periodic-report interval through pulse-digit-detection

periodic-report interval

To configure periodic reporting parameters for gateway resource entities, use the **periodic-report interval**command in voice-class configuration mode. To disable the periodic reporting parameters configuration, use the **no** form of this command.

periodic-report interval *seconds*
no periodic-report interval *seconds*

Syntax Description

seconds	Periodic interval, in seconds. The range is from 30 to 21600.

Command Default

The periodic interval report parameters are disabled.

Command Modes

Voice-class configuration mode (config-class)

Command History

Release	Modification
15.1(2)T	This command was introduced.

Usage Guidelines

Use the **periodic-report interval** command to periodically report the status of the monitoring resources to the external entity. The triggering takes place based on the preconfigured interval value. You can use the statistics collected by this method of reporting to collect information on resource usage.

Examples

The following example shows how to configure a resource group to trigger reporting every 180 seconds:

```
Router> enable
Router# configure terminal
Router(config)# voice class resource-group 1
Router(config-class)# periodic-report interval 180
```

Related Commands

Command	Description
debug rai	Enables debugging for Resource Allocation Indication (RAI).
rai target	Configures the SIP RAI mechanism.
resource (voice)	Configures parameters for monitoring resources, use the resource command in voice-class configuration mode.
show voice class resource-group	Displays the resource group configuration information for a specific resource group or all resource groups.
voice class resource-group	Enters voice-class configuration mode and assigns an identification tag number for a resource group.

permit hostname (SIP)

To store hostnames used during validation of initial incoming INVITE messages, use the **permit hostname** command in SIP-UA configuration mode or voice class tenant configuration mode. To remove a stored hostname, use the **no** form of this command.

permit hostname dns: *domain-name*
no permit hostname

Syntax Description

dns: *domain-name*	Domain name in DNS format. Domain names can be up to 30 characters in length; domain names exceeding 30 characters will be truncated.

Command Modes

SIP-UA configuration

Voice class tenant configuration (config-class)

Command History

Release	Modification
12.4(9)T	This command was introduced.
15.6(2)T and IOS XE Denali 16.3.1	This command is now available under voice class tenants.
Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models.

Usage Guidelines

The **permit hostname** command allows you to specify hostnames in FQDN (fully qualified domain name) format used during validation of incoming initial INVITE messages. The length of the hostname can be up to 30 characters; hostnames exceeding 30 characters will be truncated. You can store up to 10 hostnames by repeating the **permit hostname** command.

Once configured, initial INVITEs with a hostname in the requested Universal Resource Identifier (URI) are compared to the configured list of hostnames. If there is a match, the INVITE is processed; if there is a mismatch, a "400 Bad Request - Invalid Host" is sent, and the call is rejected.

Note Before Software Release 12.4(9)T, hostnames in incoming INVITE-request messages were only validated when they were in IPv4 format; now you can specify hostnames in fully qualified domain name (FQDN) format.

Examples

The following example show you how to set the hostname to sip.example.com:

```
Router(config)# sip-ua
Router(conf-sip-ua)# permit hostname dns:sip.example.com
```

phone context

To filter out uniform resource identifiers (URIs) that do not contain a phone-context field that matches the configured pattern, use the **phone context** command in voice URI class configuration mode. To remove the pattern, use the **no** form of this command.

phone context *phone-context-pattern*
no phone context

Syntax Description

phone-context-pattern	Cisco IOS regular expression pattern to match against the phone context field in a SIP or TEL URI. Can be up to 32 characters.

Command Default

No default behavior or values

Command Modes

Voice URI class configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Usage Guidelines

- Use this command with at least one other pattern-matching command, such as **host**, **phone number**, or **user-id**; using it alone does not result in any matches on the voice class.
- You cannot use this command if you use the **pattern** command in the voice class. The **pattern** command matches on the entire URI, whereas this command matches only a specific field.

Examples

The following example sets a match on the phone context in the URI voice class:

```
voice class uri 10 tel
 phone number ^408
 phone context 555
```

Related Commands

Command	Description
destination uri	Specifies the voice class to use for matching the destination URI that is supplied by a voice application.
host	Matches a call based on the host field in a SIP URI.
incoming uri	Specifies the voice class used to match a VoIP dial peer to the URI of an incoming call.
pattern	Matches a call based on the entire SIP or TEL URI.
phone number	Matches a call based on the phone number field in a TEL URI.

Command	Description
show dialplan incall uri	Displays which dial peer is matched for a specific URI in an incoming voice call.
show dialplan uri	Displays which outbound dial peer is matched for a specific destination URI.
user-id	Matches a call based on the user-id field in the SIP URI.
voice class uri	Creates or modifies a voice class for matching dial peers to calls containing a SIP or TEL URI.

phone number

To match a call based on the phone-number field in a telephone (TEL) uniform resource identifier (URI), use the **phone number** command in voice URI class configuration mode. To remove the pattern, use the **no** form of this command.

phone number *phone-number-pattern*
no phone number

Syntax Description

phone-number-pattern	Cisco IOS regular expression pattern to match against the phone-number field in a TEL URI. Can be up to 32 characters.

Command Default

No default behavior or values

Command Modes

Voice URI class configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Usage Guidelines

- Use this command only in a voice class for TEL URIs.
- You cannot use this command if you use the **pattern** command in the voice class. The **pattern** command matches on the entire URI, whereas this command matches only a specific field.

Examples

The following example defines a voice class that matches on the phone number field in a TEL URI:

```
voice class uri r101 tel
 phone number ^408
```

Related Commands

Command	Description
debug voice uri	Displays debugging messages related to URI voice classes.
destination uri	Specifies the voice class to use for matching the destination URI that is supplied by a voice application.
incoming uri	Specifies the voice class used to match a VoIP dial peer to the URI of an incoming call.
pattern	Matches a call based on the entire SIP or TEL URI.
phone context	Filters out URIs that do not contain a phone-context field that matches the configured pattern.
voice class uri	Creates or modifies a voice class for matching dial peers to calls containing a SIP or TEL URI.

phone-proxy (dial peer)

To configure the phone proxy for the related dial peer, use the **phone-proxy** command in dial peer configuration mode. To remove the phone proxy for the related dial peer use the **no** form of the command.

phone-proxy *phone-proxy-name* **signal-addr ipv4** *ipv4-address* **cucm ipv4** *ipv4-address*

Syntax Description

phone-proxy-name	Name of the specific phone proxy.
signal-addr ipv4 *ipv4-address*	Specifies the SIP signal IPv4 address of the access side.
cucm ipv4 *ipv4-address*	Specifies the call manager server IPv4 address.

Command Modes

Dial peer configuration (config-dial-peer)

Command History

Release	Modification
15.3(3)M	This command was introduced.

Usage Guidelines

Example

The following example shows how to configure a phone proxy for the related dial peer:

```
Device(config)# dial-peer voice 1 voip
Device(config-dial-peer)# phone-proxy pp signal-addr ipv4 10.0.0.8 cucm ipv4 198.51.100.1
```

pickup direct

To define a feature code for a Feature Access Code (FAC) to access Pickup Direct on an analog phone, use the **pickup direct**command in STC application feature access-code configuration mode. To return the code to its default, use the **no** form of this command.

pickup direct *keypad-character*
no pickup direct

Syntax Description

keypad-character	Character string that can be dialed on a telephone keypad (0-9, *, #). Default: 6. Before Cisco IOS Release 12.4(20)YA, this is a single character. In Cisco IOS Release 12.4(20)YA and later releases, the string can be any of the following: • A single character (0-9, *, #) • Two digits (00-99) • Two to four characters (0-9, *, #) and the leading or ending character must be an asterisk (*) or number sign (#)

Command Default

The default value is 6.

Command Modes

STC application feature access-code configuration (config-stcapp-fac)

Command History

Release	Modification
12.4(2)T	This command was introduced.
12.4(20)YA	The length of the *keypad-character* argument was changed to 1 to 4 characters.
12.4(22)T	This command was integrated into Cisco IOS Release 12.4(22)T.

Usage Guidelines

This command changes the value of the feature code for Pickup Direct from the default (6) to the specified value.

In Cisco IOS Release 12.4(20)YA and later releases, if the length of the *keypad-character* argument is at least two characters and the leading or ending character of the string is an asterisk (*) or a number sign (#), phone users are not required to dial a prefix to access this feature. Typically, phone users dial a feature access code (FAC) consisting of a prefix plus a feature code, for example **6. If the feature code is 78#, the phone user dials only 78#, without the FAC prefix, to access the corresponding feature.

In Cisco IOS Release 12.4(20)YA and later releases, if you attempt to configure this command with a value that is already configured for another feature code, a speed-dial code, or the Redial FSD, you receive a message. If you configure a duplicate code, the system implements the first matching feature in the order of precedence shown in the output of the **show stcapp feature codes** command.

In Cisco IOS Release 12.4(20)YA and later releases, if you attempt to configure this command with a value that precludes or is precluded by another FAC, a speed-dial code, or the Redial FSD, you receive a message. If you configure a feature code to a value that precludes or is precluded by another code, the system always executes the call feature with the shortest code and ignores the longer code. For example, #1 will always preclude #12 and #123. You must configure a new value for the precluded code in order to enable phone user access to that feature.

To display a list of all FACs, use the **show stcapp feature codes** command.

Note This FAC is not supported by Cisco Unified Communications Manager.

Examples

The following example shows how to change the value of the feature code for Pickup Direct from the default (6). This configuration also changes the value of the prefix for all FACs from the default (**) to ##. With this configuration, a phone user must press ##3 on the keypad and then the ringing extension number to pick up an incoming call.

```
Router(config)# stcapp feature access-code
Router(config-stcapp-fac)# prefix ##
Router(config-stcapp-fac)# pickup direct 3
Router(config-stcapp-fac)# exit
```

Related Commands

Command	Description
pickup group	Defines a feature code for a feature access code (FAC) to Group Call Pickup from another group.
pickup local	Defines a feature code for a feature access code (FAC) to Group Call Pickup from the local group.
prefix (stcapp-fac)	Defines the prefix for feature access codes (FACs).
show stcapp feature codes	Displays all feature access codes (FACs).
stcapp feature access-code	Enables feature access codes (FACs) in STC application and enters STC application feature access-code configuration mode for changing values of the prefix and features codes from the default.

pickup group

To define a feature code for a feature access code (FAC) to access Group Call Pickup on an analog phone, use the **pickup group** command in STC application feature access-code configuration mode. To return the code to its default, use the **no** form of this command.

pickup group *keypad-character*
no pickup group

Syntax Description

keypad-character	Character string that can be dialed on a telephone keypad (0-9, *, #). Default: 4. Before Cisco IOS Release 12.4(20)YA, this is a single character. In Cisco IOS Release 12.4(20)YA and later releases, the string can be any of the following: • A single character (0-9, *, #) • Two digits (00-99) • Two to four characters (0-9, *, #) and the leading or ending character must be an asterisk (*) or number sign (#)

Command Default

The default value is 4.

Command Modes

STC application feature access-code configuration (config-stcapp-fac)

Command History

Release	Modification
12.4(2)T	This command was introduced.
12.4(20)YA	The length of the *keypad-character* argument was changed to 1 to 4 characters.
12.4(22)T	This command was integrated into Cisco IOS Release 12.4(22)T.

Usage Guidelines

This command changes the value of the feature code for Pickup Direct from the default (4) to the specified value.

In Cisco IOS Release 12.4(20)YA and later releases, if the length of the *keypad-character* argument is at least two characters and the leading or ending character of the string is an asterisk (*) or a number sign (#), phone users are not required to dial a prefix to access this feature. Typically, phone users dial a special feature access code (FAC) consisting of a prefix plus a feature code, for example **4. If the feature code is 78#, the phone user dials only 78#, without the FAC prefix, to access the corresponding feature.

In Cisco IOS Release 12.4(20)YA and later releases, if you attempt to configure this command with a value that is already configured for another feature code, a speed-dial code, or the Redial FSD, you receive a message. If you configure a duplicate code, the system implements the first matching feature in the order of precedence shown in the output of the **show stcapp feature codes** command.

In Cisco IOS Release 12.4(20)YA and later releases, if you attempt to configure this command with a value that precludes or is precluded by another feature code, a speed-dial code, or the Redial FSD, you receive a message. If you configure a feature code to a value that precludes or is precluded by another code, the system always executes the call feature with the shortest code and ignores the longer code. For example, #1 will always preclude #12 and #123. You must configure a new value for the precluded code in order to enable phone user access to that feature.

To display a list of all FACs, use the **show stcapp feature codes** command.

Examples

The following example shows how to change the value of the feature code for Pickup Direct from the default (4). This configuration also changes the value of the prefix for all FACs from the default (**) to ##. After these values are configured, a phone user must press ##3 on the keypad, then the pickup-group number for the ringing extension number to pick up the incoming call.

```
Router(config)# stcapp feature access-code
Router(config-stcapp-fac)# prefix ##
Router(config-stcapp-fac)# pickup direct 3
Router(config-stcapp-fac)# exit
```

Related Commands

Command	Description
pickup direct	Defines a feature code for a feature access code (FAC) for Direct Call Pickup of a ringing extension number.
pickup local	Defines a feature code for a feature access code (FAC) for Group Call Pickup to pick up an incoming call from the local group.
prefix (stcapp-fac)	Defines the prefix for feature access codes (FACs).
show stcapp feature codes	Displays all feature access codes (FACs).
stcapp feature access-code	Enables feature access codes (FACs) and enters STC application feature access-code configuration mode for changing values of the prefix and features codes from the default.

pickup local

To define a a feature code for a Feature Access Code (FAC) to access Group Call Pickup for a local group on an analog phone, use the **pickup local** command in STC application feature access-code configuration mode. To return the code to its default, use the **no** form of this command.

pickup local *keypad-character*
no pickup local

Syntax Description

keypad-character	Character string that can be dialed on a telephone keypad. Default: 3. Before Cisco IOS Release 12.4(20)YA, this is a single character. In Cisco IOS Release 12.5(20)YA and later releases, the string can be any o the following: • A single character (0-9, *, #) • Two digits (00-99) • Two to four characters (0-9, *, #) and the leading or ending character must be an asterisk (*) or number sign (#)

Command Default

The default value is 3.

Command Modes

STC application feature access-code configuration (config-stcapp-fac)

Command History

Release	Modification
12.4(2)T	This command was introduced.
12.4(20)YA	The length of the *keypad-character* argument was changed to 1 to 4 characters.
12.4(22)T	This command was integrated into Cisco IOS Release 12.4(22)T.

Usage Guidelines

This command changes the value of the feature code for Local Group Pickup from the default (3) to the specified value.

In Cisco IOS Release 12.4(20)YA and later releases, if the length of the *keypad-character* argument is at least two characters and the leading or ending character of the string is an asterisk (*) or a number sign (#), phone users are not required to dial a prefix to access this feature. Typically, phone users dial a special feature access code (FAC) consisting of a prefix plus a feature code, for example **3. If the feature code is 78#, the phone user dials only 78#, without the FAC prefix, to access the corresponding feature.

In Cisco IOS Release 12.4(20)YA and later releases, if you attempt to configure this command with a value that is already configured for another feature code or speed-dial code, or for the Redial FSD, you receive a message. If you configure a duplicate code, the system implements the first matching feature in the order of precedence shown in the output of the **show stcapp feature codes** command.

In Cisco IOS Release 12.4(20)YA and later releases, if you attempt to configure this command with a value that precludes or is precluded by another feature code or speed-dial code, or by the Redial FSD, you receive a message. If you configure a feature code to a value that precludes or is precluded by another code, the system always executes the call feature with the shortest code and ignores the longer code. For example, #1 will always preclude #12 and #123. You must configure a new value for the precluded code in order to enable phone user access to that feature.

To display a list of all FACs, use the **show stcapp feature codes** command.

Examples

The following example shows how to change the value of the feature code for Pickup Direct from the default (3). This configuration also changes the value of the prefix for all FACs from the default (**) to ##. With this configuration, a phone user must press ##9 on the keypad to pick up an incoming call in the same group as this extension number.

```
Router(config)# stcapp feature access-code
Router(config-stcapp-fac)# prefix ##
Router(config-stcapp-fac)# pickup local 9
Router(config-stcapp-fac)# exit
```

Related Commands

Command	Description
pickup direct	Defines a feature code for a feature access code (FAC) for Direct Call Pickup of a ringing extension number.
pickup group	Defines a feature code for a feature access code (FAC) for Group Call Pickup to pick up an incoming call from another group.
prefix (stcapp-fac)	Defines the prefix for feature access codes (FACs).
show stcapp feature codes	Displays all feature access codes (FACs).
stcapp feature access-code	Enables feature access codes (FACs) in STC application and enters STC application feature access-code configuration mode for changing values of the prefix and features codes from the default.

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 *milliseconds*	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 *milliseconds*	(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 200 ms.
minimum	(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: • **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** -- 40 ms. Use when there are high jitter conditions in the network.
nominal *milliseconds*	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. 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 60. 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 200 ms. • For medium-complexity codec configurations, the highest nominal value is 150 ms.

nominal *milliseconds* (continued)	For conference calls when you are using DSPware version 4.1.33 or a later version, the following values are allowed: • The first decoder stream can be assigned a nominal value as high as 200 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). When the playout-delay mode is configured for fixed operation and setting the expected jitter buffer size with the nominal value, the minimum effective value for the playout delay will depend on the codec in use and the configured minimum value. • When the **playout-delay minimum low** is configured the minimum actual jitter buffer size will be 30ms even when setting the nominal to a value lower than 30msec. • When the **playout-delay minimum default**, the minimum jitter buffer size when running in fixed mode will be 60ms. When fixed mode is configured, there is a 10msec added to the nominal value when setting the jitter buffer when configured for G.729 and a 5ms added using G.711 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 200 ms for the first decoder stream and 150 ms for subsequent decoder streams. **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.

Command Default

fax --300 milliseconds**maximum**--200 milliseconds**minimum**--default (40 milliseconds)**nominal**--60 milliseconds

Command Modes

Dial peer configuration (config-dial-peer)

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.
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.

Release	Modification
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 *milliseconds*	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 *milliseconds*	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 *milliseconds*	Initial (and minimum allowed) delay time that the DSP inserts before playing out voice packets, in milliseconds. Range is from 40 to 200. Default is 80.

Command Default

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
 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 1/0/0:

```
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.

Command Default

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.

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.

Command Default

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	Thiscommand 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.

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.

police profile

To apply the media bandwidth policing profile to a media class, use the **police profile** command in media class configuration mode. To disable the configuration, use the **no** form of this command.

police profile *tag*
no police profile

Syntax Description

tag	Media profile police tag. The range is from 1 to 10000.

Command Default

The media bandwidth policing profile is not applied to a media class.

Command Modes

Media class configuration (cfg-mediaclass)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

Applying the media bandwidth policing profile at the dial peer level involves two actions; applying the profile for a media class and then applying the corresponding media class to a dial peer. Use the **police profile** command to apply the media bandwidth policing profile to a media class.

Examples

The following example shows how to apply the media bandwidth policing profile to a media class:

```
Router> enable
Router# configure terminal
Router(config)# media class 1
Router(cfg-mediaclass)# police profile 1
```

Related Commands

Command	Description
media-class	Applies the media class at the dial peer level.
snmp-server enable traps voice media-policy	Enables SNMP media policy voice traps at the global level.
snmp enable peer-trap media-policy	Enables SNMP media policy voice traps at the dial peer level.

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

neighbor -port	Port number of the neighbor. This number is used for exchanging Annex G messages. The default port number is 2099.

Command Default

2099

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. 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.

Usage Guidelines

When cofiguring the **no port** command the *neighbor-port*argument is not used.

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

Cisco AS5300 and Cisco AS5800
port *controller-number*:**D**
no port *controller-number*:**D**

Cisco uBR92x Series
port *slot*/*subunit*/*port*
no port *slot*/*subunit*/*port*

Syntax Description

slot -number	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 the VIC has been installed.
port	Voice port number. Valid entries are 0 and 1.

slot -number	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.
subunit -number	Subunit on the VIC in which the voice port is located. Valid entries are 0 and 1.
port	Voice port number. Valid entries are 0 and 1.
slot	Router location in which the voice port adapter is installed. Valid entries are 0 and 3.
port	Voice interface card location. Valid entries are 0 and 3.
ds0 -group-number	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.

controller -number	The T1 or E1 controller.
:D	Indicates the D channel associated with the ISDN PRI.

slot/subunit/port	The analog voice port. Valid entries for the *slot/subunit/port* are as follows: • *slot* -- A router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • *subunit* -- A VIC in which the voice port is located. Valid entries are 0 and 1. (The VIC fits into the voice network module.) • *port*-- An analog voice port number. Valid entries are 0 and 1.

Command Default

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.
12.4(22)T	Support for IPv6 was added.

Usage Guidelines

This command enables 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 EC feature on the Cisco AS5300.

Examples

The following example associates POTS dial peer 10 with voice port 1, which is located on subunit 0 and accessed through port 0:

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

The following example associates POTS dial peer 10 with voice port 0:D:

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

The following example associates POTS dial peer 10 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 inMGCP 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

port -number	Voice port or DS0-group number to be used as an MGCP endpoint associated with an MGCP profile.

Command Default

No default behavior or values

Command Modes

MGCP profile configuration

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

This command is used when values for an MGCP profile are configured.

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.

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.

Examples

The following example associates an analog voice port with an MGCP profile on a Cisco uBR925 platform:

```
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 (supplementary-service)

To enter the supplementary-service voice-port configuration mode for associating a voice port with STC application supplementary-service features, use the **port** command in supplementary-service configuration mode. To cancel the association, use the **no** form of this command.

port *port*
no port *port*

Syntax Description

port	Location of port in Cisco ISR or Cisco VG224 Analog Phone Gateway. Syntax is platform-dependent; type **?** to determine.

Command Default

This command has no default behavior or values.

Command Modes

Supplementary-service configuration (config-stcapp-suppl-serv)

Command History

Release	Modification
12.4(20)YA	This command was introduced.
12.4(22)T	This command was integrated into Cisco IOS Release 12.4(22)T.

Usage Guidelines

This command associates an analog FXS port to STC application supplementary-service features being configured.

Examples

The following example shows how to enable Hold/Resume on analog endpoints connected to port 2/0 of a Cisco VG224.

```
Router(config)# stcapp supplementary-services
Router(config-stcapp-suppl-serv)# port 2/0
Router(config-stcapp-suppl-serv-port)# hold-resume
Router(config-stcapp-suppl-serv-port)# end
```

Related Commands

Command	Description
hold-resume	Enables Hold/Resume in Feature mode on the port being configured.

port media

To specify the serial interface to which the local video codec is connected for a local video dial peer, use the port media command in video dial-peer configuration mode. To remove any configured locations from the dial peer, use the **no** form of this command.

port media *interface*
no port media

Syntax Description

interface	Serial interface to which the local codec is connected. Valid entries are 0 and 1.

Command Default

No interface is specified

Command Modes

Video dial-peer configuration

Command History

Release	Modification
12.0(5)XK	This command was introduced for ATM video dial-peer configuration on the Cisco MC3810.
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 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-range

To specify a port range for the TFTP server, use the **port-range** command in phone-proxy configuration mode. To remove the port-range, use the **no** form of the command.

port-range *min-port* *max-port*
no port-range *min-port* *max-port*

Syntax Description

min-port	First port number of the port range.
max-port	Last port number of the port range.

Command Default

No port range is specified.

Command Modes

Phone-proxy configuration mode (config-pp-pr)

Command History

Release	Modification
15.3(3)M	This command was introduced.

Usage Guidelines

Example

The following example shows how to configure a port range for the TFTP server. The first port number is 30000 and the last port number is 40000:

```
Device(config-pp-pr)# port-range 30000 40000
```

port signal

To specify the slot location of the video dialing module (VDM) and the port location of the EIA/TIA-366 interface for signaling for a local video dial peer, use the port signal command in video dial-peer configuration mode. To remove any configured locations from the dial peer, use the **no** form of this command.

port signal *slot/port*
no port signal

Syntax Description

slot/	Slot location of the VDM. Valid values are 1 and 2.
port	Port location of the EIA/TIA-366 interface.

Command Default

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:

```
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.

Command Default

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.
show pots status	Displays the settings of the physical characteristics and other information on the telephone interfaces of a Cisco 800 series router.

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.

Command Default

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).
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.
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pots dialing-method

To specify how the router collects and sends digits dialed on your connected telephones, fax machines, or modems, use the **pots dialing-method**command in global configuration mode. To disable the specified dialing method, use the **no** form of this command.

pots dialing-method {**overlap** | **enblock**}
no pots dialing-method {**overlap** | **enblock**}

Syntax Description

overlap	The router sends each digit dialed in a separate message.
enblock	The router collects all digits dialed and sends the digits in one message.

Command Default

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.
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-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.

Command Default

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

interval	Interval, in milliseconds. Range is from 50 to 2000.

Command Default

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).
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.
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.

Command Default

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.

Command Default

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.
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 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 the table below to configure call-forwarding service.

Command Default

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 thefunctional forwarding method, use the DTMF keypad commands in the table below to configure the call-forwarding service.

Table 1: DTMF Keypad Commands for Call-Forwarding Service

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

Task	DTMF Keypad Command[1]
Activate CFNR	****61*** *number* #
Deactivate CFNR	**#61#**
Activate CFB	****67*** *number* #
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.

Command Default

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

number	Prefix filter numbers, up to a maximum of eight characters.

Command Default

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

number	Prefix, up to a maximum of five digits.

Command Default

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.

Command Default

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

interval	Number from 0 to 10 (seconds).

Command Default

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).
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 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.

Command Default

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.
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).
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

seconds	Delay, in seconds, before signaling begins. Range is from 0 to 10. Default is 1.

Command Default

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

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.

Examples

The following example sets a predial delay value of 3 seconds on the FXO port:

```
voice-port 1/0/0
 pre-dial delay 3
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timing delay -duration	Configures delay dial signal duration for a specified voice port.

preference (dial-peer)

To indicate the preferred order of an outbound 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

value	An integer from 0 to 10. A lower number indicates a higher preference. The default is 0, which is the highest preference.

Command Default

The longest matching dial peer supersedes the preference value.

Command Modes

Dial-peer configuration (dial-peer)

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 routers.
12.0(4)T	This command was modified to support Voice over Frame Relay(VoFR) dial peers on the Cisco 2600 series and Cisco 3600 series routers.
15.1(3)T	This command was modified. Support for matching different pattern types was modified.
Cisco IOS XE Amsterdam 17.2.1r	Introduced support for YANG models.

Usage Guidelines

This command applies to Plain Old Telephone Service(POTS), VoIP, VoFR, and Voice over ATM(VoATM) dial peers.

Use this command to indicate the preferred order for matching dial peers in a hunt group. Setting a preference enables the desired dial peer to be selected when multiple dial peers within a hunt group are matched for a dial string.

Note If POTS and voice-network peers are mixed in the same hunt group, the POTS dial peers must have priority over the voice-network dial peers.

The hunting algorithm preference is configurable. For example, to specify that a call processing sequence go to destination A, then to destination B, and finally to destination C, you would assign preferences (0 being the highest preference) to the destinations in the following order:

- Preference 0 to A
- Preference 1 to B

- Preference 2 to C

Use this command only on the same pattern type. For example, destination uri and destination-pattern are two different pattern types. By default, destination uri has higher preference than destination-pattern.

Examples

The following example shows how to set 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 5550150
 preference 1
 exit
dial-peer voice 20 pots
 destination-pattern 5550150
 preference 2
 exit
dial-peer voice 30 vofr
 destination-pattern 5550150
 preference 3
 exit
```

The following examples shows different dial peer configurations:

```
Dialpeer        destpat         preference              session-target
1               4085550148      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 4085550148, the order of attempts is 1, 2, 3, 5, 4:

```
Dialpeer        destpat         preference
1               408555          0
2               4085550148      1
3               4085550       0
4     4085550    0
```

The following example shows how to set POTS dial peer 10 for the destination-pattern to a preference of 0, POTS dial peer 20 for the destination uri to a preference of 1. Though destination-pattern has higher preference than destination uri, destination uri takes preference:

```
dial-peer voice 10 pots
 destination-pattern 5550158
 preference 0
 exit
dial-peer voice 20 pots
 destination uri 5550158
 preference 1
```

exit

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.

Command	Description
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.
destination uri	Specifies the voice class used to match a dial peer to the destination uniform resource identifier (URI).
dtmf-relay (Voice over Frame Relay)	Enables the generation of FRF.11 Annex A frames for a dial peer.
session protocol	Establishes a session protocol for calls between the local and remote routers via the packet network.
session target	Specifies a network-specific address for a specified dial peer or destination gatekeeper.
signal-type	Sets the signaling type to be used when connecting to a dial peer.

preemption enable

To enable preemption capability on a trunk group, use the **preemption enable** command in trunk group configuration mode. To disable preemption capabilities, use the **no** form of this command.

preemption enable
no preemption enable

Syntax Description

This command has no arguments or keywords.

Command Default

Preemption is disabled on the trunk group.

Command Modes

Trunk group configuration

Command History

Release	Modification
12.4(4)XC	This command was introduced.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.

Examples

The following command example enables preemption capabilities on trunk group test:

```
Router(config)# trunk group test
Router(config-trunk-group)# preemption enable
```

Related Commands

Command	Description
isdn integrate all	Enables integrated mode on an ISDN PRI interface.
max-calls	Sets the maximum number of calls that a trunk group can handle.
preemption guard timer	Defines time for a DDR call and allows time to clear the last call from the channel.
preemption level	Sets the preemption level of the selected outbound dial peer. Voice calls can be preempted by a DDR call with higher preemption level.
preemption tone timer	Defines the expiry time for the preemption tone for the outgoing call being preempted by a DDR backup call.

preemption guard timer

To define the time for a DDR call and to allow time to clear the last call from the channel, use the **preemption guard timer** command in trunk group configuration mode. To disable the preemption guard time, use the **no** form of this command.

preemption guard timer *value*
no preemption guard timer

Syntax Description

value	Number, in milliseconds for the preemption guard timer. The range is 60 to 500. The default is 60.

Command Default

No preemption guard timer is configured.

Command Modes

Trunk group configuration

Command History

Release	Modification
12.4(4)XC	This command was introduced.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.

Examples

The following set of commands configures a 60-millisecond preemption guard timer on the trunk group dial2.

```
Router(config)# trunk group dial2
Router(config-trunk-group)# preemption enable
Router(config-trunk-group)# preemption guard timer 60
```

Related Commands

Command	Description
isdn integrate all	Enables integrated mode on an ISDN PRI interface.
max-calls	Sets the maximum number of calls that a trunk group can handle.
preemption enable	Enables preemption capabilities on a trunk group.
preemption level	Sets the preemption level of the selected outbound dial-peer. Voice calls can be preempted by a DDR call with higher preemption level.
preemption tone timer	Sets the expiry time for the preemption tone for the outgoing call being preempted by a DDR backup call.

preemption level

To set the precedence for voice calls to be preempted by a dial-on demand routing (DDR) call for the trunk group, use the **preemption level** command in dial-peer configuration mode. To restore the default preemption level setting, use the **no** form of this command

preemption level {**flash-override** | **flash** | **immediate** | **priority** | **routine**}
no preemption level

Syntax Description

flash-override	Sets the precedence for voice calls to preemption level 0 (highest).
flash	Sets the precedence for voice calls to preemption level 1.
immediate	Sets the precedence for voice calls to preemption level 2.
priority	Sets the precedence for voice calls to preemption level 3.
routine	Sets the precedence for voice calls to preemption level 4 (lowest). This is the default.

Command Default

The preemption level default is **routine** (lowest).

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.4(4)XC	This command was introduced.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.

Examples

The following command example sets a preemption level of flash (level 1) on POTS dial-peer 20:

```
Router(config)# dial-peer voice 20 pots
Router(config-dial-peer)# preemption level flash
```

Related Commands

Command	Description
dialer preemption level	Sets the precedence for voice calls to be preempted by a DDR call for the dialer map.
isdn integrate all	Enables integrated mode on an ISDN PRI interface.
max-calls	Sets the maximum number of calls that a trunk group can handle.
preemption enable	Enables preemption capabilities on a trunk group.
preemption guard timer	Defines time for a DDR call and allows time to clear the last call from the channel.

Command	Description
preemption tone timer	Defines the expiry time for the preemption tone for the outgoing call being preempted by a DDR backup call.

preemption tone timer

To set the expiry time for the preemption tone for the outgoing call being preempted by a DDR backup call, use the **preemption tone timer** command in trunk group configuration mode. To clear the expiry time, use the **no** form of this command.

preemption tone timer *seconds*
no preemption tone timer

Syntax Description

seconds	Length of preemption tone, in seconds. Range: 4 to 30. Default: 10.

Command Default

No preemption tone timer is configured.

Command Modes

Trunk group configuration

Command History

Release	Modification
12.4(4)XC	This command was introduced.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.

Examples

The following set of commands configures a 20-second preemption tone timer on trunk group dial2.

```
Router(config)# trunk group dial2
Router(config-trunk-group)# preemption enable
Router(config-trunk-group)# preemption tone timer 20
```

Related Commands

Command	Description
isdn integrate all	Enables integrated mode on an ISDN PRI interface.
max-calls	Sets the maximum number of calls that a trunk group can handle.
preemption enable	Enables preemption capabilities on a trunk group.
preemption level	Sets the preemption level of the selected outbound dial peer. Voice calls can be preempted by a DDR call with higher preemption level.

prefix

To specify the prefix of the dialed digits for a dial peer, use the **prefix** command in dial-peer configuration mode. To disable this feature, use the **no** form of this command.

prefix *string*
no prefix

Syntax Description

string	Integers that represent the prefix of the telephone number associated with the specified dial peer. Valid values are 0 through 9 and a comma (,). Use a comma to include a pause in the prefix.

Command Default

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

prefix *****	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.

Command Default

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.

prefix (stcapp-fac)

To define a prefix for feature access codes (FACs) used with the SCCP telephony control (STC) application, use the **prefix**command in STC application feature access-code configuration mode. To return the prefix to its default, use the **no** form of this command.

prefix *prefix-string*
no prefix

Syntax Description

prefix-string	String of one to five characters that can be dialed on a telephone keypad. String must start with an asterisk (*) or a number sign (#). Default is **.

Command Default

The default value is **.

Command Modes

STC application feature access-code configuration (stcapp-fac)

Command History

Release	Modification
12.4(2)T	This command was introduced.

Usage Guidelines

This command modifies the FAC prefix from the default (**) to the specified character string.

Use the **show stcapp feature codes** command to display a list of all FACs.

Examples

The following example shows how to change the prefix for FACs from the default value (**) to two number signs (##).

```
Router(config)# stcapp feature access-code
Router(stcapp-fac)# prefix ##
Router(stcapp-fac)#
```

Related Commands

Command	Description
call forward all	Defines the feature code in the feature access code (FAC) for forwarding all calls.
call forward cancel	Defines the feature code in the feature access code (FAC) for cancelling Call Forward All.
pickup direct	Defines the feature code in the feature access code (FAC) for Directed Call Pickup.
pickup group	Defines the feature code in the feature access code (FAC) for call pickup from another group.
pickup local	Defines the feature code in the feature access code (FAC) for call pickup from the local group.

Command	Description
show stcapp feature codes	Displays all feature access codes (FACs) and all feature speed-dials (FSDs).
stcapp feature access-code	Enables feature access codes (FACs) in STC application and enters STC application feature access-code configuration mode for changing values of the prefix and features codes from the default.

prefix (stcapp-fsd)

To define a prefix for feature speed dials (FSDs) used with the SCCP telephony control (STC) application, use the **prefix** command in STC application feature speed-dial configuration mode. To return the prefix to its default, use the **no** form of this command.

prefix *prefix-string*
no prefix

Syntax Description

prefix-string	String of one to five characters (0-9, *, #) that can be dialed on a telephone keypad. String must begin with asterisk (*) or number sign(#). Default is *.

Command Default

The default value is *.

Command Modes

STC application feature speed-dial configuration (stcapp-fsd)

Command History

Release	Modification
12.4(2)T	This command was introduced.

Usage Guidelines

This command is used with the STC application, which enables certain features on analog FXS endpoints that use Skinny Client Control Protocol (SCCP) for call control. Phone users must dial the feature speed-dial (FSD) prefix string before dialing an FSD speed-dial that dials a telephone number. For example, to dial the telephone number that is stored in speed-dial position 3, a phone user dials *2.

Use this command only if you want to change the prefix from its default (*).

The **show stcapp feature codes** command displays the FSD prefix and all FSD speed-dials.

The following example shows how to change the prefix for FSDs from the default value (*) to three asterisks (***). After this value is configured, a phone user must press***2 on the keypad to dial speed-dial number 2.

```
Router(config)# stcapp feature speed-dial
Router(stcapp-fsd)# prefix ***
Router(stcapp-fsd)# speed dial from 2 to 7
Router(stcapp-fsd)# redial 9
Router(stcapp-fsd)# voicemail 8
Router(stcapp-fsd)# exit
```

Related Commands

Command	Description
redial	Defines an speed-dial code to dial again the most-recently dialed number on this phone line.
show stcapp feature codes	Displays all feature access codes (FACs) and all feature speed-dials (FSDs).
speed dial	Designates a range of feature speed-dials (FSDs) in STC application.

Command	Description
stcapp feature access-code	Enables feature speed-dials (FSDs) in STC application and enters STC application feature speed-dial configuration mode for changing values of the prefix and speed-dial codes from the default.
voicemail (stcapp-fsd)	Defines an speed-dial code to dial the voice-mail number.

preloaded-route

To enable preloaded route support for VoIP Session Initiation Protocol (SIP) calls, use the **preloaded-route**command in SIP configuration mode or voice class tenant configuration mode. To reset to the default, use the **no** form of this command.

preloaded-route [**sip-server**] **service-route system**
no preloaded-route

Syntax Description

sip-server	(Optional) Adds SIP server information to the Route header.
service-route	Adds the Service-Route information to the Route header.
system	Specifies that the preloaded route support for VoIP Session Initiation Protocol (SIP) calls use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

Command Default

Route support is not enabled.

Command Modes

SIP configuration (conf-serv-sip)

Voice class tenant configuration (config-class)

Command History

Release	Modification
12.4(22)YB	This command was introduced.
15.0(1)M	This command was integrated into Cisco IOS Release 15.0(1)M.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: **system**.

Usage Guidelines

The **voice-class preloaded-route**command, in dial-peer configuration mode, takes precedence over the **preloaded-route** command in SIP configuration mode. However, if the **voice-class preloaded-route** command is configured with the **system** keyword, the gateway uses the global settings configured by the **preloaded-route**command.

Enter SIP configuration mode after entering voice-service VoIP configuration mode, as shown in the "Examples" section.

Examples

The following example shows how to configure the system to include SIP server and Service-Route information in the Route header:

```
voice service voip
sip
 preloaded-route sip-server service-route
```

The following example shows how to configure the system to include only Service-Route information in the Route header:

```
voice service voip
```

```
sip
 preloaded-route service-route
```

The following example shows how to configure the system to include only Service-Route information in the Route header in voice class tenant configuration mode:

```
Router(config-class)# preloaded-route service-route system
```

Related Commands

Command	Description
sip	Enters SIP configuration mode from voice-service VoIP configuration mode.
voice -class preloaded-route	Enables preloaded route support for dial-peer SIP calls.

presence

To enable presence service and enter presence configuration mode, use the **presence** command in global configuration mode. To disable presence service, use the **no** form of this command.

presence
no presence

Syntax Description

This command has no arguments or keywords.

Command Default

Presence service is disabled.

Command Modes

Global configuration (config)

Command History

Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command enables the router to perform the following presence functions:

- Process presence requests from internal lines to internal lines. Notify internal subscribers of any status change.
- Process incoming presence requests from a SIP trunk for internal lines. Notify external subscribers of any status change.
- Send presence requests to external presentities on behalf of internal lines. Relay status responses to internal lines.

Examples

The following example shows how to enable presence and enter presence configuration mode to set the maximum subscriptions to 150:

```
Router(config)# presence
Router(config-presence)# max-subscription 150
```

Related Commands

Command	Description
allow watch	Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.
debug presence	Displays debugging information about the presence service.
max-subscription	Sets the maximum number of concurrent watch sessions that are allowed.
presence enable	Allows the router to accept incoming presence requests.

Command	Description
server	Specifies the IP address of a presence server for sending presence requests from internal watchers to external presence entities.
show presence global	Displays configuration information about the presence service.
show presence subscription	Displays information about active presence subscriptions.

presence call-list

To enable Busy Lamp Field (BLF) monitoring for call lists and directories on phones registered to the Cisco Unified CME router, use the **presence call-list**command in ephone, presence, or voice register pool configuration mode. To disable BLF indicators for call lists, use the **no** form of this command.

presence call-list
no presence call-list

Syntax Description

This command has no arguments or keywords.

Command Default

BLF monitoring for call lists is disabled.

Command Modes

Ephone configuration (config-ephone)
Presence configuration (config-presence)
Voice register pool configuration (config-register pool)

Command History

Release	Modification
12.4(11)XJ	This command was introduced.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command enables a phone to monitor the line status of directory numbers listed in a directory or call list, such as a missed calls, placed calls, or received calls list. Using this command in presence mode enables the BLF call-list feature for all phones. To enable the feature for an individual SCCP phone, use this command in ephone configuration mode. To enable the feature for an individual SIP phone, use this command in voice register pool configuration mode.

If this command is disabled globally and enabled in voice register pool or ephone configuration mode, the feature is enabled for that voice register pool or ephone.

If this command is enabled globally, the feature is enabled for all voice register pools and ephones regardless of whether it is enabled or disabled on a specific voice register pool or ephone.

To display a BLF status indicator, the directory number associated with a telephone number or extension must have presence enabled with the **allow watch** command.

For information on the BLF status indicators that display on specific types of phones, see the Cisco Unified IP Phone documentation for your phone model.

Examples

The following example shows the BLF call-list feature enabled for ephone 1. The line status of a directory number that appears in a call list or directory is displayed on phone 1 if the directory number has presence enabled.

```
Router(config)# ephone 1
Router(config-ephone)# presence call-list
```

Related Commands

Command	Description
allow watch	Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.
blf-speed-dial	Enables BLF monitoring for a speed-dial number on a phone registered to Cisco Unified CME.
presence	Enables presence service and enters presence configuration mode.
show presence global	Displays configuration information about the presence service.

presence enable

To allow incoming presence requests, use the **presence enable** command in SIP user-agent configuration mode. To block incoming requests, use the **no** form of this command.

presence enable
no presence enable

Syntax Description

This command has no arguments or keywords.

Command Default

Incoming presence requests are blocked.

Command Modes

SIP UA configuration (config-sip-ua)

Command History

Release	Modification
12.4(11)XJ	This command was introduced.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command allows the router to accept incoming presence requests (SUBSCRIBE messages) from internal watchers and SIP trunks. It does not impact outgoing presence requests.

Examples

The following example shows how to allow incoming presence requests:

```
Router(config)# sip-ua
Router(config-sip-ua)# presence enable
```

Related Commands

Command	Description
allow subscribe	Allows internal watchers to monitor external presence entities (directory numbers).
allow watch	Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.
max-subscription	Sets the maximum number of concurrent watch sessions that are allowed.
show presence global	Displays configuration information about the presence service.
show presence subscription	Displays information about active presence subscriptions.
watcher all	Allows external watchers to monitor internal presence entities (directory numbers).

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 *timeslot -range*	Specifies a single range of timeslot values in the PRI goup. 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) Specifes that the operation of the D channel timeslot on this controller is the NFAS D backup.
none	(Optional) Specifes 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 *range*	Specifies the provisioned NFAS interface value. Valid values range from 0 to 32.
nfas-group *number*	Specifies the NFAS group and the NFAS group number. Valid values range from 0 to 31.
iua *as -name*	Binds the Non-Facility Associated Signaling (NFAS) group to the IDSN User Adaptation Layer (IUA) application server (AS).

Command Default

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 Dchannel3/0:1
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:

```
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-addresspbx-ip-host-name*} **pbx-port** *number*
no pri-group nec-fusion {*pbx-ip-addresspbx-ip-host-name*} **pbx-port** *number*

Syntax Description

pbx -ip-address	IP address of the NEC PBX.
pbx -ip-host-name	Host name of the NEC PBX.
pbx -port *number*	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.

Command Default

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.

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** **nfas_int** *number* **nfas_group** *number* [**service mgcp**] | **none** **nfas_int** *number* **nfas_group** *number* [**service mgcp**] | **primary** **nfas_int** *number* **nfas_group** *number* [{**iua** *as-name* | **rlm-group** *number* | **service mgcp**}]} | **service mgcp**}] [**voice-dsp**]
no pri-group timeslots *timeslot-range* [{**nfas_d** {**backup** **nfas_int** *number* **nfas_group** *number* [**service mgcp**] | **none** **nfas_int** *number* **nfas_group** *number* [**service mgcp**] | **primary** **nfas_int** *number* **nfas_group** *number* [{**iua** *as-name* | **rlm-group** *number* | **service mgcp**}]} | **service mgcp**}] [**voice-dsp**]

Syntax Description

timeslot-range	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. **Note** Groups of time slot ranges separated by commas (1-4,8-23 for example) are also accepted.
nfas_d	(Optional) Configures the operation of the ISDN PRI D channel.
backup	The D-channel time slot is used as the Non-Facility Associated Signaling (NFAS) D backup.
service mgcp	(Optional) Configures the service type as Media Gateway Control Protocol (MGCP) service.
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.
nfas_int *number*	Specifies the provisioned NFAS interface as a value. The NFAS interface range is from 0 to 44.
nfas_group *number*	Specifies the NFAS group. The NFAS group number range is from 0 to 31.
iua *as-name*	(Optional) Configures the ISDN User Adaptation Layer (IUA) application server (AS) name.
rlm-group *number*	(Optional) Specifies the Redundant Link Manager (RLM) group and releases the ISDN PRI signaling channel. The RLM group number range is from 0 to 255.
voice-dsp	(Optional) Configures an ISDN PRI group for voice applications by using the Digital Signal Processor (DSP).

Command Default

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** keyword.

Command Modes

Controller configuration (config-controller)

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 the 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.
12.4(16)b	This command was modified to ensure that the NFAS primary interface is configured before the NFAS backup or NFAS none interfaces are configured.
12.4(24)T	Support was extended to provide backup functionality for the NFAS interface in MGCP backhaul mode. With this support, if the primary interface fails, the backup can become active and calls can be maintained.
15.1(3)T	This command was modified. The **voice-dsp** keywordwas added.

Usage Guidelines

The **pri-group** command supports the use of DS0 time slots for Signaling System 7 (SS7) links, and, therefore, enables 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 by 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. When 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 switch type, or a 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 have the same configuration as that of 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** 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 the 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.

Note You must first configure the NFAS primary D channel before configuring the NFAS backup or NFAS none interfaces. If this order is not followed, this message is displayed: NFAS backup and NFAS none interfaces are not allowed to be configured without primary. First configure primary D channel. To remove the NFAS primary D channel after the NFAS backup or NFAS none interfaces are configured, you must remove the NFAS backup or NFAS none interfaces first, and then remove the NFAS primary D channel.

The **voice-dsp**keyword is available only on 1-Port and 2-Port HWIC on ISR-G2 (Cisco 2911, Cisco 2921, Cisco 2951, Cisco 3925, Cisco 3925E, Cisco 3945, and Cisco 3945E). This keyword is not available on controller T1 0/1/0 on Voice/WAN(VWIC) interface card.

Examples

The following example shows how to configure a 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 shows how to configure an ISDN PRI on T1 slot 1, port 0, and configure 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 shows how to configure 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 shows how to configure 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 shows how to configure 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
```

The following example shows how to configure T1 controller 0/2/1 for a PRI with the voice applications option:

```
Router(config)#controller T1 0/2/1
Router(config-controller)#pri-group timeslots 1-24
Router(config-controller)#pri-group timeslots 1-24 voice-dsp
```

Related Commands

Command	Description
controller	Configures a T1 or E1 controller and enters controller configuration mode.

Command	Description
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 releases the B channels when there is no active signaling channel.
show isdn nfas group	Displays all the members of a specified NFAS group or all NFAS groups.

primary (gateway accounting file)

To set the primary location for storing the call detail records (CDRs) generated for file accounting, use the **primary**command in gateway accounting file configuration mode. To reset to the default, use the **no** form of this command.

primary {**ftp** *path/filename* **username** *username* **password** *password* | **ifs** *device***:***filename*}
no primary {**ftp** | **ifs**}

Syntax Description

ftp *path /filename*	Name and location of the file on an external FTP server. Filename is limited to 25 characters.
ifs *device* **:** *filename*	Name and location of the file in flash memory or other internal file system on this router. Values depend on storage devices available on the router, for example flash or slot0. Filename is limited to 25 characters.
username *username*	User ID for authentication.
password *password*	Password user enters for authentication.

Command Default

Call records are saved to **flash:cdr**.

Command Modes

Gateway accounting file configuration (config-gw-accounting-file)

Command History

Release	Modification
12.4(15)XY	This command was introduced.
12.4(20)T	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command specifies the name and location of the primary file where CDRs are stored during the file accounting process. The filename you assign is appended with the gateway hostname and time stamp at the time the file is created to make the filename unique.

For example, if you specify the filename cdrtest1 on a router with the hostname cme-2821, a file is created with the name cdrtest1.cme-2821.2007_10_28T22_21_41.000, where 2007_10_28T22_21_41.000 is the time that the file was created.

Limit the filename you assign with this command to 25 characters, otherwise it could be truncated when the accounting file is created because the full filename, including the appended hostname and timestamp, is limited to 63 characters.

If the file transfer to this primary device fails, the file accounting process retries the primary device up to the number of times defined by the **maximum retry-count** command and then switches over to the secondary device defined with the **secondary** command.

To manually switch back to the primary device when it becomes available, use the **file-acct reset** command. The system does not automatically switch back to the primary device.

A syslog warning message is generated when flash becomes full.

Examples

The following example shows the primary location of the accounting file is set to an external FTP server and the filename is cdrtest1:

```
gw-accounting file
 primary ftp server1/cdrtest1 username bob password temp
 secondary flash ifs:cdrtest2
 maximum buffer-size  25
 maximum retry-count 3
 maximum fileclose-timer 720
 cdr-format compact
```

The following examples show how the accounting file is named when it is created. The router hostname and time stamp are appended to the filename that you assign with this command:

```
cme-2821(config)# primary ftp server1/cdrtest1 username bob password temp
```

The name of the accounting file that is created has the following format:

cdrtest1.cme-2821.06_04_2007_18_44_51.785

Related Commands

Command	Description
file-acct flush	Manually flushes the CDRs from the buffer to the accounting file.
file-acct reset	Manually switches back to the primary device for file accounting.
maximum retry-count	Sets the maximum number of times the router attempts to connect to the primary file device before switching to the secondary device.
secondary	Sets the backup location for storing CDRs if the primary location becomes unavailable.

privacy

To set privacy support at the global level as defined in RFC 3323, use the **privacy** command in voice service voip sip configuration mode or voice class tenant configuration mode. To remove privacy support as defined in RFC 3323, use the **no** form of this command.

privacy {**pstn** | *privacy-option* [**critical**]} **[system]**
no privacy

Syntax Description

pstn	Requests that the privacy service implements a privacy header using the default Public Switched Telephone Network (PSTN) rules for privacy (based on information in Octet 3a). When selected, this becomes the only valid option.
privacy-option	The privacy support options to be set at the global level. The following keywords can be specified for the *privacy-option* argument: • **header** -- Requests that privacy be enforced for all headers in the Session Initiation Protocol (SIP) message that might identify information about the subscriber. • **history** -- Requests that the information held in the history-info header is hidden outside the trust domain. • **id** -- Requests that the Network Asserted Identity that authenticated the user be kept private with respect to SIP entities outside the trusted domain. • **session** -- Requests that the information held in the session description is hidden outside the trust domain. • **user** -- Requests that privacy services provide a user-level privacy function. **Note** The keywords can be used alone, altogether, or in any combination with each other, but each keyword can be used only once.
critical	(Optional) Requests that the privacy service performs the specified service or fail the request. **Note** This optional keyword is only available after at least one of the *privacy-option* keywords (**header**, **history**, **id**, **session**, or **user**) has been specified and can be used only once per command.
system	Specifies that the privacy support use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

Command Default

Privacy support is disabled.

Command Modes

Voice service voip sip configuration (conf-serv-sip)

Voice class tenant configuration (config-class)

Command History

Release	Modification
12.4(15)T	This command was introduced.

Release	Modification
12.4(22)T	The **history** keyword was added to provide support for the history-info header information.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: **system**.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Use the **privacy** command to instruct the gateway to add a Proxy-Require header set to a value supported by RFC 3323 in outgoing SIP request messages.

Use the **privacy critical** command to instruct the gateway to add a Proxy-Require header with the value set to critical. If a user agent sends a request to an intermediary that does not support privacy extensions, the request fails.

Examples

The following example shows how to set the privacy to PSTN:

```
Router> enable

Router# configure
 terminal
Router(config)# voice
 service
 voip

Router(conf-voi-serv)# sip
Router(conf-serv-sip)# privacy
 pstn
```

The following example shows how to set privacy in the voice class tenant configuration mode:

```
Router(config-class)# privacy system
```

Related Commands

Command	Description
asserted-id	Sets the privacy level and enables either PAI or PPI privacy headers in outgoing SIP requests or response messages.
calling-info pstn-to-sip	Specifies calling information treatment for PSTN-to-SIP calls.
clid (voice-service-voip)	Passes the network-provided ISDN numbers in an ISDN calling party information element screening indicator field, removes the calling party name and number from the calling-line identifier in voice service voip configuration mode, or allows a presentation of the calling number by substituting for the missing Display Name field in the Remote-Party-ID and From headers.
voice-class sip privacy	Sets privacy support at the dial-peer configuration level as defined in RFC 3323.

privacy (supplementary-service)

To prevent phones on a shared line from joining active calls, use the **privacy** command in supplementary-service voice-port configuration mode. To return to the default behavior, use the **no** form of this command.

privacy {**on** | **off**}
no privacy

Syntax Description

on	Prevents other phones on the shared line to join active calls.
off	Allows other phones on the shared line to join active calls.

Command Default

The **no privacy** command implies that a port does not decide on its privacy status. It is not the gateway but the Cisco Unified CM that decides on the privacy status of a port.

Command Modes

Supplementary-service voice-port configuration mode (config-stcapp-suppl-serv-port)

Command History

Release	Modification
15.1(3)T	This command was introduced.

Usage Guidelines

The **privacy** command enables privacy support on analog endpoints that are connected to Foreign Exchange Station (FXS) ports on a Cisco IOS Voice Gateway, such as a Cisco Integrated Services Router (ISR) or Cisco VG224 Analog Phone Gateway.

Use the **privacy** command to prevent other phones on the shared line to join active calls.

Examples

The following example shows how to turn on privacy support on port 2/4 on a Cisco VG224:

```
Router(config)# stcapp supplementary-services
Router(config-stcapp-suppl-serv)# port 2/4
Router(config-stcapp-suppl-serv-port)# privacy on
Router(config-stcapp-suppl-serv-port)# end
```

Related Commands

Command	Description
stcapp supplementary-services	Enters supplementary-service configuration mode for configuring STCAPP supplementary-service features on an FXS port.

privacy-policy

To configure the privacy header policy options at the global level, use the **privacy-policy** command in voice service VoIP SIP configuration mode or voice class tenant configuration mode. To disable privacy header policy options, use the **no** form of this command.

privacy-policy {**passthru** | **send-always** | **strip** {**diversion** | **history-info**} **[system]**}
no privacy-policy {**passthru** | **send-always** | **strip** {**diversion** | **history-info**} **[system]**}

Syntax Description

passthru	Passes the privacy values from the received message to the next call leg.
send-always	Passes a privacy header with a value of None to the next call leg, if the received message does not contain privacy values but a privacy header is required.
strip	Strips the diversion or history-info headers received from the next call leg.
diversion	Strips the diversion headers received from the next call leg.
history-info	Strips the history-info headers received from the next call leg.
system	Specifies that the privacy header policy options use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

Command Default

No privacy-policy settings are configured.

Command Modes

Voice service VoIP SIP configuration (conf-serv-sip)

Voice class tenant configuration (config-class)

Command History

Release	Modification
12.4(22)YB	This command was introduced.
15.0(1)M	This command was integrated into Cisco IOS Release 15.0(1)M.
15.1(2)T	This command was modified. The **strip**, **diversion**, and **history-info** keywords were added.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: **system**. This command is now available under voice class tenants.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

If a received message contains privacy values, use the **privacy-policy passthru** command to ensure that the privacy values are passed from one call leg to the next. If the received message does not contain privacy values but the privacy header is required, use the **privacy-policy send-always** command to set the privacy header to None and forward the message to the next call leg. If you want to strip the diversion and history-info from the headers received from the next call leg, use the **privacy-policy strip** command. You can configure the system to support all the options at the same time.

Examples

The following example shows how to enable the pass-through privacy policy:

```
Router> enable

Router# configure
 terminal
Router(config)# voice
 service
 voip

Router(conf-voi-serv)# sip
Router(conf-serv-sip)# privacy-policy passthru
```

The following example shows how to enable the send-always privacy policy:

```
Router(config-class)# privacy-policy send-always system
```

The following example shows how to enable the strip privacy policy:

```
Router> enable

Router# configure
 terminal
Router(config)# voice
 service
 voip

Router(conf-voi-serv)# sip
Router(conf-serv-sip)# privacy-policy strip diversion
Router(conf-serv-sip)# privacy-policy strip history-info
```

The following example shows how to enable the pass-through, send-always privacy, and strip policies:

```
Router> enable

Router# configure
 terminal
Router(config)# voice
 service
 voip

Router(conf-voi-serv)# sip
Router(conf-serv-sip)# privacy-policy passthru
Router(conf-serv-sip)# privacy-policy send-always
Router(conf-serv-sip)# privacy-policy strip diversion
Router(conf-serv-sip)# privacy-policy strip history-info
```

The following example shows how to enable the send-always privacy policy in the voice class tenant configuration mode:

Related Commands

Command	Description
asserted-id	Sets the privacy level and enables either PAID or PPID privacy headers in outgoing SIP requests or response messages.
voice-class sip privacy-policy	Configures the privacy header policy options at the dial-peer configuration level.

probing interval

To configure the time interval between probing messages sent by the router, use the **probing interval** command. To reset the time interval to the default number, use the **no** form of this command.

probing interval[{**keepalive** | **negative**}] *seconds*

Syntax Description

keepalive	(optional) Configures the time interval between probing messages when the session is in a keepalive state. Range is from 1 to 255 seconds. Default is 5 seconds.
negative	(optional) Configures the time interval between probing messages when the session is in a negative state. Range is from 1 to 20 seconds. Default is 5 seconds.
seconds	Number of seconds between probing message.

Command Default

The default is 120 seconds between probing messages when the session is in a normal state and 5 seconds between probing messages when the session is in a negative state.

Command Modes

uc wsapi configuration mode.

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

Use this command to configure the time interval between probing messages sent by the router.

Examples

The following example sets an interval of 180 seconds for a normal session and 10 seconds when the session is in a negative state.

```
Router(config)# uc wsapi
Router(config-uc-wsapi)# probing interval keepalive 180
Router(config-uc-wsapi)# probing interval negative 10
```

Related Commands

Command	Description
message-exchange	Sets the maximum number of failed message responses before the provider stops sending messages.
probing max-failure	Sets the number of messages that the system will send without receiving a reply before the system unregisters the application.

probing max-failures

To configure the maximum number of probing messages that the system attempts to send to the application, and the application does not respond to before the system stops the session and unregisters the application, use the **probing max-failures** command. To reset the maximum to the default number, use the **no** form of this command.

probing max-failures *number*
no probing max-failures *number*

Syntax Description

number	Maximum number of messages allowed before the system stops the session and unregisters the application. Range is from 1 to 5. Default is 3.

Command Default

The default is 3.

Command Modes

uc wsapi configuration mode

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

Use this command to set the maximum number of probing messages sent by the system that the application does not respond to before the system stops the session and unregisters the application session.

Examples

The following example sets the maximum number of failed messages to 5.

```
Router(config)# uc wsapi
Router(config-uc-wsapi)# probing max-failures 5
```

Related Commands

Command	Description
message-exchange	Sets the maximum number of failed message attempts before the provider stops sending messages.
probing interval	Sets the time interval between probing messages.

progress_ind

To configure an outbound dial peer on a Cisco IOS voice gateway or Cisco Unified Border Element to override and remove or replace the default progress indicator (PI) in specified call messages, use the **progress_ind** command in dial peer voice configuration mode. To disable removal or replacement of the default PI in specific call messages, use the **no** form of this command.

progress_ind {{**alert** | **callproc**} {**enable** *pi-number* | **disable** | **strip** [*strip-pi-number*]} | {**connect** | **disconnect** | **progress** | **setup**} {**enable** *pi-number* | **disable**}}
no progress_ind {**alert** | **callproc** | **connect** | **disconnect** | **progress** | **setup**}

Syntax Description

alert	Specifies that the configuration applies to call Alert messages.
callproc	Specifies that the configuration applies to Session Initiation Protocol (SIP) 183 Session In Progress (Call_Proceeding) messages.
connect	Specifies that the configuration applies to call Connect messages.
disconnect	Specifies that the configuration applies to call Disconnect messages.
progress	Specifies that the configuration applies to call progress messages.
setup	Specifies that the configuration applies to call setup messages.
enable	Enables user-specified configuration of the progress indicator on the specified call message type.
pi -number	Specifies the PI to be used in place of the default PI. The following are acceptable PI values according to the call message type: • Alert, Connect, Progress, and SIP 183 Session In Progress messages: 1, 2, or 8. • Disconnect messages: 8. • Setup messages: 0, 1, or 3.
disable	Disables user-specified configuration of the progress indicator on the specified call message type.
strip	Configures the dial peer to remove all or specific progress indicators in the specified call message type. **Note** This option applies only to call Alert message on POTS dial peers or to call Proceeding messages on VoIP dial peers.
strip-pi -number	(optional) Specifies that only a specific PI is to be removed from the specified call message. The value can be 1, 2, or 8.

Command Default

This command is disabled on the outbound dial peer and the default progress indicator that is received in the incoming call message is passed intact (it is not intercepted, modified, or removed).

Command Modes

Dial peer voice configuration (conf-dial-peer)

Command History

Release	Modification
12.1(3)XI	This command was introduced on the 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. Support was added for setup messages from a POTS dial peer.
12.2(2)XA	This command was implemented on the Cisco AS5350 and Cisco AS5400.
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.
15.0(1)XA	This command was modified. Support was added for stripping of PIs in call Alert and SIP 183 Session In Progress (Call_Proceeding) messages.
15.1(1)T	This command was integrated into Cisco IOS Release 5.1(1)T.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Before configuring the **progress_ind** command on an outbound dial peer, you must configure a destination pattern on the dial peer. To configure a destination pattern for an outbound dial peer, use the **destination-pattern** command in a dial peer voice configuration mode. Once you have set a destination pattern on the dial peer, you can then use the **progress_ind** command, also in dial peer voice configuration mode, to override and replace or remove the default PI in specific call message types.

You can use the **progress_ind** command to configure replacement behavior on outbound dial peers on a Cisco IOS voice gateway or CUBEto ensure proper end-to-end signaling of VoIP calls. You can also use this command to configure removal (stripping) of PIs on outbound dial peers on Cisco IOS voice gateways or CUBEs, such as when configuring a Cisco IOS SIP gateway (or SIP-SIPCUBE) to not generate another SIP 183 Session In Progress messages.

For messages that contain multiple PIs, behavior that is configured using the **progress_ind** command overrides only the first PI in the message. Also, configuring a replacement PI will not result in an override of the default PI in call progress messages if the Progress message is sent after a backward cut-through event, such as when an Alert message with a PI of 8 was sent before the Progress message.

Use the **no progress_ind** command in dial peer voice configuration mode to disable PI override configurations on a dial peer on a Cisco IOS voice gateway or CUBE.

Examples

The following example shows how to configure POTS dial peer 3 to override default PIs in call progress and Connect messages and replace them with a PI of 1:

```
Router(config)# dial-peer voice 3 pots
Router(config-dial-peer)# destination-pattern 555
```

```
Router(config-dial-peer)# progress_ind progress enable 1
Router(config-dial-peer)# progress_ind connect enable 1
```

The following example configures outbound VoIP dial peer 1 to override SIP 183 Session In Progress messages and to strip out any PIs with a value of 8:

```
Router(config)# dial-peer voice 1 voip
Router(config-dial-peer)# destination-pattern 777
Router(config-dial-peer)# progress_ind callproc strip 8
```

Related Commands

Command	Description
destination-pattern	Specifies the destination pattern (prefix or full E.164 phone number) to be used on an outbound dial peer.

protocol mode

To configure the Cisco IOS Session Initiation Protocol (SIP) stack, use the **protocol mode**command in SIP user-agent configuration mode. To disable the configuration, use the **no** form of this command.

protocol mode {**ipv4** | **ipv6** | **dual-stack** [**preference** {**ipv4** | **ipv6**}]}
no protocol mode

Syntax Description

ipv4	Specifies the IPv4-only mode.
ipv6	Specifies the IPv6-only mode.
dual-stack	Specifies the dual-stack (that is, IPv4 and IPv6) mode.
preference {**ipv4** | **ipv6**	(Optional) Specifies the preferred dual-stack mode, which can be either IPv4 (the default preferred dual-stack mode) or IPv6.

Command Default

No protocol mode is configured. The Cisco IOS SIP stack operates in IPv4 mode when the **no protocol mode** or **protocol mode ipv4** command is configured.

Command Modes

SIP user-agent configuration (config-sip-ua)

Command History

Release	Modification
12.4(22)T	This command was introduced.
15.1(1)T	This command was integrated into Cisco IOS Release 15.1(1)T.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

The **protocol mode** command is used to configure the Cisco IOS SIP stack in IPv4-only, IPv6-only, or dual-stack mode. For dual-stack mode, the user can (optionally) configure the preferred family, IPv4, or IPv6.

For a particular mode (for example, IPv6-only), the user can configure any address (for example, both IPv4 and IPv6 addresses) and the system will not hide or restrict any commands on the router. SIP chooses the right address for communication based on the configured mode on a per-call basis.

For example, if the domain name system (DNS) reply has both IPv4 and IPv6 addresses and the configured mode is IPv6-only (or IPv4-only), the system discards all IPv4 (or IPv6) addresses and tries the IPv6 (or IPv4) addresses in the order they were received in the DNS reply. If the configured mode is dual-stack, the system first tries the addresses of the preferred family in the order they were received in the DNS reply. If all the addresses fail, the system tries addresses of the other family.

Examples

The following example configures dual-stack as the protocol mode:

```
Router(config-sip-ua)# protocol mode dual-stack
```

The following example configures IPv6 only as the protocol mode:

```
Router(config-sip-ua)# protocol mode ipv6
```

The following example configures IPv4 only as the protocol mode:

```
Router(config-sip-ua)# protocol mode ipv4
```

The following example configures no protocol mode:

```
Router(config-sip-ua)# no protocol mode
```

Related Commands

Command	Description
sip ua	Enters SIP user-agent configuration mode.

protocol rlm port

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

protocol rlm port *port-number*
no protocol rlm port *port-number*

Syntax Description

port -number	RLM port number. See the table below for the port number choices.

Command Default

3000

Command Modes

RLM configuration

Command History

Release	Modification
11.3(7)	This command was introduced.

Usage Guidelines

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

Table 2: 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.

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

provider

To configure and enable a service provider, use the **provider** command. To remove the provider, use the **no** form of this command.

provider [{**xcc** | **xsvc** | **xcdr** | **xmf**}]
no provider [{**xcc** | **xsvc** | **xcdr** | **xmf**}]

Syntax Description

xcc	(optional) Enables the XCC service provider.
xsvc	(optional) Enables the XSVC service provider.
xcdr	(optional) Enables the XCDR service provider.
xmf	(optional) Enables the XMF service provider.

Command Default

No default behavior or values.

Command Modes

uc wsapi configuration mode

uc secure-wsapi

Command History

Release	Modification
15.2(2)T	This command was introduced.
15.3(2)T	**xmf** keyword was added.
Cisco IOS XE Everest 16.6.1	Added support for **xcc** and **xsvc** service providers in secure mode.

Usage Guidelines

Use this command to enable a service provider.

Note You can enable only **xcc** and **xsvc** service providers in secure mode.

Examples

The following example enables the XCC service provider in nonsecure mode.

```
Router(config)# uc wsapi
Router(config-uc-wsapi)# provider xcc
Router(config-uc-wsapi-xcc)# no shutdown
```

Examples

The following example enables the XCC service provider in secure mode.

```
Router(config)# uc secure-wsapi
Router(config-uc-wsapi)# provider xcc
Router(config-uc-wsapi-xcc)# no shutdown
```

Related Commands

Command	Description
remote-url	Specifies the URL of the application.
source-address	Specifies the IP address of the provider.
uc wsapi	Enters nonsecure Cisco Unified Communication IOS services configuration mode.
uc secure-wsapi	Enters secure Cisco Unified Communication IOS services configuration mode.

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.

Command Default

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
```

proxy (media-profile)

To configure IP address or hostname of a WebSocket proxy server in CUBE, use the **proxy** command in media profile configuration mode. To remove the configuration, use the **no** form of this command.

proxy { **host** *host* **port** *port* | **ipv4** *ip-address* **port** *port* }
no proxy { **host** *host* **port** *port* | **ipv4** *ip-address* **port** *port* }

Syntax Description

host	WebSocket proxy server hostname.
ipv4 *ip-address*	Host IP address of the WebSocket proxy server.
port *port*	WebSocket proxy server port.

Command Default

Disabled by default.

Command Modes

Media Profile configuration mode (cfg-mediaprofile)

Command History

Release	Modification
Cisco IOS XE Bengaluru 17.6.1a	This command was introduced on Cisco Unified Border Element.

Usage Guidelines

If there’s a proxy between the WebSocket speech server and CUBE, the IP address or hostname of the proxy must be configured in media-profile. The **proxy** command configures the host IP address of the proxy server or the hostname in media profile configuration mode.

If a proxy server is configured, the WebSocket connection must be established with the proxy server itself. It is not possible to establish a direct connection with the speech server.

Note **port** *port* is an optional configuration parameter.

Examples

The following is a sample configuration for **proxy (media-profile)** in CUBE:

```
router(cfg-mediaprofile)#proxy ?
host WebSocket proxy server hostname
ip WebSocket proxy server IP address

router(cfg-mediaprofile)#proxy host
router(cfg-mediaprofile)#proxy host abc.com ?
port WebSocket proxy server port
<cr> <cr>

router(cfg-mediaprofile)#proxy host abc.com port ?
<0-65535> proxy server port

router(cfg-mediaprofile)#proxy host abc.com port 3578


router(cfg-mediaprofile)#proxy ipv4 ?
```

```
A.B.C.D Specify IP address of proxy server

router(cfg-mediaprofile)#proxy ip 1.1.1.1 ?
port WebSocket proxy server port
<cr> <cr>

router(cfg-mediaprofile)#proxy ip 1.1.1.1 port ?
<0-65535> proxy server port

router(cfg-mediaprofile)#proxy ip 1.1.1.1 port 3456
```

Related Commands

Command	Description
media profile stream-service	Enables stream service on CUBE.
connection (media-profile)	Configures idle timeout and call threshold for a media profile.
source-ip (media-profile)	Configures local source IP address of a WebSocket connection.
media class	Applies the media class at the dial peer level.

pulse-digit-detection

To enable pulse digit detection at the beginning of a call, use the **pulse-digit-detection** command in voice-port configuration mode. To disable pulse digit detection, use the **no** form of this command.

pulse-digit-detection
no pulse-digit-detection

Syntax Description

This command has no arguments or keywords.

Command Default

Pulse digit detection is enabled.

Command Modes

Voice-port configuration (config-voiceport)

Command History

Release	Modification
15.0(1)M	This command was introduced.

Usage Guidelines

Pulse digit detection is disabled at the beginning of a call for any Foreign Exchange Station (FXS) voice port not configured with the **no pulse-digit-detection** command. By default, pulse digit detection is enabled.

Note Users should configure the **no pulse-digit-detection** command only if their equipment generates pulse digits in error when initiating an outbound call.

Examples

The following example shows how to disable pulse digit detection on voice port 2/0/0:

```
Device> enable
Device# configure terminal
Device(config)# voice-port 2/0/0
Device(config-voiceport)# no pulse-digit-detection
Device(config-voiceport)# end
```

Related Commands

Command	Description
timing pulse	Specifies the pulse dialing rate for a specified voice port.