



Cisco IOS Voice Commands: P through Q

This chapter contains commands to configure and maintain Cisco IOS voice applications. The commands are presented in alphabetical order. Some commands required for configuring voice 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 Configuration Guide*.

package persistent

To configure the package type used when reporting persistent events for a multifrequency (MF) tone channel-associated signaling (CAS) endpoint type using a specific Media Gateway Control Protocol (MGCP) profile, use the **package persistent** command in MGCP profile configuration mode. To disable the persistent status, use the **no** form of this command.

package persistent *package-name*

no package persistent *package-name*

Syntax Description

package-name Valid package names are ms-package and mt-package.

Defaults

ms-package

Command Modes

MGCP profile configuration

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T.
12.2(11)T	This command was implemented on the Cisco AS5300 and Cisco AS5850.

Usage Guidelines

This command is used when configuring values for a MGCP profile.

This command is used only with MF trunks (gateway voice ports configured with the **dial-type mf** command in voice-port configuration mode). Because the same persistent event can be defined in different MGCP packages, you may need to configure the **package persistent** command to tell the gateway which package to use when reporting persistent events to the call agent for the endpoints in this MGCP profile. For example, a T1 may be configured as an MF trunk, but there is more than one MGCP package that applies to an MF trunk. An *ans* (call answer) event must be mapped to the appropriate package for call-agent notification. This command allows different T1s to be configured for different CAS protocols.

The MS package is used with certain PBX direct inward dial (DID) and direct outward dial (DOD) trunks with wink-start or ground-start signaling as indicated in RFC 3064 (MGCP CAS Packages).

The MT package is a subset of the MS package, and it is used with certain operator services on terminating MF trunks on trunking gateway endpoints, as described in PacketCable PSTN Gateway Call Signaling Protocol Specification (TGCP) PKT-SP-TGCP-D02-991028, December 1, 1999.

Examples

The following example enables event persistence for the MT package:

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# package persistent mt-package
```

Related Commands	Command	Description
	mgcp	Starts and allocates resources for the MGCP daemon.
	mgcp profile	Initiates MGCP profile mode to create and configure an MGCP profile associated with one or more endpoints or to configure the default profile.

paging (ephone-dn)

To set paging numbers that can be called to broadcast an audio page to a group of Cisco IP phones, use the **paging** command in ephone-dn configuration mode. To disable this feature, use the **no** form of this command.

paging [**ip** *multicast-address* **port** *udp-port-number*]

no paging [**ip** *multicast-address* **port** *udp-port-number*]

Syntax Description

ip	(Optional) IP multicast.
<i>multicast-address</i>	(Optional) IP multicast address to use to multicast voice packets for audio paging; for example, 224.0.1.1. Note that multicast addresses always take the form of 224.x.x.x.
port	(Optional) UDP port.
<i>udp-port-number</i>	(Optional) UDP port number to use in association with the IP multicast address. The default is the Skinny Client Protocol port 2000.

Defaults

A Cisco IP phone directory number is not configured as a paging number

Command Modes

Ephone-dn configuration

Command History

Release	Modification
12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	This command was implemented on the Cisco 1760.

Usage Guidelines

This command configures the ephone-dn number to act as an extension number to use to broadcast audio paging to idle Cisco IP phones. Cisco IP phones must be associated with the paging directory number (ephone-dn) using the ephone-dn tag number of the paging ephone-dn or are included indirectly through a paging group from another paging ephone-dn.

When the optional **ip** keyword followed by the *multicast-address* argument is used, the paging is set for multicast paging. If an IP multicast address is not configured, IP phones are paged individually using IP unicast transmission (to a maximum of ten IP phones). The recommended operation is with an IP multicast address. When multiple paging extensions are configured, each extension should use a unique IP multicast address.

Examples

The following example configures IP multicast paging:

```
Router(config)# ephone-dn 20
Router(config-ephone-dn) number 2000
Router(config-ephone-dn) paging ip 224.0.1.1 port 2000
```

The configuration is as follows:

```
ephone-dn 20
number 2000
paging ip 224.0.1.20 port 2000
```

```
ephone-dn 21
number 2001
paging ip 224.0.1.21 port 2000
```

```
ephone 1
button 1:1
paging-dn 20
```

```
ephone 2
button 1:2
paging-dn 20
```

```
ephone 3
button 1:3
paging-dn 21
```

```
ephone 4
button 1:4
paging-dn 21
```

In this example, paging calls to 2000 are multicast to Cisco IP phones (ephones) 1 and 2; paging calls to 2001 go to ephones 3 and 4.

**Note**

The maximum number of unique IP address and router physical interfaces (or subinterfaces) combinations supported for output of audio paging voice packets is ten. Paging using a single IP multicast address that requires output on three different Ethernet interfaces represents use of three counts out of the maximum ten. The limit of ten is likely to be exceeded only if unicast addressing is used for the paging ephone-dn or if individual IP phones are configured for unicast paging support only.

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
paging-dn (ephone)	Sets an audio paging directory number for each Cisco IP phone.
paging group (ephone-dn)	Sets the audio paging directory number for a large combined group.

paging group (ephone-dn)

To set the audio paging directory number for a large combined group, use the **paging group** command in ephone-dn configuration mode. To remove a paging group, use the **no** form of this command.

paging group *paging-ephone-dn-tag-list, paging-ephone-dn-tag-list*

no paging group *paging-ephone-dn-tag-list, paging-ephone-dn-tag-list*

Syntax Description

<i>paging</i>	A comma-separated list of directory number (DN) tags that are each configured as paging directory numbers. You can include up to ten paging Cisco IP phone DN tags separated by commas; for example, 4, 6, 7, 8.
<i>-phone-dn-tag-list</i>	

Defaults

By default, paging is disabled on all Cisco IP phones

Command Modes

Ephone-dn configuration

Command History

Release	Modification
12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	This command was This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	This command was implemented on the Cisco 1760.

Usage Guidelines

Use this command to combine small sets of phones associated with individual paging directory numbers (ephone-dns) into a large combined group so that a page can be sent to large numbers of phones at once. To remove a paging group, use the **no** form of the command. All ephone-dn tags included in the list must have the **paging** command set.

The use of paging groups allows phones to participate in a small local paging set (for example, paging to four phones in a company's shipping and receiving department) but also supports company-wide paging when needed (for example by combining the paging sets for shipping and receiving, with paging sets for accounting, customer support, and sales into a paging group).

Examples

The following example sets paging groups:

```
ephone-dn 20
number 2000
paging ip 224.0.1.20 port 2000
```

```

ephone-dn 21
number 2001
paging ip 224.0.1.21 port 2000

ephone-dn 22
number 2002
paging ip 224.0.2.22 port 2000
paging group 20,21

ephone 1
button 1:1
paging-dn 20

ephone 2
button 1:2
paging-dn 20

ephone 3
button 1:3
paging-dn 21

ephone 4
button 1:4
paging-dn 21

ephone 5
button 1:5
paging-dn 22

```

In this example, paging calls to 2000 go to Cisco IP phones (ephones) 1 and 2, and paging calls to 2001 go to ephones 3 and 4. Calls to 2002 go to ephones 1, 2, 3, 4, and 5. Ephones 1 and 2 are included in paging directory number (ephone-dn) 22 through membership of ephone-dn 20 in the paging group. Ephones 3 and 4 are included in paging ephone-dn 22 through membership of ephone-dn 21 in the paging group. Ephone 5 is directly subscribed to paging-dn 22. Note that multicast addresses always take the form of 224.x.x.x.

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.v
paging (ephone-dn)	Sets paging numbers that can be called in order to broadcast an audio page to a group of Cisco IP phones.
paging-dn (ephone)	Sets an audio paging directory number for each Cisco IP phone.

paging-dn (ephone)

To set an audio paging directory number for each Cisco IP phone, use the **paging-dn** command in ephone configuration mode. To disable this feature, use the **no** form of this command.

```
paging-dn paging-dn-number-tag {multicast | unicast}
```

```
no paging-dn paging-dn-number-tag {multicast | unicast}
```

Syntax Description

<i>paging-dn-number-tag</i>	Directory tag number of the paging directory number (ephone-dn) to associate with the Cisco IP phone. Paging calls to the specified ephone-dn sends the page to the Cisco IP phone.
multicast	Multicast paging for groups. By default, audio paging is transmitted to the Cisco IP phone using multicast.
unicast	Unicast paging for a single Cisco IP phone. This keyword indicates that the Cisco IP phone is not capable of receiving audio paging through multicast and requests that the phone receives the audio paging through a unicast transmission directed to the individual phone.
Note	The number of phones supported through unicast is limited to a maximum of ten phones.

Defaults

Paging is disabled on all Cisco IP phones.

Command Modes

Ephone configuration

Command History

Release	Modification
12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	This command was implemented on the Cisco 1760.

Usage Guidelines

Use this command to set an audio paging directory number (DN) for each Cisco IP phone. The audio paging feature operates in a fashion similar to intercom, but provides only one-way voice, with no press-to-answer option. A DN is created, which is associated with a certain number of local IP phones. The paging extension number is configured using the existing **number** command under ephone-dn configuration mode. Multiple paging DNs can be supported for each system. The paging number can be dialed from anywhere, including on-net calls. The paging audio stream is heard on all selected Cisco IP phones that are in the idle state through the speakerphone mode. The IP phone display shows the “name” information associated with the paging DN used to activate the page. During an active paging, incoming or outgoing call for an answered or initiated call, disconnects the IP phone from the paging output.

The paging mechanism supports audio distribution using IP multicast, replicated unicast, and a mixture of both (so that multicast is used where possible, and unicast is used with specific phones that cannot be reached through multicast).

Each Cisco IP phone can be associated with only one paging directory number (paging-dn); however, paging-dns may be grouped in order to join groups of IP phones together. Only single-level grouping is supported (no support for groups of groups). This allows for paging to IP phones for individual departments (for example, sales, support, shipping, and accounting) and then allows these sets to be combined into a group for “all employees” or “everyone in building 2”. Any number of phones may be added into the same paging set using multicast. A Cisco IP phone (ephone) may directly belong to only a single paging set. A paging set consists of all phones configured with the same paging-dn. Each paging set uses a DN.

Examples

The following example sets up an ephone-dn for multicast paging:

```
ephone-dn 22
name Paging Shipping
number 5001
paging ip 224.1.1.10 port 2000

ephone 4
mac 0030.94c3.8724
button 1:1 2:2
paging-dn 22 multicast
```

This example creates a paging number for 5001 on ephone-dn 22 and adds ephone 4 as a member of the paging set. Multicast is set for the paging-dn. Note that multicast addresses always take the form of 224.x.x.x.



Note

For unicast paging to all phones, omit the IP multicast address in the ephone-dn configuration. For unicast paging to a specific phone using an ephone-dn configured for multicast, add the **unicast** keyword after the **paging-dn** command in ephone configuration mode.

Each ephone-dn used for paging can support a maximum of ten distinct targets (IP addresses and interfaces). A multicast address counts as a single target for each physical interface in use (regardless of the number of phones connected via the interface). Each unicast target counts as a single target, such that paging that does not use multicast at all is limited to paging ten phones. For example, ten IP phones paged through multicast on Fast Ethernet interface 0/1.1 plus five IP phones paged through multicast on FastEthernet interface 0/1.2, is counted as two targets.

For simultaneous paging to more than one paging ephone-dn, Cisco recommends that you use different IP multicast addresses (not just different port numbers) for paging configuration. Note that multicast addresses always take the form of 224.x.x.x.

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
number	Configures a valid number for the Cisco IP phone.
paging (ephone-dn)	Sets paging numbers that can be called in order to broadcast an audio page to a group of Cisco IP phones.
paging group (ephone-dn)	Sets the audio paging directory number for a large combined group.

pattern direct (vm-integration)

To configure the dual-tone multifrequency (DTMF) digit pattern forwarding necessary to activate the voice-mail system when the user presses the messages button on the phone, use the **pattern direct** command in voice-mail integration configuration mode. To disable DTMF digit pattern forwarding when the user presses the messages button on the phone, use the **no** form of this command.

```
pattern direct tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
[tag3 {CGN | CDN | FDN}] [last-tag]
```

```
no pattern direct tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
[tag3 {CGN | CDN | FDN}] [last-tag]
```

Syntax Description

<i>tag1</i>	Alphanumeric string fewer than four DTMF digits in length. The alphanumeric string consists of a combination of four letters (A, B, C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system's integration file, immediately preceding the number of the calling party, the number of the called party, or a forwarding number.
<i>tag2, tag3</i>	(Optional) See <i>tag1</i> . The Cisco IOS Telephony Service router supports a maximum of four tags.
<i>last-tag</i>	(Optional) See <i>tag1</i> . This tag indicates the end of the pattern.
CGN	Calling number (CGN) information is sent to the voice-mail system.
CDN	Called number (CDN) information is sent to the voice-mail system.
FDN	Forwarding number (FDN) information is sent to the voice-mail system.

Defaults

This command is disabled

Command Modes

Voice-mail integration configuration

Command History

Release	Modification
12.2(2)XT	For Cisco IOS Telephony Service, this command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(8)T	For Cisco IOS Telephony Service, this command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	For Cisco IOS Telephony Service, this command was implemented on the Cisco 2600XM and Cisco 2691 routers.
12.2(11)T	For Cisco IOS Telephony Service, this command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.
12.2(13)T	This command was implemented in Cisco Survivable Site Remote Telephony, Version 2.02.

Usage Guidelines

The **pattern direct** command is used to configure the sequence of DTMF digits passed to a voice-mail system attached to the Cisco IOS Telephony Service router through one or more voice ports. When a call is placed directly from a Cisco IP phone attached to the Cisco IOS Telephony Service router, the voice-mail system expects to receive a sequence of DTMF digits at the beginning of the call that identify the mailbox of the user calling the voice-mail system accompanied by a string of digits that indicate that the caller is attempting to access the designated mailbox in order to retrieve messages.

Although it is unlikely that you will use multiple instances of the **CGN**, **CDN**, or **FDN** keyword in a single command line, it is permissible to do so.

Examples

The following example sets the DTMF pattern for a calling number (CGN) for a direct call to the voice-mail system:

```
Router(config) vm-integration
Router(config-vm-integration) pattern direct 2 CGN *
```

Related Commands

Command	Description
pattern ext-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern ext-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension attempts to connect to an extension that does not answer and the call is forwarded to voice mail.
pattern trunk-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.
pattern trunk-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.
vm-integration	Enters voice-mail integration configuration and enables voice-mail integration with DTMF and the analog voice-mail system.

pattern ext-to-ext busy (vm-integration)

To configure the dual-tone multifrequency (DTMF) digit pattern forwarding necessary to activate the voice-mail system when an internal extension attempts to connect to a busy extension and the call is forwarded to voice-mail system, use the **pattern ext-to-ext busy** command in voice-mail integration configuration mode. To disable DTMF digit pattern forwarding when an internal extension calls a busy extension and the call is forwarded to a voice-mail system, use the **no** form of this command.

```
pattern ext-to-ext busy tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
[tag3 {CGN | CDN | FDN}] [last-tag]
```

```
no pattern ext-to-ext busy tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
[tag3 {CGN | CDN | FDN}] [last-tag]
```

Syntax Description

<i>tag1</i>	Alphanumeric string fewer than four DTMF digits in length. The alphanumeric string consists of a combination of four letters (A, B, C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system's integration file, immediately preceding the number of the calling party, the number of the called party, or a forwarding number.
<i>tag2, tag3</i>	(Optional) See <i>tag1</i> . The Cisco IOS Telephony Service router supports a maximum of four tags.
<i>last-tag</i>	(Optional) See <i>tag1</i> . This tag indicates the end of the pattern.
CGN	Calling number (CGN) information is sent to the voice-mail system.
CDN	Called number (CDN) information is sent to the voice-mail system.
FDN	Forwarding number (FDN) information is sent to the voice-mail system.

Defaults

This command is disabled

Command Modes

Voice-mail integration configuration

Command History

Release	Modification
12.2(2)XT	For Cisco IOS Telephony Service, this command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(8)T	For Cisco IOS Telephony Service, this command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	For Cisco IOS Telephony Service, this command was implemented on the Cisco 2600XM and Cisco 2691 routers.
12.2(11)T	For Cisco IOS Telephony Service, this command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.
12.2(13)T	This command was implemented in Cisco Survivable Site Remote Telephony, Version 2.02.

Usage Guidelines

The **pattern ext-to-ext busy** command is used to configure the sequence of DTMF digits passed to a voice-mail system attached to the Cisco IOS Telephony Service router through one or more voice ports. When a call is routed to the voice-mail system by call forward on busy from a Cisco IP phone attached to the Cisco IOS Telephony Service router, the voice-mail system expects to receive a sequence of digits that identify the mailbox associated with the forwarding phone together with digits that identify the extension number of the calling IP phone.

Although it is unlikely that you will use multiple instances of the **CGN**, **CDN**, or **FDN** keyword in a single command line, it is permissible to do so.

Examples

The following example sets the DTMF pattern for a local call that is forwarded on busy to the voice-mail system:

```
Router(config) vm-integration
Router(config-vm-integration) pattern ext-to-ext busy 7 FDN * CGN *
```

Related Commands

Command	Description
pattern direct	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the messages button on the phone.
pattern ext-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern trunk-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.
pattern trunk-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.
vm-integration	Enters voice-mail integration configuration and enables voice-mail integration with DTMF and analog voice-mail system.

pattern ext-to-ext no-answer (vm-integration)

To configure the dual-tone multifrequency (DTMF) digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail, use the **pattern ext-to-ext no-answer** command in voice-mail integration configuration mode. To disable DTMF digit forwarding, use the **no** form of this command.

```
pattern ext-to-ext no-answer tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
[tag3 {CGN | CDN | FDN}] [last-tag]
```

```
no pattern ext-to-ext no-answer tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
[tag3 {CGN | CDN | FDN}] [last-tag]
```

Syntax Description		
<i>tag1</i>		Alphanumeric string fewer than four DTMF digits in length. The alphanumeric string consists of a combination of four letters (A, B, C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system's integration file, immediately preceding the number of the calling party, the number of the called party, or a forwarding number.
<i>tag2, tag3</i>		(Optional) See <i>tag1</i> . The Cisco IOS Telephony Service router supports a maximum of four tags.
<i>last-tag</i>		(Optional) See <i>tag1</i> . This tag indicates the end of the pattern.
cg n		Calling number (CGN) information is sent to the voice-mail system.
cd n		Called number (CDN) information is sent to the voice-mail system.
fd n		Forwarding number (FDN) information is sent to the voice-mail system.

Defaults This command is disabled

Command Modes Voice-mail integration configuration

Command History	Release	Modification
	12.2(2)XT	For Cisco IOS Telephony Service, this command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
	12.2(8)T	For Cisco IOS Telephony Service, this command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
	12.2(8)T1	For Cisco IOS Telephony Service, this command was implemented on the Cisco 2600XM and Cisco 2691 routers.
	12.2(11)T	For Cisco IOS Telephony Service, this command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.
	12.2(13)T	This command was implemented in Cisco Survivable Site Remote Telephony, Version 2.02.

Usage Guidelines

The **pattern ext-to-ext no-answer** command is used to configure the sequence of DTMF digits passed to a voice-mail system attached to the Cisco IOS Telephony Service router through one or more voice ports. When a call is routed to the voice-mail system by call forward on no-answer from an IP phone attached to the Cisco IOS Telephony Service router, the voice-mail system expects to receive a sequence of digits that identify the mailbox associated with the forwarding phone together with digits that identify the extension number of the calling IP phone.

Although it is unlikely that you will use multiple instances of the **CGN**, **CDN**, or **FDN** keyword in a single command line, it is permissible to do so.

Examples

The following example sets the DTMF pattern for a local call that is forwarded on no answer to the voice-mail system:

```
Router(config) vm-integration
Router(config-vm-integration) pattern ext-to-ext no-answer 5 FDN * CGN *
```

Related Commands

Command	Description
pattern direct	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the messages button on the phone.
pattern ext-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern trunk-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.
pattern trunk-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.
vm-integration	Enters voice-mail integration configuration and enables voice-mail integration with DTMF and analog voice-mail system.

pattern trunk-to-ext busy (vm-integration)

To configure the dual-tone multifrequency (DTMF) digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches a busy extension and the call is forwarded to voice-mail system, use the **pattern trunk-to-ext busy** command in voice-mail integration configuration mode. To disable DTMF digit pattern forwarding when an external trunk call reaches a busy extension and the call is forwarded to a voice-mail system, use the **no** form of this command.

```
pattern trunk-to-ext busy tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
    [tag3 {CGN | CDN | FDN}] [last-tag]
```

```
no pattern trunk-to-ext busy tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
    [tag3 {CGN | CDN | FDN}] [last-tag]
```

Syntax Description

<i>tag1</i>	Alphanumeric string fewer than four DTMF digits in length. The alphanumeric string consists of a combination of four letters (A, B, C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system's integration file, immediately preceding the number of the calling party, the number of the called party, or a forwarding number.
<i>tag2, tag3</i>	(Optional) See <i>tag1</i> . The Cisco IOS Telephony Service router supports a maximum of four tags.
<i>last-tag</i>	(Optional) See <i>tag1</i> . This tag indicates the end of the pattern.
cg n	Calling number (CGN) information is sent to the voice-mail system.
cd n	Called number (CDN) information is sent to the voice-mail system.
fd n	Forwarding number (FDN) information is sent to the voice-mail system.

Defaults

This command is disabled

Command Modes

Voice-mail integration configuration

Command History

Release	Modification
12.2(2)XT	For Cisco IOS Telephony Service, this command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(8)T	For Cisco IOS Telephony Service, this command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	For Cisco IOS Telephony Service, this command was implemented on the Cisco 2600XM and Cisco 2691 routers.
12.2(11)T	For Cisco IOS Telephony Service, this command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.
12.2(13)T	This command was implemented in Cisco Survivable Site Remote Telephony, Version 2.02.

Usage Guidelines

The **pattern trunk-to-ext busy** command is used to configure the sequence of DTMF digits passed to a voice-mail system attached to the Cisco IOS Telephony Service router through one or more voice ports. When a call is routed to the voice-mail system by call forward on busy from an IP phone attached to the Cisco IOS Telephony Service router, the voice-mail system expects to receive a sequence of digits that identify the mailbox associated with the forwarding phone together with digits that indicate that the call originated from a PSTN or VoIP caller.

Although it is unlikely that you will use multiple instances of the **CGN**, **CDN**, or **FDN** keyword in a single command line, it is permissible to do so.

Examples

The following example sets the DTMF pattern for call forwarding when an external trunk call reaches a busy extension and the call is forwarded to the voice-mail system:

```
Router(config) vm-integration
Router(config-vm-integration) pattern trunk-to-ext busy 6 FDN * CGN *
```

Related Commands

Command	Description
pattern direct	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the messages button on the phone.
pattern ext-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern ext-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern trunk-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.
vm-integration	Enters voice-mail integration configuration and enables voice-mail integration with DTMF and analog voice-mail system.

pattern trunk-to-ext no-answer (vm-integration)

To configure the dual-tone multifrequency (DTMF) digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to a voice-mail system, use the **pattern trunk-to-ext no-answer** command in voice-mail integration configuration mode. To disable DTMF digit pattern forwarding when an external trunk call reaches an unanswered extension and the call is forwarded to a voice-mail system, use the **no** form of this command.

```
pattern trunk-to-ext no-answer tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
[tag3 {CGN | CDN | FDN}] [last-tag]
```

```
no pattern trunk-to-ext no-answer tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
[tag3 {CGN | CDN | FDN}] [last-tag]
```

Syntax Description

<i>tag1</i>	Alphanumeric string fewer than four DTMF digits in length. The alphanumeric string consists of a combination of four letters (A, B, C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system's integration file, immediately preceding the number of the calling party, the number of the called party, or a forwarding number.
<i>tag2, tag3</i>	(Optional) See <i>tag1</i> . The Cisco IOS Telephony Service router supports a maximum of four tags.
<i>last-tag</i>	(Optional) See <i>tag1</i> . This tag indicates the end of the pattern.
cg n	Calling number (CGN) information is sent to the voice-mail system.
cd n	Called number (CDN) information is sent to the voice-mail system.
fd n	Forwarding number (FDN) information is sent to the voice-mail system.

Defaults

This command is disabled

Command Modes

Voice-mail integration configuration

Command History

Release	Modification
12.2(2)XT	For Cisco IOS Telephony Service, this command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(8)T	For Cisco IOS Telephony Service, this command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	For Cisco IOS Telephony Service, this command was implemented on the Cisco 2600XM and Cisco 2691 routers.
12.2(11)T	For Cisco IOS Telephony Service, this command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.
12.2(13)T	This command was implemented in Cisco Survivable Site Remote Telephony, Version 2.02.

Usage Guidelines

The **pattern trunk-to-ext no-answer** command is used to configure the sequence of DTMF digits passed to a voice-mail system attached to the Cisco IOS Telephony Service router through one or more voice ports. When a call is routed to the voice-mail system by call forward on no answer from an IP phone attached to the Cisco IOS Telephony Service router, the voice-mail system expects to receive a sequence of digits that identify the mailbox associated with the forwarding phone together with digits that indicate that the call originated from a PSTN or VoIP caller.

Although it is unlikely that you will use multiple instances of the **CGN**, **CDN**, or **FDN** keyword in a single command line, it is permissible to do so.

Examples

The following example sets the DTMF pattern for call forwarding when an external trunk call reaches an unanswered extension and the call is forwarded (FDN) to a voice-mail system:

```
Router(config) vm-integration
Router(config-vm-integration) pattern trunk-to-ext no-answer 4 FDN * CGN *
```

Related Commands

Command	Description
pattern direct	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the messages button on the phone.
pattern ext-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern ext-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern trunk-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.
vm-integration	Enters voice-mail integration configuration and enables voice-mail integration with DTMF and analog voice-mail system.

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 reset the playout buffer to the default, use the **no** form of this command.

playout-delay { **fax** *milliseconds* | **maximum** *milliseconds* | **minimum** { **default** | **low** | **high** } | **nominal** *milliseconds* }

no **playout-delay** { **fax** | **maximum** | **minimum** | **nominal** }

Syntax Description

fax <i>milliseconds</i>	Amount of playout delay that the jitter buffer should apply to fax calls, in milliseconds. Range is from 0 to 700. Default is 300.
maximum <i>milliseconds</i>	(Adaptive mode only) Upper limit of the jitter buffer, or the highest value to which the adaptive delay is set, in milliseconds. Range 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.) Default is 200. 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	<p>(Adaptive mode only) Lower limit of the jitter buffer, or the lowest value to which the adaptive delay is set, in milliseconds. Values are as follows:</p> <ul style="list-style-type: none"> • default—40 ms. Use when there are normal jitter conditions in the network. This is the default. • low—10 ms. Use when there are low jitter conditions in the network. • high—80 ms. Use when there are high jitter conditions in the network.
nominal <i>milliseconds</i>	<p>Amount of playout delay applied at the beginning of a call by the jitter buffer in the gateway, in milliseconds. In fixed mode, this is also the maximum size of the jitter buffer throughout the call.</p> <p>Range is from 0 to 1500, although this value depends on the type of DSP and how the voice card is configured for codec complexity. Default is 200.</p> <p>For non-conference calls when you are using DSPware version 4.1.33 or a later version, the following values are allowed. If the voice card is configured for high codec complexity, the highest value that can be configured for the nominal keyword for compressed codecs is 250 ms. For medium-complexity codec configurations, the highest nominal value is 150 ms.</p> <p>For conference calls when you are using DSPware version 4.1.33 or a later version, the first decoder stream can be assigned a nominal value as high as 250 ms (high-complexity codec) or 150 ms (medium-complexity codec). Subsequent decoder streams are limited to the highest nominal value of 150 ms (high-complexity) or 80 ms (medium-complexity).</p> <p>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 for the first decoder stream and 150 ms for subsequent decoder streams.</p> <p>Note With DSPware versions earlier than 4.1.33, the highest nominal value that can be configured is 150 ms for high-complexity codec configurations and analog modules. The highest nominal value for medium-complexity codec configurations is 80 ms.</p>

Defaults

fax—300 milliseconds
maximum—200 milliseconds
minimum—default (40 milliseconds)
nominal—200 milliseconds

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was implemented on the Cisco 2600 series and Cisco 3600 series.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.1(3)XI	This command was implemented on the Cisco ICS7750.

Release	Modification
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T. Support for dial-peer configuration mode was added on the following platforms: Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco MC3810, Cisco AS5200, Cisco AS5300, Cisco AS5400, and Cisco AS5800. The minimum keyword was introduced.
12.2(13)T	The fax keyword was introduced.
12.2(13)T8	DSPware version 4.1.33 was implemented.

Usage Guidelines

Before Cisco IOS Release 12.1(5)T, this command was used 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 at which a voice packet is received at the jitter buffer on the DSP and the time at which 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 this 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 is 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 sample 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 default adaptive mode with a minimum playout delay of 10 ms and a maximum playout delay of 60 ms on VoIP dial peer 80. The size of the jitter buffer is adjusted up and down on the basis of the amount of jitter that the DSP finds, but is never smaller than 10 ms and never larger than 60 ms.

```

dial-peer 80 voip
  playout-delay minimum low
  playout-delay maximum 60

```

Related Commands

Command	Description
codec complexity	Specifies call density and codec complexity based on the codec standard you are using.
playout-delay (voice-port)	Tunes the playout buffer to accommodate packet jitter caused by switches in the WAN.
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 reset the playout buffer to the default, use the **no** form of this command.

playout-delay { **fax** | **maximum** | **nominal** } *milliseconds*

no playout-delay { **fax** | **maximum** | **nominal** }

Syntax Description

fax <i>milliseconds</i>	Amount of playout delay that the jitter buffer should apply to fax calls, in milliseconds. Range is from 0 to 700. Default is 300.
maximum <i>milliseconds</i>	Delay time that the digital signal processor (DSP) allows before starting to discard voice packets, in milliseconds. Range is from 40 to 320. Default is 160.
nominal <i>milliseconds</i>	Initial (and minimum allowed) delay time that the DSP inserts before playing out voice packets, in milliseconds. Range is from 40 to 240. Default is 80.

Defaults

fax—300 milliseconds
maximum—160 milliseconds
nominal—80 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was implemented on the Cisco 2600 series and Cisco 3600 series.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.2(13)T	The fax keyword was added.

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.

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 at which a voice packet is received at the jitter buffer on the DSP and the time at which 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 incoming packet delay that is created 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.


Note

The minimum limit for playout delay is configured using the **playout-delay (dial-peer)** command.

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 GapFillWith..., ReceiveDelay, LostPackets, EarlyPackets, and LatePackets. The following is sample 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

```

playout-delay (voice-port)

```

LostPackets=0
EarlyPackets=0
LatePackets=86

```

Examples

The following example sets nominal playout delay to 80 ms and maximum playout delay to 160 ms on voice port 1/1 on a Cisco MC3810:

```

voice-port 1/1
  playout-delay nominal 80
  playout-delay maximum 160

```

The following example sets nominal playout delay to 80 ms and maximum playout delay to 160 ms on voice port 1/0/0 on the Cisco 2600 series or Cisco 3600 series:

```

voice-port 1/0/0
  playout-delay nominal 80
  playout-delay maximum 160

```

Related Commands

Command	Description
playout-delay (dial-peer)	Tunes the playout buffer on DSPs to accommodate packet jitter caused by switches in the WAN.
playout-delay mode	Selects fixed or adaptive mode for playout delay from the jitter buffer on digital signal processors.
show call active	Shows active call information for voice calls or fax transmissions in progress.
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 reset to the default, use the **no** form of this command.

playout-delay mode {adaptive | fixed}

no playout-delay mode

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

Adaptive jitter buffer size

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.1(5)T	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, Cisco MC3810, and Cisco ICS 7750. The no-timestamps keyword was removed.

Usage Guidelines

Before Cisco IOS Release 12.1(5)T, this command was used 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.



Tip

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 this 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.

■ playout-delay mode (dial-peer)

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 sets adaptive playout-delay mode with a high (80 ms) minimum delay on a VoIP dial peer 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 reset to the default, use the **no** form of this command.

playout-delay mode {adaptive | fixed}

no playout-delay mode

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

Adaptive jitter buffer size

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was implemented on the Cisco 2600 and Cisco 3600 series.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.1(3)XI	This command was implemented on the Cisco ICS 7750. The keyword mode was introduced.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T and the no-timestamps keyword was removed.

Usage Guidelines

Before Cisco IOS Release 12.1(5)T, this command was used only in voice-port configuration mode. For Cisco IOS Release 12.1(5)T and later releases, in most cases playout delay should be used 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.



Tip

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.

■ playout-delay mode (voice-port)

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 sets fixed mode on a Cisco 3640 voice port with a nominal delay of 80 ms.

```
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 (Annex G neighbor BE)

To configure the port number of the neighbor that is used for exchanging Annex G messages, use the **port** command in Annex G Neighbor BE configuration mode. To remove the port number, use the **no** form of this command.

port *neighbor-port*

no port

Syntax Description	<i>neighbor-port</i>	Port number of the neighbor. This number is used for exchanging Annex G messages. The default port number is 2099.
---------------------------	----------------------	--

Defaults	2099
-----------------	------

Command Modes	Annex G Neighbor BE Configuration
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Command History	Release	Modification
	12.2(2)XA	This command was introduced.
	12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T. Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 is not included in this release.
	12.2(2)XB1	This command was implemented on the Cisco AS5850.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T. This command is supported on the Cisco AS5300, Cisco AS5350, and Cisco AS5400 in this release.

Examples The following example sets a neighbor BE to port number 2010.

```
Router(config-annexg-neigh)# port 2010
```

Related Commands	Command	Description
	advertise (annex g)	Controls the types of descriptors that the BE advertises to its neighbors.
	cache	Configures the local BE to cache the descriptors received from its neighbors.
	id	Configures the local ID of the neighboring BE.
	query-interval	Configures the interval at which the local BE will query the neighboring BE.

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 and Cisco 3700 Series

```
port slot-number/port
no port slot-number/port
```

Cisco 2600 series, Cisco 3600 Series, and Cisco 7200 Series

```
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

```
port slot/port
no port slot/port
```

Cisco AS5300

```
port controller-number:D
no port controller-number:D
```

Cisco AS5800

```
port {shelf/slot/port:D | shelf/slot/parent:port:D}
no port {shelf/slot/port:D | shelf/slot/parent:port:D}
```

Cisco uBR92x Series

```
port slot/subunit/port
no port slot/subunit/port
```

Syntax Description

Cisco 1750 and Cisco 3700 Series

<i>slot-number</i>	Number of the slot in the router in which the voice interface card (VIC) is installed. Valid entries are from 0 to 2, depending on the slot in which it has been installed.
<i>port</i>	Voice port number. Valid entries are 0 and 1.

Cisco 2600 series, Cisco 3600 Series, and Cisco 7200 Series

<i>slot-number</i>	Number of the slot in the router in which the VIC is installed. Valid entries are from 0 to 3, depending on the slot in which it has been installed.
<i>subunit-number</i>	Subunit on the VIC in which the voice port is located. Valid entries are 0 and 1.
<i>port</i>	Voice port number. Valid entries are 0 and 1.
<i>slot</i>	Router location in which the voice port adapter is installed. Valid entries are 0 and 3.
<i>port</i>	Voice interface card location. Valid entries are 0 and 3.
<i>ds0-group-no</i>	Specifies the 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.

Cisco MC3810

<i>slot/port</i>	<p>The <i>slot</i> variable specifies the number of the slot in the router in which the VIC is installed. The only valid entry is 1.</p> <p>The <i>port</i> variable specifies the voice port number. Valid ranges are as follows:</p> <ul style="list-style-type: none"> • Analog voice ports: from 1 to 6. • Digital T1: from 1 to 24. • Digital E1: from 1 to 15, and from 17 to 31.
------------------	--

Cisco AS5300

<i>controller-number</i>	Specifies the T1 or E1 controller.
:D	Indicates the D channel associated with ISDN PRI.

Cisco AS5800

<i>shelfslot/port</i>	Specifies the T1 or E1 controller on the T1 card. Valid entries for the <i>shelf</i> argument are from 0 to 9999. Valid entries for the <i>slot</i> argument are from 0 to 11. Valid entries for the <i>port</i> argument are from 0 to 11.
<i>shelfslotparent:port</i>	Specifies the T1 controller on the T3 card. Valid entries for the <i>shelf</i> argument are from 0 to 9999. Valid entries for the <i>slot</i> variable are from 0 to 11. Valid entries for the <i>port</i> argument are from 1 to 28. The value for the <i>parent</i> argument is always 0.
:D	Indicates the D channel associated with ISDN PRI.

Cisco uBR92x series

<i>slot/subunit/port</i>	<p>Specifies the analog voice port. Valid entries for the <i>slot/subunit/port</i> are as follows:</p> <ul style="list-style-type: none"> • <i>slot</i>—specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • <i>subunit</i>—specifies a VIC in which the voice port is located. Valid entries are 0 and 1. (The VIC fits into the voice network module.) • <i>port</i>—specifies an analog voice port number. Valid entries are 0 and 1.
--------------------------	--

port (dial-peer)

Defaults

No port is configured.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
11.3(3)T	This command was implemented on the Cisco 2600 series.
11.3(1)MA	This command was implemented on the Cisco MC3810.
12.0(3)T	This command was integrated into Cisco IOS Release 12.0(3)T and implemented on the Cisco AS5300.
12.0(4)T	This command was implemented on the Cisco uBR924.
12.0(7)T	This command was implemented on the Cisco AS5800.
12.2(8)T	This command was implemented on the following platforms: Cisco 1751, Cisco 3725, and Cisco 3745.
12.2(13)T	This command was integrated into Cisco IOS Release 12.2(13)T. This command does not support the extended echo canceller (EC) feature on the Cisco AS5300 or the Cisco AS5800.

Usage Guidelines

This command is used for calls that come from a telephony interface to select an incoming dial peer and for calls that come from the VoIP network to match a port with the selected outgoing dial peer.

This command applies only to POTS peers.



Note

This command does not support the extended echo canceller (EC) feature on the Cisco AS5300 or the Cisco AS5800.

Examples

The following example associates POTS dial peer 10 of a Cisco 3600 series router 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 POTS dial peer 10 of a Cisco MC3810 with voice port 0, which is located in slot 1:

```
dial-peer voice 10 pots
  port 1/0
```

The following example associates POTS dial peer 10 of a Cisco AS5300 with voice port 0:D:

```
dial-peer voice 10 pots
  port 0:D
```

The following example associates POTS dial peer 10 of a Cisco AS5800 with voice port 1/0/0:D (T1 card):

```
dial-peer voice 10 pots
  port 1/0/0:D
```

Related Commands

Command	Description
prefix	Specifies the prefix of the dialed digits for a dial peer.

port (MGCP profile)

To associate a voice port with the Media Gateway Control Protocol (MGCP) profile that is being configured, use the **port** command in MGCP profile configuration mode. To disassociate the voice port from the profile, use the **no** form of this command.

port *port-number*

no port *port-number*

Syntax Description	<i>port-number</i>	Voice port or DS0-group number to be used as an MGCP endpoint associated with an MGCP profile.
---------------------------	--------------------	--

Defaults	No default behavior or values
-----------------	-------------------------------

Command Modes	MGCP profile configuration
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Command History	Release	Modification
	12.2(2)XA	This command was introduced as the voice-port (MGCP profile) command.
	12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T.
	12.2(8)T	This command was renamed the port (MGCP profile) command.

Usage Guidelines	<p>This command is used when values for an MGCP profile are configured.</p> <p>This command associates a voice port with the MGCP profile that is being defined. To associate multiple voice ports with a profile, repeat this command with different voice port arguments.</p> <p>This command is not used when the default MGCP profile is configured because the values in the default profile configuration apply to all parameters that have not been otherwise configured for a user-defined MGCP profile.</p>
-------------------------	--

Examples	<p>The following example associates an analog voice port with an MGCP profile on a Cisco uBR925 platform:</p>
-----------------	---

```
Router(config)# mgcp profile ny110ca
Router(config-mgcp-profile)# port 0
```

Related Commands	Command	Description
	mgcp	Starts and allocates resources for the MGCP daemon.
	mgcp profile	Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile.

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	<i>interface</i>	Serial interface to which the local codec is connected. Valid entries are 0 and 1.
---------------------------	------------------	--

Defaults	No interface is specified
-----------------	---------------------------

Command Modes	Video dial-peer configuration
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Command History	Release	Modification
	12.0(5)XK	This command was introduced for ATM video dial-peer configuration on the Cisco MC3810.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0(7)T.	

Examples The following example specifies serial interface 0 as the specified interface for the codec on a Cisco MC3810 local video dial peer 10:

```
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

<i>slot/</i>	Slot location of the VDM. Valid values are 1 and 2.
<i>port</i>	Port location of the EIA/TIA-366 interface. The Cisco MC3810 VDM has only one port, so the <i>port</i> 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.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0(7)T.

Examples

The following example sets up the VDM and EIA/TIA-366 interface locations for the local video dial peer designated as 10 on a Cisco MC3810:

```
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, 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

Remote, in which 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	This command was introduced on the Cisco 800 series.

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	Configures 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, use the **no** form of this command.

pots country *country*

no pots country *country*

Syntax Description

country 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.

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	Command	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 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.

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 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.

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.
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 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

<i>interval</i>	Interval, in milliseconds. Range is from 50 to 2000.
-----------------	--

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.

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 ms 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 Delay, in milliseconds. Range is from 0 to 1000.

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.

Usage Guidelines

This command applies to Cisco 800 series routers.

Examples

The following example specifies that a telephone port can be rung 100 ms 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 scheme, use the **no** form of this command.

pots encoding {**alaw** | **ulaw**}

no pots encoding {**alaw** | **ulaw**}

Syntax Description	alaw	A-law. International Telecommunication Union Telecommunication Standardization Section (ITU-T) PCM encoding scheme used to represent analog voice samples as digital values.
	ulaw	Mu-law. 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.

Usage Guidelines This command applies to Cisco 800 series routers.
Europe typically uses a-law. North America typically uses u-law.

Examples The following example specifies a-law 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 forwarding-method

To configure the type of call-forwarding method to be used for Euro-ISDN (formerly NET3) switches, use the **pots forwarding-method** command in global configuration mode. To turn forwarding off, use the **no** form of this command.

pots forwarding-method {keypad | functional}

no pots forwarding-method {keypad | functional}

Syntax Description

keypad	Gives forwarding control to the Euro-ISDN switch.
functional	Gives forwarding control to the router. If you select this method, use the dual-tone multifrequency (DTMF) keypad commands listed in Table 24 to configure call-forwarding service.

Defaults

Forwarding is off

Command Modes

Global configuration

Command History

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

Usage Guidelines

Use this command to select the type of forwarding method to be used for Euro-ISDN switches. This command does not affect any other switch types.

You can select one or more call-forwarding services at a time, but keep the following Euro-ISDN switch characteristics in mind:

- Call forward unconditional (CFU) redirects a call without restriction and takes precedence over other call-forwarding service types.
- Call forward busy (CFB) redirects a call to another number if the dialed number is busy.
- Call forward no reply (CFNR) forwards a call to another number if the dialed number does not answer within a specified period of time.

If all three call-forwarding services are enabled, CFU overrides CFB and CFNR. The default is that no call-forwarding service is selected.

If you select the functional forwarding method, use the DTMF keypad commands in [Table 24](#) to configure the call-forwarding service.

Table 24 DTMF Keypad Commands for Call-Forwarding Service

Task	DTMF Keypad Command ¹
Activate CFU	**21*number#
Deactivate CFU	#21#

Table 24 DTMF Keypad Commands for Call-Forwarding Service (continued)

Task	DTMF Keypad Command ¹
Activate CFNR	**61* <i>number</i> #
Deactivate CFNR	#61#
Activate CFB	**67* <i>number</i> #
Deactivate CFB	#67#

1. Where *number* is the telephone number to which your calls are forwarded.

When you enable or disable the call-forwarding service, it is enabled or disabled for four basic services: speech, audio at 3.1 kilohertz (kHz), telephony at 3.1 kHz, and telephony at 7 kHz. You should hear a dial tone after you enter the DTMF keypad command when the call-forwarding service is successfully enabled for at least one of the four basic services. If you hear a busy tone, the command is invalid or the switch does not support that service.

Examples

The following example gives forwarding control to the router:

```
pots forwarding-method functional
```

Related Commands

Command	Description
pots prefix filter	Sets a filter that prevents a dial prefix from being added to a dialed number when the digits in the dialed number match the filter.
pots prefix number	Sets a prefix to be added to a called telephone number for analog or modem calls.

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.

Usage Guidelines

This command applies to Cisco 800 series routers.

Examples

The following example sets the line type to type1:

```
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 prefix filter

To set a filter that prevents a dial prefix from being added to a dialed number when the digits in the dialed number match the filter, use the **pots prefix filter** command in global configuration mode. To remove the filter, use the **no** form of this command.

pots prefix filter *number*

no pots prefix filter *number*

Syntax Description

<i>number</i>	Prefix filter numbers, up to a maximum of eight characters.
---------------	---

Defaults

No default filter is set.

Command Modes

Global configuration

Command History

Release	Modification
12.2(2)T	This command was introduced on the Cisco 803 and Cisco 804.

Usage Guidelines

The **pots prefix filter** command is used to set a filter for prefix dialing. A maximum of ten filters can be set. Once the maximum number of filters have been configured, an additional filter is not accepted nor does it overwrite any of the existing filters.

To configure a new filter, remove at least one filter using the **no pots prefix filter** command.

You can set matching criteria for the filter using the * wildcard character. For example, if you configure the filter 1* and a dialed number starts with 1, the called number is not prefixed. Prefix filters can be of variable length. All configured prefix filters are compared to the number dialed, up to the length of the prefix filter. If there is a match, no prefix is added to the dialed number.

Examples

The following example configures five filters that prevent dial prefixes from being added to dialed numbers:

```
pots prefix filter 192
pots prefix filter 1
pots prefix filter 9
pots prefix filter 0800
pots prefix filter 08456
```

With these filters configured, a prefix is *not* added to the following dialed numbers:

```
192      Directory calls
100      Operator services
999      Emergency services
0800...  Toll-free calls
08456... Calls on an Energis network information controller
```

Related Commands	Command	Description
	pots forwarding-method	Configures the type of forwarding method to be used for Euro-ISDN (formerly NET3) switches.
	pots prefix number	Sets a prefix to be added to a called telephone number for analog or modem calls.

pots prefix number

To set a prefix to be added to a called telephone number for analog or modem calls, use the **pots prefix number** command in global configuration mode. To remove the prefix, use the **no** form of this command.

pots prefix number *number*

no pots prefix number *number*

Syntax Description

<i>number</i>	Prefix, up to a maximum of five digits.
---------------	---

Defaults

No prefix is associated with the called number for analog or modem calls

Command Modes

Global configuration

Command History

Release	Modification
12.2(2)T	This command was introduced on the Cisco 803 and Cisco 804.

Usage Guidelines

Only one prefix can be configured using this command. If a prefix already exists, the next prefix configured with this command overwrites the old prefix. Prefixes can be of variable length, up to five digits. The **no pots prefix number** command removes the prefix.

As numbers are dialed on the keypad, a comparison is made to the configured prefix filter. When a match is determined, the number is dialed without adding the prefix. In the unlikely event that the prefix filter has more digits than the dialed number, and the dialed number matches the first digits of the prefix filter, the prefix is not added to the dialed number. For example, if the prefix filter is 5554000 and you dial 555 and stop, the router considers the called number to be 555 and does not add a prefix to the number. This event is unlikely to occur because the number of digits in dialed numbers is typically greater than the number of digits in prefix filters.

Examples

The following example sets the prefix to 12345:

```
pots prefix number 12345
```

This prefix is added to any number dialed for analog or modem calls that do not match the prefix filter.

Related Commands

Command	Description
pots prefix filter	Sets a filter that prevents a dial prefix from being added to a dialed number when the digits in the dialed number match the filter.

pots ringing-freq

To specify the frequency on the Cisco 800 series router at which connected telephones, fax machines, or modems ring, use the **pots ringing-freq** command in global configuration mode. To disable the specified 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.

Usage Guidelines This command applies to Cisco 800 series routers.

Examples The following example sets the ringing frequency to 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

<i>interval</i>	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.

Usage Guidelines

This command applies to Cisco 800 series routers.

Examples

The following example sets the interval of silence to 10 seconds:

```
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 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

Local (router supplies the tones)

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco 800 series.

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 sets the tone source to remote:

```
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 reset to the default, use the **no** form of the command.

pre-dial delay *seconds*

no pre-dial delay

Syntax Description	<i>seconds</i>	Delay, in seconds, before signaling begins. Range is from 0 to 10. Default is 1.
---------------------------	----------------	--

Defaults	1 second
-----------------	----------

Command Modes	Voice-port configuration
----------------------	--------------------------

Command History	Release	Modification
	11.(7)T	This command was introduced on the Cisco 3600 series.
	12.0(2)T	This command was integrated into Cisco IOS Release 12.0(2)T.

Usage Guidelines	<p>This command applies to Cisco 3600 series routers.</p> <p>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. This command allows a signaling delay.</p>
-------------------------	---

Examples	<p>The following example sets a predial delay value of 3 seconds on the FXO port of a Cisco 3600 series router:</p> <pre>voice-port 1/0/0 pre-dial delay 3</pre>
-----------------	--

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 (dial-peer)

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, use the **no** form of this command.

preference *value*

no preference

Syntax Description	<i>value</i>	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.
	12.0(3)T	This command was integrated into Cisco IOS Release 12.0(3)T and implemented on the Cisco 2600 series and Cisco 3600 series.
	12.0(4)T	This command was modified to support VoFR dial peers on the Cisco 2600 series and Cisco 3600 series.

Usage Guidelines	<p>This command applies to POTS, VoIP, VoFR, and VoATM dial peers on the Cisco MC3810.</p> <p>Use this 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.</p>
-------------------------	--



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 sets POTS dial peer 10 to a preference of 1, POTS dial peer 20 to 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:

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 is 1, 2, 3, 5, 4:

Dialpeer	destpat	preference
1	408555	0
2	4085551048	1
3	4085551	0
44085551.....	0

If the number dialed is 4085551048, the order is 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.

Command	Description
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.

preference (ephone-dn)

To set preference order for the directory number associated with a Cisco IP phone, use the **preference** command in ephone-dn configuration mode. To reset to the default, use the **no** form of this command.

preference *preference-order*

no preference *preference-order*

Syntax Description	<i>preference-order</i>	Preference order. Range is from 0 to 10, where 0 is the highest preference and 10 is the lowest preference.
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Defaults	0 (highest preference)
-----------------	------------------------

Command Modes	Ephone-dn configuration
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Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
	12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
	12.2(8)T1	This command was implemented on the Cisco 2600XM and Cisco 2691.
	12.2(11)T	This command was implemented on the Cisco 1760.

Usage Guidelines	This command sets preference order for the directory number (ephone-dn) associated with a Cisco IP phone. Use this command to indicate the preference order for matching dial peers in Cisco IP phone virtual dial-peer 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.
-------------------------	--

Examples	The following example sets a preference of 2 for the directory number 3000:
-----------------	---

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 3000
Router(config-ephone-dn)# preference 2
```

The configuration is as follows:

```
ephone-dn 4
  number 1222
  preference 0
!
!
ephone-dn 5
```

■ preference (ephone-dn)

```
number 1222
preference 1
```

In this example, number 1222 under directory number (ephone-dn) 4 has a higher preference than the number 1222 under ephone-dn 5.

Related Commands

Command	Description
ephone	Enters ephone configuration mode.
ephone-dn	Enters ephone-dn configuration mode.
huntstop	Sets the huntstop attribute for the dial-peers associated with the Cisco IP phone lines.
name	Configures a username associated with a directory number.
number	Configures a valid number for the Cisco IP phone.
preference (dial-peer)	Indicates the preferred order of a dial peer within a hunt group.
telephony-service	Enables Cisco IOS Telephony Service and enters telephony-service configuration mode.

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	<i>string</i>	Integers that represent the prefix of the telephone number associated with the specified dial peer. Valid values 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.
	12.0(4)XJ	This command was implemented on the Cisco AS5300. It and modified for store-and-forward fax.
	12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.
	12.2(4)T	This command was implemented on the Cisco 1750.
	12.2(8)T	This command was implemented on the following platforms: Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745.
	12.2(13)T	This command was supported in Cisco IOS Release 12.2(13)T and implemented on the Cisco 2600XM, Cisco ICS7750, and Cisco VG200.

Usage Guidelines Use this 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,
```

The following example specifies a prefix of 5120002:

```
Router(config-dial-peer)# prefix 5120002
```

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 to be used for a dial peer.

prefix (Annex G)

To restrict the prefixes for which the gatekeeper should query the Annex G border element (BE), use the **prefix** command in gatekeeper border element configuration mode.

```
prefix prefix* [seq | blast]
```

Syntax Description

<i>prefix*</i>	Prefix for which BEs should be queried.
seq	(Optional) Queries are sent out to the neighboring BEs sequentially.
blast	(Optional) Queries are sent out to the neighboring BEs simultaneously.

Defaults

Any time a remote zone query occurs, the BE is also queried.

Command Modes

Gatekeeper border element configuration

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T. Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 is not included in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

By default, the gatekeeper sends all remote zone requests to the BE. Use this command only if you want to restrict the queries to the BE to a specific prefix or set of prefixes.

Examples

The following example directs the gatekeeper to query the BE using a prefix of 408.

```
Router(config-gk-annexg)# prefix 408* seq
```

Related Commands

Command	Description
h323-annexg	Enables the BE on the gatekeeper and enters border element configuration mode.

pri-group timeslots

To specify an ISDN PRI group on a channelized T1 or E1 controller, and to release the ISDN PRI signaling time slot, use the **pri-group timeslots** command in controller configuration mode. To remove or change the ISDN PRI configuration, use the **no** form of this command.

```
pri-group timeslots timeslot-range [nfas_d {backup | none | primary {nfas_int number | nfas_group number | rlm-group number}}] | service]
```

```
no pri-group timeslots timeslot-range [nfas_d {backup | none | primary {nfas_int number | nfas_group number | rlm-group number}}] | service]
```

Syntax Description	<p><i>timeslot-range</i></p> <p>A value or range of values for time slots on a T1 or E1 controller that consists of an ISDN PRI group. Use a hyphen to indicate a range.</p> <p>Note Groups of time slot ranges separated by commas (1-4,8-23 for example) are also accepted.</p> <hr/> <p>nfas_d {backup none primary}</p> <p>(Optional) Configures the operation of the ISDN PRI D channel.</p> <ul style="list-style-type: none"> • backup—The D-channel time slot is used as the Non-Facility Associated Signaling (NFAS) D backup. • none—The D-channel time slot is used as an additional B channel. • primary—The D-channel time slot is used as the NFAS D primary. The primary keyword requires further interface and group configuration: <ul style="list-style-type: none"> – nfas_int <i>number</i>—Specifies the provisioned NFAS interface as a value; value is a number from 0 to 8. – nfas_group <i>number</i>—Specifies the NFAS group. – rlm-group <i>number</i>—Specifies the Redundant Link Manager (RLM) group and release the ISDN PRI signaling channel. <hr/> <p>primary {nfas_int <i>number</i> nfas_group <i>number</i> rlm-group <i>number</i>}</p> <hr/> <p>service</p> <p>(Optional) Configures service type mgcp for Media Gateway Control Protocol service.</p>
---------------------------	---

Defaults No ISDN PRI group is configured. The switch type is automatically set to the National ISDN switch type (**primary-ni** keyword) when the **pri-group timeslots** command is configured with the **rlm-group** subkeyword.

Command Modes Controller configuration

Command History	Release	Modification
	11.0	This command was introduced.
	11.3	This command was enhanced to support NFAS.
	12.0(2)T	This command was implemented on the Cisco MC3810 multiservice concentrator.
	12.0(7)XK	This command was implemented on 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.
	12.2(8)B	This command was modified with the rlm-group subkeyword to support release of the ISDN PRI signaling channels.
	12.2(15)T	The modifications in Cisco IOS Release 12.2(8)B were integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

The **pri-group** command supports the use of DS0 time slots for Signaling System 7 (SS7) links, and therefore the coexistence of SS7 links and PRI voice and data bearer channels on the same T1 or E1 span. In these configurations, the command applies to voice applications.

In SS7-enabled Voice over IP (VoIP) configurations when an RLM group is configured, High-Level Data Link Control (HDLC) resources allocated for ISDN signaling on a digital subscriber line (DSL) interface are released and the signaling slot is converted to a bearer channel (B24). The D channel will be running on IP. The chosen D-channel time slot can still be used as a B channel by using the **isdn rlm-group** interface configuration command to configure the NFAS groups.

NFAS allows a single D channel to control multiple PRI interfaces. Use of a single D channel to control multiple PRI interfaces frees one B channel on each interface to carry other traffic. A backup D channel can also be configured for use when the primary NFAS D channel fails. When a backup D channel is configured, any hard system failure causes a switchover to the backup D channel and currently connected calls remain connected.

NFAS is supported only with a channelized T1 controller and, as a result, must be ISDN PRI capable. Once the channelized T1 controllers are configured for ISDN PRI, only the NFAS primary D channel must be configured; its configuration is distributed to all members of the associated NFAS group. Any configuration changes made to the primary D channel will be propagated to all NFAS group members. The primary D channel interface is the only interface shown after the configuration is written to memory.

The channelized T1 controllers on the router must also be configured for ISDN. The router must connect to either an AT&T 4ESS, Northern Telecom DMS-100 or DMS-250, or National ISDN switch type.

The ISDN switch must be provisioned for NFAS. The primary and backup D channels should be configured on separate T1 controllers. The primary, backup, and B-channel members on the respective controllers should be the same configuration as that configured on the router and ISDN switch. The interface ID assigned to the controllers must match that of the ISDN switch.

You can disable a specified channel or an entire PRI interface, thereby taking it out of service or placing it into one of the other states that is passed in to the switch using the **isdn service** interface configuration command.

In the event that a controller belonging to an NFAS group is shut down, all active calls on the controller that is shut down will be cleared (regardless of whether the controller is set to primary, backup, or none), and one of the following events will occur:

- If the controller that is shut down is configured as the primary and no backup is configured, all active calls on the group are cleared.
- If the controller that is shut down is configured as the primary, and the active (In service) D channel is the primary and a backup is configured, then the active D channel changes to the backup controller.
- If the controller that is shut down is configured as the primary, and the active D channel is the backup, then the active D channel remains as backup controller.
- If the controller that is shut down is configured as the backup, and the active D channel is the backup, then the active D channel changes to the primary controller.

The expected behavior in NFAS when an ISDN D channel (serial interface) is shut down is that ISDN Layer 2 should go down but keep ISDN Layer 1 up, and that the entire interface will go down after the amount of seconds specified for timer T309.

**Note**

The active D channel changeover between primary and backup controllers happens only when one of the link fails and not when the link comes up. The T309 timer is triggered when the changeover takes place.

Examples

The following example configures T1 controller 1/0 for PRI and for the NFAS primary D channel. This primary D channel controls all the B channels in NFAS group 1.

```
controller t1 1/0
  framing esf
  linecode b8zs
  pri-group timeslots 1-24 nfas_d primary nfas_int 0 nfas_group 1
```

The following example specifies ISDN PRI on T1 slot 1, port 0, and configures voice and data bearer capability on time slots 2 through 6:

```
isdn switch-type primary-4ess
controller t1 1/0
  framing esf
  linecode b8zs
  pri-group timeslots 2-6
```

The following example configures a standard ISDN PRI interface:

```
! Standard PRI configuration:
controller t1 1
  pri-group timeslots 1-23 nfas_d primary nfas_int 0 nfas_group 0
  exit

! Standard ISDN serial configuration:
interface serial1:23
! Set ISDN parameters:
  isdn T309 4000
  exit
```

The following example configures a dedicated T1 link for SS7-enabled VoIP:

```
controller T1 1
  pri-group timeslots 1-23 nfas_d primary nfas_int 0 nfas_group 0
  exit
```

```

! In a dedicated configuration, we assume the 24th timeslot will be used by ISDN.
! Serial interface 0:23 is created for configuring ISDN parameters.
interface Serial:24
! The D channel is on the RLM.
 isdn rlm 0
 isdn T309 4000
exit

```

The following example configures a shared T1 link for SS7-enabled VoIP. The **rlm-group 0** portion of the **pri-group timeslots** command releases the ISDN PRI signaling channel.

```

controller T1 1
 pri-group timeslots 1-3 nfas_d primary nfas_int 0 nfas_group 0 rlm-group 0
 channel group 23 timeslot 24
end

! D-channel interface is created for configuration of ISDN parameters:
interface Dchannel1
 isdn T309 4000
end

```

Related Commands

Command	Description
controller	Configures a T1 or E1 controller and enters controller configuration mode.
interface Dchannel	Specifies an ISDN D-channel interface for VoIP applications that require release of the ISDN PRI signaling time slot for RLM configurations.
interface serial	Specifies a serial interface created on a channelized E1 or channelized T1 controller for ISDN PRI signaling.
isdn rlm-group	Specifies the RLM group number that ISDN will start using.
isdn switch-type	Specifies the central office switch type on the ISDN PRI interface.
isdn timer t309	Changes the value of the T309 timer to clear network connections and release the B channels when there is no signaling channel active, that is, when the D channel has failed and cannot recover by switching to an alternate D channel. Calls remain active and able to transfer data when the D channel fails until the T309 timer expires. The T309 timer is canceled when D-channel failover succeeds.
show isdn nfas group	Displays all the members of a specified NFAS group or all NFAS groups.

pri-group (pri-slt)

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

```
pri-group [timeslots timeslot-range [nfas_d [backup | none | primary [nfas_int number]]
  [nfas-group number [iua as-name]]]
```

```
no pri-group
```

Syntax Description

timeslots <i>timeslot-range</i>	Specifies a single range of timeslot values in the PRI group. For T1, the allowable range is from 1 to 23. For E1, the allowable range is from 1 to 31.
nfas_d	Specifies the operation of the D channel timeslot.
backup	(Optional) Specifies that the operation of the D channel timeslot on this controller is the NFAS D backup.
none	(Optional) Specifies that the D channel timeslot is used as an additional B channel.
primary	Specifies that the D channel timeslot on this controller in NFAS D.
nfas_int <i>range</i>	Specifies the provisioned NFAS interface value. Valid values range from 0 to 32.
nfas-group <i>number</i>	Specifies the NFAS group and the NFAS group number. Valid values range from 0 to 31.
iua <i>as-name</i>	Binds the Non-Facility Associated Signaling (NFAS) group to the ISDN User Adaptation Layer (IUA) application server (AS).

Defaults

No ISDN-PRI group is configured.

Command Modes

Controller configuration

Command History

Release	Modification
12.2(11)T	This command was introduced.
12.2(15)T	This command was integrated on the Cisco 2420, Cisco 2600 series, Cisco 3600 series, and Cisco 3700 series; and Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 network access server (NAS) platforms.

Usage Guidelines

The **pri-group (pri-slt)** command provides another way to bind a D channel to a specific IUA AS. This option allows the RLM group to be configured at the **pri-group** level instead of in the D channel configuration. For example, a typical configuration would look like the following:

```
controller t1 1/0/0
```

```
pri-group timeslots 1-24 nfas_d pri nfas_int 0 nfas_group 1 iua asname
```

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

When configuring NFAS, you use an extended version of the **pri-group** command to specify the following values for the associated channelized T1 controllers configured for ISDN:

- The range of PRI timeslots to be under the control of the D channel (timeslot 24).
- The function to be performed by timeslot 24 (primary D channel, backup, or none); the latter specifies its use as a B channel.
- The group identifier number for the interface under the control of a particular D channel.

The **iua** keyword is used to bind an NFAS group to the IUA AS.

When binding the D channel to an IUA AS, the *as-name* must match the name of an AS set up during IUA configuration.

Before you can modify a PRI group on a Media Gateway Controller (MGC), you must first shut down the D channel.

The following shows how to shut down the D channel:

```
Router# configure terminal
```

```
Enter configuration commands, one per line. End with CNTL/Z.
```

```
Router(config)# interface Dchannel13/0:1
```

```
Router(config-if)# shutdown
```

Examples

The following example configures the NFAS primary D channel on one channelized T1 controller, and binds the D channel to an IUA AS. This example uses the Cisco AS5400 and applies to T1, which has 24 timeslots and is used mainly in North America and Japan:

```
Router(config-controller)# pri-group timeslots 1-23 nfas-d primary nfas-int 0 nfas-group 1 iua as5400-4-1
```

The following example applies to E1, which has 32 timeslots and is used by the rest of the world:

```
Router(config-controller)# pri-group timeslots 1-31 nfas-d primary nfas-int 0 nfas-group 1 iua as5400-4-1
```

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

```
Router(config)# controller E1 4/1  
Router(config-controller)# pri-group timeslots 1-7,16
```

In the following example, the **rlm-timeslot** keyword automatically creates interface serial 4/7:11 (4/7:0:11 if you are using the CT3 card) for the D channel object on a Cisco AS5350. You can choose any timeslot other than 24 to be the virtual container for the D channel parameters for ISDN.

```
Router(config-controller)# pri-group timeslots 1-23 nfas-d primary nfas-int 0 nfas-group 0 rlm-timeslot 3
```

Related Commands

Command	Description
isdn switch-type	Configures the Cisco 2600 series 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

<i>pbx-ip-address</i>	IP address of the NEC PBX.
<i>pbx-ip-host-name</i>	Host name of the NEC PBX.
pbx-port <i>number</i>	Port number for the PBX. Range is from 49152 to 65535. Default is 55000. If this value is already in use, the next greater value is used.

Defaults

PBX port number: 55000

Command Modes

Controller configuration

Command History

Release	Modification
12.0(7)T	This command was introduced on the Cisco AS5300.
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 directs 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 VoIP gateway, use the **progress_ind** command in dial-peer configuration mode. To reset to the default, 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 <i>pi-number</i>	Progress indicator that is sent in all messages of the specified type from the outbound dial peer. For setup messages from 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 following platforms: Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco 7500 series, Cisco MC3810, Cisco AS5300, and Cisco AS5800.
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.
12.2(2)XA	This command was implemented on the Cisco AS5400 and Cisco AS5350.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

This 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 does 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 POTS dial peer 3:

```
dial-peer voice 3 pots
 destination-pattern 55275
 progress_ind progress enable 1
 progress_ind connect enable 1
```

Related Commands

Command	Description
destination-pattern	Specifies the telephone number that is used to identify the outbound dial peer for the call.
dial-peer voice	Enters dial-peer configuration mode and configures a VoIP or POTS dial peer.

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	<i>port-number</i>	RLM port number. See Table 87 for the port number choices.
--------------------	--------------------	--

Defaults	3000
----------	------

Command Modes	RLM configuration
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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.

Command	Description
shutdown (RLM)	Shuts down all of the links under the RLM group.
timer	Overwrites the default setting of timeout values.

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.

Usage Guidelines If the multimedia interface is not enabled using this command or if no gatekeeper is available, starting the proxy allows it to attempt to locate these resources. No calls are accepted until the multimedia interface and the gatekeeper are found.

Examples The following example turns on the proxy feature:

```
proxy h323
```

query-interval

To configure the interval at which the local border element (BE) queries the neighboring BE, use the **query-interval** command in Annex G Neighbor BE Configuration mode. To remove the interval, use the **no** form of this command.

query-interval *query-interval*

no query-interval

Syntax Description

query-interval Frequency, in minutes, at which this BE should query the specified neighbor BE for descriptors. Default is 30. A value of 0 disables periodic querying.

Note If caching is disabled, this value is ignored.

Defaults

30 minutes

Command Modes

Annex G Neighbor BE Configuration

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

Use this command to configure the interval at which the local BE queries the neighboring BE. Use this command only if you want a query interval other than 30 minutes.

Examples

The following example sets the query interval to 45 minutes:

```
Router(config-annexg-neighbor)# query-interval 45
```

Related Commands

Command	Description
emulate	Configures the local BE to cache the descriptors received from its neighbors. If caching is enabled, the neighbors are queried at the specified interval for their descriptors.
local	Configures the identifier for the neighbor BE.
session transport	Configures the neighbor's port number that is used for exchanging Annex G messages.