time*

no group *group-name* **timer transfer** *time*

Caution Do not change this parameter unless instructed to do so by Cisco technical support. There are relationships between group parameters that can cause sessions to fail if not set correctly.

Syntax Description

group-name	Session-group name.
timer transfer *time*	Maximum number of milliseconds RUDP will wait for a transfer request. The range is 0 to 65535 milliseconds.

Defaults

2000

Command Modes

Backhaul session manager configuration

Command History

Release	Modification
12.1(1)T	This command was introduced.

Examples

To set the state transfer timer for Group5 to 1800, see the following example:

```
Router(config-bsm)# group group5 timer transfer-state 1800
```

Related Commands

Command	Description
group timer cumulative-ack	Configures cumulative acknowledgment timeout.
group timer keepalive	Configures keepalive (or null segment) timeout.
group timer retransmit	Configures retransmission timeout.

gw-accounting

To enable Voice over IP (VoIP) gateway-specific accounting and define the accounting method, use the **gw-accounting** command in global configuration mode. To disable gateway-specific accounting, use the **no** form of this command.

gw-accounting {**h323** [**vsa**] | **syslog** | **voip**}

no gw-accounting {**h323** [**vsa**] | **syslog** | **voip**}

Syntax Description

h323	Enables standard H.323 accounting using Internet Engineering Task Force (IETF) RADIUS attributes.
vsa	(Optional) Enables H.323 accounting using RADIUS vendor specific attributes (VSAs).
syslog	Enables the system logging facility to output accounting information in the form of a system log message.
voip	Enables generic gateway-specific accounting.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
11.3(6)NA2	This command was introduced for the Cisco 2500 and 3600 series routers and the AS5300 universal access server.
12.0(7)T	The **vsa** keyword was added.
12.1(1)T	The **voip** keyword was added.

Usage Guidelines

To collect basic start-stop connection accounting data, the gateway must be configured to support gateway-specific H.323 accounting functionality. The **gw-accounting** command enables you to send accounting data to the RADIUS server in one of four ways:

- Using standard IETF RADIUS accounting attribute/value (AV) pairs—This method is the basic method of gathering accounting data (connection accounting) according to the specifications defined by the IETF. Use the **gw-accounting h323** command to configure the standard IETF RADIUS method of applying H.323 gateway-specific accounting. Table 20 shows the IETF RADIUS attributes supported.

Table 20 Supported IETF RADIUS Accounting Attributes

Number	Attribute	Description
30	Called-Station-Id	Allows the network access server to send the telephone number that the user called as part of the Access-Request packet (using Dialed Number Identification [DNIS] or similar technology). This attribute is supported only on ISDN, and modem calls on the Cisco AS5200 and Cisco AS5300 universal access server if used with ISDN PRI.
31	Calling-Station-Id	Allows the network access server to send the telephone number that the call came from as part of the Access-Request packet (using Automatic Number Identification or similar technology). This attribute has the same value as "remote-addr" from TACACS+. This attribute is supported only on ISDN, and modem calls on the Cisco AS5200 and Cisco AS5300 universal access server if used with ISDN PRI.
42	Acct-Input-Octets	Indicates how many octets have been received from the port over the course of the accounting service being provided.
43	Acct-Output-Octets	Indicates how many octets have been sent to the port over the course of delivering the accounting service.
44	Acct-Session-Id	Indicates a unique accounting identifier that makes it easy to match start and stop records in a log file. Acct-Session-Id numbers restart at 1 each time the router is power-cycled or the software is reloaded.
47	Acct-Input-Packets	Indicates how many packets have been received from the port over the course of this service being provided to a framed user.
48	Acct-Output-Packets	Indicates how many packets have been sent to the port in the course of delivering this service to a framed user.

For more information about RADIUS and the use of IETF-defined attributes, see the *Cisco IOS Security Configuration Guide*.

- Overloading the Acct-Session-Id field—Attributes that cannot be mapped to standard RADIUS are packed into the Acct-Session-Id attribute field as ASCII strings separated by the character "/". The Acct-Session-Id attribute is defined to contain the RADIUS account session ID, which is a unique identifier that links accounting records associated with the same login session for a user. To support additional fields, we have defined the following string format for this field:

```
<session id>/<call leg setup time>/<gateway id>/<connection id>/<call origin>/
<call type>/<connect time>/<disconnect time>/<disconnect cause>/<remote ip address>
```

Table 21 shows the field attributes that you use with the overloaded session-ID method and a brief description of each.

Table 21 Field Attributes in Overloaded Acct-Session ID

Field Attribute	Description
Session-Id	Specifies the standard RADIUS account session ID.
Setup-Time	Provides the Q.931 setup time for this connection in Network Time Protocol (NTP) format. NTP time formats are displayed as%H: %M: %S %k %Z %tw %tn %td %Y where: %H is hour (00 to 23). %M is minutes (00 to 59). %S is seconds (00 to 59). %k is milliseconds (000 to 999). %Z is timezone string. %tw is day of week (Saturday through Sunday). %tn is month name (January through December). %td is day of month (01 to 31). %Y is year including century (for example, 1998).
Gateway-Id	Indicates the name of the underlying gateway in the form “gateway.domain_name.”
Call-Origin	Indicates the origin of the call relative to the gateway. Possible values are **originate** and **answer**.
Call-Type	Indicates the call leg type. Possible values are **telephony** and **VoIP**.
Connection-Id	Specifies the unique global identifier used to correlate call legs that belong to the same end-to-end call. The field consists of 4 long words (128 bits). Each long word is displayed as a hexadecimal value and is separated by a space character.
Connect-Time	Provides the Q.931 connect time for this call leg, in NTP format.
Disconnect-Time	Provides the Q.931 disconnect time for this call leg, in NTP format.
Disconnect-Cause	Specifies the reason a call was taken offline as defined in the Q.931 specification.
Remote-Ip-Address	Indicates the address of the remote gateway port where the call is connected.

Because of the limited size of the Acct-Session-Id string, it is not possible to embed very many information elements in it. Therefore, this feature supports only a limited set of accounting information elements.

Use the **gw-accounting h323** command to configure the overloaded session ID method of applying H.323 gateway-specific accounting.

- Using vendor-specific RADIUS attributes—The IETF draft standard specifies a method for communicating vendor-specific information between the network access server and the RADIUS server by using the vendor-specific attribute (Attribute 26). Vendor-specific attributes (VSAs) allow vendors to support their own extended attributes not suitable for general use. The Cisco RADIUS implementation supports one vendor-specific option using the format recommended in the specification. The Cisco vendor-ID is 9, and the supported option has vendor-type 1, which is named "cisco-avpair." The value is a string of the format:

```
protocol: attribute sep value *
```

"Protocol" is a value of the Cisco "protocol" attribute for a particular type of authorization. "Attribute" and "value" are an appropriate attribute/value (AV) pair defined in the Cisco TACACS+ specification, and "sep" is "=" for mandatory attributes and "*" for optional attributes. This allows the full set of features available for TACACS+ authorization to also be used for RADIUS.

The VSA fields and their ASCII values are listed in Table 22.

Table 22 VSA Fields and Their ASCII Values

IETF RADIUS Attribute	Vendor-Specific Company Code	Subtype Number	Attribute Name	Description
26	9	23	h323-remote-address	Indicates the IP address of the remote gateway.
26	9	24	h323-conf-id	Identifies the conference ID.
26	9	25	h323-setup-time	Indicates the setup time for this connection in Coordinated Universal Time (UTC) formerly known as Greenwich Mean Time (GMT) and Zulu time.
26	9	26	h323-call-origin	Indicates the origin of the call relative to the gateway. Possible values are originating and terminating (answer).
26	9	27	h323-call-type	Indicates the call leg type. Possible values are **telephony** and **VoIP**.
26	9	28	h323-connect-time	Indicates the connection time for this call leg in UTC.
26	9	29	h323-disconnect-time	Indicates the time this call leg was disconnected in UTC.
26	9	30	h323-disconnect-cause	Specifies the reason a connection was taken offline per the Q.931 specification.
26	9	31	h323-voice-quality	Specifies the impairment factor (ICPIF) affecting voice quality for a call.
26	9	33	h323-gw-id	Indicates the name of the underlying gateway.

Use the **gw-accounting h323 vsa** command to configure the VSA method of applying H.323 gateway-specific accounting.

- Using syslog records—The syslog accounting option exports the information elements associated with each call leg through a system log message, which can be captured by a syslog daemon on the network. The syslog output consists of the following:

```
<server timestamp> <gateway id> <message number> : <message label> : <list of AV
pairs>
```

The syslog messages fields are listed in Table 23.

Table 23 Syslog Mesage Output Fields

Field	Description
server timestamp	The time stamp created by the server when it receives the message to log.
gateway id	The name of the gateway that emits the message.
message number	The number assigned to the message by the gateway.
message label	A string used to identify the message category.
list of AV pairs	A string that consists of <attribute name> <attribute value> pairs separated by commas.

Use the **gw-accounting syslog** command to configure the syslog record method of gathering H.323 accounting data.

Examples

The following example configures basic H.323 accounting using IETF RADIUS attributes:

```
gw-accounting h323
```

The following example configures H.323 accounting using VSA RADIUS attributes:

```
gw-accounting h323 vsa
```

gw-type-prefix

To configure a technology prefix in the gatekeeper, use the **gw-type-prefix** command in gatekeeper configuration mode. To remove the technology prefix, use the **no** form of this command.

gw-type-prefix *type-prefix* [[**hopoff** *gkid1*] [**hopoff** *gkid2*] [**hopoff** *gkidn*] [**seq** | **blast**]] [**default-technology**] [[**gw ipaddr** *ipaddr* [*port*]]]

no gw-type-prefix *type-prefix* [[**hopoff** *gkid1*] [**hopoff** *gkid2*] [**hopoff** *gkidn*] [**seq** | **blast**]] [**default-technology**] [[**gw ipaddr** *ipaddr* [*port*]]]

Syntax Description

type-prefix	A technology prefix is recognized and is stripped before checking for the zone prefix. It is strongly recommended that you select technology prefixes that do not lead to ambiguity with zone prefixes. Do this by using the # character to terminate technology prefixes, for example, 3#.
hopoff *gkid*	(Optional) Use this option to specify the gatekeeper where the call is to hop off, regardless of the zone prefix in the destination address. The *gkid* argument refers to a gatekeeper previously configured using the zone local or zone remote comment. You can enter this keyword and argument multiple times to configure redundant gatekeepers for a given technology prefix.
seq | **blast**	(Optional) If you list multiple hopoffs, this indicates that the LRQs should be sent sequentially or simultaneously (blast) to the gatekeepers according to the order in which they were listed. The default is to send them sequentially.
default-technology	(Optional) Gateways registering with this prefix option are used as the default for routing any addresses that are otherwise unresolved.
gw ipaddr *ipaddr* [*port*]	(Optional) Use this option to indicate that the gateway is incapable of registering technology prefixes. When it registers, it adds the gateway to the group for this type prefix, just as if it had sent the technology prefix in its registration. This parameter can be repeated to associate more than one gateway with a technology prefix.

Defaults

By default, no technology prefix is defined, and LRQs are sent sequentially to all the gatekeepers listed.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(6)NA2	This command was introduced on the Cisco 2500 and 3600 series routers and the AS5300 universal access server.
12.1(1)T	This command was modified to allow the user to specify multiple hopoffs.
12.1(2)T	This command was modified to allow the user to specify whether LRQs should be sent simultaneously or sequentially to the gatekeepers.

Usage Guidelines

More than one gateway can register with the same technology prefix. In such cases, a random selection is made of one of them.

You do not have to define a technology prefix to a gatekeeper if there are gateways configured to register with that prefix and if there are no special flags (**hopoff** *gkid* or **default-technology**) that you want to associate with that prefix.

You need to configure the gateway type prefix of all remote technology prefixes that will be routed through this gatekeeper.

Examples

The following example defines two gatekeepers for technology zone 3:

```
gw-type-prefix 3#* hopoff c2600-1-gk hopoff c2514-1-gk
```

Related Commands

Command	Description
show gatekeeper gw-type-prefix	Displays the list of currently defined technology zones and the gatekeepers responsible for each.
zone prefix	Configures the gatekeeper with knowledge of its own prefix and the prefix of any remote zone.

h225 timeout tcp establish

To set the H.225 TCP timeout value for Voice over IP (VoIP) dial peers, use the **h225 timeout tcp establish** command in voice class configuration mode. To set the timeout value to its default, use the **no** form of this command.

h225 timeout tcp establish *seconds*

no h225 timeout tcp establish

Syntax Description

seconds	Specifies the number of seconds for the timeout. Possible values are 0 to 30. The default is 15. If you specify 0, the H.225 TCP timer is disabled.

Defaults

The default timeout value is 15 seconds.

Command Modes

Voice class configuration

Command History

Release	Modification
12.1(2)T	This command was introduced on the Cisco 1700, 2500, 2600, 3600, 7200 series routers, AS5300 universal access server, uBR900 series, and uBR924.

Examples

The following example configures a timeout of 10 seconds, which is associated with the H.323 voice class labeled 1:

```
voice class h323 1
 h225 timeout tcp establish 10
```

Related Commands

Command	Description
voice class h323	Establishes an H.323 voice class.

h323 asr

To enable application-specific routing (ASR) and specify the maximum bandwidth for a proxy, use the **h323 asr** command in interface configuration mode. To remove a bandwidth setting but keep ASR enabled, use **no** form of this command.

h323 asr [**bandwidth** *max-bandwidth*]

no h323 asr [**bandwidth** *max-bandwidth*]

Syntax Description

bandwidth *max-bandwidth*	(Optional) Maximum bandwidth on the interface. The value ranges are from 1 to 10,000,000 kbps. If you do not specify the *max-bandwidth*, the value defaults to the bandwidth on the interface. If you specify *max-bandwidth* as a value greater than the interface bandwidth, the bandwidth defaults to the interface bandwidth.

Defaults

ASR is disabled.

Command Modes

Interface configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced on the Cisco 2500 and 3600 series routers.
12.0(3)T	This command was integrated into Cisco IOS Release 12.0(3)T.

Usage Guidelines

This command is independent of the **h323 interface** command.

This command is not supported on Frame Relay or ATM interfaces for the Cisco MC3810 multiservice concentrator.

Note Specifying the **no h323 asr bandwidth** *max-bandwidth* command removes the bandwidth setting but leaves ASR enabled. You must enter the **no h323 asr** command to disable ASR.

Examples

The following example enables ASR and specifies a maximum bandwidth of 10,000 kbps:

```
h323 asr bandwidth 10000
```

h323 call start

To force the H.323 Version 2 gateway to use Fast Connect or Slow Connect procedures for all H.323 calls, use the **h323 call start** command in voice-service configuration mode. To restore the default condition, use the **no** form of this command.

h323 call start {**fast** | **slow**}

no h323 call start

Syntax Description

fast	Gateway uses H.323 Version 2 (Fast Connect) procedures.
slow	Gateway uses H.323 Version 1 (Slow Connect) procedures.

Defaults

The default is **fast**.

Command Modes

Voice-service configuration

Command History

Release	Modification
12.1(3)XI	This command was introduced on the Cisco 2600, 3600, and 7200 series routers, the AS5300 universal access server and AS5800 universal gateways, and on the Cisco MC3810 multiservice concentrator.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.

Usage Guidelines

In Cisco IOS Release 12.1(3)XI and later releases, H.323 Voice over IP (VoIP) gateways by default use H.323 Version 2 (Fast Connect) for all calls including those initiating RSVP. Previously, gateways used only Slow Connect procedures for RSVP calls. To enable Cisco IOS Release 12.1(3)XI gateways to be backward compatible with earlier releases of Cisco IOS Release 12.1 T, the **h323 call start** command forces the originating gateway to initiate calls using Slow Connect.

This **h323 call start** command is configured as part of the global voice-service configuration for VoIP services. It does not take effect unless the **call start system** voice-class configuration command is configured in the VoIP dial peer.

Examples

The following example selects Slow Connect procedures for the gateway:

```
voice service voip
 h323 call start slow
```

Related Commands

Command	Description
call rsvp-sync	Enables synchronization between RSVP and the H.323 voice signaling protocol.
call rsvp-sync resv-timer	Sets the timer for RSVP reservation setup.
call start	Selects whether the H.323 gateway uses Fast Connect or Slow Connect procedures for the specific VoIP dial peer.
debug call rsvp-sync events	Displays the events that occur during RSVP synchronization.
show call rsvp-sync conf	Displays the RSVP synchronization configuration.
show call rsvp-sync stats	Displays statistics for calls that attempted RSVP reservation.
voice service	Enters voice-service configuration mode and specifies the voice encapsulation type.

h323 gatekeeper

To specify the gatekeeper associated with a proxy and to control how the gatekeeper is discovered, use the **h323 gatekeeper** command in interface configuration mode. To disassociate the gatekeeper, use the **no** form of this command.

h323 gatekeeper [**id** *gatekeeper-id*] {**ipaddr** *ipaddr* [*port*] | **multicast**}

no h323 gatekeeper [**id** *gatekeeper-id*] {**ipaddr** *ipaddr* [*port*] | **multicast**}

Syntax Description

id *gatekeeper-id*	(Optional) The *gatekeeper-id* argument specifies the gatekeeper name. Typically, this is a Domain Name Server (DNS) name, but it can also be a raw IP address in dotted form. If this parameter is specified, gatekeepers that have either the default or explicit flags set for the subnet of the proxy will respond. If this parameter is not specified, only those gatekeepers with the default subnet flag will respond.
ipaddr *ipaddr* [*port*]	If this parameter is specified, the gatekeeper discovery message will be unicast to this address and, optionally, the port specified.
multicast	If this parameter is specified, the gatekeeper discovery message will be multicast to the well-known RAS multicast address and port.

Defaults

No gatekeeper is configured for the proxy.

Command Modes

Interface configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced on the Cisco 2500 and 3600 series routers.

Usage Guidelines

You must enter the **h323 interface** and **h323 h323-id** commands before using this command. The **h323 gatekeeper** command must be specified on your Cisco IOS platform or the proxy will not go online. The proxy will use the interface address as its RAS signaling address.

Examples

The following example sets up a unicast discovery to a gatekeeper whose name is unknown:

```
h323 gatekeeper ipaddr 192.168.5.2
```

The following example sets up a multicast discovery for a gatekeeper of a particular name:

```
h323 gatekeeper id gk.zone5.com multicast
```

Related Commands

Command	Description
h323 h323-id	Registers an H.323 proxy alias with a gatekeeper.
h323 interface	Specifies the interface from which the proxy will take its IP address.

h323-gateway voip bind srcaddr

To designate a source IP address for the voice gateway, use the **h323-gateway voip bind srcaddr** command in interface configuration mode. To remove the source IP address, use the **no** form of the command.

h323-gateway voip bind srcaddr *ip-address*

no h323-gateway voip bind srcaddr

Syntax Description

ip-address	Specifies the source IP address in dotted-decimal notation.

Defaults

No default behaviors or values.

Command Modes

Interface configuration

Command History

Release	Modification
12.1(2)T	This command was introduced on the Cisco 1700, 2500, 2600, 3600, and 7200 series routers, the AS5300 universal access server, and the uBR924.

Usage Guidelines

You can issue this command on any interface in the router. You do not have to issue it on the interface that you defined as the voice gateway interface (although it may be more convenient to do so). Issuing the command for one interface assigns the source IP address for the entire router.

Examples

The following example assigns a source IP address of 10.1.1.1:

```
h323-gateway voip bind srcaddr 10.1.1.1
```

h323-gateway voip h323-id

To configure the H.323 name of the gateway that identifies this gateway to its associated gatekeeper, use the **h323-gateway voip h323-id** command in interface configuration mode. To disable this defined gateway name, use the **no** form of this command.

h323-gateway voip h323-id *interface-id*

no h323-gateway voip h323-id *interface-id*

Syntax Description

interface-id	H.323 name (ID) used by this gateway when this gateway communicates with its associated gatekeeper. Usually, this ID is the name of the gateway with the gatekeeper domain name appended to the end: name@domain-name.

Defaults

No gateway identification is defined.

Command Modes

Interface configuration

Command History

Release	Modification
11.3(6)NA2	This command was introduced on the Cisco 2500 and 3600 series routers and the Cisco AS5300 universal access server.

Examples

The following example configures Ethernet interface 0.0 as the gateway interface. In this example, the gateway ID is GW13@cisco.com.

```
interface Ethernet0/0
 ip address 172.16.53.13 255.255.255.0
 h323-gateway voip interface
 h323-gateway voip id GK15.cisco.com ipaddr 172.16.53.15 1719
 h323-gateway voip h323-id GW13@cisco.com
 h323-gateway voip tech-prefix 13#
```

Related Commands

Command	Description
h323-gateway voip id	Defines the name and location of the gatekeeper for this gateway.
h323-gateway voip interface	Configures an interface as an H.323 interface.
h323-gateway voip tech-prefix	Defines the technology prefix that the gateway will register with the gatekeeper.

h323-gateway voip id

To define the name and location of the gatekeeper for a specific gateway, use the **h323-gateway voip id** command in interface configuration mode. To disable this gatekeeper identification, use the **no** form of this command.

h323-gateway voip id *gatekeeper-id* {**ipaddr** *ip-address* [*port-number*] | **multicast**} [**priority** *number*]

no h323-gateway voip id *gatekeeper-id* {**ipaddr** *ip-address* [*port-number*] | **multicast**} [**priority** *number*]

Syntax Description

gatekeeper-id	Indicates the H.323 identification of the gatekeeper. This value must exactly match the gatekeeper ID in the gatekeeper configuration. The recommended format is *name.doman-name*.
ipaddr	Indicates that the gateway will use an IP address to locate the gatekeeper.
ip-address	Defines the IP address used to identify the gatekeeper.
port-number	(Optional) Defines the port number used.
multicast	Indicates that the gateway will use multicast to locate the gatekeeper.
priority *number*	(Optional) The priority of this gatekeeper. The range is 1 through 127, and the default value is 127.

Defaults

No gatekeeper identification is defined.

Command Modes

Interface configuration

Command History

Release	Modification
11.3(6)NA2	This command was introduced on the Cisco 2500 and 3600 series routers and the Cisco AS5300 universal access server.
12.0(7)T	The **priority** *number* keyword and argument option was added.

Usage Guidelines

This command tells the H.323 gateway associated with this interface which H.323 gatekeeper to talk to and where to locate it. The gatekeeper ID configured here must exactly match the gatekeeper ID in the gatekeeper configuration.

You can configure up to two alternate gatekeepers.

The IP address of the gatekeeper does not have to be explicit; you can also use the multicast option. Multicasting saves bandwidth by forcing the network to replicate packets only when necessary. The multicast option, shown below, notifies every gatekeeper in the LAN using a universal address, 224.0.1.41.

```
h323-gateway voip id GK1 multicast
h323-gateway voip id GK2 ipaddr 172.18.193.65 1719
```

Examples

The following example configures Ethernet interface 0.0 as the gateway interface and defines a specific gatekeeper for it. In this example, the gatekeeper ID is GK15.cisco.com and its IP address is 172.16.53.15 (using port 1719).

```
interface Ethernet0/0
 ip address 172.16.53.13 255.255.255.0
 h323-gateway voip interface
 h323-gateway voip id GK15.cisco.com ipaddr 172.16.53.15 1719
 h323-gateway voip h323-id GW13@cisco.com
 h323-gateway voip tech-prefix 13#
```

Related Commands

Command	Description
h323-gateway voip h323-id	Configures the H.323 name of the gateway that identifies this gateway to its associated gatekeeper.
h323-gateway voip interface	Configures an interface as an H.323 interface.
h323-gateway voip tech-prefix	Defines the technology prefix that the gateway will register with the gatekeeper.

h323-gateway voip interface

To configure an interface as an H.323 gateway interface, use the **h323-gateway voip interface** command in interface configuration mode. To disable H.323 gateway functionality for an interface, use the **no** form of this command.

h323-gateway voip interface

no h323-gateway voip interface

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Interface configuration

Command History

Release	Modification
11.3(6)NA2	This command was introduced on the Cisco 2500 and 3600 series routers and the AS5300 universal access server.

Examples

The following example configures Ethernet interface 0.0 as the gateway interface. In this example, the **h323-gateway voip interface** command configures this interface as an H.323 interface.

```
interface Ethernet0/0
 ip address 172.16.53.13 255.255.255.0
 h323-gateway voip interface
 h323-gateway voip id GK15.cisco.com ipaddr 172.16.53.15 1719
 h323-gateway voip h323-id GW13@cisco.com
 h323-gateway voip tech-prefix 13#
```

Related Commands

Command	Description
h323-gateway voip h323-id	Configures the H.323 name of the gateway that identifies this gateway to its associated gatekeeper.
h323-gateway voip id	Defines the name and location of the gatekeeper for this gateway.
h323-gateway voip tech-prefix	Defines the technology prefix that the gateway will register with the gatekeeper.

h323-gateway voip tech-prefix

To define the technology prefix that the gateway will register with the gatekeeper, use the **h323-gateway voip tech-prefix** command in interface configuration mode. To disable this defined technology prefix, use the **no** form of this command.

h323-gateway voip tech-prefix *prefix*

no h323-gateway voip tech-prefix *prefix*

Syntax Description

prefix	Defines the numbers used as the technology prefixes. Each technology prefix can contain up to 11 characters. Although not strictly necessary, a pound symbol (#) is frequently used as the last digit in a technology prefix. Valid characters are 0 though 9, the pound symbol (#), and the asterisk (*).

Defaults

Disabled

Command Modes

Interface configuration

Command History

Release	Modification
11.3(6)NA2	This command was introduced on the Cisco 2500 and 3600 series routers and the Cisco AS5300 universal access server.

Usage Guidelines

This command defines a technology prefix that the gateway will then register with the gatekeeper. Technology prefixes can be used as a discriminator so that the gateway can tell the gatekeeper that a certain technology is associated with a particular call (for example, 15# could mean a fax transmission), or it can be used like an area code for more generic routing. No standard currently defines what the numbers in a technology prefix mean. By convention, technology prefixes are designated by a pound symbol (#) as the last character.

Note Cisco gatekeepers use the asterisk (*) as a reserved character. If you are using Cisco gatekeepers, do not use the asterisk as part of the technology prefix.

Examples

The following example configures Ethernet interface 0.0 as the gateway interface. In this example, the technology prefix is defined as 13#.

```
interface Ethernet0/0
 ip address 172.16.53.13 255.255.255.0
 h323-gateway voip interface
 h323-gateway voip id GK15.cisco.com ipaddr 172.16.53.15 1719
 h323-gateway voip h323-id GW13@cisco.com
 h323-gateway voip tech-prefix 13#
```

Related Commands

Command	Description
h323-gateway voip h323-id	Configures the H.323 name of the gateway that identifies this gateway to its associated gatekeeper.
h323-gateway voip id	Defines the name and location of the gatekeeper for this gateway.
h323-gateway voip interface	Configures an interface as an H.323 interface.

h323 h323-id

To register an H.323 proxy alias with a gatekeeper, use the **h323 h323-id** command in interface configuration mode. To remove an H.323 proxy alias, use the **no** form of this command.

h323 h323-id *h323-id*

no h323 h323-id *h323-id*

Syntax Description

h323-id	Specifies the name of the proxy. It is recommended that this name be a fully qualified e-mail ID, with the domain name being the same as that of its gatekeeper.

Defaults

No H.323 proxy alias is registered.

Command Modes

Interface configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced on the Cisco 2500 and 3600 series routers.
12.0(3)T	This command was integrated into Cisco IOS Release 12.0(3)T.

Usage Guidelines

Each entry registers a specified H.323 ID proxy alias to a gatekeeper. Typically, these aliases are either simple text strings or legitimate e-mail IDs.

Note You must enter the **h323 interface** command before using this command. The **h323 h323-id** command must be entered on the same interface as the **h323 gatekeeper** command. The proxy will not go online without the **h323 interface** command.

Examples

The following example registers an H.323 proxy alias called proxy1@zone5.com with a gatekeeper:

```
h323 h323-id proxy1@zone5.com
```

Related Commands

Command	Description
h323 gatekeeper	Specifies the gatekeeper associated with a proxy and controls how the gatekeeper is discovered.
h323 interface	Specifies the interface from which the proxy will take its IP address.

h323 interface

To select an interface whose IP address will be used by the proxy to register with the gatekeeper, use the **h323 interface** command in interface configuration mode. To use the default port, use the **no h323 interface** command and then the **h323 interface** command.

h323 interface [*port-number*]

no h323 interface [*port-number*]

Syntax Description

port-number	(Optional) The port number the proxy will listen on for incoming call setup requests. Range is 1 to 65,356. The default port number for the proxy is 11,720 in -isx- or -jsx- Cisco IOS images. The default port number for the proxy is 1720 in -ix- Cisco IOS images, which do not contain the VoIP gateway.

Defaults

Default port number is image dependent as described in the Syntax Description.

Command Modes

Interface configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced on the Cisco 2500 and 3600 series routers.
12.1(5)T	The ability to specify the proxy port number was added on the 2600, 3600, and 7200 series routers and on the Cisco MC3810 multiservice concentrator.

Usage Guidelines

At proxy startup, Cisco IOS software checks for the presence of the VoIP gateway subsystem. If the subsystem is found to be present, the proxy code opens and listens for call setup requests on the new port. The proxy then registers this port with the gatekeeper.

Examples

The following example shows how to configure Ethernet interface 0 for incoming call setup requests:

```
interface ethernet0
 h323 interface
```

Related Commands

Command	Description
bandwidth	Specifies the maximum aggregate bandwidth for H.323 traffic from a zone to another zone, within a zone, or for a session in a zone.
bandwidth remote	Specifies the total bandwidth for H.323 traffic between this gatekeeper and any other gatekeeper.
h323 qos	Enables QoS on the proxy.
h323 t120	Enables the T.120 capabilities on your router and specifies bypass or proxy mode.

h323 qos

To enable quality of service (QoS) on the proxy, use the **h323 qos** command in interface configuration mode. To disable QoS, use the **no** form of this command.

h323 qos {**ip-precedence** *value* | **rsvp** {**controlled-load** | **guaranteed-qos**}}

no h323 qos {**ip-precedence** *value* | **rsvp** {**controlled-load** | **guaranteed-qos**}}

Syntax Description

ip-precedence *value*	Specifies that RTP streams should set their IP precedence bits to the specified value.
rsvp controlled-load	Specifies controlled load class of service.
rsvp guaranteed-qos	Specifies guaranteed QoS class of service.

Defaults

No QoS is configured.

Command Modes

Interface configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced on the Cisco 2500 and 3600 series routers.

Usage Guidelines

You must execute the **h323 interface** command before using this command.

Both IP precedence and RSVP QoS can be configured by invoking this command twice with the two different QoS forms.

Examples

The following example enables QoS on the proxy:

```
interface Ethernet0
 ip address 172.21.127.38 255.255.255.192
 no ip redirects
 ip rsvp bandwidth 7000 7000
 ip route-cache same-interface
 fair-queue 64 256 1000
 h323 interface
 h323 qos rsvp controlled-load
 h323 h323-id px1@zone1.com
 h323 gatekeeper ipaddr 172.21.127.39
```

Related Commands

Command	Description
h323 interface	Specifies the interface from which the proxy will take its IP address.

h323 t120

To enable the T.120 capabilities on your router and to specify bypass or proxy mode, use the **h323 t120** command in interface configuration mode.

h323 t120 {**bypass** | **proxy**}

Syntax Description

bypass	Bypass mode. In this mode, the H.245 Open Logical Channel messages for T.120 data channels are passed unmodified through the proxy, and TCP connections for T.120 are established directly between the two endpoints of the H.323 call.
proxy	Proxy mode. In this mode, T.120 features function properly.

Defaults

Bypass mode

Command Modes

Interface configuration

Command History

Release	Modification
12.1(5)T	This command was introduced on on the Cisco 2600, 3600, and 7200 series routers and the Cisco MC3810 multiservice concentrator.

Usage Guidelines

The **no** form of this command has no function—the only possible commands are **h323 t120 bypass** and **h323 t120 proxy**.

Examples

The following example shows how to enable the T.120 capabilities:

```
proxy h323
interface ethernet0
 h323 t120 proxy
```

Related Commands

Command	Description
bandwidth	Specifies the maximum aggregate bandwidth for H.323 traffic from a zone to another zone, within a zone, or for a session in a zone.
bandwidth remote	Specifies the total bandwidth for H.323 traffic between this gatekeeper and any other gatekeeper.
h323 interface	Defines which port the proxy will listen on.

huntstop

To disable all dial-peer hunting if a call fails when using hunt groups, use the **huntstop** command in dial-peer configuration mode. To reenable dial-peer hunting, use the **no** form of this command.

huntstop

no huntstop

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(5)T	This command was introduced on the Cisco MC3810 multiservice concentrator.
12.0(7)XK	Support for this command was extended to the Cisco 2600 and 3600 series routers.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

Once you enter this command, no further hunting is allowed if a call fails on the specified dial peer.

Note This command can be used with all types of dial peers.

Examples

The following example shows how to disable dial-peer hunting on a specific dial peer:

```
dial peer voice 100 vofr
 huntstop
```

The following example shows how to reenable dial-peer hunting on a specific dial peer:

```
dial peer voice 100 vofr
 no huntstop
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice-related encapsulation.

icpif

To specify the Impairment/Calculated Planning Impairment Factor (ICPIF) for calls sent by a dial peer, use the **icpif** command in dial-peer configuration mode. To restore the default value, use the **no** form of this command.

icpif *integer*

no icpif *integer*

Syntax Description

integer	Integer, expressed in equipment impairment factor units, that specifies the ICPIF value. Valid entries are 0 to 55. The default is 20.

Defaults

20

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series routers.
12.0(7)XK	This command was first supported on the Cisco MC3810 multiservice concentrator.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

This command is applicable only to Voice over IP (VoIP) dial peers.

Use the **icpif** command to specify the maximum acceptable impairment factor for the voice calls sent by the selected dial peer.

Examples

The following example disables the **icpif** command:

```
dial-peer voice 10 voip
 icpif 0
```

idle-voltage

To specify the idle voltage on an Foreign Exchange Station (FXS) voice port, use the **idle-voltage** command in voice-port configuration mode. To restore the default idle voltage, use the **no** form of this command.

idle-voltage {**high** | **low**}

no idle-voltage

Syntax Description

high	The talk-battery (tip-to-ring) voltage is high (–48V) when the FXS port is idle.
low	The talk-battery (tip-to-ring) voltage is low (–24V) when the FXS port is idle.

Defaults

The idle voltage is –24V.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.0(4)T	This command was introduced on the Cisco MC3810 multiservice concentrator.

Usage Guidelines

Some fax equipment and answering machines require a –48V idle voltage to be able to detect an off-hook condition in a parallel phone.

If the idle voltage setting is **high**, the talk battery reverts to –24V whenever the voice port is active (off hook).

The **idle-voltage** command applies only to FXS voice ports on Cisco MC3810 multiservice concentrators.

Examples

The following example sets the idle voltage to –48V on voice port 1/1 on a Cisco MC3810 multiservice concentrator:

```
voice-port 1/1
 idle-voltage high
```

The following example restores the default idle voltage (–24V) on voice port 1/1 on a Cisco MC3810 multiservice concentrator:

```
voice-port 1/1
 no idle-voltage
```

Related Commands

Command	Description
show voice port	Displays voice port configuration information.

ignore

To configure the North American E&M or E&M MELCAS voice port to ignore specific receive bits, use the **ignore** command in voice-port configuration mode. To restore the default value, use the **no** form of this command.

ignore {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**}

no ignore {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**}

Syntax Description

rx-a-bit	Ignores the receive A bit.
rx-b-bit	Ignores the receive B bit.
rx-c-bit	Ignores the receive C bit.
rx-d-bit	Ignores the receive D bit.

Defaults

The default is mode-dependent:

- North American E&M:
  - The receive B, C, and D bits are ignored.
  - The receive A bit is not ignored.
- E&M MELCAS:
  - The receive A bit is ignored.
  - The receive B, C, and D bits are not ignored.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810 multiservice concentrator.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

The **ignore** command applies to E&M digital voice ports associated with T1/E1 controllers. Repeat the command for each receive bit to be configured. Use this command with the **define** command.

Examples

To configure voice port 1/1 on a Cisco MC3810 multiservice concentrator to ignore receive bits A, B, and C and to monitor receive bit D, enter the following commands:

```
voice-port 1/1
 ignore rx-a-bit
 ignore rx-b-bit
```

```
 ignore rx-c-bit
 no ignore rx-d-bit
```

To configure voice port 1/0/0 on a Cisco 3600 series router to ignore receive bits A, C, and D and to monitor receive bit B, enter the following commands:

```
voice-port 1/0/0
 ignore rx-a-bit
 ignore rx-c-bit
 ignore rx-d-bit
 no ignore rx-b-bit
```

Related Commands

Command	Description
condition	Manipulates the signaling bit pattern for all voice signaling types.
define	Defines the transmit and receive bits for North American E&M and E&M MELCAS voice signaling.
show voice port	Displays configuration information for voice ports.

image encoding

To select a specific encoding method for fax images associated with an MMoIP dial peer, use the **image encoding** command in dial-peer configuration mode. To restore the default value, use the **no** form of this command.

image encoding {**mh** | **mr** | **mmr** | **passthrough**}

no image encoding {**mh** | **mr** | **mmr** | **passthrough**}

Syntax Description

mh	Specifies Modified Huffman image encoding. This is the IETF standard.
mr	Specifies Modified Read image encoding.
mmr	Specifies Modified Modified Read image encoding.
passthrough	Specifies that the image will not be modified by an encoding method.

Defaults

passthrough

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

Use the **image encoding** command to specify an encoding method for e-mail fax TIFF images for a specific MMoIP dial peer. This command applies primarily to the on-ramp MMoIP dial peer. Although you can optionally create an off-ramp dial peer and configure a particular image encoding value for that off-ramp call leg, store and forward fax ignores the off-ramp MMoIP setting and sends the file using Modified Huffman encoding.

There are four available encoding methods:

- Modified Huffman (MH)—One-dimensional data compression scheme that compresses data in only one direction (horizontal). Modified Huffman compression does not allow the transmission of redundant data. This encoding method produces the largest image file size.
- Modified Read (MR)—Two-dimensional data compression scheme (used by fax devices) that handles the data compression of the vertical line and that concentrates on the space between lines and within given characters.
- Modified Modified Read (MMR)—Data compression scheme used by newer Group 3 fax devices. This encoding method produces the smallest possible image file size and is slightly more efficient than Modified Read.
- Passthrough—No encoding method will be applied to the image—meaning that the image will be encoded by whatever encoding method is used by the fax device.

The IETF standard for sending fax TIFF images is Modified Huffman encoding with fine or standard resolution. RFC 2301 requires that compliant receivers support TIFF images with MH encoding and fine or standard resolution. If a receiver supports features beyond this minimal requirement, you might want to configure the Cisco AS5300 universal access server to send enhanced-quality documents to that receiver.

The primary reason to use a different encoding scheme from MH is to save network bandwidth. MH ensures interoperability with all Internet fax devices, but it is the least efficient of the encoding schemes for sending fax TIFF images. For most images, MR is more efficient than MH, and MMR is more efficient than MR. If you know that the recipient is capable of receiving more efficient encodings than just MH, store and forward fax allows you to send the most efficient encoding that the recipient can process. For end-to-end closed networks, you can choose any encoding scheme because the off-ramp gateway can process MH, MR, and MMR.

Another factor to consider is the viewing software. Many viewing applications (for example, those that come with Windows 95 or Windows NT) are able to display MH, MR, and MMR. Therefore you should decide, based on the viewing application and the available bandwidth, which encoding scheme is right for your network.

This command applies to both on-ramp and off-ramp store and forward fax functions.

Examples

The following example selects Modified Modified Read as the encoding method for fax TIFF images sent by MMoIP dial peer 10:

```
dial-peer voice 10 mmoip
 image encoding mmr
```

Related Commands

Command	Description
image resolution	Specifies a particular fax image resolution for a specific MMoIP dial peer.

image resolution

To specify a particular fax image resolution for a specific MMoIP dial peer, use the **image resolution** command in dial-peer configuration mode. To restore the default value, use the **no** form of this command.

image resolution {**fine** | **standard** | **superfine** | **passthrough**}

no image resolution {**fine** | **standard** | **superfine** | **passthrough**}

Syntax Description

fine	Configures the fax TIFF image resolution to be 204-by-196 pixels per inch.
standard	Configures the fax TIFF image resolution to be 204-by-98 pixels per inch.
superfine	Configures the fax TIFF image resolution to be 204-by-391 pixels per inch.
passthrough	Indicates that the resolution of the fax TIFF image will not be altered.

Defaults

passthrough

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

Use the **image resolution** command to specify a specific resolution (in pixels per inch) for e-mail fax TIFF images sent by the specified MMoIP dial peer. This command applies primarily to the on-ramp MMoIP dial peer. Although you can optionally create an off-ramp dial peer and configure a particular image resolution value for that off-ramp call leg, store and forward fax ignores the off-ramp MMoIP setting and sends the file using fine resolution.

This command enables you to increase or decrease the resolution of a fax TIFF image, thereby changing not only the resolution but also the size of the fax TIFF file. The IETF standard for sending fax TIFF images is Modified Huffman encoding with fine or standard resolution. The primary reason to configure a different resolution is to save network bandwidth.

This command applies to both on-ramp and off-ramp store and forward fax functions.

Examples

The following example selects the fine resolution (meaning 204-by-196 pixels per inch) for e-mail fax TIFF images associated with MMoIP dial peer 10:

```
dial-peer voice 10 mmoip
 image encoding mh
 image resolution fine
```

Related Commands

Command	Description
image encoding	Selects a specific encoding method for fax images associated with an MMoIP dial peer.

impedance

To specify the terminating impedance of a voice-port interface, use the **impedance** command in voice-port configuration mode. To restore the default value, use the **no** form of this command.

impedance {**600c** | **600r** | **900c** | **complex1** | **complex2**}

no impedance {**600c** | **600r** | **900c** | **complex1** | **complex2**}

Syntax Description

600c	Specifies 600 ohms (complex).
600r	Specifies 600 ohms (real).
900c	Specifies 900 ohms (complex).
complex1	Specifies complex 1.
complex2	Specifies complex 2.

Defaults

600r

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series routers.

Usage Guidelines

Use the **impedance** command to specify the terminating impedance of an Foreign Exchange Office (FXO) voice-port interface. The impedance value selected needs to match the specifications from the specific telephony system to which it is connected. Different countries often have different standards for impedance. CO switches in the United States are predominantly 600r. PBXs in the United States are normally either 600r or 900c.

If the impedance is set incorrectly (if there is an impedance mismatch), there will be a significant amount of echo generated (which could be masked if the **echo-cancel** command has been enabled). In addition, gains might not work correctly if there is an impedance mismatch.

Configuring the impedance on a voice port will change the impedance on both voice ports of a VPM card. This voice port must be shut down and then opened for the new value to take effect.

Examples

The following example configures an FXO voice port on the Cisco 3600 series router for a terminating impedance of 600 ohms (real):

```
voice-port 1/0/0
 impedance 600r
```

The following example configures an E&M voice port on the Cisco MC3810 multiservice concentrator for a terminating impedance of 900 ohms (complex):

```
voice-port 1/1
 impedance 900c
```

incoming called-number

To specify a digit string that can be matched by an incoming call to associate the call with a dial peer, use the **incoming called-number** command in dial-peer configuration mode. To reset the default value, use the **no** form of this command.

incoming called-number *string*

no incoming called-number *string*

Syntax Description

string	Specifies the incoming called telephone number. Valid entries are any series of digits that specify the E.164 telephone number.

Defaults

No incoming called number is defined.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series routers.
11.3NA	This command was introduced on the Cisco AS5800 universal gateway.
12.0(4)XJ	This command was modified for store and forward fax.
12.0(7)XK	This command was first supported on the Cisco MC3810 multiservice concentrator platform.
12.1(2)T	This command was integrated into the Cisco IOS Release 12.1(2)T.

Usage Guidelines

When a Cisco device (such as a Cisco AS5300 universal access server or Cisco AS5800 universal gateway) is handling both modem and voice calls, it needs to be able to identify the service type of the call—meaning whether the incoming call to the server is a modem or a voice call. When the access server handles only modem calls, the service type identification is handled through modem pools. Modem pools associate calls with modem resources based on the called number (DNIS). In a mixed environment, in which the server receives both modem and voice calls, you need to identify the service type of a call by using the **incoming called-number** command.

If you do not use the **incoming called-number** command, the server attempts to resolve whether an incoming call is a modem or voice call based on the interface over which the call comes. If the call comes in over an interface associated with a modem pool, the call is assumed to be a modem call; if a call comes in over a voice port associated with a dial peer, the call is assumed to be a voice call.

By default, there is no called number associated with the dial peer, which means that incoming calls will be associated with dial peers based on matching calling number with answer address, call number with destination pattern, or calling interface with configured interface.

Use the **incoming called-number** command to define the destination telephone number for a particular dial peer. For the on-ramp POTS dial peer, this telephone number is the DNIS number of the incoming fax call. For the off-ramp MMoIP dial peer, this telephone number is the destination fax machine telephone number.

This command applies to both Voice over IP (VoIP) and POTS dial peers and applies to both on-ramp and off-ramp store and forward fax functions.

This command is also used to provide a matching VoIP dial peer on the basis of called number when fax or modem pass-through with named service events (NSEs) is defined globally on a terminating gateway.

You can ensure that all calls will match at least one dial peer by using the following commands:

```
Router(config)# dial-peer voice tag voip
Router(config-dial-peer)# incoming called-number .
```

Examples

The following example configures calls coming in to the server with a called number of 3799262 as being voice calls:

```
dial peer voice 10 pots
 incoming called-number 3799262
```

The following example configures the number (310) 555-9261 as the incoming called number for MMoIP dial peer 10:

```
dial-peer voice 10 mmoip
 incoming called-number 3105559261
```

info-digits

To automatically prepend two information digits to the beginning of a dialed number associated with the given POTS dial peer, use the **info-digits** command in dial-peer configuration mode. To keep the router from automatically prepending the two-digit information numbers to the beginning of the POTS dial peer, use the **no** form of this command.

info-digits *string*

no info-digits

Syntax Description

string	Specifies the two-digit prefix that the router will automatically prepend to the dialed number for the given POTS dial peer. **Note** This string cannot contain any more or any less than two digits.

Defaults

No default behavior or values.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.2(1)T	This command was introduced on Cisco 2600 series, Cisco 3600 series, and Cisco 3700 series routers and on Cisco AS5300 series universal access servers.

Usage Guidelines

This command is designed to prepend a pair of information digits to the beginning of the dialed number string for the POTS dial peer that will enable you to dynamically redirect the outgoing call. The **info-digits** command is only available for POTS dial peers.

Examples

The following example prepends the information number string 91 to the beginning of the dialed number for POTS dial peer 10:

```
dial-peer voice 10 pots
 info-digits 91
```

information-type

To select a particular information type for either an Mail Message over IP (MMoIP) or Plain Old Telephone Service (POTS) dial peer, use the **information-type** command in dial-peer configuration mode. To reset the default value for this command, use the **no** form of this command.

information-type {**fax** | **voice**}

no information-type {**fax** | **voice**}

Syntax Description

fax	Indicates that the information type has been set to store and forward fax.
voice	Indicates that the information type has been set to voice.

Defaults

Voice

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series routers.
12.0(4)XJ	This command was modified for store and forward fax.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

This command applies to both on-ramp and off-ramp store and forward fax functions.

Examples

The following example sets the information type for MMoIP dial peer 10 to fax:

```
dial-peer voice 10 mmoip
 information-type fax
```

input gain

To configure a specific input gain value, use the **input gain** command in voice-port configuration mode. To disable the selected amount of inserted gain, use the **no** form of this command.

input gain *decibels*

no input gain *decibels*

Syntax Description

decibels	Specifies, in decibels, the amount of gain to be inserted at the receiver side of the interface. Acceptable values are integers from –6 to 14.

Defaults

Zero (0) decibels

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
11.3(1)MA	This command was first supported on the Cisco MC3810 multiservice concentrator.

Usage Guidelines

A system-wide loss plan must be implemented using both the **input gain** and **output attenuation** commands. Other equipment (including PBXs) in the system must be considered when creating a loss plan. The default value for this command assumes that a standard transmission loss plan is in effect, meaning that there must be an attenuation of –6 dB between phones. Connections are implemented to provide –6 dB of attenuation when the **input gain** and **output attenuation** commands are configured with the default value of 0 dB.

You cannot increase the gain of a signal to the Public Switched Telephone Network (PSTN), but you can decrease it. If the voice level is too high, you can decrease the volume by either decreasing the input gain or increasing the output attenuation.

You can increase the gain of a signal coming into the router. If the voice level is too low, you can increase the input gain by using the **input gain** command.

Examples

The following example configures a 3-dB gain to be inserted at the receiver side of the interface in the Cisco 3600 series router:

```
port 1/0/0
 input gain 3
```

The following example configures a 3-dB gain to be inserted at the receiver side of the interface in the Cisco MC3810 multiservice concentrator:

```
port 1/1
 input gain 3
```

Related Commands

Command	Description
output attenuation	Configures a specific output attenuation value for a voice port.

interface (RLM server)

To define the IP addresses of the Redundant Link Manager (RLM) server, use the **interface** command in interface configuration mode. To disable this function, use the **no** form of this command.

interface *name-tag*

no interface *name-tag*

Syntax Description

name-tag	Name to identify the server configuration so that multiple entries of server configuration can be entered.

Defaults

Disabled

Command Modes

Interface configuration

Command History

Release	Modification
11.3(7)	This command was introduced.

Usage Guidelines

Each server can have multiple entries of IP addresses or aliases.

Examples

The following example shows how to configure the access server interfaces for RLM servers named Loopback1 and Loopback2:

```
interface Loopback1
 ip address 10.1.1.1 255.255.255.255
 interface Loopback2
 ip address 10.1.1.2 255.255.255.255
rlm group 1
 server r1-server
 link address 10.1.4.1 source Loopback1 weight 4
 link address 10.1.4.2 source Loopback2 weight 3
```

Related Commands

Command	Description
clear interface	Resets the hardware logic on an interface.
clear rlm group	Clears all RLM group time stamps to zero.
link (RLM)	Specifies the link preference.
protocol rlm port	Reconfigures the port number for the basic RLM connection for the whole rlm-group.
retry keepalive	Allows consecutive keepalive failures a certain amount of time before the link is declared down.
server (RLM)	Defines the IP addresses of the server.
show rlm group statistics	Displays the network latency of the RLM group.
show rlm group status	Displays the status of the RLM group.
show rlm group timer	Displays the current RLM group timer values.
shutdown (RLM)	Shuts down all of the links under the RLM group.
timer	Overwrites the default setting of timeout values.

ip precedence (dial-peer)

To set IP precedence (priority) for packets sent by the dial peer, use the **ip precedence** command in dial-peer configuration mode. To restore the default value, use the **no** form of this command.

ip precedence *number*

no ip precedence *number*

Syntax Description

number	Integer specifying the IP precedence value. Valid entries are from 0 to 7. A value of 0 means that no precedence (priority) has been set.

Defaults

The default value for this command is zero (0).

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)NA	This command was introduced on the Cisco 2500, 3600 series routers and the Cisco AS5300 universal access server.

Usage Guidelines

Use the **ip precedence** (dial-peer) command to configure the value set in the IP precedence field when voice data packets are sent over the IP network. This command should be used if the IP link utilization is high and the quality of service for voice packets needs to have a higher priority than other IP packets. The **ip precedence** (dial-peer) command should also be used if RSVP is not enabled and the user would like to give voice packets a higher priority than other IP data traffic.

This command applies to Voice over IP (VoIP) peers.

Examples

The following example sets the IP precedence to 5:

```
dial-peer voice 10 voip
 ip precedence 5
```

ip udp checksum

To calculate the UDP checksum for voice packets sent by the dial peer, use the **ip udp checksum** command in dial-peer configuration mode. To disable this feature, use the **no** form of this command.

ip udp checksum

no ip udp checksum

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series routers.

Usage Guidelines

Use the **ip udp checksum** command to enable UDP checksum calculation for each of the outbound voice packets. This command is disabled by default to speed up the transmission of the voice packets. If you suspect that the connection has a high error rate, you should enable the **ip udp checksum** command to prevent corrupted voice packets forwarded to the digital signal processor (DSP).

This command applies to Voice over IP (VoIP) peers.

Examples

The following example calculates the UDP checksum for voice packets sent by dial peer 10:

```
dial-peer voice 10 voip
 ip udp checksum
```

Related Commands

Command	Description
loop-detect	Enables loop detection for T1 for Voice over ATM, Voice over Frame Relay, and Voice over HDLC.

isdn bind-l3

To configure the ISDN serial interface for backhaul, use the **isdn bind-l3** command in interface configuration mode. To disable backhaul on the interface, use the **no** form of this command.

isdn bind-l3 *set-name*

no isdn bind-l3 *set-name*

Syntax Description

set-name	Session-set name.

Defaults

No default behavior or values.

Command Modes

Interface configuration

Command History

Release	Modification
12.1(1)T	This command was introduced.

Examples

To configure the ISDN serial interface for backhaul for the set named Set1, see the following example:

```
Router(config-if)# isdn bind-l3 set1
```

isdn contiguous-bchan

To configure contiguous bearer channel handling on an E1 PRI interface, use the **isdn contiguous-bchan** command in interface configuration mode. To disable the contiguous B-channel handling, use the **no** form of this command.

isdn contiguous-bchan

no isdn contiguous-bchan

Syntax Description

This command has no arguments or keywords.

Defaults

By default, contiguous B channel handling is disabled.

Command Modes

Interface configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced on the Cisco 2500, 3600, and 7200 series routers and on the Cisco MC3810 multiservice concentrator.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

Use the **isdn contiguous-bchan** command to specify contiguous bearer channel handling so that B channels 1 through 30, skipping 16, map to time slots 1 through 31. This is available for E1 PRI interfaces only, when the **primary-qsig** switch type option is configured by using the **isdn switch-type** command.

Examples

The following example shows the command configuration on the E1 interface of a Cisco 3660 series router E1 interface:

```
interface Serial5/0:15
 no ip address
 ip mroute-cache
 no logging event link-status
 isdn switch-type primary-qsig
 isdn overlap-receiving
 isdn incoming-voice voice
 isdn continguous-bchan
```

Related Commands

Command	Description
isdn switch-type primary-qsig	Configures the primary-qsig switch type for PRI support.

isdn global-disconnect

To allow passage of "release" and "release complete" messages over the voice network, use the **isdn global-disconnect** command in interface configuration mode. To disable the passage of these messages, use the **no** form of this command.

isdn global-disconnect

no isdn global-disconnect

Syntax Description

This command has no arguments or keywords.

Defaults

Passage of messages is disabled by default; "release" and "release complete" messages terminate locally by default.

Command Modes

Interface configuration

Command History

Release	Modification
12.1(2)T	This command was introduced on the Cisco 2600, 3600, and 7200 series routers and on the Cisco MC3810 multiservice concentrator.

Usage Guidelines

Enter this command under the isdn interface with switch type bri-qsig or pri-qsig. Use the **isdn global-disconnect** command to allow passage of "release" and "release complete" messages end-to-end across the network. This is required for certain types of QSIG PBXs whose software or features require either Facility or User Info IEs in those messages to be passed end-to-end between the PBXs. All QSIG interfaces that connect the PBXs to the routers must have this command enabled. This command is available when using the BRI QSIG or PRI QSIG switch type in either master or slave mode.

Examples

The following example shows command configuration on the T1 PRI interface of a Cisco 3660 series router:

```
interface Serial5/0:23
 no ip address
 ip mroute-cache
 no logging event link-status
 isdn switch-type primary-qsig
 isdn global-disconnect
 isdn overlap-receiving
 isdn incoming-voice voice
```

Related Commands

Command	Description
isdn protocol-emulate	Configures the interface to serve as either the QSIG slave or the QSIG master.
isdn switch-type	Configures the switch type for BRI or PRI support.

isdn i-number

To configure several terminal devices to use one subscriber line, use the **isdn i-number** command in interface configuration mode.

isdn i-number *n ldn*

Syntax Description

n	Subscriber line 1, 2, or 3, as specified in the NTT specification.
ldn	LDN assigned to the router plain old telephone service (POTS) port.

Defaults

Each terminal device uses one subscriber line.

Command Modes

Interface configuration

Command History

Release	Modification
12.1.(2)XF	This command was introduced on the Cisco 800 series routers.

Usage Guidelines

Enter the **interface bri** command before entering the **isdn i-number** command.

Examples

The following example shows screen output for two LDNs configured under BRI interface 0:

```
interface bri0
 isdn i-number 1 5551234
 isdn i-number 2 5556789
 exit
dial-peer voice 1 pots
 destination-pattern 5551234
 exit
dial-peer voice 2 pots
 destination-pattern 5556789
 exit
```

Related Commands

Command	Description
interface bri	Specifies a BRI interface and enters interface configuration mode.

isdn network-failure-cause

To specify the cause code to pass to the PBX when a call cannot be placed or completed because of internal network failures, use the **isdn network-failure-cause** command in interface configuration mode. To unconfigure the use of this cause code, use the **no** form of this command.

isdn network-failure-cause *value*

no isdn network-failure-cause *value*

Syntax Description

value	Number from 1 to 127. See Table 24 for a list of failure cause code values.

Defaults

No default behavior or values.

Command Modes

Interface configuration

Command History

Release	Modification
12.1(2)T	This command was introduced to the Cisco IOS 12.1(2)T on the Cisco 2600, 3600, and 7200 series routers and on the Cisco MC3810 multiservice concentrator.

Usage Guidelines

The PBX can reroute calls based on the cause code returned by the router.

This command allows the original cause code to be changed to the value specified if the original cause code is not one of the following:

- NORMAL_CLEARING (16)
- USER_BUSY (17)
- NO_USER_RESPONDING (18)
- NO_USER_ANSWER (19)
- NUMBER_CHANGED (22)
- INVALID_NUMBER_FORMAT (28)
- UNSPECIFIED_CAUSE (31)
- UNASSIGNED_NUMBER (1)

Table 24 describes the cause codes.

Table 24 ISDN Failure Cause Codes

Failure Cause Code	Meaning
1	Unallocated or unassigned number.
2	No route to specified transit network.
3	No route to destination.
6	Channel unacceptable.
7	Call awarded and being delivered in an established channel.
16	Normal call clearing.
17	User busy.
18	No user responding.
19	No answer from user (user alerted).
21	Call rejected.
22	Number changed.
26	Nonselected user clearing.
27	Destination out of order.
28	Invalid number format.
29	Facility rejected.
30	Response to status enquiry.
31	Normal, unspecified.
34	No circuit/channel available.
38	Network out of order.
41	Temporary failure.
42	Switch congestion.
43	Access information discarded.
44	Requested channel not available.
45	Preempted.
47	Resources unavailable, unspecified.
49	Quality of service unavailable.
50	Requested facility not subscribed.
52	Outgoing calls barred.
54	Incoming calls barred.
57	Bearer capability not authorized.
58	Bearer capability not available now.
63	Service or option not available, unspecified.
65	Bearer capability not implemented.
66	Channel type not implemented.
69	Requested facility not implemented.
70	Only restricted digital information bearer capability is available.

Table 24 ISDN Failure Cause Codes (continued)

Failure Cause Code	Meaning
79	Service or option not implemented, unspecified.
81	Invalid call reference value.
82	Identified channel does not exist.
83	Suspended call exists, but this call ID does not.
84	Call ID in use.
85	No call suspended.
86	Call with requested call ID is cleared.
88	Incompatible destination.
91	Invalid transit network selection.
95	Invalid message, unspecified.
96	Mandatory information element missing.
97	Message type nonexistent or not implemented.
98	Message not compatible with call state or message type nonexistent or not implemented.
99	Information element nonexistent or not implemented.
100	Invalid information element contents.
101	Message not compatible with call state.
102	Recovery on timer expiry.
111	Protocol error, unspecified.
127	Interworking, unspecified.

Examples

The following is an example of specifying a cause code to pass to a PBX when a call cannot be placed or completed of internal network failures:

```
isdn network-failure-cause 28
```

ivr autoload

To load files from a particular TFTP server (as indicated by a defined URL), use the **ivr autoload** command in global configuration mode. To disable this function, use the **no** form of this command.

ivr autoload url *location*

no ivr autoload url *location*

Syntax Description

url	Indicates that a URL is used to locate the index file that contains a list of all available audio files.
location	Specifies the URL of the index file.

Defaults

No URL is defined.

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)T	This command was introduced on the Cisco 2600 and 3600 series routers and the Cisco AS5300 universal access server.

Usage Guidelines

The index file contains a list of audio files (URL) that can be downloaded from the TFTP server. Use the **ivr autoload** command to download audio files from TFTP to memory. The command only starts up a background process. The background process (loader) does the actual down loading of the files.

The background process first reads the index file from either Flash or TFTP. It parses the files line by line looking for the URL. It ignores lines that start with # as comment lines. Once it has a correct URL, it tries to read that .au file into memory and creates a media object. If there are any errors during the reading of the file, it retries the configured number of times. If the mode is set to "verbose," the loader logs the transaction to console. Once parsing has reached the end of the index file, the background process exits memory.

Perform the following checks before initiating the background process. If one of the checks fail, it indicates the background process is not started, and instead you will see an error response to the command.

- Check if any prompt is being actively used (IVR is actively playing some prompts). If there are active prompts, the command fails, displaying the following error message (.au files are also referred to as prompts):

```
command is not allowed when prompts are active
```

- Check if there is already a background process in progress. If there is a process, the command fails, displaying the following error:

```
previous autoload command is still in progress
```

- Check if an earlier **ivr autoload** command has already been configured. If an **ivr autoload** command has already been configured, the user sees the following response when the command is issued:

```
previous command is being replaced
```

- When the **no ivr autoload** command is issued, if there was already an **ivr autoload** command in progress, it will be aborted.

The audio files (prompts) loaded using the **ivr autoload** command are not dynamically swapped out of memory. They are considered as autoloaded prompts as opposed to “dynamic” prompts. (See the **ivr prompt memory** command for details on dynamic prompts.)

Examples

The following example loads audio files from the TFTP server (located at //jurai/mgindi/tclware/index4):

```
ivr autoload url tftp://jurai/mgindi/tclware/index4
```

The index file for this example index4 is shown as follows:

```
more index4
tftp://jurai/mgindi/tclware/au/en/en_one.au
tftp://jurai/mgindi/tclware/au/ch/ch_one.au
tftp://jurai/mgindi/tclware/au/ch/ch_one.au
```

An example of an index file on Flash is shown as follows:

```
flash:index
```

Related Commands

Command	Description
ivr prompt memory	Configures the maximum amount of memory that the dynamic audio files (prompts) occupy in memory.

ivr autoload retry

To specify the number of times that the system will try to load audio files from TFTP to memory when there is an error, use the **ivr autoload retry** command in global configuration mode. To disable this function, use the **no** form of this command.

ivr autoload retry *number*

no ivr autoload retry *number*

Syntax Description

number	Number from 1 to 5. The default value is 3.

Defaults

Three times

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)T	This command was introduced on the Cisco 2600 and 3600 series routers and on the Cisco AS5300 universal access server.

Examples

The following example configures the system to try three times to load audio files:

```
ivr autoload retry 3
```

Related Commands

Command	Description
ivr prompt memory	Configures the maximum amount of memory that the dynamic audio files (prompts) occupy in memory.

ivr autoload mode

To load files from TFTP to memory using either verbose or silent mode, use the **ivr autoload mode** command in global configuration mode. To disable this function, use the **no** form of this command.

ivr autoload mode {**verbose url** *location* [**retry** *number*]} | {**silent url** *location* [**retry** *number*]}

no ivr autoload mode {**verbose url** *location* [**retry** *number*]} | {**silent url** *location* [**retry** *number*]}

Syntax Description

verbose	Displays the file transfer activity to the console. This mode is recommended while debugging.
url	Indicates that a URL is used to locate the index file that contains a list of all available audio files.
location	Specifies the URL of the index file.
retry	(Optional) Specifies the number of times the system will try to transfer a file when there are errors. This parameter applies to each file transfer.
number	(Optional) Number of times from 1 to 5. The default value is 3.
silent	Performs the file transfer in silent mode, meaning that no file transfer activity is displayed to the console.

Defaults

Silent mode

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)T	This command was introduced on the Cisco 2600 and 3600 series routers and on the Cisco AS5300 universal access server.

Usage Guidelines

The index file contains a list of audio files (URL) that can be downloaded from the TFTP server. Use the **ivr autoload** command to download audio files from TFTP to memory. The command only starts up a background process. The background process (loader) does the actual downloading of the files.

The background process first reads the index file from either Flash or TFTP. It parses the files line by line looking for the URL. It ignores lines that start with # as comment lines. Once it has a correct URL, it tries to read that.au file into memory and creates a media object. If there are any errors during the reading of the file, it retries the configured number of times. If the mode is set to **verbose**, the loader logs the transaction to console. Once parsing has reached the end of the index file, the background process exits memory.

Perform the following checks before initiating the background process. If one of the checks fails, it indicates the background process is not started, and instead you will see an error response to the command.

- Check if any prompt is being actively used (IVR is actively playing some prompts). If there are active prompts, the command fails, displaying the following error message (.au files are also referred to as prompts):

  ```
  command is not allowed when prompts are active
  ```

- Check if there is already a background process in progress. If there is a process, the command fails, displaying the following error:

  ```
  previous autoload command is still in progress
  ```

- Check if an earlier **ivr autoload** command has already been configured. If an **ivr autoload** command has already been configured, the user sees the following response when the command is issued:

  ```
  previous command is being replaced
  ```

- When the **no ivr autoload** command is issued, if there was already an **ivr autoload** command in progress, it will be aborted.

The audio files (prompts) loaded using the **ivr autoload** command are not dynamically swapped out of memory. They are considered as autoloaded prompts as opposed to "dynamic" prompts. (See the **ivr prompt memory** command for details on dynamic prompts.)

Examples

The following example configures verbose mode:

```
ivr autoload mode verbose url tftp://jurai/mgindi/tclware/index4 retry 3
```

The index file for the example index4 is shown as follows:

```
more index4
tftp://jurai/mgindi/tclware/au/en/en_one.au
tftp://jurai/mgindi/tclware/au/ch/ch_one.au
tftp://jurai/mgindi/tclware/au/ch/ch_one.au
```

The following is an example of index file on Flash:

```
flash:index
```

Related Commands

Command	Description
ivr prompt memory	Configures the maximum amount of memory that the dynamic audio files (prompts) occupy in memory.

ivr prompt memory

To configure the maximum amount of memory that the dynamic audio files (prompts) occupy in memory, use the **ivr prompt memory** command in global configuration mode. To disable the maximum memory size, use the **no** form of this command.

ivr prompt memory *size* **files** *number*

no ivr prompt memory *size* **files** *number*

Syntax Description

size	Specifies the maximum memory to be used by the free dynamic prompts, in kilobytes. Valid entries are from 128 to 16,384.
files *number*	Specifies the number of files that can stay in memory. Valid entries for the number argument are 50 to 1000.

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)T	This command was introduced on the Cisco 2600 and 3600 series routers and on the Cisco AS5300 universal access server.

Usage Guidelines

When both the *number* and *size* parameters are specified, the minimum memory out of the two will be used for memory calculations.

All the prompts that are not autoloaded or fixed are considered dynamic. Dynamic prompts are loaded in to memory from TFTP or Flash, as and when they are needed. When they are actively used for playing prompts, they are considered to be in “active” state. However, once the prompt playing is complete, these prompts are no longer active and are considered to be in “free” state.

The free prompts either stay in memory or are removed from memory depending on the availability of space in memory for these free prompts. The **prompt-mem** command essentially specifies a maximum memory to be used for these free prompts.

The free prompts are saved in memory and are queued in a waitQ. When the waitQ is full (either because the totally memory occupied by the free prompts exceeds the maximum configured value or the number of files in the waitQ exceeds maximum configured), oldest free prompts are removed from memory.

Examples

The following example shows how to use the **ivr prompt memory** command:

```
ivr prompt memory 2048 files 500
```

Related Commands

Command	Description
ivr autoload	Loads files from a particular TFTP server.
show call prompt-mem-usage	Displays the memory site use by prompts.

line-power

To configure the BRI port to supply line power to the terminal equipment (TE), use the **line-power** command in interface configuration mode. To disable the line power supply, use the **no** form of this command.

line-power

no line-power

Syntax Description

This command has no arguments or keywords.

Defaults

The BRI port does not supply line power.

Command Modes

Interface configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810 multiservice concentrator.
12.0(4)T	This command was integrated into the Cisco IOS Release 12.0(4)T.
12.1(3)XI	This command was added for the Cisco 2600 and Cisco 3600 series router.

Usage Guidelines

This command is supported only if an installed BRI voice module (BVM) or BRI VIC is equipped to supply line power (phantom power).

This command is used only on a BRI port that is operating in NT mode. A BRI port that is operating in TE mode is automatically disabled as a source of line power, and the **line-power** command is rejected.

When you use the **line-power** command, the line power provision is activated on a BRI port if the port is equipped with the hardware to supply line power. When you enter the **no line-power** command, the line power provision is deactivated on a BRI port.

Examples

The following example configures a BRI port to supply power to an attached TE device:

```
interface bri 1
 line-power
```

line-termination

To set the line termination on an E1 controller, use the **line-termination** command in controller configuration mode. To restore the default value, use the **no** form of this command.

line-termination { **75-ohm** | **120-ohm** }

no line-termination

Syntax Description

75-ohm	Matches the balanced BNC 75-ohm interface.
120-ohm	Matches the unbalanced twisted-pair 120-ohm interface.

Defaults

The default value is 120-ohm.

Command Modes

Controller configuration

Command History

Release	Modification
12.0(7)XR	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

This command applies to E1 controllers only.

Examples

The following example shows how to set controller E1 0/0 to a line-termination of 75-ohm:

```
controller e1 0/0
 line-termination 75-ohm
```

link (RLM)

To enable a Redundant Link Manager (RLM) link, use the **link** command in RLM configuration mode. To disable this function, use the **no** form of this command.

link {**hostname** *name* | **address** *ip-address*} **source** *loopback-source* **weight** *factor*

no link {**hostname** *name* | **address** *ip-address*} **source** *loopback-source* **weight** *factor*

Syntax Description

hostname *name*	RLM host name. If host name is used, RLM will look up the DNS server periodically for the host name configured until lookup is successful or the configuration is removed.
address *ip-address*	IP address of the link.
source *loopback-source*	Loopback interface source. We recommend you use the loopback interface as the source, so that it is independent of the hardware condition. Also, the source interface should be different in every link to avoid falling back to the same routing path. If you intend to use the same routing path for the failover, a single link is sufficient to implement it.
weight *factor*	An arbitrary number that sets link preference. The higher the weighting factor number assigned, the higher priority it gets to become the active link. If all entries have the same weighting factor assigned, all links will be treated equally. There is no preference among servers according to the assumption that only one server will accept the connection requests at any given time. Otherwise, preferences are extended across all servers.

Defaults

Disabled

Command Modes

RLM configuration

Command History

Release	Modification
11.3(7)	This command was introduced.

Usage Guidelines

This command is a preference-weighted multiple entries command. Within the same server, the link preference is specified in weighting.

Examples

The following example specifies the RLM group (network access server), the device name, and the link addresses and their weighting preferences:

```
rlm group 1
 server r1-server
  link address 10.1.4.1 source Loopback1 weight 4
  link address 10.1.4.2 source Loopback2 weight 3
```

Related Commands

Command	Description
clear interface	Resets the hardware logic on an interface.
clear rlm group	Clears all RLM group time stamps to zero.
interface	Defines the IP addresses of the server, configures an interface type, and enters interface configuration mode.
protocol rlm port	Reconfigures the port number for the basic RLM connection for the whole rlm-group.
retry keepalive	Allows consecutive keepalive failures a certain amount of time before the link is declared down.
server (RLM)	Defines the IP addresses of the server.
show rlm group statistics	Displays the network latency of the RLM group.
show rlm group status	Displays the status of the RLM group.
show rlm group timer	Displays the current RLM group timer values.
shutdown (RLM)	Shuts down all of the links under the RLM group.
timer	Overwrites the default setting of timeout values.

protocol rlm port

To configure the RLM port number, use the **protocol rlm port** RLM configuration command. To disable this function, use the **no** form of this command.

protocol rlm port *port-number*

no protocol rlm port *port-number*

Syntax Description

port-number	RLM port number. See Table 87 for the port number choices.

Defaults

3000

Command Modes

RLM configuration

Command History

Release	Modification
11.3(7)	This command was introduced.

Usage Guidelines

The port number for the basic RLM connection can be reconfigured for the entire RLM group. Table 87 lists the default RLM port numbers.

Table 25 Default RLM Port Number

Protocol	Port Number
RLM	3000
ISDN	Port[RLM]+1

Related Commands

Command	Description
clear interface	Resets the hardware logic on an interface.
clear rlm group	Clears all RLM group time stamps to zero.
interface	Defines the IP addresses of the server, configures an interface type, and enters interface configuration mode.
link (RLM)	Specifies the link preference.
retry keepalive	Allows consecutive keepalive failures a certain amount of time before the link is declared down.
server (RLM)	Defines the IP addresses of the server.
show rlm group statistics	Displays the network latency of the RLM group.
show rlm group status	Displays the status of the RLM group.
show rlm group timer	Displays the current RLM group timer values.
shutdown (RLM)	Shuts down all of the links under the RLM group.
timer	Overwrites the default setting of timeout values.

loopback

To set the loopback method for testing a T1 or E1 interface, use the **loopback** command in controller configuration mode. To restore the default value, use the **no** form of this command.

loopback {**diagnostic** | **local** {**payload** | **line**} | **remote** {**v54 channel-group** *channel-number* | **iboc** | **esf** {**payload** | **line**}}}

no loopback

Syntax Description

diagnostic	Loops the outgoing transmit signal back to the receive signal.
local	Places the interface into local loopback mode.
payload	Places the interface into external loopback mode at the payload level.
line	Places the interface into external loopback mode at the line level.
remote	Keeps the local end of the connection in remote loopback mode.
v54 channel-group	Activates a V.54 channel-group loopback at the remote end. Available for both T1 and E1 facilities.
channel-number	Specifies the channel number range (from 0 to 1) for the V.54 channel-group loopback.
iboc	Sends an in band bit oriented code to the far end to cause it to go into line loopback.
esf	Specifies Extended Super Frame (ESF) as the T1 or E1 frame type. Only available under T1 or E1 controllers when ESF is configured on the controller. The following are keywords: • **payload**—Activates remote payload loopback by sending Facility Data Link (FDL) code. FDL is a 4-kbps out-of-band signaling channel in ESF. • **line**—Activates remote line loopback by sending FDL code.

Defaults

No loopback is configured.

Command Modes

Controller configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced as a controller configuration command for the Cisco MC3810 multiservice concentrator.
12.0(5)T and 12.0(5)XK	The command was introduced as an ATM interface configuration command for the Cisco 2600 and 3600 series router.
12.0(5)XE	The command was introduced as an ATM interface configuration command for the Cisco 7200 and 7500 series.

Release	Modification
12.0(5)XK and 12.0(7)T	The command was introduced as a controller configuration command for the Cisco 2600 and 3600 series router.
12.1(1)T	The command was modified as a controller configuration command for the Cisco 2600 series.

Usage Guidelines

You can use a loopback test on lines to detect and distinguish equipment malfunctions caused either by the line and channel service unit/digital service unit (CSU/DSU) or by the interface. If correct data transmission is not possible when an interface is in loopback mode, the interface is the source of the problem.

Examples

The following example shows how to set the diagnostic loopback method on controller T1 0/0:

```
controller t1 0/0
 loopback diagnostic
```

The following example shows how to set the payload loopback method on controller E1 0/0:

```
controller e1 0/0
 loopback local payload
```

loop-detect

To enable loop detection for T1, use the **loop-detect** command in controller configuration mode. To cancel the loop detect operation, use the **no** form of this command.

loop-detect

no loop-detect

Syntax Description

This command has no arguments or keywords.

Defaults

Loop detection is disabled.

Command Modes

Controller configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810 multiservice concentrator.

Usage Guidelines

This command applies to Voice over Frame Relay and Voice over ATM on the Cisco MC3810 multiservice concentrator.

Examples

The following example configures loop detection for controller T1 0:

```
controller t1 0
 loop-detect
```

Related Commands

Command	Description
loopback (interface)	Diagnoses equipment malfunctions between an interface and a device.

loss-plan

To specify the analog-to-digital gain offset for an analog Foreign Exchange Office (FXO) or Foreign Exchange Station (FXS) voice port, use the **loss-plan** command in voice-port configuration mode. To restore the default value, use the **no** form of this command.

loss-plan {**plan1** | **plan2** | **plan3** | **plan4** | **plan5** | **plan6** | **plan7** | **plan8** | **plan9**}

no loss-plan

Syntax Description

plan1	FXO: A-D gain = 0 dB, D-A gain = 0 dB. FXS: A-D gain = –3 dB, D-A gain = –3 dB.
plan2	FXO: A-D gain = 3 dB, D-A gain = 0 dB. FXS: A-D gain = 0 dB, D-A gain = –3 dB.
plan3	FXO: A-D gain = –3 dB, D-A gain = 0 dB. FXS: Not applicable.
plan4	FXO: A-D gain = –3 dB, D-A gain = –3 dB. FXS: Not applicable.
plan5	FXO: Not applicable. FXS: A-D gain = –3 dB, D-A gain = –10 dB.
plan6	FXO: Not applicable. FXS: A-D gain = 0 dB, D-A gain = –7 dB.
plan7	FXO: A-D gain = 7 dB, D-A gain = 0 dB. FXS: A-D gain = 0 dB, D-A gain = –6 dB.
plan8	FXO: A-D gain = 5 dB, D-A gain = –2 dB. FXS: Not applicable.
plan9	FXO: A-D gain = 6 dB, D-A gain = 0 dB. FXS: Not applicable.

Defaults

FXO: A-D gain = 0 dB, D-A gain = 0 dB (loss plan 1)

FXS: A-D gain = –3 dB, D-A gain = –3 dB (loss plan 1)

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810 multiservice concentrator.
12.0(7)XK	The following additional signal level choices were added: plan 3, plan 4, plan 8, and plan 9.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

The **loss-plan** command sets the analog signal level difference (offset) between the analog voice port and the digital signal processor (DSP). Each loss plan specifies a level offset in both directions—from the analog voice port to the DSP (A-D) and from the DSP to the analog voice port (D-A).

Use this command to obtain the required levels of analog voice signals to and from the DSP.

The **loss-plan** command is supported only on Cisco MC3810 multiservice concentrators, on FXO and FXS analog voice ports.

Examples

The following example configures FXO voice port 1/6 for a –3 dB offset from the voice port to the DSP and for a 0 dB offset from the DSP to the voice port:

```
voice-port 1/6
 loss-plan plan3
```

The following example configures FXS voice port 1/1 for a 0 dB offset from the voice port to the DSP and for a –7 dB offset from the DSP to the voice port:

```
voice-port 1/1
 loss-plan plan6
```

Related Commands

Command	Description
impedance	Specifies the terminating impedance of a voice port interface.
input gain	Specifies the gain applied by a voice port to the input signal from the PBX or other customer premises equipment.
output attenuation	Specifies the attenuation applied by a voice port to the output signal toward the PBX or other customer premises equipment.

lrq forward-queries

To enable a gatekeeper to forward Location Requests (LRQs) that contain E.164 addresses that match zone prefixes controlled by remote gatekeepers, use the **lrq forward-queries** command in gatekeeper configuration mode. To disable this function, use the **no** form of this command.

lrq forward-queries

no lrq forward-queries

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Gatekeeper configuration

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco 2500, 3600 series routers and on the Cisco MC3810 multiservice concentrator.

Usage Guidelines

LRQ forwarding is dependent on a Cisco nonstandard field that first appeared in Cisco IOS Release 12.0(3)T. This means that any LRQ received from a non-Cisco gatekeeper or any gatekeeper running a Cisco IOS software image prior to Cisco IOS Release 12.0(3)T will not be forwarded.

The routing of E.164-addressed calls is dependent on the configuration of zone prefix tables (for example, area code definitions) on each gatekeeper. Each gatekeeper is configured with a list of prefixes controlled by itself and by other remote gatekeepers. Calls are routed to the zone that manages the matching prefix. Thus, in the absence of a directory service for such prefix tables, you, the network administrator, may have to define extensive lists of prefixes on all the gatekeepers in your administrative domain.

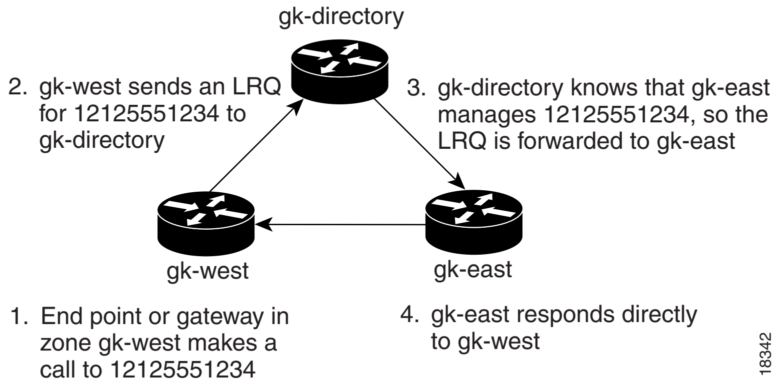
To simplify this task, you can select one of your gatekeepers as the “directory” gatekeeper and configure that gatekeeper with the complete list of prefixes and the **lrq forward-queries** command. You can then simply configure all the other gatekeepers with their own prefixes and the wildcard prefix “*” for your directory gatekeeper.

This command affects only the forwarding of LRQs for E.164 addresses. LRQs for H.323-ID addresses are never forwarded.

Examples

The following example shows how this command is used to simplify configuration by selecting one gatekeeper as the directory gatekeeper. Refer to Figure 5.

Figure 5 Example Scenario with Directory Gatekeeper and Two Remote Gatekeepers



Configuration on gk-directory

On the directory gatekeeper called gk-directory, identify all the prefixes for all the gatekeepers in your administrative domain:

```
zone local gk-directory cisco.com
zone remote gk-west cisco.com 172.16.1.1
zone remote gk-east cisco.com 172.16.2.1

zone prefix gk-west 1408.......
zone prefix gk-west 1415.......
zone prefix gk-west 1213.......
zone prefix gk-west 1650.......

zone prefix gk-east 1212.......
zone prefix gk-east 1617.......

lrq forward-queries
```

Configuration on gk-west

On the gatekeeper called gk-west, configure all the locally managed prefixes for that gatekeeper:

```
zone local gk-west cisco.com
zone remote gk-directory cisco.com 172.16.2.3

zone prefix gk-west 1408.......
zone prefix gk-west 1415.......
zone prefix gk-west 1213.......
zone prefix gk-west 1650.......
zone prefix gk-directory *
```

Configuration on gk-east

On the gatekeeper called gk-east, configure all the locally managed prefixes for that gatekeeper:

```
zone local gk-east cisco.com
zone remote gk-directory cisco.com 172.16.2.3

zone prefix gk-east 1212.......
zone prefix gk-east 1617.......
zone prefix gk-directory *
```

Now when an endpoint or gateway in zone gk-west makes a call to 12125551234, gk-west will send an LRQ for that E.164 address to gk-directory, which forwards the LRQ to gk-east. Gatekeeper gk-east responds directly to gk-west.

Related Commands

Command	Description
lrq reject-unknown-prefix	Enables the gatekeeper to reject all LRQs for zone prefixes that are not configured.

lrq reject-unknown-prefix

To enable the gatekeeper to reject all Location Requests (LRQs) for zone prefixes that are not configured, use the **lrq reject-unknown-prefix** command in gatekeeper configuration mode. To reenable the gatekeeper to accept and process all incoming LRQs, use the **no** form of this command.

lrq reject-unknown-prefix

no lrq reject-unknown-prefix

Syntax Description

This command has no arguments or keywords.

Defaults

The gatekeeper accepts and processes all incoming LRQs.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(6)NA2	This command was introduced on the Cisco 2500 and 3600 series routers.
12.0(3)T	This command was integrated into Cisco IOS Release 12.0(3)T.

Usage Guidelines

Use the **lrq reject-unknown-prefix** command to configure the gatekeeper to reject any incoming LRQs for a destination E.164 address that does not match any of the configured zone prefixes.

Whether or not you enable the **lrq reject-unknown-prefix** command, the following is true when the E.164 address matches a zone prefix:

- If the matching zone prefix is local (that is, controlled by this gatekeeper), the LRQ is serviced.
- If the matching zone prefix is remote (that is, controlled by some other gatekeeper), the LRQ is rejected.

If you do not enable the **lrq reject-unknown-prefix** command and the target address does not match any known local or remote prefix, the default behavior is to attempt to service the call using one of the local zones. If this default behavior is not suitable for your site, configure the **lrq reject-unknown-prefix** command on your router to force the gatekeeper to reject such requests.

Examples

Consider the following gatekeeper configuration:

```
zone local gk408 cisco.com
zone local gk415 cisco.com
zone prefix gk408 1408.......
zone prefix gk415 1415.......
lrq reject-unknown-prefix
```

In this sample configuration, the gatekeeper is configured to manage two zones. One zone contains gateways with interfaces in the 408 area code, and the second zone contains gateways in the 415 area code. Then using the **zone prefix** command, the gatekeeper is configured with the appropriate prefixes so that calls to those area codes hop off in the optimal zone.

Now say some other zone has been erroneously configured to route calls to the 212 area code to this gatekeeper. When the LRQ for a number in the 212 area code arrives at this gatekeeper, the gatekeeper fails to match the area code, and the LRQ is rejected.

If this was your only site that had any gateways in it and you wanted your other sites to route all calls that require gateways to this gatekeeper, you can undo the **lrq reject-unknown-prefix** command by simply using the **no lrq reject-unknown-prefix** command.Now when the gatekeeper receives an LRQ for the address 12125551234, it will attempt to find an appropriate gateway in either one of the zones gk408 or gk415 to service the call.

Related Commands

Command	Description
lrq forward-queries	Enables a gatekeeper to forward LRQs that contain E.164 addresses that match zone prefixes controlled by remote gatekeepers.

lrq timeout blast window

To configure the timeout window for use when sending multiple Location Requests (LRQs) (either sequentially or simultaneously), use the **lrq timeout blast window** command in gatekeeper configuration mode. To return to the default value, use the **no** form of this command.

lrq timeout blast window *seconds*

no lrq timeout blast window

Syntax Description

seconds	The duration of the window, in seconds. Possible values are 1 through 10. The default is 6.

Defaults

By default, the duration of the window is 6 seconds.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
12.1(2)T	This command was introduced on the Cisco 2500, 2600, 3600, and 7200 series routers and on the Cisco MC3810 multiservice concentrator.

Examples

The following example sets the window to 3 seconds:

```
lrq timeout blast window 3
```

Related Commands

Command	Description
gatekeeper gw-type-prefix	Sets the gatekeepers responsible for each technology prefix.
zone prefix	Adds a prefix to a gatekeeper’s zone list.

lrq timeout seq delay

To configure the delay for use when sending Location Requests (LRQs) sequentially, use the **lrq timeout seq delay** command in gatekeeper configuration mode. To return to the default value, use this **no** form of the command.

lrq timeout seq delay *value*

no lrq timeout seq delay

Syntax Description

value	The duration of the delay, in 100 millisecond units. Possible values are 1 through 10. The default is 5 (500 ms or 0.5 seconds).

Defaults

By default, the duration of the window is five 100 millisecond units (500 ms or 0.5 seconds)

Command Modes

Gatekeeper configuration

Command History

Release	Modification
12.1(2)T	This command was introduced on the Cisco 2500, 2600, 3600, and 7200 series routers, and on the Cisco MC3810 multiservice concentrator.

Examples

The following example sets the window to 300 milliseconds:

```
lrq timeout seq delay 3
```

Related Commands

Command	Description
gatekeeper gw-type-prefix	Sets the gatekeepers responsible for each technology prefix.
zone prefix	Adds a prefix to a gatekeeper's zone list.

max-conn

To specify the maximum number of allowed connections for a particular Voice over IP (VoIP) or POTS dial peer, use the **max-conn** command in dial-peer configuration mode. To set an unlimited number of connections for this dial peer, use the **no** form of this command.

max-conn *number*

no max-conn

Syntax Description

number	Specifies the maximum number of connections for this dial peer. Valid values for this field are 1 to 2,147,483,647.

Defaults

The **no** form of this command is the default, meaning an unlimited number of connections.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series roter.
12.0(4)XJ	This command was modified for store and forward fax on the Cisco AS5300 universal access server.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

This command applies to both VoIP and POTS dial peers. Use the **max-conn** command to define the maximum number of connections used simultaneously on the Cisco AS5300 universal access server to send fax-mail.

This command applies to off-ramp store and forward fax functions.

Examples

The following example configures the maximum number of connections for VoIP dial peer 10 as 5:

```
dial-peer voice 10 voip
 max-conn 5
```

Related Commands

Command	Description
mta receive maximum-recipients	Specifies the maximum recipients for all SMTP connections.

max-connection

To set the maximum number of simultaneous connections to be used for communication with a settlement provider, use the **max-connection** command in settlement configuration mode. To reset to the default value of this command, use the **no** form of this command.

max-connection *number*

no max-connection *number*

Syntax Description

number	Specifies the maximum number of HTTP connections to a settlement provider.

Defaults

The default is 10 connections.

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced on the Cisco 2600 and Cisco 3600 series routers and on the Cisco AS5300 universal access server.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Examples

The following command sets the maximum number of simultaneous connections to 10:

```
settlement 0
 max-connection 10
```

Related Commands

Command	Description
connection-timeout	Configures the time that a connection is maintained after completing a communication exchange.
customer-id	Sets the customer identification.
device-id	Specifies a gateway associated with a settlement provider.
encryption	Sets the encryption method to be negotiated with the provider.
response-timeout	Configures the maximum time to wait for a response from a server.
retry-delay	Sets the time between attempts to connect with the settlement provider.
retry-limit	Sets the maximum number of connection attempts to the provider.
session-timeout	Sets the interval for closing the connection when there is no input or output traffic.
settlement	Enters settlement configuration mode and specifies the attributes specific to a settlement provider.

Command	Description
shutdown	Brings up the settlement provider.
type	Configures an SAA-RTR operation type.
url	Configures the ISP address.

max-forwards

To set the maximum number of proxy or redirect servers that can forward the request, use the **max-forwards** command in the SIP user agent configuration mode. To reset the default value, use the **no** form of this command.

max-forwards *number*

no max-forwards

Syntax Description

number	Number of hops. Possible values are 1 through 15. The default is 6.

Defaults

The default number of hops is 6.

Command Modes

SIP user agent configuration

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco 2600 and Cisco 3600 series routers and on the Cisco AS5300 universal access server.

Usage Guidelines

To reset this command to the default value, you can also use the **default** command.

Examples

The following is an example of forwarding requests to proxy or redirect servers:

```
sip-ua
 max-forwards 2
```

max-redirects

To set the maximum number of redirect servers that the user agent allows, use the **max-redirects** command in dial-peer configuration mode. To reset the default value, use the **no** form of this command.

max-redirects *number*

no max-redirects

Syntax Description

number	Maximum number of redirect servers that a call can traverse. Possible values are 1 through 10.

Defaults

The default number of redirects is 1.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.1(1)T	This command was introduced on the Cisco 2600 and Cisco 3600 series routers and on the Cisco AS5300 universal access server.

Examples

The following is an example of setting the maximum number of redirect servers that the user agent allows:

```
dial-peer voice 102 voip
 max-redirects 2
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice-related encapsulation.

mdn

To request that a message disposition notice (MDN) be generated when the message is processed ("opened"), use the **mdn** command in dial-peer configuration mode. To restore the default value, use the **no** form of this command.

mdn

no mdn

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access router.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

Message disposition notification is an e-mail message that is generated and sent to the sender when the message is opened by the receiver. Use the **mdn** command to request that an e-mail response message be sent to the sender when the e-mail that contains the fax TIFF image has been opened.

This command applies to on-ramp store and forward fax functions.

Examples

The following example requests that a message disposition notice be generated by the recipient:

```
dial-peer voice 10 mmoip
 mdn
```

Related Commands

Command	Description
mta receive generate-mdn	Specifies that the off-ramp gateway process a response MDN from an SMTP server.
mta send return-receipt-to	Specifies the address where MDNs are sent.

mgcp

To allocate resources for the media gateway control protocol (MGCP) and start the MGCP daemon, use the **mgcp** command in global configuration mode. To terminate all calls, release all allocated resources, and stop the MGCP daemon, use the **no** form of this command.

mgcp [*port*]

no mgcp

Syntax Description

port	(Optional) Specifies a UDP port for the MGCP gateway. Valid values are 1025 through 65,535. If no port is specified, the command defaults to UDP port 2427.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
12.1(1)T	This command was introduced for the Cisco AS5300 universal access server.
12.1(3)T	Support for this command was extended to the Cisco 3660, Cisco uBR924, and Cisco 2600 series router router platforms.

Usage Guidelines

Once you start the MGCP daemon using the **mgcp** command, you can suspend it (for example, for maintenance) by using the **mgcp block-newcalls** command. When you are ready to resume normal MGCP operations, use the **no mgcp block-newcalls** command. Use the **no mgcp** command only if you intend to terminate all MGCP applications and protocols.

When the MGCP daemon is not active, all MGCP messages are ignored.

Examples

The following example shows how to initiate the MGCP daemon:

```
mgcp
```

Related Commands

Command	Description
debug mgcp	Enables debugging on MGCP.
mgcp ip-tos	Terminates all MGCP activity in an orderly manner.
mgcp request retries	Specifies the number of times to retry sending the **mgcp** command.

mgcp block-newcalls

To block new calls while maintaining existing calls, use the **mgcp block-newcalls** command in global configuration mode. To resume media gateway control protocol (MGCP) operation, use the **no** form of this command.

mgcp block-newcalls

no mgcp block-newcalls

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
12.1(1)T	This command was introduced for the Cisco AS5300 universal access server.
12.1(3)T	Support for this command was extended to the Cisco 3660, Cisco uBR924, and Cisco 2600 series router router platforms.

Usage Guidelines

This command is valid only if the **mgcp** command is enabled.

Once you issue this command, all requests for new connections (CreateConnection requests) are denied. All existing calls will be maintained until participants terminate them or you use the **no mgcp** command. When the last active call is terminated, the MGCP daemon will be terminated and all resources allocated to it will be released. The **no mgcp block-newcalls** command returns the router to normal MGCP operations.

Examples

The following example shows how to prevent the gateway from receiving new calls:

```
mgcp block-newcalls
```

Related Commands

Command	Description
mgcp	Allocates resources for the MGCP and starts the daemon.

mgcp call-agent

To configure the call agent (media gateway controller) address, use the **mgcp call-agent** command in global configuration mode. To unconfigure the call agent address, use the **no** form of this command.

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

no mgcp call-agent

Syntax Description

ip-address | *host-name*	Specifies the IP address or domain name of the call agent.
port	(Optional) Specifies the User Datagram Protocol (UDP) port for the call agent to use. Valid values are 1025 through 65,535. If a port is not specified, the default is UDP 2427.
service-type *type*	(Optional) Specifies the type of gateway control service to be supported by the call agent. Valid values are **mgcp** and **sgcp**.

Defaults

MGCP service-type

Command Modes

Global configuration

Command History

Release	Modification
12.1(1)T	This command was introduced for the Cisco AS5300 universal access server.
12.1(3)T	The **service-type** *type* option was added.

Usage Guidelines

Use this command on any platform and media gateway.

If you do not specify a UDP port from the command line, media gateway control protocol (MGCP) will use 2427 as the default call agent UDP port.

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

Examples

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

```
mgcp call-agent 209.165.200.225 service-type sgcp
mgcp call-agent 209.165.200.225 5530 service-type mgcp
mgcp call-agent igloo service-type sgcp
mgcp call-agent igloo 2009 service-type mgcp
```

Related Commands

Command	Description
mgcp	Starts the MGCP daemon.

mgcp codec

To select the default codec type and its optional packetization period value, use the **mgcp codec** command in global configuration mode. To set the parameters to their default values, use the **no** form of this command.

mgcp codec *type* [**packetization-period** *value*]

no mgcp codec

Syntax Description

type	Specifies the types of codec supported. Valid codecs are G711alaw, G711ulaw, G723ar53, G723ar63, G723r53, G723r63, G729ar8, G729br8, and G729r8.
packetization-period *value*	(Optional) This parameter is useful when the preferred compression algorithm and packetization period parameter is not provided by the Media Gateway Controller. The value range depends on the type of codec selected. For example, the range for G729r8 is 10 to 220 in increments of 10. For G711ulaw, the range is 10 to 20 in increments of 10. For G723ur53, the range is 30 to 330 in increments of 10.

Defaults

G711ulaw

Command Modes

Global configuration

Command History

Release	Modification
12.1(1)T	This command was introduced for the Cisco AS5300 universal access server.
12.1(3)T	Support for this command was extended to the Cisco 3660, Cisco uBR924, and Cisco 2600 series router router platforms.

Examples

This example shows how to specify the default codec type:

```
mgcp codec g711alaw
```

The following example specifies the codec type and sets the packetization period:

```
mgcp codec g729r8 packetization-period 150
```

Related Commands

Command	Description
mgcp	Starts the MGCP daemon.

mgcp default-package

To configure the default package capability type for the media gateway, use the **mgcp default-package** command in global configuration mode. This command does not support a **no** form. To change the default package, use the **mgcp default-package** command with a different, actively supported package.

Residential Gateways

mgcp default-package {**line-package** | **dtmf-package** | **gm-package**}

Trunking Gateways

mgcp default-package {**as-package** | **dtmf-package** | **gm-package** | **rtp-package** | **trunk-package**}

Syntax Description

as-package	Announcement server package.
dtmf-package	DTMF package.
gm-package	Generic media package.
line-package	Line package.
rtp-package	RTP package.
trunk-package	Trunk package.

Defaults

For residential gateways (RGWs): **line-package**

For trunking gateways (TGWs): **trunk-package**

Command Modes

Global configuration

Command History

Release	Modification
12.1(1)T	This command was introduced for the Cisco AS5300 universal access server.
12.1(3)T	The **line-package** keyword and a distinction between residential and trunking gateways were added.

Usage Guidelines

This command is helpful when the Media Gateway Controller does not provide the package capability to be used for the given connection.

Before selecting a package as the default, use the **show mgcp** command to ensure that the package is actively supported. If the package you want does not appear in the display, use the **mgcp package-capability** command to add the package to the supported list.

If only one package is actively supported, it becomes the default package.

Examples

The following example shows how to set the default package:

```
mgcp default-package as-package
! The announcement server package type will be the new default package type.
```

Related Commands

Command	Description
mgcp	Starts the MGCP daemon.
mgcp package-capability	Includes a specific MGCP package that is supported by the gateway.

mgcp dtmf-relay

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

mgcp dtmf-relay {**codec** | **low-bit-rate**} **mode** {**cisco** | **out-of-band**}

no mgcp dtmf-relay

Syntax Description

codec	Specifies use of either a G.711 or a G.726 codec.
low-bit-rate	Specifies a low-bit-rate codec other than G.711 and G.726.
mode	Specifies the mode.
cisco	This mode removes the DTMF tone from the voice stream and sends FRF.11 with a special payload 121 for the DTMF digits.
out-of-band	This mode removes the DTMF tone from the voice stream and does not send FRF.11.

Defaults

Non compressed codecs are disabled.

Command Modes

Controller configuration

Command History

Release	Modification
12.1(3)T	This command was first supported by media gateway control protocol (MGCP) on the Cisco 2600 and Cisco 3600 series routers, the Cisco AS5300 universal access server, and the Cisco uBR924 cable access router.

Usage Guidelines

Use this command to access an announcement server or a voice mail server that does not have the capability to decode RTP packets containing DTMF digits. When the **mgcp dtmf-relay** command is active, the DTMF digits are removed from the voice stream and carried by FRF.11 so that the server can decode the digits.

Examples

The following example shows how to remove the DTMF tone from the voice stream and send FRF.11 with a special payload for the DTMF digits:

```
mgcp dtmf-relay codec mode cisco
```

Related Commands

Command	Description
mgcp	Allocates resources of the MGCP and starts the daemon.

mgcp ip-tos

To enable or disable the IP type of service (ToS) for MGCP-controlled connections, use the **mgcp ip-tos** command in global configuration mode. To set the parameters to their default values, use the **no** form of this command.

mgcp ip-tos {**high-reliability** | **high-throughput** | **low-cost** | **low-delay** | **precedence** *value*}

no mgcp ip-tos {**high-reliability** | **high-throughput** | **low-cost** | **low-delay** | **precedence** *value*}

Syntax Description

high-reliability	Specifies high-reliability ToS.
high-throughput	Specifies high-throughput ToS.
low-cost	Specifies low-cost ToS.
low-delay	Specifies low-delay ToS.
precedence *value*	Specifies the value of the IP precedence bit. Valid values are from 0 to 7. The default IP precedence value is 3.

Defaults

Services disabled; precedence is 3.

Command Modes

Global configuration

Command History

Release	Modification
12.1(1)T	This command was introduced for the Cisco AS5300 universal access server.
12.1(3)T	Support for this command was extended to the Cisco 3660, Cisco uBR924, and Cisco 2600 series router router platforms.

Usage Guidelines

Only one of the high-reliability, high-throughput, low-cost, or low-delay parameters can be enabled at any given time. Enabling one parameter disables any other that was active. Enabling one parameter has no effect on the precedence parameter. When you configure a new value for the precedence parameter, the old value is erased.

The **no** form of the **mgcp ip-tos** command disables the first four parameters and sets the precedence parameter back to 3.

Examples

In the following example, activating the **low-delay** keyword disables the other three parameters.

```
mgcp ip-tos high-reliability
mgcp ip-tos high-throughput
mgcp ip-tos low-cost
mgcp ip-tos low-delay
mgcp ip-tos precedence 4
```

Related Commands

Command	Description
mgcp	Starts the MGCP daemon.

mgcp max-waiting-delay

To specify the media gateway control protocol (MGCP) maximum waiting delay (MWD), use the **mgcp max-waiting-delay** command in global configuration mode. To restore the default value, use the **no** form of this command.

mgcp max-waiting-delay *milliseconds*

no mgcp max-waiting-delay

Syntax Description

milliseconds	The number of milliseconds to wait after restart. The valid range is 0 to 600,000 milliseconds (600 seconds).

Defaults

3000 milliseconds

Command Modes

Global configuration

Command History

Release	Modification
12.1(1)T	This command was introduced for the Cisco AS5300 universal access server.
12.1(3)T	Support for this command was extended to the Cisco 3660, Cisco uBR924, and Cisco 2600 series router router platforms.

Usage Guidelines

Use the maximum waiting delay to send out an RSIP message to the call agent with the restart method. This command helps prevent traffic bottlenecks caused by MGCP gateways all trying to connect at the same time after a restart.

Examples

The following example shows how to set the MGCP maximum waiting delay to 600 milliseconds:

```
mgcp max-waiting-delay 600
```

Related Commands

Command	Description
mgcp	Starts the MGCP daemon.
mgcp restart-delay	Configures the graceful tear down method sent in the RSIP message.

mgcp modem passthru

To enable the gateway to send and receive modem and fax data, use the **mgcp modem passthru** command in controller configuration mode. To disable support for modem and fax data, use the **no** form of this command.

mgcp modem passthru {**cisco** | **ca**}

no mgcp modem passthru

Syntax Description

cisco	When the gateway detects a modem/fax tone, the gateway switches the codec to G.711 to allow the analog data to pass through.
ca	When the gateway detects a modem/fax tone, the gateway alerts the call agent to switch the codec to G.711 to allow the analog data to pass through.

Defaults

ca

Command Modes

Controller configuration

Command History

Release	Modification
12.1(3)T	This command was added to MGCP.

Usage Guidelines

When the **cisco** keyword is activated and the gateway detects a modem/fax tone, the gateway switches the codec to G.711 then sends the analog data to a remote gateway. The remote gateway also switches the codec on its side of the call to G.711 to allow the analog data to pass through.

When the **ca** keyword is activated and the gateway detects a modem/fax tone, the gateway alerts the call agent to switch the codec to G.711 to allow the analog data to pass through. The call agent must send an MDCX signal to the G.711 codec for successful data pass-through.

Examples

The following example configures a gateway to send and receive modem or fax data:

```
mgcp modem passthru cisco
```

Related Commands

Command	Description
mgcp	Starts the MGCP daemon.

mgcp package-capability

To specify a media gateway control protocol (MGCP) package capability for a gateway, use the **mgcp package-capability** command in global configuration mode. To remove a specific MGCP package capability from the list of capabilities, use the **no** form of this command.

All Residential Gateways

mgcp package-capability {**line-package** | **dtmf-package** | **gm-package** | **rtp-package**}

no mgcp package-capability {**line-package** | **dtmf-package** | **gm-package** | **rtp-package**}

Cisco AS5300 Universal Access Server

mgcp package-capability {**trunk-package** | **dtmf-package** | **gm-package** | **rtp-package** | **as-package** | **script-package**}

no mgcp package-capability {**trunk-package** | **dtmf-package** | **gm-package** | **rtp-package** | **as-package** | **script-package**}

All Other Trunking Gateways

mgcp package-capability {**trunk-package** | **dtmf-package** | **gm-package** | **rtp-package** | **as-package**}

no mgcp package-capability {**trunk-package** | **dtmf-package** | **gm-package** | **rtp-package** | **as-package**}

Syntax Description

line-package	Line package.
trunk-package	Trunk package.
dtmf-package	DTMF package.
gm-package	Generic media package.
rtp-package	RTP package.
as-package	Announcement server package.
script-package	Script package.

Defaults

For all residential gateways (RGWs): **line-package**

For all trunking gateways (TGWs): **trunk-package**

Command Modes

Global configuration

Command History

Release	Modification
12.1(1)T	This command was introduced for the Cisco AS5300 universal access server.
12.1(3)T	The command was expanded to the Cisco uBR924, Cisco 2600 series router router, and Cisco 3660 platforms. The **line-package**, **rtp-package**, and **script-package** keywords were added.

Usage Guidelines

Select packages that are supported by your call agent. Events specified in the MGCP messages from the call agent must belong to one of the supported packages. Otherwise, the connection requests are refused by the media gateway.

Use this command before specifying a default package using the **mgcp default-package** command. Specify at least one package to have a default.

Enter each package as a separate command.

Examples

The following example shows how to specify an MGCP package capability for a gateway:

```
mgcp package-capability trunk-package
mgcp package-capability dtmf-package
mgcp package-capability script-package
mgcp default-package trunk-package
```

Related Commands

Command	Description
mgcp	Starts the MGCP daemon.
mgcp default-package	Configures the default package capability type for the media gateway.

mgcp playout

To tune the jitter buffer packet size attempted for MGCP-controlled connections, use the **mgcp playout** command in global configuration mode. To restore the default value, use the **no** form of this command.

mgcp playout {**adaptive** *init-value min-value max-value* | **fixed** *init-value*}

no mgcp playout {**adaptive** | **fixed**}

Syntax Description

adaptive *init-value min-value max-value*	Specifies a user-defined variable range (in milliseconds) for the jitter buffer packet size. The range for each value is 4 to 250. The default values are: *init-value* 60, *min-value* 4, and *max-value* 200. Note that *init-value* must be between *min-value* and *max-value*.
fixed *init-value*	Specifies a fixed size (in milliseconds) for the jitter buffer packet size. Valid values are from 4 to 250.

Defaults

adaptive 60 4 200

No default for **fixed**.

Command Modes

Global configuration

Command History

Release	Modification
12.1(1)T	This command was introduced for the Cisco AS5300 universal access server.
12.1(3)T	Support for this command was extended to the Cisco 3660, Cisco uBR924, and Cisco 2600 series router router platforms.

Examples

The following example illustrates a jitter buffer configuration with an initial playout of 100, a minimum buffer size of 50, and a maximum buffer size of 150:

```
mgcp playout adaptive 100 50 150
```

The following example illustrates setting the jitter buffer to a fixed playout of 120:

```
mgcp playout fixed 120
```

Related Commands

Command	Description
mgcp	Starts the MGCP daemon.

mgcp quality-threshold

To set the jitter buffer size threshold, latency threshold, and packet-loss threshold parameters, use the **mgcp quality-threshold** command in global configuration mode. To restore the default parameter values, use the **no** form of this command.

mgcp quality-threshold {**hwm-jitter-buffer** *value* | **hwm-latency** *value* | **hwm-packet-loss** *value* | **lwm-jitter-buffer** *value* | **lwm-latency** *value* | **lwm-packet-loss** *value*}

no mgcp quality-threshold {**hwm-jitter-buffer** | **hwm-latency** | **hwm-packet-loss** | **lwm-jitter-buffer** | **lwm-latency** | **lwm-packet-loss**}

Syntax Description

hwm-jitter-buffer *value*	Specifies the high-water-mark jitter buffer size. Valid range is from 100 to 200 the default value is 150 milliseconds.
hwm-latency *value*	Specifies the high-water-mark latency value. Valid range is from 250 to 400 milliseconds, and the default value is 300 milliseconds.
hwm-packet-loss *value*	Specifies the high-water-mark packet-loss value. Valid range is from 5000 to 25,000 milliseconds, and the default value is 10000 milliseconds.
lwm-jitter-buffer *value*	Specifies the low-water-mark jitter buffer size. Valid range is from 4 to 60 milliseconds, and the default value is 30 milliseconds.
lwm-latency *value*	Specifies the low-water-mark latency value. Valid range is from 125 to 200 milliseconds, and the default value is 150 milliseconds.
lwm-packet-loss *value*	Specifies the low-water-mark packet-loss value. Valid range is from 1 to 3000 milliseconds, and the default value is 1000 milliseconds.

Defaults

The voice quality parameter defaults are (in milliseconds): **hwm-jitter-buffer** 150, **hwm-latency** 300, **hwm-packet-loss** 10,000, **lwm-jitter-buffer** 30, **lwm-latency** 150, and **lwm-packet-loss** 1000.

Command Modes

Global configuration

Command History

Release	Modification
12.1(1)T	This command was introduced for the Cisco AS5300 universal access server.
12.1(3)T	Support for this command was extended to the Cisco 3660, Cisco uBR924, and Cisco 2600 series router router platforms.

Usage Guidelines

The following parameters impact the quality of voice calls:

- Jitter buffer (storage area containing active call voice packets that have been received from the network and are waiting to be decoded and played)
- Latency (network delay in sending/receiving packets)
- Packet loss (number of packets lost per 100,000 packets for a given call)

For good voice quality, the system should perform below the **lwm** values. As the values go higher, voice quality degrades. The system generates a report when the values go above the **hwm** levels. Set the **hwm** and **lwm** values sufficiently apart so that you receive reports on poor performance, but not so close together that you receive too much feedback.

Enter each parameter as a separate command.

Examples

The following example shows how the different parameters can be set to new values:

```
mgcp quality-threshold hwm-jitter-buffer 100
mgcp quality-threshold hwm-latency 250
mgcp quality-threshold hwm-packet-loss 5000
mgcp quality-threshold lwm-jitter-buffer 50
mgcp quality-threshold lwm-latency 200
mgcp quality-threshold lwm-packet-loss 20
```

Related Commands

Command	Description
mgcp	Starts the MGCP daemon.
mgcp playout	Tunes the jitter buffer packet size.

mgcp request retries

To specify the number of times to retry sending the **mgcp** command, use the **mgcp request retries** command in global configuration mode. To restore the default value, use the **no** form of this command.

mgcp request retries *count*

no mgcp request retries

Syntax Description

count	Specifies the number of times a Notify message is resent to the Call Agent before the request is dropped. The valid range is 1 to 10.

Defaults

Three times

Command Modes

Global configuration

Command History

Release	Modification
12.1(1)T	This command was introduced for the Cisco AS5300 universal access server.
12.1(3)T	Support for this command was extended to the Cisco 3660, Cisco uBR924, and Cisco 2600 series router router platforms.

Usage Guidelines

This command applies to a trunking gateway.

Examples

The following example shows that the system will try to send the **mgcp** command 10 times before dropping the request:

```
mgcp request retries 10
```

Related Commands

Command	Description
mgcp	Starts the MGCP daemon.
mgcp request timeout	Specifies how long the system waits for a response to a request.

mgcp request timeout

To specify how long the system waits for a response to a request, use the **mgcp request timeout** command in global configuration mode. To restore the default value, use the **no** form of this command.

mgcp request timeout *timeout*

no mgcp request timeout

Syntax Description

timeout	Specifies the number of milliseconds to wait for a response to a request. The valid range is 1 to 10,000 (10 seconds).

Defaults

500 milliseconds

Command Modes

Global configuration

Command History

Release	Modification
12.1(1)T	This command was introduced for the Cisco AS5300 universal access server.
12.1(3)T	Support for this command was extended to the Cisco 3660, Cisco uBR924, and Cisco 2600 series router router platforms.

Examples

The following example configures the system to wait 40 milliseconds for a reply to a request:

```
mgcp request timeout 40
```

Related Commands

Command	Description
mgcp	Starts the MGCP daemon.
mgcp request retries	Specifies the number of times to retry sending the **mgcp** command.

mgcp restart-delay

To select the delay value sent in the Restart in Progress (RSIP) graceful tear down, use the **mgcp restart-delay** command in global configuration mode. To restore the default value, use the **no** form of this command.

mgcp restart-delay *seconds*

no mgcp restart-delay

Syntax Description

seconds	Specifies the restart delay value in seconds. The valid range is from 0 to 600.

Defaults

Zero (0) seconds

Command Modes

Global configuration

Command History

Release	Modification
12.1(1)T	This command was introduced for the Cisco AS5300 universal access server.
12.1(3)T	Support for this command was extended to the Cisco 3660, Cisco uBR924, and Cisco 2600 series router router platforms.

Usage Guidelines

Use the restart value to send a RSIP message that indicates when the connection in the gateway will be torn down.

Examples

The following example sets the restart delay to 30 seconds:

```
mgcp restart-delay 30
```

Related Commands

Command	Description
mgcp	Starts the MGCP daemon.
mgcp max-waiting-delay	Specifies the MGCP MWD after a restart.

mgcp sdp simple

To initiate a subset of the SDP protocol, use the **mgcp sdp simple** command in controller configuration mode. To return to the full set of SDP protocol fields, use the **no** form of this command.

mgcp sdp simple

no mgcp sdp simple

Syntax Description

This command has no arguments or keywords.

Defaults

no mgcp sdp simple

Command Modes

Controller configuration

Command History

Release	Modification
12.1(3)T	This command was added to the media gateway control protocol (MGCP).

Usage Guidelines

When the **mgcp sdp simple** command is enabled, the gateway will not generate three SDP fields: time, session name, and other (username, session id, sdp version, network type, address type, or address). Certain call agents require this modified SDP protocol to send data through the network.

Examples

The following example configures the modified SDP protocol:

```
mgcp sdp simple
```

Related Commands

Command	Description
mgcp	Starts the MGCP daemon.

mgcp vad

To set the default VAD parameter for the media gateway control protocol (MGCP), use the **mgcp vad** command in global configuration mode. To disable the VAD parameter, use the **no** form of this command.

mgcp vad

no mgcp vad

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.1(1)T	This command was introduced for the Cisco AS5300 universal access server.
12.1(3)T	Support for this command was extended to the Cisco 3660, Cisco uBR924, and Cisco 2600 series router router platforms.

Usage Guidelines

Use the MGCP VAD parameter to tell the MGCP gateway to turn silence suppression on or off.

Examples

The following example turns silence suppression on:

```
mgcp vad
```

Related Commands

Command	Description
mgcp	Starts the MGCP daemon.

microcode reload controller

To reload the firmware and FPGA from the command-line interface (CLI) without reloading the Cisco IOS image, use the **microcode reload controller** command in privileged EXEC mode.

microcode reload controller {**t1** | **e1**} {*x*/*y*}

Syntax Description

t1	Specifies T1.
e1	Specifies E1.
x/*y*	Controller slot and unit numbers.

Defaults

No microcode reload activity is initiated.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(2)XH	This command was introduced on the Cisco 2600 and Cisco 3600 series routers.

Usage Guidelines

Configurations such as channel groups, TDM connections, and loopbacks and so on in the running configuration are restored after this command is issued. If either of the controller ports on the WIC is in a looped state before this command is issued, the looped condition is dropped. You have to reinitiate the loopbacks from the remote end by doing “no loop” and “loop remote” from the controller configuration. If the BERT test is running, that test will be aborted for the microcode reload.

mmoip aaa global-password

To define a password to be used with CiscoSecure for Windows NT when using store and forward fax, use the **mmoip aaa global-password** command in global configuration mode. To restore the default state, use the **no** form of this command.

mmoip aaa global-password *password*

no mmoip aaa global-password *password*

Syntax Description

password	Character string used to define the password for CiscoSecure for Windows NT to be used with store and forward fax. The maximum length is 64 alphanumeric characters.

Defaults

No password is defined.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

CiscoSecure for Windows NT might require a separate password in order to complete authentication, no matter what security protocol you use. This command defines the password to be used with CiscoSecure for Windows NT. All records on the Windows NT server use this defined password.

This command applies to on-ramp store and forward fax functions when using a modem card. It is not used with voice feature cards.

Examples

The following example defines a password (password) when CiscoSecure for Windows NT is used with store and forward fax:

```
mmoip aaa global-password password
```

mmoip aaa method fax accounting

To define the name of the method list to be used for authentication, authorization, and accounting (AAA) accounting with store and forward fax, use the **mmoip aaa method fax accounting** command in global configuration mode. To restore the default value, use the **no** form of this command.

mmoip aaa method fax accounting *method-list-name*

no mmoip aaa method fax accounting *method-list-name*

Syntax Description

method-list-name	Character string used to name a list of accounting methods to be used with store and forward fax.

Defaults

No AAA accounting method list is defined.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

This command defines the name of the AAA accounting method list to be used with store and forward fax. The method list itself, which defines the type of accounting services provided for store and forward fax, is defined using the **aaa accounting** global configuration command. Unlike standard AAA (where each defined method list can be applied to specific interfaces and lines), the AAA accounting method lists used in store and forward fax are applied globally on the Cisco AS5300 universal access server.

After the accounting method lists have been defined, they are enabled by using the **mmoip aaa receive-accounting enable** command.

This command applies to both on-ramp and off-ramp store and forward fax functions when using a modem card. It is not used with voice feature cards.

Examples

The following example defines a AAA accounting method list (called sherman) to be used with store and forward fax:

```
aaa new-model
mmoip aaa method fax accounting sherman
```

Related Commands

Command	Description
aaa accounting	Enables AAA accounting of requested services for billing or security purposes when you use RADIUS or TACACS+.
mmoip aaa receive-accounting enable	Enables on-ramp store and forward fax AAA accounting services.

mmoip aaa method fax authentication

To define the name of the method list to be used for authentication, authorization, and accounting (AAA) authentication with store and forward fax, use the **mmoip aaa method fax authentication** command in global configuration mode. To restore the default state, use the **no** form of this command.

mmoip aaa method fax authentication *method-list-name*

no mmoip aaa method fax authentication *method-list-name*

Syntax Description

method-list-name	Character string used to name a list of authentication methods to be used with store and forward fax.

Defaults

No AAA authentication method list is defined.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

This command defines the name of the AAA authentication method list to be used with store and forward fax. The method list itself, which defines the type of authentication services provided for store and forward fax, is defined using the **aaa authentication** global configuration command. Unlike standard AAA (where each defined method list can be applied to specific interfaces and lines), AAA authentication method lists used with store and forward fax are applied globally on the Cisco AS5300 universal access server.

After the authentication method lists have been defined, they are enabled by using the **mmoip aaa receive-authentication enable** command.

This command applies to both on-ramp and off-ramp store and forward fax functions.

Examples

The following example defines a AAA authentication method list (called xyz) to be used with store and forward fax:

```
aaa new-model
mmoip aaa method fax authentication xyz
```

Related Commands

Command	Description
aaa accounting	Enables AAA accounting of requested services for billing or security purposes when you use RADIUS or TACACS+.
mmoip aaa receive-authentication enable	Enables on-ramp store and forward fax AAA authentication services.

mmoip aaa receive-accounting enable

To enable on-ramp authentication, authorization, and accounting (AAA) accounting services, use the **mmoip aaa receive-accounting enable** command in global configuration mode. To restore the default state, use the **no** form of this command.

mmoip aaa receive-accounting enable

no mmoip aaa receive-accounting enable

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1T)	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

This command enables AAA accounting services if a AAA accounting method list has been defined using both the **aaa accounting** command and the **mmoip aaa method fax accounting** command.

This command applies to on-ramp store and forward fax functions.

Examples

The following example enables a AAA accounting method list (called xyz) to be used with inbound store and forward fax. In this example, store and forward fax is being configured to track start and stop connection accounting records.

```
aaa new-model
mmoip aaa method fax accounting xyz
aaa accounting connection sherman stop-only radius
mmoip aaa receive-accounting enable
```

Related Commands

Command	Description
aaa accounting	Enables AAA accounting of requested services for billing or security purposes when you use RADIUS or TACACS+.
mmoip aaa method fax accounting	Defines the name of the method list to be used for AAA accounting with store and forward fax.

mmoip aaa receive-authentication enable

To enable on-ramp authentication, authorization, and accounting (AAA) authentication services, use the **mmoip aaa receive-authentication enable** command in global configuration mode. To restore the default state, use the **no** form of this command.

mmoip aaa receive-authentication enable

no mmoip aaa receive-authentication enable

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

This command enables AAA authentication services if an AAA authentication method list has been defined using both the **aaa authentication** command and the **mmoip aaa method fax authentication** command.

This command applies to on-ramp store and forward fax functions.

Examples

The following example enables a AAA authentication method list (called xyz) to be used with inbound store and forward fax. In this example, RADIUS authentication (and if the RADIUS server fails, then local authentication) is being configured for store and forward fax.

```
aaa new-model
mmoip aaa method fax authentication xyz
aaa authentication login peabody radius local
mmoip aaa receive-authentication enable
```

Related Commands

Command	Description
aaa accounting	Enables AAA accounting of requested services for billing or security purposes when you use RADIUS or TACACS+.
mmoip aaa method fax authentication	Defines the name of the method list to be used for AAA authentication with store and forward fax.

mmoip aaa receive-id primary

To specify the primary location where the authentication, authorization, and accounting (AAA) protocol retrieves its account identification information for on-ramp faxing, use the **mmoip aaa receive-id primary** command in global configuration mode. To restore the default state, which means that the account identification source is undefined, use the **no** form of this command.

mmoip aaa receive-id primary {**ani** | **dnis** | **gateway** | **redialer-id** | **redialer-dnis**}

no mmoip aaa receive-id primary {**ani** | **dnis** | **gateway** | **redialer-id** | **redialer-dnis**}

Syntax Description

ani	Indicates that AAA uses the calling party telephone number (automatic number identification or ANI) as the AAA account identifier.
dnis	Indicates that AAA uses the called party telephone number (dialed number identification service or DNIS) as the AAA account identifier.
gateway	Indicates that AAA uses the router-specific name derived from the host name and domain name as the AAA account identifier, displayed in the following format: *router-name.domain-name*.
redialer-id	Indicates that AAA uses the account string returned by the external redialer device as the AAA account identifier. In this case, the redialer ID is either the redialer serial number or the redialer account number.
redialer-dnis	Indicates that AAA uses the called party telephone number (dialed number identification service or DNIS) as the AAA account identifier captured by the redialer if a redialer device is present.

Defaults

No account identification source is defined.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

Normally, when AAA is being used for simple user authentication, AAA uses the username information defined in the user profile for authentication. With store and forward fax, you can specify that the ANI, DNIS, gateway ID, redialer ID, or redialer DNIS be used to identify the user for authentication. This command defines what AAA uses for the primary identifier for inbound or on-ramp user authentication with store and forward fax.

Store and forward fax allows you to define either a primary or a secondary identifier. (You configure the secondary identifier using the **mmoip aaa receive-id secondary** command.)

AAA does not use these methods sequentially—meaning that if the primary identifier is defined and AAA cannot authenticate the primary identifier information, it will not use the secondary identifier for authentication. Authentication simply fails.

Defining only the secondary identifier enables you to service two different scenarios simultaneously—for example, if you are offering fax services to two different companies, one of which uses redialers and the other does not. In this case, configure the **mmoip aaa receive-id primary** command to use the redialer DNIS, and configure the **mmoip aaa receive-id secondary** command to use ANI. With this configuration, when a user dials in and the redialer DNIS is not null, the redialer DNIS is used as the authentication identifier. If a user dials in and the redialer DNIS is null, ANI is used as the authentication identifier.

This command applies to on-ramp store and forward fax functions.

Examples

The following example defines the DNIS captured by the redialer as the AAA authentication identifier for store and forward fax:

```
aaa new-model
mmoip aaa receive-id primary redialer-dnis
```

Related Commands

Command	Description
mmoip aaa receive-id secondary	Specifies the secondary location where AAA retrieves its account identification information for on-ramp faxing if the primary identifier has not been defined.

mmoip aaa receive-id secondary

To specify the secondary location where the authentication, authorization, and accounting (AAA) protocol retrieves its account identification information for on-ramp faxing if the primary identifier has not been defined, use the **mmoip aaa receive-id secondary** command in global configuration mode. To restore the default state, which means that the account identification source is undefined, use the **no** form of this command.

mmoip aaa receive-id secondary {**ani** | **dnis** | **gateway** | **redialer-id** | **redialer-dnis**}

no mmoip aaa receive-id secondary {**ani** | **dnis** | **gateway** | **redialer-id** | **redialer-dnis**}

Syntax Description

ani	Indicates that AAA uses the calling party telephone number (automatic number identification or ANI) as the AAA account identifier.
dnis	Indicates that AAA uses the called party telephone number (dialed number identification service or DNIS) as the AAA account identifier.
gateway	Indicates that AAA uses the router-specific name derived from the host name and domain name as the AAA account identifier, displayed in the following format: *router-name.domain-name*.
redialer-id	Indicates that AAA uses the account string returned by the external redialer device as the AAA account identifier. In this case, the redialer ID is either the redialer serial number or the redialer account number.
redialer-dnis	Indicates that AAA uses the called party telephone number (dialed number identification service or DNIS) as the AAA account identifier captured by the redialer if a redialer device is present.

Defaults

No account identification source is defined.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

Normally, when AAA is being used for simple user authentication, AAA uses the username information defined in the user profile for authentication. With store and forward fax, you can specify that the ANI, DNIS, gateway ID, redialer DNIS, or redialer ID be used to identify the user for authentication. This command defines what AAA uses for the secondary identifier for inbound or on-ramp user authentication with store and forward fax if the primary identifier has not been defined.

Store and forward fax allows you to define either a primary or a secondary identifier. (You configure the primary identifier using the **mmoip aaa receive-id primary** command.)

AAA does not use these methods sequentially—meaning that if the primary identifier is defined and AAA cannot match the primary identifier information, it will not use the secondary identifier for authentication. Authentication simply fails.

Defining only the secondary identifier enables you to service two different scenarios simultaneously—for example, if you are offering fax services to two different companies, one of which uses redialers and the other does not. In this case, configure the **mmoip aaa receive-id primary** command to use the redialer DNIS, and configure the **mmoip aaa receive-id secondary** command to use ANI. With this configuration, when a user dials in and the redialer DNIS is not null, the redialer DNIS is used as the authentication identifier. If a user dials in and the redialer DNIS is null, ANI is used as the authentication identifier.

This command applies to on-ramp store and forward fax functions.

Examples

The following example defines the DNIS captured by the redialer as the secondary AAA authentication identifier for store and forward fax:

```
aaa new-model
mmoip aaa receive-id secondary redialer-dnis
```

Related Commands

Command	Description
mmoip aaa receive-id primary	Specifies the primary location where AAA retrieves its account identification information for on-ramp faxing.

mmoip aaa send-accounting enable

To enable off-ramp authentication, authorization, and accounting (AAA) accounting services, use the **mmoip aaa send-accounting enable** command in global configuration mode. To restore the default state, use the **no** form of this command.

mmoip aaa send-accounting enable

no mmoip aaa send-accounting enable

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

This command enables AAA accounting services if an AAA accounting method list has been defined using both the **aaa accounting** command and the **mmoip aaa method fax accounting** command.

This command applies to off-ramp store and forward fax functions when using a modem card. It is not used with voice feature cards.

Examples

The following example enables an AAA accounting method list (called xyz) to be used with outbound store and forward fax. In this example, store and forward fax is being configured to track start and stop connection accounting records.

```
aaa new-model
mmoip aaa method fax accounting xyz
aaa accounting connection sherman stop-only radius
mmoip aaa send-accounting enable
```

Related Commands

Command	Description
aaa accounting	Enables AAA accounting of requested services for billing or security purposes when you use RADIUS or TACACS+.
mmoip aaa method fax accounting	Defines the name of the method list to be used for AAA accounting with store and forward fax.

mmoip aaa send-authentication enable

To enable off-ramp authentication, authorization, and accounting (AAA) authentication services, use the **mmoip aaa send-authentication enable** command in global configuration mode. To restore the default value, use the **no** form of this command.

mmoip aaa send-authentication enable

no mmoip aaa send-authentication enable

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

This command enables AAA authentication services if an AAA authentication method list has been defined using both the **aaa authentication** command and the **mmoip aaa method fax authentication** command.

This command applies to off-ramp store and forward fax functions.

Examples

The following example enables an AAA authentication method list (called xyz) to be used with outbound store and forward fax. In this example, RADIUS authentication (and if the RADIUS server fails, then local authentication) is being configured for store and forward fax.

```
aaa new-model
mmoip aaa method fax authentication xyz
aaa authentication login peabody radius local
mmoip aaa send-authentication enable
```

Related Commands

Command	Description
aaa accounting	Enables AAA accounting of requested services for billing or security purposes when you use RADIUS or TACACS+.
mmoip aaa method fax authentication	Defines the name of the method list to be used for AAA authentication with store and forward fax.

mmoip aaa send-id primary

To specify the primary location where the authentication, authorization, and accounting (AAA) protocol retrieves its account identification information for off-ramp faxing, use the **mmoip aaa send-id primary** command in global configuration mode. To restore the default state, which means that account identification source is undefined, use the **no** form of this command.

mmoip aaa send-id primary {**account-id** | **envelope-from** | **envelope-to** | **gateway**}

no mmoip aaa send-id primary {**account-id** | **envelope-from** | **envelope-to** | **gateway**}

Syntax Description

account-id	Indicates that AAA uses the account username from the originating fax-mail system as the AAA account identifier. This means that the off-ramp gateway uses the account identifier in the X-account ID field of the e-mail header. Using this attribute offers end-to-end authentication and accounting tracking.
envelope-from	Indicates that AAA uses the account username from the fax-mail header as the AAA account identifier.
envelope-to	Indicates that AAA uses the recipient derived from the fax-mail header as the AAA account identifier.
gateway	Indicates that AAA uses the router-specific name derived from the host name and domain name as the AAA account identifier, displayed in the following format: *router-name.domain-name*.

Defaults

No account identification source is defined.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

Normally, when AAA is being used for simple user authentication, AAA uses the username information defined in the user profile for authentication. With store and forward fax, you can specify that the account ID, username, or recipient name from the e-mail header information be used to identify the user for authentication. This command defines what AAA uses for the primary identifier for outbound or off-ramp user authentication with store and forward fax.

Store and forward fax allows you to define either a primary or a secondary identifier. (You configure the secondary identifier using the **mmoip aaa send-id secondary** command.) AAA extracts the authentication identifier information from the defined sources. If the field is blank (meaning undefined), AAA will use the secondary identifier source if configured. The secondary identifier is used only when the primary identifier is null. In this case, when AAA sees that the primary identifier is null, it will check to see if a secondary identifier has been defined and use that value for user authentication.

AAA does not use these methods sequentially—meaning that if the primary identifier is defined and AAA cannot authenticate the primary identifier information, it will not use the secondary identifier for authentication. Authentication simply fails.

When you enable authentication, the on-ramp gateway inserts whatever value you configure for the **mmoip aaa receive-id primary** command in the X-account ID field of the e-mail header. This X-account ID field contains the value that is used for authentication and accounting by the on-ramp gateway. For example, if the **mmoip aaa receive-id primary** command is set to **gateway**, the on-ramp gateway name (for example, hostname.domain-name) is inserted in the X-account ID field of the e-mail header of the fax-mail message.

If you want to use this configured gateway value in the X-account ID field, you must configure the **mmoip aaa send-id primary** command with the **account-id** keyword. This particular keyword enables store and forward fax to generate end-to-end authentication and accounting tracking records. If you do not enable authentication on the on-ramp gateway, the X-account ID field is left blank.

This command applies to off-ramp store and forward fax functions.

Examples

The following example defines the recipient name as defined in the envelope-to field of the e-mail header to be used as the AAA authentication identifier for store and forward fax:

```
aaa new-model
mmoip aaa send-id primary envelope-to
```

Related Commands

Command	Description
mmoip aaa send-id primary	Specifies the primary location where AAA retrieves its account identification information for off-ramp faxing.
mmoip aaa send-id secondary	Specifies the secondary location where AAA retrieves its account identification information for off-ramp faxing.

mmoip aaa send-id secondary

To specify the secondary location where the authentication, authorization, and accounting (AAA) protocol retrieves its account identification information for off-ramp faxing, use the **mmoip aaa send-id secondary** command in global configuration mode. To restore the default state, which means that account identification source is undefined, use the **no** form of this command.

mmoip aaa send-id secondary {**account-id** | **envelope-from** | **envelope-to** | **gateway**}

no mmoip aaa send-id secondary {**account-id** | **envelope-from** | **envelope-to** | **gateway**}

Syntax Description

account-id	Indicates that AAA uses the account username from the originating fax-mail system as the AAA account identifier. This means that the off-ramp gateway uses the account identifier in the x-account ID field of the e-mail header. Using this attribute offers end-to-end authentication and accounting tracking.
envelope-from	Indicates that AAA uses the account username from the fax-mail header as the AAA account identifier.
envelope-to	Indicates that AAA uses the recipient derived from the fax-mail header as the AAA account identifier.
gateway	Indicates that AAA uses the router-specific name derived from the host name and domain name as the AAA account identifier, displayed in the following format: *router-name.domain-name*.

Defaults

No account identification source is defined.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

Normally, when AAA is being used for simple user authentication, AAA uses the username information defined in the user profile for authentication. With store and forward fax, you can specify that the account ID, username, or recipient name from the e-mail header information be used to identify the user for authentication. This command defines what AAA uses for the secondary identifier for outbound or off-ramp user authentication with store and forward fax.

Store and forward fax allows you to define either a primary or a secondary identifier. (You configure the primary identifier using the **mmoip aaa send-id primary** command.) AAA extracts the authentication identifier information from the defined sources. If the field is blank (meaning undefined), AAA will use the secondary identifier source if configured. The secondary identifier is used only when the primary identifier is null. In this case, when AAA sees that the primary identifier is null, it will check to see if a secondary identifier has been defined and use that value for user authentication.

AAA does not use these methods sequentially—meaning that if the primary identifier is defined and AAA cannot match the primary identifier information, it will not use the secondary identifier for authentication. Authentication simply fails.

When you enable authentication, the on-ramp gateway inserts whatever value you configure for the **mmoip aaa receive-id secondary** command in the X-account ID field of the e-mail header (if store-and-forward uses the defined secondary identifier). This X-account ID field contains the value that is used for authentication and accounting by the on-ramp gateway. For example, if the **mmoip aaa receive-id secondary** command is set to **gateway**, the on-ramp gateway name (for example, hostname.domain-name) is inserted in the X-account ID field of the e-mail header of the fax-mail message.

If you want to use this configured gateway value in the X-account ID field, you must configure the **mmoip aaa send-id secondary** command with the **account-id** keyword. This particular keyword enables store and forward fax to generate end-to-end authentication and accounting tracking records. If you do not enable authentication on the on-ramp gateway, the X-account ID field is left blank.

This command applies to off-ramp store and forward fax functions.

Examples

The following example defines the recipient name as defined in the envelope-to field of the e-mail header to be used as the AAA authentication identifier for store and forward fax:

```
aaa new-model
mmoip aaa send-id secondary envelope-to
```

Related Commands

Command	Description
mmoip aaa send-id primary	Specifies the primary location where AAA retrieves its account identification information for off-ramp faxing.
mmoip aaa send-id secondary	Specifies the secondary location where AAA retrieves its account identification information for off-ramp faxing.

mode

To set the mode of the T1/E1 controller and enter specific configuration commands for each mode type, use the **mode** command in controller configuration mode. To restore the default mode of the controller, use the **no** form of this command.

mode {**atm** | **cas**}

no mode {**atm** | **cas**}

Syntax Description

atm	Places the controller into ATM mode and creates an ATM interface (ATM 0) on the Cisco MC3810 multiservice concentrator. When ATM mode is enabled, no channel groups, channel-associated signaling (CAS) groups, CCS groups, or clear channels are allowed because ATM occupies all the DS0s on the T1/E1 trunk. When you set the controller to ATM mode, the controller framing is automatically set to ESF for T1 or CRC4 for E1. The linecode is automatically set to B8ZS for T1 or HDBC for E1. When you remove ATM mode by entering the **no mode atm** command, ATM interface 0 is deleted. ATM mode is supported only on controller 0 (T1 or E1 0).
cas	Places the controller into CAS mode, which allows you to create channel groups, CAS groups, and clear channels (both data and CAS modes). CAS mode is supported on both controller 0 and controller 1.

Defaults

No mode is configured.

Command Modes

Controller configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810 multiservice concentrator.

Usage Guidelines

This command applies to the Cisco MC3810 multiservice concentrator with the digital voice module (DVM) installed.

When no mode is selected, channel groups and clear channels (data mode) can be created using the **channel group** and **tdm-group** commands, respectively.

On the Cisco MC3810 multiservice concentrator, some DS0s are used exclusively for different signaling modes. The DS0 channels have the following limitations when mixing different applications (such as voice and data) on the same network trunk:

- On E1 controllers, DS0 16 is used exclusively for either CAS or CCS, depending on which mode is configured.
- On T1 controllers, DS0 24 is used exclusively for CCS.

Examples

The following example configures ATM mode on controller T1 0. This is required for Voice over ATM.

```
controller T1 0
 mode atm
```

The following example configures CAS mode on controller T1 1:

```
controller T1 1
 mode cas
```

Related Commands

Command	Description
channel-group	Defines the time slots that belong to each T1 or E1 circuit.
tdm-group	Configures a list of time slots for creating clear channel groups (pass-through) for TDM cross-connect.

mode ccs

To configure the T1/E1 controller to support common channel signaling (CCS) cross-connect or CCS frame forwarding, use the **mode ccs** command in controller configuration mode. To disable support for CCS cross-connect or CCS frame forwarding on the controller, use the **no** form of this command.

mode ccs {**cross-connect** | **frame-forwarding**}

no mode ccs {**cross-connect** | **frame-forwarding**}

Syntax Description

cross-connect	Enables CCS cross-connect on the controller.
frame-forwarding	Enables CCS frame forwarding on the controller.

Defaults

No CCS mode is configured.

Command Modes

Controller configuration

Command History

Release	Modification
12.0(2)T	This command was introduced on the Cisco MC3810 multiservice concentrator.
12.1(2)XH	This command was introduced on the Cisco 2600 series and Cisco 3600 series routers.
12.1(3)T	The modifications in 12.1(2)XH were integrated into 12.1(3)T.

Examples

To enable CCS cross-connect on controller T1 1, enter the following commands:

```
controller T1 1
 mode ccs cross-connect
```

To enable CCS frame forwarding on controller T1 1, enter the following commands:

```
controller T1 1
 mode ccs frame-forwarding
```

Related Commands

Command	Description
ccs connect	Configures a CCS connection on an interface configured to support CCS frame forwarding.

modem passthrough (dial-peer)

To configure modem passthrough over VoIP for a specific dial peer, use the **modem passthrough** command in dial-peer configuration mode. To disable modem passthrough for a specific dial peer, use the **no modem passthrough** command.

modem passthrough {**system** | **nse** [**payload-type** *number*] **codec** {**g711ulaw** | **g711alaw**} [**redundancy**]}

no modem passthrough

Syntax Description

system	Defaults to the global configuration.
nse	Named signaling event.
payload-type	(Optional) NSE payload type.
number	(Optional) The value of the payload type (96–119).
codec	Voice compression for speech or audio signals. Codec selections for upspeed. The upspeed method is the method used to dynamically change the codec type and speed to meet network conditions. This means that you might move to a faster codec when you have both voice and data calls, and then slow down when there is only voice traffic.
g711ulaw	Codec G.711 u-Law 64000 bits per second for T1.
g711alaw	Codec G.711 A-Law 64000 bits per second for E1.
redundancy	(Optional) Packet redundancy (RFC 2198) for modem traffic.

Defaults

Defining **system** as the method in dial peer points to the voice service Voice over IP (VoIP) configuration default and is intended to simplify gateway configuration. The default is **modem passthrough system**. As required, the gateway defaults to **no modem passthrough**.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.1(3)T	This command was introduced for the Cisco AS5300 universal access server.

Usage Guidelines

Use the **modem passthrough** dial-peer configuration command to configure modem passthrough over VoIP for a specific dial peer. The payload type is an optional parameter for the **nse** keyword. Use the same **payload-type** *number* for both the originating gateway and the terminating gateway. The **payload-type** *number* can be set from 96 to 119. If you do not specify the **payload-type** *number*, the *number* defaults to 100.

Use the same codec type for both the originating gateway and the terminating gateway. **g711ulaw** codec is required for T1, and **g711alaw** codec is required for E1.

The **redundancy** keyword is an optional parameter for sending redundant packets for modem traffic.

When the **system** keyword is enabled, the following parameters are not available: **nse**, **payload-type**, **codec**, and **redundancy**. The **system** keyword overrides the configuration for the dial peer, and the values from the global configuration are used.

Examples

The following example shows how modem passthrough over VoIP is configured for a specific dial peer in dial-peer configuration mode:

```
modem passthrough nse codec g711ulaw redundancy
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode.

modem passthrough (voice-service)

To configure modem passthrough over VoIP for the Cisco AS5300 universal access server, use the **modem passthrough** command in voice-service configuration mode. To disable modem passthrough, use the **no** form of this command.

modem passthrough nse [**payload-type** *number*] **codec** {**g711ulaw** | **g711alaw**} [**redundancy**] [**maximum-sessions** *value*]

no modem passthrough

Syntax Description

nse	Named signaling event.
payload-type	(Optional) NSE payload type.
number	(Optional) The value of the payload type. The number can be from 96 to 119.
codec	Codec selections for upspeed. The upspeed method is the method used to dynamically change the codec type and speed to meet network conditions. This means that you might move to a faster codec when you have both voice and data calls and then slow down when there is only voice traffic.
g711ulaw	Codec G.711 u-Law 64000 bits per second for T1.
g711alaw	Codec G.711 A-Law 64000 bits per second for E1.
redundancy	(Optional) Packet redundancy (RFC 2198) for modem traffic.
maximum-sessions	(Optional) Maximum number of simultaneous modem passthrough sessions.
value	(Optional) The number of simultaneous modem pass through sessions. The minimum value is 1, and the maximum value is 26. The default is 16 sessions.

Defaults

Disabled

Command Modes

Voice-service configuration

Command History

Release	Modification
12.1(3)T	This command was introduced for the Cisco AS5300 universal access server.

Usage Guidelines

Use the **modem passthrough** command to configure modem passthrough over Voice over IP (VoIP) for the Cisco AS5300 universal access server. The default behavior for the **voice service voip** command is **no modem passthrough**.

The payload type is an optional parameter for the **nse** keyword. Use the same **payload-type** *number* for both the originating gateway and the terminating gateway. The **payload-type** *number* can be set from 96 to 119. If you do not specify the **payload-type** *number*, the *number* defaults to 100.

Use the same codec type for both the originating gateway and the terminating gateway. **g711ulaw** codec is required for T1, and **g711alaw** codec is required for E1.

The **redundancy** keyword is an optional parameter for sending redundant packets for modem traffic.

The **maximum-sessions** keyword is an optional parameter for the **modem passthrough** command. This parameter determines the maximum number of simultaneous modem passthrough sessions. The recommended *value* for the **maximum-sessions** keyword is 16. The value can be set from 1 to 26.

When using the **voice service voip** and **modem passthrough nse** commands on a terminating gateway to globally set up fax or modem pass-through with NSEs, you must also ensure that each incoming call will be associated with a VoIP dial peer to retrieve the global fax or modem configuration. You associate calls with dial peers by using the **incoming called-number** command to specify a sequence of digits that incoming calls can match. You can ensure that all calls will match at least one dial peer by using the following commands:

```
Router(config)# dial-peer voice tag voip
Router(config-dial-peer)# incoming called-number .
```

Examples

The following example shows modem pass-though configuration in voice-service configuration mode for NSE payload type 101 using codec G.711:

```
modem passthrough nse payload-type 101 codec g711ulaw redundancy maximum-sessions 1
```

Related Commands

Command	Description
incoming called-number	Defines an incoming called number to match a specific dial peer.
voice service voip	Enters voice-service configuration mode and specifies the voice encapsulation type.

mta receive aliases

To specify a host name accepted as a Simple Mail Transfer Protocol (SMTP) alias for off-ramp faxing, use the **mta receive aliases** command in global configuration mode. To disable this alias, use the **no** form of this command.

mta receive aliases *string*

no mta receive aliases *string*

Syntax Description

string	Specifies the host name or IP address to be used as an alias for the SMTP server. If you specify an IP address to be used as an alias, you must enclose the IP address in brackets as follows: [xxx.xxx.xxx.xxx].

Defaults

Enabled with an empty string

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 and Cisco AS5800 universal access servers.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

This command creates an accept or reject alias list. The first alias is used by the mailer to identify itself in SMTP banners and when generating its own RFC 822 Received: header.

Note This command does not automatically include reception for a domain IP address—it must be explicitly added. To explicitly add a domain IP address, use the following format: **mta receive aliases** [*ip-address*]. Use the IP address of the Ethernet or the FastEthernet interface of the off-ramp gateway.

This command applies to on-ramp store and forward fax functions.

Examples

The following example specifies the host name seattle-fax-offramp.example.com as the alias for the SMTP server:

```
mta receive aliases seattle-fax-offramp.example.com
```

The following example specifies the IP address 172.166.0.0 as the alias for the SMTP server:

```
mta receive aliases [172.16.0.0]
```

Related Commands

Command	Description
mta receive generate-mdn	Specifies that the off-ramp gateway process a response MDN from an SMTP server.
mta receive maximum-recipients	Specifies the maximum number of recipients for all SMTP connections.

mta receive generate-mdn

To specify that the off-ramp gateway process a response message disposition notice (MDN) from a Simple Mail Transfer Protocol (SMTP) server, use the **mta receive generate-mdn** command in global configuration mode. To disable message delivery notice generation, use the **no** form of this command.

mta receive generate-mdn

no mta receive generate-mdn

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 and Cisco AS5800 universal access servers.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

When message delivery notification is enabled on a sending Cisco AS5300 universal access server, the device inserts a flag in the off-ramp message e-mail header, requesting that the receiving Cisco AS5300 universal access server generate the message delivery notification and return that message to the sender when the e-mail message that contains the fax image is opened. Use the **mta receive generate-mdn** command to enable the receiving device—the off-ramp gateway—to process the response message delivery notification.

Depending on the configuration, usage, and features of the mailers used at a site, it might be desirable to enable or disable MDN generation. (DSN generation cannot be disabled.)

Specifications for MDN are described in RFC 2298.

This command applies to off-ramp store and forward fax functions.

Examples

The following example enables the receiving device to generate message delivery notices:

```
mta receive generate-mdn
```

Related Commands

Command	Description
mdn	Requests that a message disposition notice be generated when the fax-mail message is processed (opened).

Command	Description
mta receive aliases	Specifies a host name accepted as an SMTP alias for off-ramp faxing.
mta receive maximum-recipients	Specifies the maximum number of recipients for all SMTP connections.

mta receive maximum-recipients

To specify the maximum number of recipients for all Simple Mail Transfer Protocol (SMTP) connections, use the **mta receive maximum-recipients** command in global configuration mode. To restore the default value, use the **no** form of this command.

mta receive maximum-recipients *number*

no mta receive maximum-recipients

Syntax Description

number	Specifies the maximum number of recipients for all SMTP connections. Valid entries are from 0 to 1024.

Defaults

The default is 0 recipients, meaning that incoming mail messages will not be accepted; therefore, no faxes are sent by the off-ramp gateway.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 and Cisco AS5800 universal access servers.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

This command applies to off-ramp store and forward fax functions.

Note Unless the sending mailer supports the X-SESSION SMTP service extension, each incoming SMTP connection will be allowed to send only to one recipient and thus consume only one outgoing modem.

Use the **mta receive maximum-recipients** command to configure the maximum number of modems that you want to allocate for fax usage at any one time. You can use this command to limit the resource usage on the gateway. When the value for the *number* argument is set to 0, no new connections can be established. This is particularly useful when preparing to shut down the system.

Examples

The following example defines 10 as the maximum number of recipients for all SMTP connections:

```
mta receive maximum-recipients 10
```

Related Commands

Command	Description
mta receive aliases	Specifies a host name accepted as an SMTP alias for off-ramp faxing.
mta receive generate-mdn	Specifies that the off-ramp gateway process a response MDN from an SMTP server.

mta send mail-from

To specify the mail-from address (also called the RFC 821 envelope-from address or the Return-Path address), use the **mta send mail-from** command in global configuration mode. To disable this return path information, use the **no** form of this command.

mta send mail-from {**hostname** *string* | **username** *string* | **username s**}

no mta send mail-from {**hostname** *string* | **username** *string* | **username s**}

Syntax Description

hostname *string*	Text string that specifies the Simple Mail Transfer Protocol (SMTP) host name or IP address. If you specify an IP address, you must enclose the IP address in brackets as follows: [xxx.xxx.xxx.xxx].
username *string*	Text string that specifies the sender username.
username s	Wildcard that specifies that the username will be derived from the calling number.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 and Cisco AS5800 universal access servers.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

Use the **mta send mail-from** command to designate the sender of the fax TIFF attachment. This value is equivalent to the return path information in an e-mail message.

The postmaster address, configured with the **mta send postmaster** command, is used if the mail-from address is blank.

This command applies to on-ramp store and forward fax functions.

Examples

The following example specifies that the mail-from username information will be derived from the calling number of the sender:

```
mta send mail-from username $s$
```

Related Commands

Command	Description
mta send origin-prefix	Adds information to the e-mail prefix header.
mta send postmaster	Defines where an e-mail message should be delivered (the mail server postmaster account) if it cannot be delivered to the defined destination.
mta send return-receipt-to	Specifies the address where MDNs are sent.
mta send server	Specifies a destination mail server or servers.
mta send subject	Specifies the subject header of the e-mail message.

mta send origin-prefix

To add information to the e-mail prefix header, use the **mta send origin-prefix** command in global configuration mode. To disable the defined string, use the **no** form of this command.

mta send origin-prefix *string*

no mta send origin-prefix *string*

Syntax Description

string	Text string that adds comments to the e-mail prefix header. If this string contains more than one word, the string value should be contained within quotation marks ("abc xyz").

Defaults

Null string

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 and Cisco AS5800 universal access servers.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

Normally, the store and forward fax feature provides the slot and port number from which the e-mail came in the e-mail prefix header information. Use this command to append the defined text string to the front of the e-mail prefix header information. This test string is a prefix string that is appended with the modem port and slot number and passed in the originator_comment field of the esmtp_client_engine_open() call. Eventually, this ends up in the received header field of the fax-mail message, for example:

```
Received (test onramp Santa Cruz slot1 port15) by router-5300.cisco.com for
<test-test@cisco.com> (with Cisco NetWorks); Fri, 25 Dec 1998 001500 -0800
```

In other words, using the command **mta send origin-prefix dog** causes the Received header to contain the following information:

```
Received (dog, slot 3 modem 8) by as5300-sj.example.com....
```

This command applies to on-ramp store and forward fax functions.

Examples

The following example shows how to add information to the e-mail prefix header:

```
mta send origin-prefix "Cisco-Powered Fax System"
```

Related Commands

Command	Description
mta send mail-from	Specifies the mail-from address (also called the RFC 821 envelope-from address or the Return-Path address).
mta send postmaster	Defines where an e-mail message should be delivered (the mail server postmaster account) if it cannot be delivered to the defined destination.
mta send return-receipt-to	Specifies the address where MDNs are sent.
mta send server	Specifies a destination mail server or servers.
mta send subject	Specifies the subject header of the e-mail message.

mta send postmaster

To define where an e-mail message should be delivered (the mail server postmaster account) if it cannot be delivered to the defined destination, use the **mta send postmaster** command in global configuration mode. To disable this defined postmaster, use the **no** form of this command.

mta send postmaster *e-mail-address*

no mta send postmaster *e-mail-address*

Syntax Description

e-mail-address	E-mail address that defines where this e-mail message should be delivered (the mail server postmaster account) if it cannot be delivered to the defined destination.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 and Cisco AS5800 universal access servers.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

If you have configured the Cisco AS5300 universal access server to generate DSNs and MDNs but you have not configured the sender information (using the **mta send mail-from** command) or the Simple Mail Transfer Protocol (SMTP) server, DSNs and MDNs are delivered to the e-mail address determined by this command.

The address defined by this command is used as the **mta send mail-from** address if the evaluated string is blank. An address, such as fax-administrator@example.com, is recommended (where example.com is replaced with your domain name, and fax-administrator is aliased to the person responsible for the operation of the AS5300 universal access server fax functions). At some sites, this may be the same person as the e-mail postmaster, but at most sites this is likely to be a different person and thus should be a different e-mail address.

This command applies to on-ramp store and forward fax functions.

Examples

The following example configures the e-mail address fax-admin@example.com as the sender for all incoming faxes. Thus, any returned DSNs will be delivered to fax-admin@example.com if the mail-from field is otherwise blank.

```
mta send postmaster fax-admin@example.com
```

Related Commands

Command	Description
mta send mail-from	Specifies the mail-from address (also called the RFC 821 envelope-from address or the Return-Path address).
mta send origin-prefix	Adds information to the e-mail prefix header.
mta send return-receipt-to	Specifies the address where MDNs are sent.
mta send server	Specifies a destination mail server or servers.
mta send subject	Specifies the subject header of the e-mail message.

mta send return-receipt-to

To specify the address where message disposition notices (MDNs) are sent, use the **mta send return-receipt-to** command in global configuration mode. To restore the default value, use the **no** form of this command.

mta send return-receipt-to {**hostname** *string* | **username** *string* | **username s**}

no mta send return-receipt-to {**hostname** *string* | **username** *string* | **username s**}

Syntax Description

hostname *string*	Text string that specifies the Simple Mail Transfer Protocol (SMTP) host name or IP address where MDNs are sent. If you specify an IP address, you must enclose the IP address in brackets as follows: [xxx.xxx.xxx.xxx].
username *string*	Text string that specifies the sender username where MDNs are sent.
username s	Wildcard that specifies that the username are derived from the calling number.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 and Cisco AS5800 universal access servers.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

Use the **mta send return-receipt-to** command to define where you want MDNs to be sent after the fax-mail is opened.

Note Store and forward fax supports the Eudora proprietary format, meaning that the header that store and forward fax generates is in compliance with RFC 2298 (MDN).

Note Multimedia Mail over Internet Protocol (MMoIP) dial peers must have MDN enabled to generate return receipts in off-ramp fax-mail messages.

This command applies to on-ramp store and forward fax functions.

Examples

The following example configures scoobee as the SMTP mail server to which DSNs are sent:

```
mta send return-receipt-to hostname server.com
mta send return-receipt-to username scoobee
```

Related Commands

Command	Description
mta send mail-from	Specifies the mail-from address (also called the RFC 821 envelope-from address or the Return-Path address).
mta send origin-prefix	Adds information to the e-mail prefix header.
mta send postmaster	Defines where an e-mail message should be delivered (the mail server postmaster account) if it cannot be delivered to the defined destination.
mta send server	Specifies a destination mail server or servers.
mta send subject	Specifies the subject header of the e-mail message.

mta send server

To specify a destination mail server or servers, use the **mta send server** command in global configuration mode. To disable the specified destination mail server, use the **no** form of this command.

mta send server {*host-name* | *IP-address*}

no mta send server {*host-name* | *IP-address*}

Syntax Description

host-name	Defines the host name of the destination mail server.
IP-address	Defines the IP address of the destination mail server.

Defaults

IP address defined as 0.0.0.0

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 and Cisco AS5800 universal access servers.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

Use the **mta send server** command to provide a backup destination server in case the first configured mail server is unavailable. (This command is not intended to be used for load distribution.)

You can configure up to ten different destination mail servers using the **mta send server** command. If you configure more than one destination mail server, the Cisco AS5300 universal access server attempts to contact the first mail server configured. If that mail server is unavailable, it will contact the next configured destination mail server.

DNS MX records are not used to look up host names provided to this command.

Note When you use the **mta send server** command, you should configure the Cisco AS5300 universal access server to perform name lookups using the **ip name-server** command.

This command applies to on-ramp store and forward fax functions.

Examples

The following example defines the mail servers scoobee.example.com and doogie.example.com as the destination mail servers:

```
mta send server scoobee.example.com
mta send server doogie.example.com
```

Related Commands

Command	Description
ip name-server	Specifies the address of one or more name servers to use for name and address resolution.
mta send mail-from	Specifies the mail-from address (also called the RFC 821 envelope-from address or the Return-Path address).
mta send origin-prefix	Adds information to the e-mail prefix header.
mta send postmaster	Defines where an e-mail message should be delivered (the mail server postmaster account) if it cannot be delivered to the defined destination.
mta send return-receipt-to	Specifies the address where MDNs are sent.
mta send subject	Specifies the subject header of the e-mail message.

mta send subject

To specify the subject header of the e-mail message, use the **mta send subject** command in global configuration mode. To disable this string, use the **no** form of this command.

mta send subject *string*

no mta send subject *string*

Syntax Description

string	Text string that specifies the subject header of an e-mail message.

Defaults

Null string

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 and Cisco AS5800 universal access servers.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

This command applies to on-ramp store and forward fax functions.

Note The string does not need to be enclosed in quotation marks.

Examples

The following example defines the subject header of an e-mail message as “fax attachment”:

```
mta send subject fax attachment
```

Related Commands

Command	Description
mta send mail-from	Specifies the mail-from address (also called the RFC 821 envelope-from address or the Return-Path address).
mta send origin-prefix	Adds information to the e-mail prefix header.
mta send postmaster	Defines where an e-mail message should be delivered (the mail server postmaster account) if it cannot be delivered to the defined destination.
mta send return-receipt-to	Specifies the address where MDNs are sent.
mta send server	Specifies a destination mail server or servers.

music-threshold

To specify the threshold for on-hold music for a specified voice port, use the **music-threshold** command in voice-port configuration mode. To disable this feature, use the **no** form of this command.

music-threshold *number*

no music-threshold *number*

Syntax Description

number	The on-hold music threshold in decibels (dB). Valid entries are any integer from –70 to –30.

Defaults

–38 dB

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced for the Cisco 3600 series router.
12.0(4)T	Support was added for the Cisco MC3810 multiservice concentrator.

Usage Guidelines

Use this command to specify the decibel level of music played when calls are put on hold. This command tells the firmware to pass steady data above the specified level. It only affects the operation of voice activity detection (VAD) when receiving voice.

If the value for this command is set too high, VAD interprets music-on-hold as silence, and the remote end does not hear the music. If the value for this command is set too low, VAD compresses and passes silence when the background is noisy, creating unnecessary voice traffic.

Examples

The following example sets the decibel threshold to –35 for the music played when calls are put on hold:

```
voice port 0:D
 music-threshold -35
```

The following example sets the decibel threshold to –35 for the music played when calls are put on hold on the Cisco 3600 series router:

```
voice-port 1/0/0
 music-threshold -35
```

The following example sets the decibel threshold to –35 for the music played when calls are put on hold on the Cisco MC3810 multiservice concentrator:

```
voice-port 1/1
 music-threshold -35
```

Related Commands

Command	Description
voice-port	Enters voice-port configuration mode.

network-clock base-rate

To configure the network clock base rate for universal I/O serial ports 0 and 1 on the Cisco MC3810 multiservice concentrator, use the **network-clock base-rate** command in global configuration mode. To disable the current network clock base rate, use the **no** form of this command.

network-clock base-rate {**56k** | **64k**}

no network-clock base-rate {**56k** | **64k**}

Syntax Description

56k	Sets the network clock base rate to 56 kbps.
64k	Sets the network clock base rate to 64 kbps.

Defaults

56 kbps

Command Modes

Global configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810 multiservice concentrator.

Usage Guidelines

This command applies to Voice over Frame Relay and Voice over ATM on the Cisco MC3810 multiservice concentrator.

Examples

The following example sets the network clock base rate to 64 kbps:

```
network-clock base-rate 64k
```

Related Commands

Command	Description
network-clock-select	Uses the network clock source to provide timing to the system backplane PCM bus.
network-clock-switch	Configures the switch delay time to the next priority network clock source when the current network clock source fails.

network-clock-switch

To configure the switch delay time to the next priority network clock source when the current network clock source fails, use the **network-clock-switch** command in global configuration mode. To cancel the network clock delay time selection, use the **no** form of this command.

network-clock-switch {*switch-delay* | **never**} {*restore-delay* | **never**}

no network-clock-switch

Syntax Description

switch-delay	The delay time, in seconds, before the next priority network clock source is used when the current network clock source fails. The range is from 0 to 99 seconds. The default is 10 seconds.
never	Indicates no delay time before the current network clock source recovers.
restore-delay	The delay time before the current network clock source recovers. The range is from 0 to 99 seconds.
never	Indicates no delay time, in seconds, before the next priority network clock source is used when the current network clock source fails.

Defaults

10 seconds

Command Modes

Global configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810 multiservice concentrator.

Examples

The following command switches the network clock source after 20 seconds and sets the delay time before the current network clock source recovers to 20 seconds:

```
network-clock-switch 20 20
```

Related Commands

Command	Description
network-clock-select	Uses the network clock source to provide timing to the system backplane PCM bus.

non-linear

To enable nonlinear processing in the echo canceller, use the **non-linear** command in voice-port configuration mode. To disable nonlinear processing, use the **no** form of this command.

non-linear

no non-linear

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series router.

Usage Guidelines

The function enabled by the **non-linear** command is also generally known as residual echo suppression. This command is associated with the echo canceller operation. The **echo-cancel enable** command must be enabled for the **non-linear** command to take effect. Use the **non-linear** command to shut off any signal if no near-end speech is detected.

Enabling the **non-linear** command normally improves performance, although some users might perceive truncation of consonants at the end of sentences when this command is enabled.

Examples

The following example enables nonlinear call processing on the Cisco 3600 series router:

```
voice-port 1/0/0
 non-linear
```

The following example enables nonlinear call processing on the Cisco MC3810 multiservice concentrator:

```
voice-port 1/1
 non-linear
```

Related Commands

Command	Description
echo-cancel enable	Enables the cancellation of voice that is sent out the interface and is received on the same interface.

nsap

To specify the network service access point (NSAP) address for a local video dial peer, use the **nsap** command in dial-peer configuration mode. To remove any configured NSAP address from the dial peer, use the **no** form of this command.

nsap *nsap-address*

no nsap

Syntax Description

nsap-address	A 40-digit hexadecimal number; the number must be unique on the device.

Defaults

No NSAP address for a video dial peer is configured.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(5)XK	This command was introduced for ATM video dial-peer configuration on the Cisco MC3810 multiservice concentrator.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0(9)T.

Usage Guidelines

The address must be unique on the router.

Examples

On a Cisco MC3810 multiservice concentrator, the following example sets up an NSAP address for the local video dial peer designated as 10:

```
dial-peer video 10 videocodec
 nsap 47.0091810000000002F26D4901.333333333332.02
```

Related Commands

Command	Description
dial-peer video	Defines a video ATM dial peer for a local or remote video codec, specifies video-related encapsulation, and enters dial-peer configuration mode.
show dial-peer video	Displays dial-peer configuration.

num-exp

To define how to expand a telephone extension number into a particular destination pattern, use the **num-exp** command in global configuration mode. To cancel the configured number expansion, use the **no** form of this command.

num-exp *extension-number expanded-number*

no num-exp *extension-number*

Syntax Description

extension-number	Digit or digits that define an extension number for a particular dial peer.
expanded-number	Digit or digits that define the expanded telephone number or destination pattern for the extension number listed.

Defaults

No number expansion is defined.

Command Modes

Global configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series router.
12.0(3)T	This command was supported on the Cisco AS5300 universal access server.
12.0(4)XL	This command was supported on the Cisco AS5800 universal gateway.
12.0(7)T	The modifications in 12.0(4)XL were integrated in 12.0(7)T.
12.0(7)XK	This command was supported on the Cisco MC3810 multiservice concentrator platform.
12.1(2)T	The modifications in 12.0(7)XK were integrated in 12.1(2)T.

Usage Guidelines

Use the **num-exp** global configuration command to define how to expand a particular set of numbers (for example, a telephone extension number) into a particular destination pattern. With this command, you can bind specific extensions and expanded numbers together by explicitly defining each number, or you can define extensions and expanded numbers using variables. You can also use this command to convert seven-digit numbers to numbers containing less than seven digits.

Use a period (.) as a variable or wildcard, representing a single number. Use a separate period for each number that you want to represent with a wildcard—for example, if you want to replace four numbers in an extension with wildcards, type in four periods.

Examples

The following example expands the extension number 55541 to the number 14085555541:

```
num-exp 55541 14085555541
```

The following example expands all five-digit extensions beginning with 5 such that the 5 is replaced with the digits 1408555 at the beginning of the extension number:

```
num-exp 5.... 1408555....
```

Related Commands

Command	Description
dial-peer terminator	Designates a special character to be used as a terminator for variable length dialed numbers.
forward-digits	Specifies which digits to forward for voice calls.
prefix	Specifies a prefix for a dial peer.

numbering-type

To match on a number type for a dial-peer call leg, use the **numbering-type** command in dial-peer configuration mode. To remove the numbering type for a dial-peer call leg, use the **no** form of this command.

numbering-type {**international** | **abbreviated** | **national** | **network** | **reserved** | **subscriber** | **unknown**}

no numbering-type {**international** | **abbreviated** | **national** | **network** | **reserved** | **subscriber** | **unknown**}

Syntax Description

international	Specifies international numbering type.
abbreviated	Specifies abbreviated numbering type.
national	Specifies national numbering type.
network	Specifies network numbering type.
reserved	Specifies reserved numbering type.
subscriber	Specifies subscriber numbering type.
unknown	Specifies if the numbering type is unknown.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(7)XR1	This command was introduced for Voice over IP on the Cisco AS5300 universal access server.
12.0(7)XK	This command was first supported for the following voice technologies on the following platforms: • Voice over IP (Cisco 2600 series router router, Cisco 3600 series router, Cisco MC3810 multiservice concentrator) • Voice over Frame Relay (Cisco 2600 series router router, Cisco 3600 series router, Cisco MC3810 multiservice concentrator) • Voice over ATM (Cisco 3600 series router, Cisco MC3810 multiservice concentrator)

Release	Modification
12.1(1)T	This command was first supported on the T train for the following voice technology on the following platforms: • Voice over IP (Cisco 1750, Cisco 2600 series router, Cisco 3600 series router, Cisco AS5300 universal access server, Cisco 7200 series router, and Cisco 7500 series)
12.1(2)T	This command was first supported on the T train for the following voice technologies on the following platforms: • Voice over IP (Cisco MC3810 multiservice concentrator) • Voice over Frame Relay (Cisco 2600 series router, Cisco 3600 series router, Cisco MC3810 multiservice concentrator) • Voice over ATM (Cisco 3600 series router, Cisco MC3810 multiservice concentrator)

Usage Guidelines

The **numbering-type** command is supported for plain old telephone service (POTS), VoIP, VoFR, and VoATM dial peers. The numbering type options are implemented as defined by the ITU Q.931 specification.

Examples

The following example shows how to configure a POTS dial peer for network usage:

```
dial-peer voice 100 pots
 numbering-type network
```

The following example shows how to configure a VoIP dial peer for subscriber usage:

```
dial-peer voice 200 voip
 numbering-type subscriber
```

Related Commands

Command	Description
rule	Applies a translation rule to a calling party number or a called party number for both incoming and outgoing calls.
show translation-rule	Displays the contents of all the rules that have been configured for a specific translation name.
test translation-rule	Tests the execution of the translation rules on a specific name-tag.
translate	Applies a translation rule to a calling party number or a called party number for incoming calls.
translate-outgoing	Applies a translation rule to a calling party number or a called party number for outgoing calls.
translation-rule	Creates a translation name and enters translation-rule configuration mode.
voip-incoming translation-rule	Captures calls that originate from H.323-compatible clients.

operation

To select a specific cabling scheme for E&M ports, use the **operation** command in voice-port configuration mode. To restore the default, use the **no** form of this command.

operation {**2-wire** | **4-wire**}

no operation {**2-wire** | **4-wire**}

Syntax Description

2-wire	Specifies a 2-wire E&M cabling scheme.
4-wire	Specifies a 4-wire E&M cabling scheme.

Defaults

2-wire operation

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series..
11.3(1)MA	Support was added for the Cisco MC3810 multiservice concentrator.

Usage Guidelines

This command applies to both the Cisco 3600 series router and the Cisco MC3810 multiservice concentrator.

The **operation** command affects only voice traffic. Signaling is independent of 2-wire versus 4-wire settings. If the wrong cable scheme is specified, the user might get voice traffic in only one direction.

Configuring the **operation** command on a voice port changes the operation of both voice ports on a VPM card. The voice port must be shut down and then opened again for the new value to take effect.

This command is not applicable to FXS or FXO interfaces because they are, by definition, 2-wire interfaces.

On the Cisco MC3810 multiservice concentrator, this command applies only to the analog voice module (AVM).

Examples

The following example specifies that an E&M port on the Cisco 3600 series router uses a 4-wire cabling scheme:

```
voice-port 1/0/0
 operation 4-wire
```

The following example specifies that an E&M port on the Cisco MC3810 multiservice concentrator uses a 2-wire cabling scheme:

```
voice-port 1/1
 operation 2-wire
```

output attenuation

To configure a specific output attenuation value, use the **output attenuation** command in voice-port configuration mode. To disable the selected output attenuation value, use the **no** form of this command.

output attenuation *decibels*

no output attenuation

Syntax Description

decibels	The amount of attenuation, in decibels, at the transmit side of the interface. An acceptable value is any integer from –6 to 14.

Defaults

The default value for FXO, FXS, and E&M ports is 0.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series router.
11.3(1)MA	Support was added for the Cisco MC3810 multiservice concentrator.

Usage Guidelines

A system-wide loss plan must be implemented using both the **input gain** and **output attenuation** commands. Other equipment (including PBXs) in the system must be considered when creating a loss plan. The default value for this command assumes that a standard transmission loss plan is in effect, meaning that there must be an attenuation of –6 dB between phones. Connections are implemented to provide –6 dB of attenuation when the **input gain** and **output attenuation** commands are configured with the default value of 0 dB.

You cannot increase the gain of a signal to the Public Switched Telephone Network (PSTN), but you can decrease it. If the voice level is too high, you can decrease the volume by either decreasing the input gain or increasing the output attenuation.

You can increase the gain of a signal coming into the router. If the voice level is too low, you can increase the input gain by using the **input gain** command.

Examples

On the Cisco 3600 series router, the following example configures a 3-dB gain to be inserted at the transmit side of the interface:

```
voice-port 1/0/0
 output attenuation 3
```

On the Cisco AS5300 universal access server, the following example configures a 3-dB gain to be inserted at the transmit side of the interface:

```
voice-port 0:D
 output attenuation 3
```

On the Cisco MC3810 multiservice concentrator, the following example configures a 6-dB gain to be inserted at the transmit side of the interface:

```
voice-port 1/1
 output attenuation 6
```

Related Commands

Command	Description
input gain	Configures a specific input gain value for a voice port.

playout-delay (dial-peer)

To tune the playout buffer on digital signal processors (DSPs) to accommodate packet jitter caused by switches in the WAN, use the **playout-delay** command in dial-peer configuration mode. To restore the default value, use the **no** form of this command.

playout-delay {**nominal** *milliseconds* | **maximum** *milliseconds* | **minimum** {**default** | **low** | **high**}}

no playout-delay {**nominal** *milliseconds* | **maximum** *milliseconds* | **minimum** {**default** | **low** | **high**}}

Syntax Description

nominal *milliseconds*	The **nominal** keyword represents the amount of playout delay applied at the beginning of a call by the jitter buffer in the gateway. In fixed mode, this is also the maximum size of the jitter buffer throughout the call. The *milliseconds* argument is the number of milliseconds (ms) of delay; the range accepted is from 0 to 1500, although this value depends on the type of DSP and how the voice card is configured for codec complexity. (See the **codec complexity** command.) If the voice card is configured for high codec complexity, the highest value that can be configured for **nominal** for compressed codecs is 250 ms. For medium-complexity codec configurations, the highest **nominal** value is 150 ms. Voice hardware that does not support the voice card complexity configuration (such as analog voice modules for the Cisco 3600 series router) has an upper limit of 250 ms.
maximum *milliseconds*	(Adaptive mode only) The **maximum** keyword represents the upper limit of the jitter buffer, or the highest value to which the adaptive delay will be set. The *milliseconds* argument is the number of milliseconds of delay; the range accepted is from 40 to 1700, although this value depends on the type of DSP and how the voice card is configured for codec complexity. (See the **codec complexity** command.) If the voice card is configured for high codec complexity, the highest value that can be configured for **maximum** for compressed codecs is 250 ms. For medium-complexity codec configurations, the highest **maximum** value is 150 ms. Voice hardware that does not support the voice card complexity configuration (such as analog voice modules for the Cisco 3600 series router) has an upper limit of 250 ms.

minimum	(Adaptive mode only) The **minimum** keyword represents the lower limit of the jitter buffer, or the lowest value to which adaptive delay will be set. The **low** keyword represents 10 milliseconds. Use this keyword when there are low jitter conditions in the network. The **high** keyword represents 80 milliseconds. Use this keyword when there are high jitter conditions in the network. The **default** keyword represents 40 milliseconds and is appropriate when there are normal jitter conditions in the network. This is the default value used in adaptive mode when a minimum value is not configured.

Defaults

The default for nominal is 200 milliseconds.

The default for maximum is 200 milliseconds.

The default for minimum is 40 milliseconds.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810 multiservice concentrator in the voice-port configuration mode.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series router routers.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.1(3)XI	This command was first supported on the Cisco ICS 7750.
12.1(5)T	The **minimum** keyword was introduced, and support for dial-peer configuration mode was added on the Cisco 2600 series, Cisco 3600 series, and Cisco 7200 series routers, Cisco MC3810 multiservice concentrator, and Cisco AS5200, Cisco AS5300, Cisco AS5400, and Cisco AS5800 universal access servers.

Usage Guidelines

Before Cisco IOS Release 12.1(5)T, the **playout-delay** command was configured in voice-port configuration mode. For Cisco IOS Release 12.1(5)T and later releases, in most cases playout delay should be configured in dial-peer configuration mode on the Voice over IP (VoIP) dial peer that is on the receiving end of the voice traffic that is to be buffered. This dial peer senses network conditions and relays them to the DSPs, which adjust the jitter buffer as necessary. When multiple applications are configured on the gateway, playout delay should be configured in dial-peer configuration mode. When there are numerous dial peers to configure, it might be simpler to configure playout delay on a voice port. If conflicting playout delay values have been configured on a voice port and on a dial peer, the dial-peer configuration takes precedence.

Playout delay is the amount of time that elapses between the time that a voice packet is received at the jitter buffer on the DSP and the time that it is played out to the codec. In most networks with normal jitter conditions, the defaults are adequate and you will not need to configure the **playout-delay** command.

In situations in which you want to improve voice quality by reducing jitter or you want to reduce network delay, you can configure **playout-delay** parameters. The parameters are slightly different for each of the two playout delay modes, adaptive and fixed (see the **playout-delay mode** command).

In adaptive mode, the average delay for voice packets varies depending on the amount of interarrival variation that packets have as the call progresses. The jitter buffer grows and shrinks to compensate for jitter and to keep voice packets playing out smoothly, within the maximum and minimum limits that have been configured. The maximum limit establishes the highest value to which the adaptive delay will be set. The minimum limit is the low-end threshold for the delay of incoming packets by the adaptive jitter buffer. Algorithms in the DSPs that control the growth and shrinkage of the jitter buffer are weighted toward the improvement of voice quality at the expense of network delay: jitter buffer size increases rapidly in response to spikes in network transmissions and decreases slowly in response to reduced congestion.

In fixed mode, the nominal value is the amount of playout delay applied at the beginning of a call by the jitter buffer in the gateway and is also the maximum size of the jitter buffer throughout the call.

As a general rule, if there is excessive breakup of voice due to jitter with the default playout delay settings, increase playout delay times. If your network is small and jitter is minimal, decrease playout delay times for a smaller overall delay.

When there is bursty jitter in the network, voice quality can be degraded even though the jitter buffer is actually adjusting the playout delay correctly. The constant readjustment of playout delay to erratic network conditions causes voice quality problems that are usually alleviated by increasing the minimum playout delay value in adaptive mode or by increasing the nominal delay for fixed mode.

Use the **show call active voice** command to display the current delay, as well as high- and low-water marks for delay during a call. Other fields that can help determine the size of a jitter problem are ReceiveDelay, GapFillWith..., LostPackets, EarlyPackets, and LatePackets. The following is an example of the output from the **show call active voice** command:

```
VOIP:
 ConnectionId[0xECDE2E7B 0xF46A003F 0x0 0x47070A4]
 IncomingConnectionId[0xECDE2E7B 0xF46A003F 0x0 0x47070A4]
 RemoteIPAddress=192.168.100.101
 RemoteUDPPort=18834
 RoundTripDelay=26 ms
 SelectedQoS=best-effort
 tx_DtmfRelay=inband-voice
 FastConnect=TRUE
 Separate H245 Connection=FALSE
 H245 Tunneling=FALSE
 SessionProtocol=cisco
 SessionTarget=
 OnTimeRvPlayout=417000
 GapFillWithSilence=850 ms
 GapFillWithPrediction=2590 ms
 GapFillWithInterpolation=0 ms
 GapFillWithRedundancy=0 ms
 HiWaterPlayoutDelay=70 ms
 LoWaterPlayoutDelay=29 ms
 ReceiveDelay=39 ms
 LostPackets=0
 EarlyPackets=0
 LatePackets=86
```

Examples

The following example uses the default adaptive mode for the **playout-delay command with** a minimum playout delay of 10 milliseconds and a maximum playout delay of 60 milliseconds, on the VoIP dial peer tagged 80. The size of the jitter buffer will be adjusted up and down on the basis of the amount of jitter that the DSP finds, but will never be smaller than 10 milliseconds, and never larger than 60 milliseconds.

```
dial-peer 80 voip
 playout-delay minimum low
 playout-delay maximum 60
```

Related Commands

Command	Description
playout-delay mode	Selects fixed or adaptive mode for the jitter buffer on DSPs.
show call active voice	Displays active call information for voice calls.

playout-delay (voice-port)

To tune the playout buffer to accommodate packet jitter caused by switches in the WAN, use the **playout-delay** command in voice-port configuration mode. To restore the default value, use the **no** form of this command.

playout-delay {**maximum** | **nominal**} *milliseconds*

no playout-delay {**maximum** | **nominal**}

Syntax Description

maximum	The delay time that the digital signal processor (DSP) allows before starting to discard voice packets. The default is 160 milliseconds.
nominal	The initial (and minimum allowed) delay time that the DSP inserts before playing out voice packets. The default is 80 milliseconds.
milliseconds	Playout-delay value, in milliseconds. The range for maximum playout delay is 40 to 320, and the range for nominal playout delay is 40 to 240.

Defaults

The default for maximum delay is 160 milliseconds. The default for nominal delay is 80 milliseconds.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810 multiservice concentrator.
12.0(7)XK	This command was first supported on Cisco 2600 and 3600 series routers.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

If there is excessive breakup of voice due to jitter with the default playout delay settings, increase the delay times. If your network is small and jitter is minimal, decrease the delay times to reduce delay.

Examples

The following example configures a nominal playout delay of 80 milliseconds and a maximum playout delay of 160 milliseconds on voice port 1/1 on a Cisco MC3810 multiservice concentrator:

```
voice-port 1/1
 playout-delay nominal 80
 playout-delay maximum 160
```

The following example configures a nominal playout delay of 80 milliseconds and a maximum playout delay of 160 milliseconds on voice port 1/0/0 on the Cisco 2600 or 3600:

```
voice-port 1/0/0
 playout-delay nominal 80
 playout-delay maximum 160
```

Related Commands

Command	Description
vad	Enables voice activity detection.

playout-delay mode (dial-peer)

To select fixed or adaptive mode for playout delay from the jitter buffer on digital signal processors (DSPs), use the **playout-delay mode** command in dial-peer configuration mode. To restore the default value, use the **no** form of this command.

playout-delay mode {**adaptive** | **fixed**}

no playout-delay mode {**adaptive** | **fixed**}

Syntax Description

adaptive	Jitter buffer size and amount of playout delay are adjusted during a call, on the basis of current network conditions.
fixed	Jitter buffer size does not adjust during a call; a constant playout delay is added.

Defaults

The default is the **adaptive** keyword.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.1(5)T	Support for this command was added on the Cisco 2600 and Cisco 3600 series routers, Cisco MC3810 multiservice concentrator, and the Cisco ICS 7750. In addition, the **no-timestamps** keyword was removed from the syntax.

Usage Guidelines

Before Cisco IOS Release 12.1(5)T, the **playout-delay** command was configured only in voice-port configuration mode. For Cisco IOS Release 12.1(5)T and later releases, in most cases playout delay should be configured in dial-peer configuration mode on the VoIP dial peer that is on the receiving end of the voice traffic that is to be buffered. This dial peer senses network conditions and relays them to the DSPs, which adjust the jitter buffer as necessary. When multiple applications are configured on the gateway, playout delay should be configured in dial-peer configuration mode.

Tips When there are numerous dial peers to configure, it might be simpler to configure playout delay on a voice port. If conflicting playout delay values have been configured on a voice port and on a dial peer, the dial-peer configuration takes precedence.

In most networks with normal jitter conditions, the default is adequate and you do not need to configure the **playout-delay mode** command.

The default is adaptive mode, in which the average delay for voice packets varies depending on the amount of interarrival variation that packets have as the call progresses. The jitter buffer grows and shrinks to compensate for jitter and to keep voice packets playing out smoothly, within the maximum and minimum limits that have been configured.

Select fixed mode only when you understand your network conditions well, and when you have a network with very poor quality of service (QoS) or when you are interworking with a media server or similar transmission source that tends to create a lot of jitter at the transmission source. In most situations it is better to configure adaptive mode and let the DSP size the jitter buffer according to current conditions.

Examples

The following example configures adaptive playout-delay mode with the minimum delay set at high (80 milliseconds), on a VoIP dial peer that has a tag of 80:

```
dial-peer 80 voip
 playout-delay mode adaptive
 playout-delay minimum high
```

Related Commands

Command	Description
playout-delay	Tunes the jitter buffer on DSPs for playout delay of voice packets.
show call active voice	Displays active call information for voice calls.

playout-delay mode (voice-port)

To select fixed or adaptive mode for playout delay from the jitter buffer on digital signal processors (DSPs), use the **playout-delay mode** command in voice port configuration mode. To restore the default value, use the **no** form of this command.

playout-delay mode {**adaptive** | **fixed**}

no playout-delay mode {**adaptive** | **fixed**}

Syntax Description

adaptive	Jitter buffer size and amount of playout delay are adjusted during a call, on the basis of current network conditions.
fixed	Jitter buffer size does not adjust during a call; a constant playout delay is added.

Defaults

The default is the **adaptive** keyword.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)MA	The **playout-delay** command was introduced on the Cisco MC3810 multiservice concentrator.
12.0(7)XK	The **playout-delay** command was first supported on the Cisco 2600 and Cisco 3600 series routers.
12.1(2)T	The **playout-delay** command was integrated into Cisco IOS Release 12.1(2)T.
12.1(3)XI	The keyword **mode** was introduced, and the **playout-delay mode** command was first supported on the Cisco ICS 7750.
12.1(5)T	The **no-timestamps** keyword was removed from the syntax.

Usage Guidelines

Before Cisco IOS Release 12.1(5)T, the **playout-delay** command was configured only in voice-port configuration mode. For Cisco IOS Release 12.1(5)T and later releases, in most cases playout delay should be configured in dial-peer configuration mode on the VoIP dial peer that is on the receiving end of the voice traffic that is to be buffered. This dial peer senses network conditions and relays them to the DSPs, which adjust the jitter buffer as necessary. When multiple applications are configured on the gateway, playout delay should be configured in dial-peer configuration mode.

Tips

When there are numerous dial peers to configure, it might be simpler to configure playout delay on a voice port. If conflicting playout delay values have been configured on a voice port and on a dial peer, the dial-peer configuration takes precedence.

In most networks with normal jitter conditions, the default is adequate and you do not need to configure the **playout-delay mode** command.

The default is adaptive mode, in which the average delay for voice packets varies depending on the amount of interarrival variation that packets have as the call progresses. The jitter buffer grows and shrinks to compensate for jitter and to keep voice packets playing out smoothly, within the maximum and minimum limits that have been configured.

Select fixed mode only when you understand your network conditions well, and when you have a network with very poor quality of service (QoS) or when you are interworking with a media server or similar transmission source that tends to create a lot of jitter at the transmission source. In most situations it is better to configure adaptive mode and let the DSP size the jitter buffer according to current conditions.

Examples

The following example configures fixed mode on a Cisco 3640 voice port, with a nominal delay of 80 milliseconds.

```
voice-port 1/1/0
 playout-delay mode fixed
 playout-delay nominal 80
```

Related Commands

Command	Description
playout-delay	Tunes the jitter buffer on DSPs for playout delay of voice packets.
show call active voice	Displays active call information for voice calls.

port (dial peer)

To associate a dial peer with a specific voice port, use the **port** command in dial-peer configuration mode. To cancel this association, use the **no** form of this command.

Cisco 1750 Router

port *slot-number/port*

no port *slot-number/port*

Cisco 2600 and 3600 Series Routers

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

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

Cisco MC3810 Multiservice Concentrator

port *slot/port*

no port *slot/port*

Cisco AS5300 Universal Access Server

port *controller number***:D**

no port *controller number***:D**

Cisco AS5800 Universal Gateway

port {*shelf/slot/port***:D**} | {*shelf***/***slot***/***parent***:***port***:D**}

no port {*shelf/slot/port***:D**} | {*shelf***/***slot***/***parent***:***port***:D**}

Cisco 7200 Series Routers

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

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

Syntax Description

For the Cisco 1750 Router:

slot-number	Slot number in the router in which the VIC is installed. Valid entries are from 0 to 2, depending on the slot in which it has been installed.
port	Voice port number. Valid entries are 0 or 1.

For the Cisco 2600 and 3600 Series Routers:

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

For the Cisco MC3810 Multiservice Concentrator:

*slot***/***port*	The *slot* variable specifies the slot number in the Cisco router where the voice interface card is installed. The only valid entry is 1. The *port* variable specifies the voice port number. Valid ranges are as follows: Analog voice ports: from 1 to 6. Digital T1: from 1 to 24. Digital E1: from 1 to 15, and from 17 to 31.

For the Cisco AS5300 Universal Access Server::

controller number	Specifies the T1 or E1 controller.
:D	Indicates the D channel associated with ISDN PRI.

For the Cisco AS5800 Universal Gateway:

shelf/slot/port	Specifies the T1 or E1 controller on the T1 card. Valid entries for the *shelf* variable is 0 to 9999. Valid entries for the *slot* value is 0 to 11. Valid entries for the *port* variable is 0 to 11.
*shelf***/***slot***/***parent***:***port*	Specifies the T1 controller on the T3 card. Valid entries for the *shelf* variable is 0 to 9999. Valid entries for the *slot* variable is 0 to 11. Valid entries for the *port* variable is 1 to 28. The value for the *parent* variable is always 0.
:D	Indicates the D channel associated with ISDN PRI.

For the Cisco 7200 Series Routers:

slot	Router location where the voice port adapter is installed. Valid entries are from 0 to 3.
port	Voice interface card location. Valid entries are 0 or 1.
dso-group-no	Indicates the defines DS0 group number. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.
slot-number	Indicates the slot number in the Cisco router where the voice interface card is installed. Valid entries are from 0 to 3, depending on the slot where it has been installed.

subunit-number	Indicates the subunit on the voice interface card where the voice port is located. Valid entries are 0 or 1.
port	Indicates the voice port number. Valid entries are 0 or 1.

Defaults

No port is configured.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced (Cisco 3600 series router).
11.3(3)T	Port-specific values for the Cisco 2600 were added.
11.3(1)MA	Port-specific values for the Cisco MC3810 multiservice concentrator were added.
12.0(3)T	Port-specific values for the Cisco AS5300 universal access server were added.
12.0(4)T	Support was added for the Cisco uBR924 platform.
12.0(7)T	Port-specific values for the Cisco AS5800 universal gateway were added.

Usage Guidelines

This command is used for calls incoming from a telephony interface to select an incoming dial peer and for calls coming from the VoIP network to match a port with the selected outgoing dial peer.

This command applies only to POTS peers.

Examples

The following example associates a Cisco 3600 series router POTS dial peer 10 with voice port 1, which is located on subunit 0, and accessed through port 0:

```
dial-peer voice 10 pots
 port 1/0/0
```

The following example associates a Cisco MC3810 multiservice concentrator POTS dial peer 10 with voice port 0, which is located in slot 1:

```
dial-peer voice 10 pots
 port 1/0
```

The following example associates a Cisco AS5300 universal access server POTS dial peer 10 with voice port 0:D:

```
dial-peer voice 10 pots
 port 0:D
```

The following example associates a Cisco AS5800 universal gateway POTS dial peer 10 with voice port 1/0/0:D (T1 card):

```
dial-peer voice 10 pots
 port 1/0/0:D
```

port media

To specify the serial interface to which the local video codec is connected for a local video dial peer, use the **port media** command in video dial-peer configuration mode. To remove any configured locations from the dial peer, use the **no** form of this command.

port media *interface*

no port media

Syntax Description

interface	Serial interface to which the local codec is connected. Valid entries are the numbers 0 or 1.

Defaults

No interface is specified.

Command Modes

Video dial-peer configuration

Command History

Release	Modification
12.0(5)XK	This command was introduced for ATM video dial-peer configuration on the Cisco MC3810 multiservice concentrator.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0(7)T.

Examples

On a Cisco MC3810 multiservice concentrator local video dial peer designated as 10, the following example shows serial interface 0 as the specified interface for the codec:

```
dial-peer video 10 videocodec
 port media Serial0
```

Related Commands

Command	Description
port signal	Specifies the slot location of the VDM and the port location of the EIA/TIA-366 interface for signaling.
show dial-peer video	Displays dial-peer configuration.

port signal

To specify the slot location of the video dialing module (VDM) and the port location of the EIA/TIA-366 interface for signaling for a local video dial peer, use the **port signal** command in video dial-peer configuration mode. To remove any configured locations from the dial peer, use the **no** form of this command.

port signal *slot*/*port*

no port signal

Syntax Description

slot/	Slot location of the VDM. Valid values are 1 and 2.
port	Port location of the EIA/TIA-366 interface. The Cisco MC3810 multiservice concentrator VDM has only one port, so the *port* value is always 0.

Defaults

No locations are specified.

Command Modes

Video dial-peer configuration

Command History

Release	Modification
12.0(5)XK	This command was introduced for ATM video dial-peer configuration on the Cisco MC3810 multiservice concentrator.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0(7)T.

Examples

On a Cisco MC3810 multiservice concentrator, the following example shows how to set up the VDM and EIA/TIA-366 interface locations for the local video dial peer designated as 10:

```
dial-peer video 10 videocodec
 port signal 1/0
```

Related Commands

Command	Description
port media	Specifies the serial interface to which the local video codec is connected.
show dial-peer video	Displays dial-peer configuration.

pots call-waiting

To enable the local call waiting feature on a Cisco 800 series router, use the global configuration **pots call-waiting** command in global configuration mode. To disable the local call waiting feature, use the **no** form of this command.

pots call-waiting {**local** | **remote**}

no pots call-waiting {**local** | **remote**}

Syntax Description

local	Enable call waiting on a local basis for the routers.
remote	Rely on the network provider service instead of the router to hold calls.

Defaults

The call waiting default is **remote** if the Call Waiting feature is not configured. In that case, the call holding pattern follows the settings of the service provider rather than those of the router.

Command Modes

Global configuration

Command History

Release	Modification
12.1.(2)XF	The command **pots call-waiting** was introduced on the Cisco 800 series routers.

Usage Guidelines

To display the call waiting setting, use the **show running-config** or **show pots status** command. The ISDN call waiting service is used if it is available on the ISDN line connected to the router even if local call waiting is configured on the router. That is, if the ISDN line supports call waiting, the local call waiting configuration on the router is ignored.

Examples

The following example enables local call waiting on a router:

```
pots call-waiting local
```

Related Commands

Command	Description
call-waiting	Configure Call Waiting for a specific dial peer.

pots country

To configure your connected telephones, fax machines, or modems to use country-specific default settings for each physical characteristic, use the **pots country** command in global configuration mode. To disable the use of country-specific default settings for each physical characteristic, use the **no** form of this command.

pots country *country*

no pots country *country*

Syntax Description

country	Specifies the country in which your router is located.

Defaults

A default country is not defined.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco 800 series routers.

Usage Guidelines

This command applies to the Cisco 800 series routers.

If you need to change a country-specific default setting of a physical characteristic, you can use the associated command listed in the "Related Commands" section. Enter the **pots country ?** command to get a list of supported countries and the code you must enter to indicate a particular country.

Examples

The following example specifies that the devices connected to the telephone ports use default settings specific to Germany for the physical characteristics:

```
pots country de
```

Related Commands

Command	Description
pots dialing-method	Specifies how the Cisco 800 series router collects and sends digits dialed on your connected telephones, fax machines, or modems.
pots disconnect-supervision	Specifies how a Cisco 800 series router notifies the connected telephones, fax machines, or modems when the calling party has disconnected.
pots disconnect-time	Specifies the interval in which the disconnect method is applied if telephones, fax machines, or modems connected to a Cisco 800 series router fail to detect that a calling party has disconnected.
pots distinctive-ring-guard-time	Specifies the delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).

Command	Description
pots encoding	Specifies the PCM encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router.
pots line-type	Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router.
pots ringing-freq	Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring.
pots silence-time	Specifies the interval of silence after a calling party disconnects (Cisco 800 series router).
pots tone-source	Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router.
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pots dialing-method

To specify how the router collects and sends digits dialed on your connected telephones, fax machines, or modems, use the **pots dialing-method** command in global configuration mode. To disable the specified dialing method, use the **no** form of this command.

pots dialing-method {**overlap** | **enblock**}

no pots dialing-method {**overlap** | **enblock**}

Syntax Description

overlap	The router sends each digit dialed in a separate message.
enblock	The router collects all digits dialed and sends the digits in one message.

Defaults

The default depends on the setting of the **pots country** command. For more information, see to the **pots country** command.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco 800 series routers.

Usage Guidelines

This command applies to Cisco 800 series routers.

To interrupt the collection and transmission of dialed digits, enter a pound sign (#), or stop dialing digits until the interdigit timer runs out (10 seconds).

Examples

The following example specifies that the router uses the enblock dialing method:

```
pots dialing-method enblock
```

Related Commands

Command	Description
pots country	Configures telephones, fax machines, or modems connected to a Cisco 800 series router to use country-specific default settings for each physical characteristic.
pots disconnect-supervision	Specifies how a Cisco 800 series router notifies the connected telephones, fax machines, or modems when the calling party has disconnected.
pots disconnect-time	Specifies the interval in which the disconnect method is applied if telephones, fax machines, or modems connected to a Cisco 800 series router fail to detect that a calling party has disconnected.

Command	Description
pots distinctive-ring-guard-time	Specifies the delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).
pots encoding	Specifies the PCM encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router.
pots line-type	Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router.
pots ringing-freq	Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring.
pots silence-time	Specifies the interval of silence after a calling party disconnects (Cisco 800 series router).
pots tone-source	Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router.
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pots disconnect-supervision

To specify how a router notifies the connected telephones, fax machines, or modems when the calling party has disconnected, use the **pots disconnect-supervision** command in global configuration mode. To disable the specified disconnect method, use the **no** form of this command.

pots disconnect-supervision {**osi** | **reversal**}

no pots disconnect-supervision {**osi** | **reversal**}

Syntax Description

osi	Open switching interval (OSI) is the duration for which DC voltage applied between tip and ring conductors of a telephone port is removed.
reversal	Polarity reversal of tip and ring conductors of a telephone port.

Defaults

The default depends on the setting of the **pots country** command. For more information, see to the **pots country** command.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco 800 series routers.

Usage Guidelines

This command applies to Cisco 800 series routers.

Most countries except Japan typically use the **osi** option. Japan typically uses the **reversal** option.

Examples

The following example specifies that the router uses the OSI disconnect method:

```
pots disconnect-supervision osi
```

Related Commands

Command	Description
pots country	Configures telephones, fax machines, or modems connected to a Cisco 800 series router to use country-specific default settings for each physical characteristic.
pots dialing-method	Specifies how the Cisco 800 series router collects and sends digits dialed on your connected telephones, fax machines, or modems.
pots disconnect-time	Specifies the interval in which the disconnect method is applied if telephones, fax machines, or modems connected to a Cisco 800 series router fail to detect that a calling party has disconnected.

Command	Description
pots distinctive-ring-guard-time	Specifies the delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).
pots encoding	Specifies the PCM encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router.
pots line-type	Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router.
pots ringing-freq	Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring.
pots silence-time	Specifies the interval of silence after a calling party disconnects (Cisco 800 series router).
pots tone-source	Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router.
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pots disconnect-time

To specify the interval in which the disconnect method is applied if your connected telephones, fax machines, or modems fail to detect that a calling party has disconnected, use the **pots disconnect-time** command in global configuration mode. To disable the specified disconnect interval, use the **no** form of this command.

pots disconnect-time *interval*

no pots disconnect-time *interval*

Syntax Description

interval	Number from 50 to 2000 (milliseconds).

Defaults

The default depends on the setting of the **pots country** command. For more information, see the **pots country** command.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco 800 series routers.

Usage Guidelines

This command applies to Cisco 800 series routers.

The **pots disconnect-supervision** command configures the disconnect method.

Examples

The following example specifies that the connected devices apply the configured disconnect method for 100 milliseconds after a calling party disconnects:

```
pots disconnect-time 100
```

Related Commands

Command	Description
pots country	Configures telephones, fax machines, or modems connected to a Cisco 800 series router to use country-specific default settings for each physical characteristic.
pots dialing-method	Specifies how the Cisco 800 series router collects and sends digits dialed on your connected telephones, fax machines, or modems.
pots disconnect-supervision	Specifies how a Cisco 800 series router notifies the connected telephones, fax machines, or modems when the calling party has disconnected.
pots distinctive-ring-guard-time	Specifies the delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).

Command	Description
pots encoding	Specifies the PCM encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router.
pots line-type	Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router.
pots ringing-freq	Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring.
pots silence-time	Specifies the interval of silence after a calling party disconnects (Cisco 800 series router).
pots tone-source	Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router.
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pots distinctive-ring-guard-time

To specify the delay in which a telephone port can be rung after a previous call is disconnected, use the **pots distinctive-ring-guard-time** command in global configuration mode. To disable the specified delay, use the **no** form of this command.

pots distinctive-ring-guard-time *milliseconds*

no pots distinctive-ring-guard-time *milliseconds*

Syntax Description

milliseconds	Number from 0 to 1000 (milliseconds).

Defaults

The default depends on the setting of the **pots country** command. For more information, see the **pots country** command.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco 800 series routers.

Usage Guidelines

This command applies to Cisco 800 series routers.

Examples

The following example specifies that a telephone port can be rung 100 milliseconds after a previous call is disconnected:

```
pots distinctive-ring-guard-time 100
```

Related Commands

Command	Description
pots country	Configures telephones, fax machines, or modems connected to a Cisco 800 series router to use country-specific default settings for each physical characteristic.
pots dialing-method	Specifies how the Cisco 800 series router collects and sends digits dialed on your connected telephones, fax machines, or modems.
pots disconnect-supervision	Specifies how a Cisco 800 series router notifies the connected telephones, fax machines, or modems when the calling party has disconnected.
pots disconnect-time	Specifies the interval in which the disconnect method is applied if telephones, fax machines, or modems connected to a Cisco 800 series router fail to detect that a calling party has disconnected.
pots encoding	Specifies the PCM encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router.

Command	Description
pots line-type	Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router.
pots ringing-freq	Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring.
pots silence-time	Specifies the interval of silence after a calling party disconnects (Cisco 800 series router).
pots tone-source	Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router.
ring	Sets up a distinctive ring for telephones, fax machines, or modems connected to a Cisco 800 series router.
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pots encoding

To specify the pulse code modulation (PCM) encoding scheme for your connected telephones, fax machines, or modems, use the **pots encoding** command in global configuration mode. To disable the specified PCM encoding scheme, use the **no** form of this command.

pots encoding {**alaw** | **ulaw**}

no pots encoding {**alaw** | **ulaw**}

Syntax Description

alaw	International Telecommunication Union Telecommunication Standardization Section (ITU-T) PCM encoding scheme used to represent analog voice samples as digital values.
ulaw	North American PCM encoding scheme used to represent analog voice samples as digital values.

Defaults

The default depends on the setting of the **pots country** command. For more information, see the **pots country** command.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco 800 series routers.

Usage Guidelines

This command applies to Cisco 800 series routers.

Europe typically uses the **alaw** option. North America typically uses the **ulaw** option.

Examples

The following example specifies **alaw** as the PCM encoding scheme:

```
pots encoding alaw
```

Related Commands

Command	Description
pots country	Configures telephones, fax machines, or modems connected to a Cisco 800 series router to use country-specific default settings for each physical characteristic.
pots dialing-method	Specifies how the Cisco 800 series router collects and sends digits dialed on your connected telephones, fax machines, or modems.
pots disconnect-supervision	Specifies how a Cisco 800 series router notifies the connected telephones, fax machines, or modems when the calling party has disconnected.

Command	Description
pots disconnect-time	Specifies the interval in which the disconnect method is applied if telephones, fax machines, or modems connected to a Cisco 800 series router fail to detect that a calling party has disconnected.
pots distinctive-ring-guard-time	Specifies the delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).
pots line-type	Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router.
pots ringing-freq	Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring.
pots silence-time	Specifies the interval of silence after a calling party disconnects (Cisco 800 series router).
pots tone-source	Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router.
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pots line-type

To specify the impedance of your connected telephones, fax machines, or modems, use the **pots line-type** command in global configuration mode. To disable the specified line type, use the **no** form of this command.

pots line-type {**type1** | **type2** | **type3**}

no pots line-type {**type1** | **type2** | **type3**}

Syntax Description

type1	Runs at 600 ohms.
type2	Runs at 900 ohms.
type3	Runs at 300 or 400 ohms.

Defaults

The default depends on the setting of the **pots country** command. For more information, see the **pots country** command.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco 800 series routers.

Usage Guidelines

This command applies to Cisco 800 series routers.

Examples

The following example specifies type1 as the line type:

```
pots line-type type1
```

Related Commands

Command	Description
pots country	Configures telephones, fax machines, or modems connected to a Cisco 800 series router to use country-specific default settings for each physical characteristic.
pots dialing-method	Specifies how the Cisco 800 series router collects and sends digits dialed on your connected telephones, fax machines, or modems.
pots disconnect-supervision	Specifies how a Cisco 800 series router notifies the connected telephones, fax machines, or modems when the calling party has disconnected.
pots disconnect-time	Specifies the interval in which the disconnect method is applied if telephones, fax machines, or modems connected to a Cisco 800 series router fail to detect that a calling party has disconnected.

Command	Description
pots distinctive-ring-guard-time	Specifies the delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).
pots encoding	Specifies the PCM encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router.
pots ringing-freq	Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring.
pots silence-time	Specifies the interval of silence after a calling party disconnects (Cisco 800 series router).
pots tone-source	Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router.
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pots ringing-freq

To specify the frequency on the Cisco 800 series router at which your connected telephones, fax machines, or modems ring, use the **pots ringing-freq** command in global configuration mode. To disable the specified ringing frequency, use the **no** form of this command.

pots ringing-freq {**20Hz** | **25Hz** | **50Hz**}

no pots ringing-freq {**20Hz** | **25Hz** | **50Hz**}

Syntax Description

20Hz	Connected devices ring at 20 Hz.
25Hz	Connected devices ring at 25 Hz.
50Hz	Connected devices ring at 50 Hz.

Defaults

The default depends on the setting of the **pots country** command. For more information, see the **pots country** command.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco 800 series routers.

Usage Guidelines

This command applies to Cisco 800 series routers.

Examples

The following example specifies a ringing frequency of 50 Hz:

```
pots ringing-freq 50Hz
```

Related Commands

Command	Description
pots country	Configures telephones, fax machines, or modems connected to a Cisco 800 series router to use country-specific default settings for each physical characteristic.
pots dialing-method	Specifies how the Cisco 800 series router collects and sends digits dialed on your connected telephones, fax machines, or modems.
pots disconnect-supervision	Specifies how a Cisco 800 series router notifies the connected telephones, fax machines, or modems when the calling party has disconnected.
pots disconnect-time	Specifies the interval in which the disconnect method is applied if telephones, fax machines, or modems connected to a Cisco 800 series router fail to detect that a calling party has disconnected.

Command	Description
pots distinctive-ring-guard-time	Specifies the delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).
pots encoding	Specifies the PCM encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router.
pots line-type	Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router.
pots silence-time	Specifies the interval of silence after a calling party disconnects (Cisco 800 series router).
pots tone-source	Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router.
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pots silence-time

To specify the interval of silence after a calling party disconnects, use the **pots silence-time** command in global configuration mode. To disable the specified silence time, use the **no** form of this command.

pots silence-time *interval*

no pots silence-time *interval*

Syntax Description

interval	Number from 0 to 10 (seconds).

Defaults

The default depends on the setting of the **pots country** command. For more information, see the **pots country** command.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco 800 series routers.

Usage Guidelines

This command applies to Cisco 800 series routers.

Examples

The following example specifies 10 seconds as the interval of silence:

```
pots silence-time 10
```

Related Commands

Command	Description
pots country	Configures telephones, fax machines, or modems connected to a Cisco 800 series router to use country-specific default settings for each physical characteristic.
pots dialing-method	Specifies how the Cisco 800 series router collects and sends digits dialed on your connected telephones, fax machines, or modems.
pots disconnect-supervision	Specifies how a Cisco 800 series router notifies the connected telephones, fax machines, or modems when the calling party has disconnected.
pots disconnect-time	Specifies the interval in which the disconnect method is applied if telephones, fax machines, or modems connected to a Cisco 800 series router fail to detect that a calling party has disconnected.
pots distinctive-ring-guard-time	Specifies the delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).

Command	Description
pots encoding	Specifies the PCM encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router.
pots line-type	Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router.
pots ringing-freq	Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring.
pots tone-source	Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router.
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pots tone-source

To specify the source of dial, ringback, and busy tones for your connected telephones, fax machines, or modems, use the **pots tone-source** command in global configuration mode. To disable the specified tone source, use the **no** form of this command.

pots tone-source {**local** | **remote**}

no pots tone-source {**local** | **remote**}

Syntax Description

local	Router supplies the tones.
remote	Telephone switch supplies the tones.

Defaults

The default setting is **local**.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco 800 series routers.

Usage Guidelines

This command applies to Cisco 800 series routers.

This command applies only to ISDN lines connected to a EURO-ISDN (NET3) switch.

Examples

The following example specifies **remote** as the tone source:

```
pots tone-source remote
```

Related Commands

Command	Description
pots country	Configures telephones, fax machines, or modems connected to a Cisco 800 series router to use country-specific default settings for each physical characteristic
pots dialing-method	Specifies how the Cisco 800 series router collects and sends digits dialed on your connected telephones, fax machines, or modems.
pots disconnect-supervision	Specifies how a Cisco 800 series router notifies the connected telephones, fax machines, or modems when the calling party has disconnected.
pots disconnect-time	Specifies the interval in which the disconnect method is applied if telephones, fax machines, or modems connected to a Cisco 800 series router fail to detect that a calling party has disconnected.

Command	Description
pots distinctive-ring-guard-time	Specifies the delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).
pots encoding	Specifies the PCM encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router.
pots line-type	Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router.
pots ringing-freq	Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring.
pots silence-time	Specifies the interval of silence after a calling party disconnects (Cisco 800 series router).
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pre-dial delay

To configure a delay on an Foreign Exchange Office (FXO) interface between the beginning of the off-hook state and the initiation of dual-tone multifrequency (DTMF) signaling, use the **pre-dial delay** command in voice-port configuration mode. To restore the default value, use the **no** form of the command.

pre-dial delay *seconds*

no pre-dial delay

Syntax Description

seconds	Delay, in seconds, before signaling begins. Valid values are from 0 to 10.

Defaults

The default setting is 1 second.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.(7)T	This command was introduced on the Cisco 3600 series routers.
12.0(2)T	This command was integrated into 12.0(2)T.

Usage Guidelines

This command applies to Cisco 3600 series routers.

To disable the command, set the delay to 0. When an FXO interface begins to draw loop current (off-hook state), a delay is required between the initial flow of loop current and the beginning of signaling. Some devices initiate signaling too quickly, resulting in redial attempts. The **pre-dial delay** command allows a signaling delay.

Examples

The following example sets a predial delay value of 3 seconds on the FXO port of a Cisco 3600 series router:

```
voice-port 1/0/0
 pre-dial delay 3
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timing delay-duration	Configures delay dial signal duration for a specified voice port.

preference

To indicate the preferred order of a dial peer within a hunt group, use the **preference** command in dial-peer configuration mode. To remove the preference value on the voice port, use the **no** form of this command.

preference *value*

no preference *value*

Syntax Description

value	An integer from 0 to 10, where the lower the number, the higher the preference. The default value is 0 (highest preference).

Defaults

0 (highest preference)

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810 multiservice concentrator.
12.0(3)T	This command was supported on the Cisco 2600 series router and 3600 series routers.
12.0(4)T	Support was added for VoFR dial peers on the Cisco 2600 series router and 3600 series routers.

Usage Guidelines

This command applies to plain old telephone service (POTS) dial peers, Voice over IP (VoIP) dial peers, Voice over Frame Relay (VoFR) dial peers, and Voice over ATM (VoATM) dial peers on the Cisco MC3810 multiservice concentrator.

Use the **preference** command to indicate the preference order for matching dial peers in a rotary group. Setting the preference enables the desired dial peer to be selected when multiple dial peers within a hunt group are matched for a dial string.

Note If POTS and voice-network peers are mixed in the same hunt group, the POTS dial peers must have priority over the voice-network dial peers.

Use this command with the Rotary Calling Pattern feature.

The hunting algorithm precedence is configurable. For example, if you wish a call processing sequence to go to destination A first, to destination B second, and to destination C third, you would assign preference (0 being the highest priority) to the destinations in the following order:

- Preference 0 to A
- Preference 1 to B
- Preference 2 to C

Examples

The following example configures POTS dial peer 10 with a preference of 1, POTS dial peer 20 with a preference of 2, and VoFR dial peer 30 to a preference of 3:

```
dial-peer voice 10 pots
 destination pattern 5552150
 preference 1
 exit

dial-peer voice 20 pots
 destination pattern 5552150
 preference 2
 exit

dial-peer voice 30 vofr
 destination pattern 5552150
 preference 3
 exit
```

The following examples show different dial peer configurations using the **preference** command:

```
Dialpeer        destpat         preference              session-target
1               4085551048      0 (highest)             jmmurphy-voip
2               408555          0                       sj-voip
3               408555          1 (lower)               backup-sj-voip
4               ..........      1                       0:D     (interface)
5               ..........      0                       anywhere-voip
```

If the destination number is 4085551048, the order of attempts will be 1, 2, 3, 5, 4:

```
Dialpeer        destpat         preference
1               408555          0
2               4085551048      1
3               4085551         0
4 ..............4085551.........0
```

If the number dialed is 4085551048, the order will be 2, 3, 4, 1.

Note The default behavior is that the longest matching dial peer supersedes the preference value.

Related Commands

Command	Description
called-number (dial-peer)	Enables an incoming VoFR call leg to get bridged to the correct POTS call leg when using a static FRF.11 trunk connection.
codec (dial-peer)	Specifies the voice coder rate of speech for a Voice over Frame Relay dial peer.
cptone	Specifies a regional analog voice interface-related tone, ring, and cadence setting.
destination-pattern	Specifies the prefix, the full E.164 telephone number, or an ISDN directory number (depending on the dial plan) to be used for a dial peer.
dtmf-relay (Voice over Frame Relay)	Enables the generation of FRF.11 Annex A frames for a dial peer.
session protocol	Establishes a session protocol for calls between the local and remote routers via the packet network.
session target	Specifies a network-specific address for a specified dial peer or destination gatekeeper.
signal-type	Sets the signaling type to be used when connecting to a dial peer.

prefix

To specify the prefix of the dialed digits for a dial peer, use the **prefix** command in dial-peer configuration mode. To disable this feature, use the **no** form of this command.

prefix *string*

no prefix

Syntax Description

string	Integers that represent the prefix of the telephone number associated with the specified dial peer. Valid numbers are 0 through 9, and a comma (,). Use a comma to include a pause in the prefix.

Defaults

Null string

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series routers.
12.0(4)XJ	This command was modified for store and forward fax and to add support on the Cisco AS5300 universal access server.
12.1(1)T	Modifications in Cisco IOS Release 12.0(4)XJ were integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

Use the **prefix** command to specify a prefix for a specific dial peer. When an outgoing call is initiated to this dial peer, the **prefix** *string* value is sent to the telephony interface first, before the telephone number associated with the dial peer.

If you want to configure different prefixes for dialed numbers on the same interface, you need to configure different dial peers.

This command is applicable only to plain old telephone service (POTS) dial peers. This command applies to off-ramp store and forward fax functions.

Examples

The following example specifies a prefix of 9 and then a pause:

```
dial-peer voice 10 pots
 prefix 9,
```

Related Commands

Command	Description
answer-address	Specifies the full E.164 telephone number to be used to identify the dial peer of an incoming call.
destination-pattern	Specifies either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer.

pri-group

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

pri-group timeslots *timeslot-range*

no pri-group

Syntax Description

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

Defaults

There is no ISDN-PRI group configured.

Command Modes

Controller configuration

Command History

Release	Modification
11.0	This command was introduced.
12.0(2)T	This command was introduced for the Cisco MC3810 multiservice concentrator.
12.0(7)XK	This command was introduced for the Cisco 2600 and Cisco 3600 series routers.
12.1(2)T	The modifications in Cisco IOS Release 12.0(7)XK were integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

The **pri-group** command applies to the configuration of Voice over Frame Relay and Voice over ATM on the Cisco MC3810 multiservice concentrator and the Cisco 2600 and Cisco 3600 series routers.

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

Note Only one PRI group can be configured on a controller.

Examples

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

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

Related Commands

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

pri-group nec-fusion

To configure your NEC PBX to support Fusion Call Control Signaling (FCCS), use the **pri-group nec-fusion** command in controller configuration mode. To disable FCCS, use the **no** form of this command.

pri-group nec-fusion {*pbx-ip-address* | *pbx-ip-host-name*} **pbx-port** *number*

no pri-group nec-fusion {*pbx-ip-address* | *pbx-ip-host-name*} **pbx-port** *number*

Syntax Description

pbx-ip-address	The IP address of the NEC PBX.
pbx-ip-host-name	The host name of the NEC PBX.
pbx-port *number*	Choose a port number for the PBX. The range for the PBX port is 49152 to 65535. If you do not specify a port number, the default value of 55000 will be used. If this value is already in use, the next greater value will be used.

Defaults

Port number 55000

Command Modes

Controller configuration

Command History

Release	Modification
12.0(7)T	This command was introduced on the Cisco AS5300 universal access server.
12.2(1)	This command was modified to add support for Setup messages from a POTS dial peer.

Usage Guidelines

This command is used only if the PBX in your configuration is an NEC PBX, and if you are configuring it to run FCCS and not QSIG signaling.

Examples

The following example shows how to configure this NEC PBX to use FCCS:

```
pri-group nec-fusion 172.31.255.255 pbx-port 60000
```

Related Commands

Command	Description
isdn protocol-emulate	Configures the Layer 2 and Layer 3 port protocol of a BRI voice port or a PRI interface to emulate NT (network) or TE (user) functionality.
isdn switch type	Configures the Cisco AS5300 universal access server PRI interface to support QSIG signaling.
show cdapi	Displays the CDAPI.
show rawmsg	Displays the raw messages owned by the required component.

progress_ind

To set a specific progress indicator (PI) in call Setup, Progress, or Connect messages from an H.323 Voice over IP (VoIP) gateway, use the **progress_ind** command in dial-peer configuration mode. To restore the default condition, use the **no** or **disable** forms of this command.

progress_ind {**setup** | **connect** | **progress** | **alert**} {**enable** *pi-number* | **disable**}

no progress_ind {**setup** | **connect** | **progress** | **alert**}

Note This command is not supported on VoIP gateways using session initiation protocol (SIP).

Syntax Description

setup	Sets the progress indicator for Setup messages.
connect	Sets the progress indicator for Connect messages.
progress	Sets the progress indicator for Progress messages.
alert	Sets the progress indicator for Alert messages.
enable	Enables the configuration of the progress indicator.
pi-number	The progress indicator that is sent in all messages of the specified type from the outbound dial peer. For Setup messages from plain old telephone service (POTS) or VoIP dial peers, values are 0, 1, or 3. For Progress, Connect, or Alert messages from a POTS dial peer, values are 1, 2, or 8.
disable	Disables the user configuration of the progress indicator.

Defaults

The default progress indicator from the switch is not intercepted or modified.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.1(3)XI	This command was introduced on the Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, and Cisco 7500 series routers, Cisco MC3810 multiservice concentrator, and Cisco AS5300 and Cisco AS5800 universal access servers.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.2(1)	This command was modified to add support for Setup messages from a POTS dial peer.

Usage Guidelines

The **progress_ind** command overrides the default progress indicator that is sent by the switch. This enables you to set the progress indicator at the H.323 gateway, if necessary, to ensure the proper end-to-end signaling for VoIP calls. This command sets the progress indicator only in messages from outbound dial peers that have a set destination pattern, configured by using the **destination-pattern** command. If a message contains multiple progress indicators, the **progress_ind** command overrides only the first progress indicator in the message.

The **disable** and **no** forms of the **progress_ind** command have the same result: The call messages are not intercepted by the session application, and the default progress indicator, if any, is forwarded unmodified.

Note A progress indicator that is configured by using the **progress_ind** command will not override the default progress indicator in a Progress message, if the Progress message is sent after backward cut-through has occurred (for example, because an Alert message with a progress indicator of 8 was sent before the Progress message).

Examples

The following example sets the progress indicator to 1 in Progress and Connect messages from the number 3 POTS dial peer:

```
dial-peer voice 3 pots
 destination-pattern 55275
 progress_ind progress enable 1
 progress_ind connect enable 1
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and configures a VoIP or POTS dial peer.
destination-pattern	Specifies the telephone number that is used to identify the outbound dial peer for the call.

proxy h323

To enable the proxy feature on your router, use the **proxy h323** command in global configuration mode. To disable the proxy feature, use the **no** form of this command.

proxy h323

no proxy h323

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced on the Cisco 2500 series and Cisco 3600 series routers.

Usage Guidelines

If the multimedia interface is not enabled using the **proxy h323** command, or if no gatekeeper is available, starting the proxy allows it to attempt to locate these resources. No calls will be accepted until the multimedia interface and the gatekeeper are found.

Examples

The following example turns on the proxy feature:

```
proxy h323
```