caller-id (dial peer)

To enable caller ID, use the **caller-id** command in dial peer configuration mode. To disable caller ID, use the **no** form of the command.

caller-id

no caller-id

Syntax Description

This command contains no arguments or keywords.

Command Default

Caller ID is disabled

Command Modes

Dial peer configuration

Command History

Release	Modification
12.1.(2)XF	This command was introduced on the Cisco 800 series routers.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.

Usage Guidelines

This command is available on Cisco 800 series routers that have plain old telephone service (POTS) ports. The command is effective only if you subscribe to caller ID service. If you enable caller ID on a router without subscribing to the caller ID service, caller ID information does not appear on the telephone display.

The configuration of caller ID must match the device connected to the POTS port. That is, if a telephone supports the caller ID feature, use the command **caller-id** to enable the feature. If the telephone does not support the caller ID feature, use the command default or disable the caller ID feature. Odd ringing behavior might occur if the caller ID feature is disabled when it is a supported telephone feature or enabled when it is not a supported telephone feature.

Note Specific hardware is required to provide full support for the Caller ID features. To determine support for these features in your configuration, review the appropriate hardware documentation and data sheets. This information is available on Cisco.com.

Examples

The following example enables a router to use the caller ID feature:

```
dial-peer voice 1 pots
 caller-id
```

Related Commands

Command	Description
block-caller	Configures call blocking on caller ID.
debug pots csm csm	Activates events from which an application can determine and display the status and progress of calls to and from POTS ports.
isdn i-number	Configures several terminal devices to use one subscriber line.
pots call-waiting	Enables local call waiting on a router.
registered-caller ring	Configures the Nariwake service-registered caller ring cadence.

caller-id alerting dsp-pre-alloc

To statically allocate a digital signal processor (DSP) resource for receiving caller ID information for on-hook (Type 1) caller ID at a receiving Foreign Exchange Office (FXO) voice port, use the **caller-id alerting dsp-pre-alloc** command in voice-port configuration mode. To disable the command's effect, use the **no** form of this command.

caller-id alerting dsp-pre-alloc

no caller-id alerting dsp-pre-alloc

Syntax Description

This command contains no arguments or keywords.

Command Default

No preallocation of DSP resources

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(2)XH	This command was introduced on the Cisco MC3810, Cisco 2600 series, and Cisco 3600 series.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T.

Usage Guidelines

The **caller-id alerting dsp-pre-alloc** command may be required on an FXO port if the central office uses line polarity reversal to signal the start of caller-ID information transmission. Preallocating a DSP allows the DSP to listen for caller-ID information continuously without requiring an alerting signal from the central office (CO).

This command is the FXO counterpart to the **caller-id alerting line-reversal** command, which is applied to the Foreign Exchange Station (sending) end of the caller-ID call.

Note Specific hardware is required to provide full support for the caller ID features. To determine support for these features in your configuration, review the appropriate hardware documentation and data sheets. This information is available on Cisco.com.

Examples

The following example configures a voice port where caller-ID information is received:

```
voice-port 1/0/1
  cptone US
  caller-id enable
  caller-id alerting line-reversal
  caller-id alerting dsp-pre-alloc
```

Related Commands

Command	Description
caller-id alerting line-reversal	Sets the line-reversal method of caller-ID call alerting.

caller-id alerting line-reversal

To set the line-reversal alerting method for caller-ID information for on-hook (Type 1) caller ID at a sending Foreign Exchange Station (FXS) voice port, use the **caller-id alerting line-reversal** command in voice-port configuration mode. To disable the command's effect, use the **no** form of this command.

caller-id alerting line-reversal

no caller-id alerting line-reversal

Syntax Description

This command has no arguments or keywords.

Command Default

No line-reversal alert

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(2)XH	This command was introduced.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T.

Usage Guidelines

This command is required only when the telephone device attached to an FXS port requires the line-reversal method to signal the start of a caller-ID transmission. Use it on FXS voice ports that send caller-ID information.

This command is the FXS counterpart to the **caller-id alerting dsp-pre-alloc** command, which is applied to the FXO (receiving) end of the caller-ID call with the line-reversal alerting method.

Note Specific hardware is required to provide full support for the caller ID features. To determine support for these features in your configuration, review the appropriate hardware documentation and data sheets. This information is available on Cisco.com.

Examples

The following example configures a voice port from which caller-ID information is sent:

```
voice-port 1/0/1
   cptone US
   station name  A. sample
   station number 4085550111
   caller-id alerting line-reversal
   caller-id alerting dsp-pre-alloc
```

Related Commands

Command	Description
caller-id alerting dsp-pre-alloc	At the receiving end of a line-reversal alerting caller-ID call, preallocates DSPs for caller ID calls.

caller-id alerting pre-ring

To set a 250-millisecond prering alerting method for caller ID information for on-hook (Type 1) caller ID at a sending Foreign Exchange Station (FXS) voice port, use the **caller-id alerting pre-ring** command in voice-port configuration mode. To disable the command, use the **no** form of this command.

caller-id alerting pre-ring

no caller-id alerting pre-ring

Syntax Description

This command has no arguments or keywords.

Defaults

No prering alert

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(2)XH	This command was introduced on the Cisco MC3810, Cisco 2600 series, and Cisco 3600 series.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T.

Usage Guidelines

This command is required only when the telephone device attached to an FXS port requires the prering (immediate ring) method to signal the start of caller ID transmission. Use it on FXS voice ports that send caller ID information. This command allows the FXS port to send a short prering preceding the normal ring cadence. On an FXO port, an incoming prering (immediate ring) is simply counted as a normal ring using the **caller-id alerting ring** command.

Note Specific hardware is required to provide full support for the caller ID features. To determine support for these features in your configuration, review the appropriate hardware documentation and data sheets. This information is available on Cisco.com.

Examples

The following example configures a voice port from which caller ID information is sent:

```
voice-port 1/0/1
   cptone US
   station name  A. sample
   station number 4085550111
   caller-id alerting pre-ring
```

Related Commands

Command	Description
caller-id alerting line-reversal	Enables caller ID operation and sets the line-reversal alerting type at an FXS port.
caller-id alerting ring	Enables caller ID operation and sets an alerting ring type at an FXO or FXS port.

caller-id alerting ring

To set the ring-cycle method for receiving caller ID information for on-hook (Type 1) caller ID at a receiving Foreign Exchange Office (FXO) or a sending Foreign Exchange Station (FXS) voice port, use the **caller-id alerting ring** command in voice-port configuration mode. To set the command to the default, use the **no** form of this command.

caller-id alerting ring {**1** | **2**}

no caller-id alerting ring

Syntax Description

1	Use this setting if your telephone service provider specifies it to provide caller ID alerting (display) after the first ring at the receiving station. This is the most common setting.
2	Use this setting if your telephone service provider specifies it to provide caller ID alerting (display) after the second ring. This setting is used in Australia, where the caller ID information is sent following two short rings (double-pulse ring).

Command Default

1

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(2)XH	This command was introduced.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T.

Usage Guidelines

This setting is determined by the Bellcore/Telcordia or ETSI standard that your telephone service provider uses for caller ID. Use it on FXO loop-start and ground-start voice ports where caller ID information arrives and on FXS voice ports from which caller ID information is sent.

This setting must match on the sending and receiving ends of the telephone line connection.

Note Specific hardware is required to provide full support for the caller ID features. To determine support for these features in your configuration, review the appropriate hardware documentation and data sheets. This information is available on line.

Examples

The following example configures a voice port where caller ID information is received:

```
voice-port 1/0/1
   cptone US
   caller-id alerting ring 1
```

The following example configures a voice port from which caller ID information is sent:

```
voice-port 1/0/1
   cptone northamerica
   station name A. sample
   station number 4085550111
   caller-id alerting ring 1
```

Related Commands

Command	Description
caller-id alerting line-reversal	Enables caller ID operation and sets the line-reversal alerting type at an FXS port.
caller-id alerting pre-ring	Enables caller ID operation and sets the pre-ring alerting method at an FXS port.

caller-id attenuation

To set the attenuation for caller ID at a receiving Foreign Exchange Office (FXO) voice port, use the **caller-id attenuation** command in voice-port configuration mode. To set the command to the default, use the **no** form of this command.

caller-id attenuation [*attenuation*]

no caller-id attenuation

Syntax Description

attenuation	(Optional) specifies the attenuation, in decibels (dB). Range is from 0 to 64. The default is 14.

Command Default

The default value is 14 dB, signal level of –14 dBm.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(2)XH	This command was introduced.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T.

Usage Guidelines

Use this setting to specify the attenuation for a caller ID FXO port. If the setting is not used, the attenuation is set to 14 dB, signal level of –14 dBm.

Note Specific hardware is required to provide full support for the caller ID features. To determine support for these features in your configuration, review the appropriate hardware documentation and data sheets. This information is available on line.

Examples

The following example configures a voice port where caller ID information is received:

```
voice-port 1/0/1
   cptone US
   caller-id attenuation 0
```

caller-id block

To request the blocking of the display of caller ID information at the far end of a call from calls originated at a Foreign Exchange Station (FXS) port, use the **caller-id block** command in voice-port configuration mode at the originating FXS voice port. To allow the display of caller ID information, use the **no** form of this command.

caller-id block

no caller-id block

Syntax Description

This command has no arguments or keywords.

Command Default

No blocking of caller ID information

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(2)XH	This command was introduced.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T.

Usage Guidelines

This command is used on FXS voice ports that are used to originate on-net telephone calls. This command affects all calls sent to a far-end FXS station from the configured originating FXS station. Calling number and called number are provided in the H.225 setup message for VoIP, through the H.225 Octet 3A field. Calling name information is included in a display information element.

Note Cisco-switched calls using Voice over Frame Relay (VoFR) and Voice over ATM (VoATM) carry calling party information in the Cisco proprietary setup message. For standards-based, point-to-point VoFR (FRF.11) trunks where transparent signaling is applied for FXS-to-FXO calls, only pass-through of in-band automatic number identification (ANI) is supported. ANI information is always unblocked for these communications. Interface technology using transparent channel-associated signaling (CAS) can support only ANI through Feature Group D (in-band MF signaling). The Caller ID feature cannot be used with fixed point-to-point trunk connects created using the **connection trunk** command.

Note Specific hardware is required to provide full support for the caller ID features. To determine support for these features in your configuration, review the appropriate hardware documentation and data sheets. This information is available on Cisco.com.

Examples

The following example configures a voice port from which caller ID information is sent:

```
voice-port 1/0/1
   cptone US
   station name A. sample
   station number 4085550111
   caller-id block
```

Related Commands

Command	Description
caller-id enable	Enables caller ID operation.

caller-id enable

To allow the sending or receiving of caller-ID information, use the **caller-id enable** command in voice-port configuration mode at the sending foreign exchange station (FXS) voice port or the receiving foreign exchange office (FXO) voice port. To disable the sending and receiving of caller-ID information, use the **no** form of this command.

caller-id enable [**type** {**1** | **2**}]

no caller-id enable [**type** {**1** | **2**}]

Syntax Description

type	(Optional) Indicates that the following keyword is a caller-ID type. • **1**—Type I only. Type I transmits the signal when the receiving phone is on hook. • **2**—Type II only. Type II transmits the signal when the receiving phone is off hook, for instance to display the caller ID of an incoming call when the receiving phone is busy (call-waiting caller ID).

Command Default

The sending and receiving of caller-ID information is disabled.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(2)XH	This command was introduced.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T.
12.3(7)T	The **type 1** and **type 2** keywords were added.

Usage Guidelines

This command applies to FXS voice ports that send caller-ID information and to FXO ports that receive caller-ID information. Calling number and called number are provided in the H.225.0 setup message for VoIP through the H.225.0 Octet 3A field. Calling name information is included in a display information element.

Some users that do not have caller ID type II support on their phones hear noise when type II caller ID is enabled. The **caller-id enable type 1** command allows only type I on the voice port and disables type II, so that the user does not hear this noise.

If this command is used without the optional **type** keyword, both type I and type II caller ID are enabled.

Note The **no** form of this command also clears all other caller-ID configuration settings for the voice port.

Note Cisco-switched calls using Voice over Frame Relay (VoFR) and Voice over ATM (VoATM) carry calling-party information in the Cisco-proprietary setup message. For standards-based, point-to-point VoFR (FRF.11) trunks where transparent signaling is applied for FXS-to-FXO calls, only pass-through of in-band automatic number identification (ANI) is supported. ANI information is always unblocked for these communications. Interface technology using transparent channel-associated signaling (CAS) can support only ANI through Feature Group D (in-band multifrequency signaling). Caller ID cannot be used with fixed point-to-point trunk connections created using the **connection trunk** command.

If the **station name**, **station number**, or a **caller-id alerting** command is configured on the voice port, caller ID is automatically enabled, and the **caller-id enable** command is not necessary.

Note Specific hardware is required to provide full support for the caller-ID features. To determine support for these features in your configuration, review the appropriate hardware documentation and data sheets. This information is available on line.

Examples

The following example configures a Cisco 2600 series or Cisco 3600 series router voice port at which caller-ID information is received:

```
voice-port 1/0/1
 cptone US
 caller-id enable
```

The following example configures a Cisco 2600 series or Cisco 3600 series router voice port from which caller-ID information is sent:

```
voice-port 1/0/1
 cptone northamerica
 station name A. sample
 station number 4085550111
 caller-id enable
```

The following example enables only type I caller ID on port 2/0:

```
voice-port 2/0
 caller-id enable type 1
```

Related Commands

Command	Description
caller-id alerting line-reversal	Enables caller ID operation and sets the line-reversal alerting type at an FXS port.
caller-id alerting pre-ring	Enables caller ID operation and sets the pre-ring alerting method at an FXS port.
caller-id alerting ring	Enables caller ID operation and sets an alerting ring type at an FXO or FXS port.
caller-id block	Disables the sending of caller ID information from an FXS port.
station name	Enables caller ID operation and sets the name sent from an FXS port.
station number	Enables caller ID operation and sets the number sent from an FXS port.

cancel-call-waiting

To define a feature code for a Feature Access Code (FAC) to enable the Cancel Call Waiting feature, use the **cancel-call-waiting** command in STC application feature access-code configuration mode. To reset the feature code to its default, use the **no** form of this command.

cancel-call-waiting *keypad-character*

no cancel-call-waiting

Syntax Description

keypad-character	Character string that can be dialed on a telephone keypad (0-9, *, #). Default: 8. 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

Feature code for Cancel Call Waiting is 8.

Command Modes

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

Command History

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

Usage Guidelines

This command changes the default value of the feature code for Cancel Call Waiting (8).

If you attempt to configure this command with a value that is already configured for another FAC, 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.

If you attempt to configure this command with a value that precludes or is precluded by another FAC, 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 cancel call waiting. With this configuration, a phone user must press **9 on the phone keypad to cancel call waiting.

```
Router(config)# stcapp feature access-code
Router(config-stcapp-fac)# cancel-call-waiting **9
```

Related Commands

Command	Description
prefix (stcapp-fac)	Defines the prefix for FACs.
show stcapp feature codes	Displays all FACs.

caller-number

To associate a type of ring cadence with a specific caller ID, use the **caller-number** command in dial peer voice configuration mode. To disable the type of ring cadence for a specific caller ID, use the **no** form of this command.

caller-number *number* **ring** *cadence*

no caller-number *number* **ring** *cadence*

Syntax Description

number	Caller ID for which the user wants to set the cadence. Twenty numbers along with their respective cadences may be set for each of the plain old telephone service (POTS) ports.
ring *cadence*	Ring cadence level. The three cadence levels (0, 1, and 2), which differ in duration and cadence, are as follows: • **0**—The ring cadence is 1 second on and 2 seconds off (NTT-defined regular ring). • **1**—The ring cadence is 0.25 seconds on, 0.2 seconds off, 0.25 seconds on, and 2.3 seconds off (NTT-defined nonregular ring). • **2**—The ring cadence is 0.5 seconds on, 0.25 seconds off, 0.25 seconds on, and 2 seconds off (Cisco-defined nonregular ring).

Command Default

The router does not associate any caller ID with a cadence level. Therefore, there is no distinctive ring.

Command Modes

Dial peer voice configuration

Command History

Release	Modification
12.2(8)T	This command was introduced on the Cisco 803, Cisco 804, and Cisco 813 routers.

Usage Guidelines

You can enter the **caller-number** command for each POTS port. A maximum of 20 caller IDs can be associated with distinct ring cadences. After 20 numbers per port have been set, you cannot set more numbers (and their ring cadences) for that port until you have removed any of the numbers that have already been set. To remove already-set numbers and their ring cadences, use the **no** form of the **caller-number** command.

The command must be set within each dial peer. Six dial peers are available, you can specify 20 caller IDs per port, for a maximum of 120 caller ID numbers.

Note If you have already subscribed to Nariwake service, the priority goes to the Nariwake caller ID cadence.

To disable distinctive ringing based on a caller ID number, configure the **no caller-number** command. Disabling the ringing removes the specific cadence that has been set for that particular number. If you have set 20 numbers and their ring cadences, you need to set the **no caller-number** command for each of the 20 numbers.

Use the **show running-config** command to check distinctive ringing status.

Examples

The following output examples show that three caller ID numbers and their ring cadences have been set for POTS port 1 and that five caller ID numbers and their ring cadences have been set for POTS port 2:

```
dial-peer voice 1 pots
 destination-pattern 5550102
 port 1
 no call-waiting
 ring 0
 volume 4
 caller-number 1111111 ring 2
 caller-number 2222222 ring 1
 caller-number 3333333 ring 1

dial-peer voice 2 pots
 destination-pattern 5550110
 port 2
 no call-waiting
 ring 0
 volume 2
 caller-number 4444444 ring 1
 caller-number 6666666 ring 2
 caller-number 7777777 ring 0
 caller-number 8888888 ring 1
 caller-number 9999999 ring 2
```

Related Commands

Command	Description
call-waiting	Enables call waiting.
volume	Configures the receiver volume level in the router.

calling-info pstn-to-sip

To specify calling information treatment for public switched telephone network (PSTN) to Session Initiation Protocol (SIP) calls, use the **calling-info pstn-to-sip** command in SIP user agent configuration mode. To disable calling information treatment for PSTN-to-SIP calls, use the **no** form of this command.

calling-info pstn-to-sip {**unscreened discard** | {**from** | **remote-party-id** | **asserted-id** {**name set** *name* | **number set** *number*}}}

no calling-info pstn-to-sip

Syntax Description

unscreened discard	(Optional) Specifies that the calling name and number be discarded.
from name set *name*	(Optional) Specifies that the display-name of the From header is unconditionally set to the configured ASCII string in the forwarded INVITE message.
from number set *number*	(Optional) Specifies that the user part of the From header is unconditionally set to the configured ASCII string in the forwarded INVITE message.
remote-party-id name set *name*	(Optional) Specifies that the display-name of the Remote-Party-ID header is unconditionally set to the configured ASCII string in the forwarded INVITE message.
remote-party-id number set *number*	(Optional) Specifies that the user part of the Remote-Party-ID header is unconditionally set to the configured ASCII string in the forwarded INVITE message.
asserted-id name *set name*	(Optional) Specifies that the display-name in the Asserted-ID header is unconditionally set to the configured ASCII string in the forwarded INVITE message.
asserted-id number *set number*	(Optional) Specifies that the user part in the Asserted-ID header is unconditionally set to the configured ASCII string in the forwarded INVITE message.

Command Default

This command is disabled.

Command Modes

SIP UA configuration (config-sip-ua)

Command History

Release	Modification
12.2(13)T	This command was introduced.
12.4(15)T	The **asserted-id** keyword was added.

Usage Guidelines

When a call exits the gateway, the **calling-info pstn-to-sip** treatments are applied.

Examples

The following example enables calling information treatment for PSTN-to-SIP calls and sets the company name and number:

```
Router(config-sip-ua)# calling-info pstn-to-sip from name set CompanyA
Router(config-sip-ua)# calling-info pstn-to-sip from number set 5550101
Router(config-sip-ua)# exit
Router(config)# exit

Router# show running-config
Building configuration...

.
.
.
!
sip-ua
 calling-info pstn-to-sip from name set CompanyA
 calling-info pstn-to-sip from number set 5550101
 no remote-party-id
!
.
.
.
```

Related Commands

Command	Description
asserted-id	Sets the privacy level and enables either P-Asserted-Identity (PAI) or P-Preferred-Identity (PPI) privacy headers in outgoing SIP requests or response messages.
calling-info sip-to-pstn	Specifies calling information treatment for SIP-to-PSTN calls.
debug ccsip events	Enables tracing of SIP SPI events.
debug ccsip messages	Enables tracing SIP messages exchanged between the SIP UA client and the access server.
debug isdn q931	Displays call setup and teardown of ISDN connections.
debug voice ccapi error	Enables tracing error logs in the call control API.
debug voip ccapi in out	Enables tracing the execution path through the call control API.

calling-info sip-to-pstn

To specify calling information treatment for Session Initiation Protocol (SIP) to public switched telephone network (PSTN) calls, use the **calling-info sip-to-pstn** command in SIP UA configuration mode. To disable calling information treatment for SIP-to-PSTN calls, use the **no** form of this command.

calling-info sip-to-pstn {**unscreened discard** | **name set** *name* | **number set** *number*}

no calling-info sip-to-pstn

Syntax Description

unscreened *discard*	(Optional) Specifies that the calling name and number be discarded.
name set *name*	(Optional) Specifies that the calling name be unconditionally set to the configured ASCII string in the forwarded Setup mesage.
number set *number*	(Optional) Specifies that the calling number be unconditionally set to the configured ASCII string in the forwarded Setup message.

Command Default

This command is disabled.

Command Modes

SIP UA configuration

Command History

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

Usage Guidelines

When a call enters the gateway, the **calling-info sip-to-pstn** treatments are applied.

Examples

The following example enables calling information treatment for SIP-to-PSTN calls and sets the company name to CompanyA and the number to 5550100:

```
Router(config-sip-ua)# calling-info sip-to-pstn name set CompanyA
Router(config-sip-ua)# calling-info sip-to-pstn number set 5550100
Router(config-sip-ua)# exit
Router(config)# exit

Router# show running-config
Building configuration...

.
.
.
!
sip-ua
 calling-info sip-to-pstn name set CompanyA
     calling-info sip-to-pstn number set 5550100
!
.
.
.
```

Related Commands

Command	Description
debug ccsip events	Enables tracing of SIP SPI events.
debug ccsip messages	Enables SIP SPI message tracing.
debug isdn q931	Displays call setup and teardown of ISDN connections.
debug voip ccapi in out	Enables tracing the execution path through the call control API.
calling-info pstn-to-sip	Specifies calling information treatment for PSTN-to-SIP calls.

calling-number outbound

To specify automatic number identification (ANI) to be sent out when T1-channel-associated signaling (T1-CAS) Feature Group D-Exchange Access North American (FGD-EANA) is configured as the signaling type, use the **calling-number outbound** command in dial peer or voice-port configuration mode. To disable this command, use **no** form of this command.

calling-number outbound {**range** *string1 string2* | **sequence** *string1*... *string5* | **null**}

no calling-number outbound {**range** *string1 string2* | **sequence** *string1*... *string5* | **null**}

Syntax Description

range	Generates the sequence of ANI by rotating through the specified range (*string1* to *string2*).
sequence	Configures a sequence of discrete strings (*string1*... *string5*) to be passed out as ANI for successive calls using the peer **Note** The ellipses (**...**) is entered as shown above.
null	Suppresses ANI. If used, no ANI is passed when this dial peer is selected.
string#...	Valid E.164 telephone number strings. Strings must be of equal length and cannot be more than 32 digits long.

Command Default

No outbound calling number is specified.

Command Modes

Dial peer configuration
Voice-port configuration

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco AS5300.

Usage Guidelines

This command is effective only for FGD-EANA signaling.

Examples

Use the **calling-number outbound** command to enable or disable the passing of ANI on a T1-CAS FGD-EANA configured T1 interface for outgoing calls. Syntax for this command is the same for both voice-port mode and dial peer mode. Examples are given for both modes.

calling-number outbound Range

```
calling-number outbound range string1 string2
```

The values *string1* and *string2* are valid E.164 telephone number strings. Both strings must be of the same length and cannot be more than 32 digits long. Only the last four digits are used for specifying the range (string1 to string2) and for generating the sequence of ANI by rotating through the range until *string2* is reached and then starting from *string1* again. If strings are fewer than four digits in length, then entire strings are used.

ANI is generated by using the 408555 prefix and by rotating through 0100 to 0101 for each call using this peer.

Dial peer configuration mode:

```
dial-peer voice 1 pots
 calling-number outbound range 4085550100 4085550101
 calling Number Outbound is effective only for fgd_eana signaling
```

Voice-port configuration mode:

```
voice-port 1:D
 calling-number outbound range 4085550100 4085550105
 Calling Number Outbound is effective only for fgd_eana signaling
```

calling-number outbound Sequence

```
calling-number outbound sequence string1 string2 string3
string4 string5
```

This option configures a sequence of discrete strings (*string1… string5*) to be passed out as ANI for successive calls using the peer. The limit is five strings. All strings must be valid E.164 numbers, up to 32 digits in length.

Dial peer configuration mode:

```
dial-peer voice 1 pots
 calling-number outbound sequence 6000 6006 4000 5000 5025
 Calling Number Outbound is effective only for fgd_eana signaling
```

Voice-port configuration mode:

```
voice-port 1:D
 calling-number outbound sequence 6000 6006 4000 5000 5025
 Calling Number Outbound is effective only for fgd_eana signaling
```

calling-number outbound Null

```
calling-number outbound null
```

This option suppresses ANI. If used, no ANI is passed when this dial peer is selected.

Dial peer configuration mode:

```
dial-peer voice 1 pots
 calling-number outbound null
 Calling Number Outbound is effective only for fgd_eana signaling
```

Voice-port configuration mode:

```
voice-port 1:D
 calling-number outbound null
 Calling Number Outbound is effective only for fgd_eana signaling
```

Related Commands

Command	Description
info-digits string1	Configures two information digits to be prepended to the ANI string.

capacity update interval (dial peer)

To change the capacity update for prefixes associated with this dial peer, use the **capacity update interval** command in dial peer configuration mode. To return to the default, use the **no** form of this command.

capacity update interval *seconds*

no capacity update interval *seconds*

Syntax Description

seconds	Interval, in seconds, between the sending of periodic capacity updates. This can be a number in the range 10 to 1000. The default value is 25 seconds.

Command Default

25 seconds

Command Modes

Dial peer configuration

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

The update interval should be set depending how many updates that are sent. Updates are sent more often when more calls are coming in, which can lead to data getting out of synchrony. If the interval is too short for the number of updates, the location server can be overwhelmed.

If a dial peer gets too much traffic, set the *seconds* argument to a higher value.

Examples

The following example shows that POTS dial peer 10 is having the capacity update occur every 35 seconds:

```
Router(config)# dial-peer voice 10 pots
Router(config-dial-peer)# capacity update interval 35
```

Related Commands

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

capacity update interval (trunk group)

To change the capacity update for carriers or trunk groups, use the **capacity update interval** command in trunk group configuration mode. To return to the default, use the **no** form of this command.

capacity {**carrier** | **trunk-group**} **update interval** *seconds*

no capacity {**carrier** | **trunk-group**}

Syntax Description

carrier	Carrier capacity.
trunk-group	Trunk group capacity.
seconds	Interval, in seconds, between the sending of periodic capacity updates. This can be a number in the range 10 to 1000. The default value is 25 seconds.

Command Default

25 seconds

Command Modes

Trunk group configuration

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

The update interval should be set depending how many updates that are sent. Updates are sent more often when more calls are coming in, which can lead to data getting out of synchrony. If the interval is too short for the number of updates, the location server can be overwhelmed.

If a dial peer gets too much traffic, set the *seconds* argument to a higher value.

Examples

The following example sets the capacity update for trunk group 101 to occur every 45 seconds:

```
Router(config)# trunk group 101
Router(config-trunkgroup)# capacity trunk-group update interval 45
```

Related Commands

Command	Description
trunk group	Defines the trunk group and enters trunk group configuration mode.

cap-list vfc

To add a voice codec overlay file to the capability file list, use the **cap-list vfc** command in global configuration mode. To disable a particular codec overlay file that has been added to the capability list, use the **no** form of this command.

cap-list *filename* **vfc** *slot-number*

no cap-list *filename* **vfc** *slot-number*

Syntax Description

filename	Identifies the codec file stored in voice feature card (VFC) flash memory.
slot-number	Identifies the slot where the VFC is installed. Range is 0 to 2. There is no default value.

Command Default

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
11.3NA	This command was introduced on the Cisco AS5300.

Usage Guidelines

When VCWare is unbundled, it automatically adds DSPWare to flash memory, creates both the capability and default file lists, and populates these lists with the default files for the particular version of VCWare. The capability list defines the available voice codecs for H.323 capability negotiation. Use the **cap-list vfc** command to add the indicated voice codec overlay file (defined by *filename*) to the capability file list in flash memory.

Examples

The following example adds the following codec to the list included in flash memory:

```
config terminal
 cap-list cdc-g711-1.0.14.0.bin vfc 0
```

Related Commands

Command	Description
default-file vfc	Specifies an additional (or different) file from the ones in the default file list and stored in VFC Flash memory.

card type (t1/e1)

To configure the card type, use the **card type** command in global configuration mode. To restore the default value, use the **no** form of this command.

card type {**t1** | **e1**} *slot [bay]*

no card type {**t1** | **e1**} *slot [bay]*

Syntax Description

t1	Specifies T1 connectivity of 1.544 Mbps through the telephone switching network, using AMI or B8ZS coding.
e1	Specifies a wide-area digital transmission scheme used predominantly in Europe that carries data at a rate of 2.048 Mbps.
slot	Slot (port) number of the interface.
bay	(Optional) Card interface bay number in a slot (route/switch processor [RSP] platform only). This option is not available on other platforms.

Command Default

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.0(5)XE	This command was introduced.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0(7)T.
12.3(1)	This command was integrated into Cisco IOS Release 12.3(1) and support was added for Cisco 2610XM, Cisco 2611XM, Cisco 2620XM, Cisco 2621XM, Cisco 2650XM, Cisco 2651XM, Cisco 2691, Cisco 3631, Cisco 3660, Cisco 3725, and Cisco 3745 platforms.

Usage Guidelines

Changes made using this command do not take effect unless the **reload** command is used or the router is rebooted.

Examples

The following example configures T1 data transmission on slot 1 (port 1) of the router:

```
card type t1 1
```

Related Commands

Command	Description
controller	Configures a T1 or E1 controller and enters controller configuration mode.
reload	Reloads the operating system.

card type (t3/e3)

To configure the card type on the T3 or E3 controller, use the **card type command in** global configuration mode. To restore the default value, use the **no** form of this command.

card type {**t3** | **e3**} *slot*

no card type {**t3** | **e3**} *slot*

Syntax Description

t3	Specifies T3 connectivity of 44210 kbps through the network, using B8ZS coding.
e3	Specifies a wide-area digital transmission scheme used predominantly in Europe that carries data at a rate of 34010 kbps.
slot	Slot number of the interface.

Command Default

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
12.1(1)T	This command was introduced.
12.2(11)YT	This command was integrated into Cisco IOS Release 12.2(11)YT and implemented on the following platforms: Cisco 2650XM, Cisco 2651XM, Cisco 2691, Cisco 3660 series, Cisco 3725, and Cisco 3745 routers.
12.2(15)T	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

Once a card type is issued, the user can enter the **no card type** command and then another **card type** command to configure a new card type. The user must save the configuration to the NVRAM and reboot the router in order for the new configuration to take effect.

When the router comes up, the software comes up with the new card type. Note that the software will reject the configuration associated with the old controller and old interface. The user will now have to configure the new controller and serial interface and save it.

Examples

The following example shows T3 data transmission configured in slot 1:

```
card type t3 1
```

Related Commands

Command	Description
controller	Configures a T3 or E3 controller and enters controller configuration mode.
reload	Reloads the operating system.

carrier-id (dial peer)

To specify the carrier associated with a VoIP call in a dial peer, use the **carrier-id** command in dial peer configuration mode. To delete the source carrier ID, use the **no** form of this command.

carrier-id {**source** | **target**} *name*

no carrier-id {**source** | **target**} *name*

Syntax Description

source	Indicates the carrier that the dial peer uses as a matching key for inbound dial-peer matching.
target	Indicates the carrier that the dial peer uses as a matching key for outbound dial-peer matching.
name	Specifies the ID of the carrier to use for the call. Valid carrier IDs contain a maximum of 127 alphanumeric characters.

Command Default

No default behavior or values

Command Modes

Dial peer configuration

Command History

Release	Modification
12.2(11)T	This command was introduced.

Usage Guidelines

A Gatekeeper Transaction Message Protocol (GKTMP) route server-based application at the terminating gateway uses the source carrier ID to select a target carrier that routes the call over a plain old telephone service (POTS) line.

The terminating gateway uses the target carrier ID to select a dial peer for routing the call over a POTS line.

Examples

The following example indicates that dial peer 112 should use carrier ID "east17" for outbound dial-peer matching in the terminating gateway:

```
Router(config)# dial-peer voice 112 pots
Router(config-dial-peer)# carrier-id target east17
```

The following example indicates that dial peer 111 should use carrier ID "beta23" for inbound dial-peer matching in the terminating gateway:

```
Router(config)# dial-peer voice 111 voip
Router(config-dial-peer)# carrier-id source beta23
```

Related Commands

Command	Description
translation-profile (dial peer)	Associates a translation profile with a dial peer.
trunkgroup (dial peer)	Assigns a trunk group to a source IP group or dial peer for trunk group label routing.

carrier-id (global)

To set the carrier ID for trunk groups when a local carrier ID is not configured, use the **carrier-id** command in global configuration mode. To disable the carrier ID, use the **no** form of this command.

carrier-id *name* [**cic**]

no carrier-id *name* [**cic**]

Syntax Description

name	Identifier for the carrier ID. Must be four-digit numeric carrier identification code to be advertised as a TRIP carrier family but can be alphanumeric if used otherwise.
cic	(Optional) Specifies that the carrier ID is a circuit identification code (CIC).

Command Default

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

To advertise the carrier as a TRIP carrier family, the **cic** keyword must be used. When the **cic** keyword is used, only numeric values can be accepted for the *name* value. If the **cic** keyword is not used, the *name* value can be alphanumeric but is not advertised to TRIP location servers.

Examples

The following example shows a carrier ID using the circuit identification code:

```
Router(config)# carrier-id 1234 cic
```

Related Commands

Command	Description
carrier-id (trunk group)	Configures the carrier ID locally on the trunk group.

carrier-id (trunk group)

To specify the carrier associated with a trunk group, use the **carrier-id** command in trunk group configuration mode. To delete the source carrier ID, use the **no** form of this command.

carrier-id *name* [**cic**]

no carrier-id *name* [**cic**]

Syntax Description

name	The ID of the carrier to use for the call. Valid carrier IDs contain a maximum of 127 alphanumeric characters. To be advertised as a TRIP carrier family, this must be set to a four-digit numeric carrier identification code.
cic	(Optional) Specifies that the carrier ID is a circuit identification code.

Command Default

No default behavior or values

Command Modes

Trunk group configuration

Command History

Release	Modification
12.2(11)T	This command was introduced.
12.3(1)	The **cic** keyword was added.

Usage Guidelines

In a network, calls are routed over incoming trunk groups and outgoing trunk groups. The *name* arguments identifies the carrier that handles the calls for a specific trunk group. In some cases, the same trunk group may be used to carry both incoming calls and outgoing calls.

The carrier ID configured locally on the trunk group supersedes the globally configured carrier ID.

To advertise the carrier as a TRIP carrier family, the **cic** keyword must be used. When **cic** is used, only numeric values can be accepted for the *name* value. If **cic** is not used, the *name* value can be alphanumeric but is not advertised to TRIP location servers.

Examples

The following example indicates that carrier "alpha1" carries calls for trunk group 5:

```
Router(config)# trunk group 5
Router(config-trunk-group)# carrier-id alpha1
```

The following example shows that the carrier with circuit identification code 1234 carries calls for trunk group 101. This trunk group can carry TRIP advertisements.

```
Router(config)# trunk group 101
Router(config-trunk-group)# carrier-id 1234 cic
```

Related Commands

Command	Description
carrier-id (global)	Configures the carrier ID globally for all trunk groups.
translation-profile (trunk group)	Associates a translation profile with a trunk group.
trunk group	Initiates the definition of a trunk group.

carrier-id (voice source group)

To specify the carrier associated with a VoIP call, use the **carrier-id** command in voice source group configuration mode. To delete the source carrier ID, use the **no** form of this command.

carrier-id {**source** | **target**} *name*

no carrier-id {**source** | **target**} *name*

Syntax Description

source	Indicates the carrier ID associated with an incoming VoIP call at the terminating gateway.
target	Indicates the carrier ID used by the terminating gateway to match an outbound dial peer.
name	The ID of the carrier to use for the call. Valid carrier IDs contain a maximum of 127 alphanumeric characters.

Command Default

No default behavior or values

Command Modes

Voice source group configuration

Command History

Release	Modification
12.2(11)T	This command was introduced.

Usage Guidelines

A Gatekeeper Transaction Message Protocol (GKTMP) server application at the terminating gateway uses the source carrier ID to select a target carrier that routes the call over a plain old telephone service (POTS) line. The terminating gateway uses the target carrier ID to select a dial peer for routing the call over a POTS line.

Note If an incoming H.323 VoIP call matches a source IP group that has a target carrier ID, the source IP group's target carrier ID overrides the VoIP call's H.323 setup message.

Examples

The following example indicates that voice source IP group "group1" should use carrier ID named "source3" for incoming VoIP calls and carrier ID named "target17" for outbound dial-peer matching in the terminating gateway:

```
Router(config)# voice source-group group1
Router(cfg-source-grp)# carrier-id source source3
Router(cfg-source-grp)# carrier-id target target17
```

Related Commands

Command	Description
voice source-group	Initiates the definition of a source IP group.

cause-code

To represent internal failures with former and nonstandard H.323 or Session Initiation Protocol (SIP) cause codes, use the **cause-code** command in voice service VoIP configuration mode. To use standard cause-code categories, use the **no** form of this command.

cause-code legacy

no cause-code legacy

Syntax Description

legacy	Sets the internal cause code to the former and nonstandard set of H.323 and SIP values.

Command Default

The default for SIP and H.323 is to use standard cause-code categories, so the command is disabled.

Command Modes

Voice service VoIP configuration

Command History

Release	Modification
12.2(11)T	This command was introduced.

Usage Guidelines

This command is used for backward compatibility purposes.

Examples

The following example sets the internal cause codes to the former and nonstandard set of SIP and H.323 values for backward compatibility:

```
Router(config)# voice service voip
Router(config-voi-srv)# cause-code legacy
```

Related Commands

Command	Description
show call history voice	Displays the call history table for voice calls.

ccm-manager application redundant-link port

To configure the port number for the redundant link application, use the **ccm-manager application redundant-link port** command in global configuration mode. To disable the configuration, use the **no** form of this command.

ccm-manager application redundant-link port *number*

no ccm-manager application redundant-link port

Syntax Description

port *number*	Port number for the transport protocol. The protocol may be User Data Protocol (UDP), Reliable User Datagram Protocol (RDUP), or TCP. Range is from 0 to 65535, and the specified value must not be a well-known reserved port number such as 1023. The default is 2428.

Command Default

Port number: 2428

Command Modes

Global configuration

Command History

Release	Modification
12.1(3)T	This command was introduced with Cisco CallManager Version 3.0 and the Cisco Voice Gateway 200 (VG200).
12.2(2)XA	The command was implemented on the Cisco 2600 series and Cisco 3600 series.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T.

Usage Guidelines

Use this command only when defining an application-specific port other than the default.

Examples

In the following example, the port number of the redundant link application is 2429:

```
ccm-manager application redundant-link port 2429
```

Related Commands

Command	Description
ccm-manager redundant-host	Configures the IP address or the DNS name of up to two backup Cisco CallManagers.
ccm-manager switchback	Configures the switchback mode that determines when the primary Cisco CallManager is used if it becomes available again while a backup Cisco CallManager is being used.

ccm-manager config

To specify the TFTP server from which the Media Gateway Control Protocol (MGCP) gateway downloads Cisco Unified Communications Manager (Cisco UCM) Extensible Markup Language (XML) configuration files and to enable the download of the configuration, use the **ccm-manager config** command in global configuration mode. To disable the dial-peer and server configurations, use the **no** form of this command.

ccm-manager config [**dialpeer-prefix** *prefix* | **server** {*ip-address* | *name*}]

no ccm-manager config [**dialpeer-prefix** *prefix* | **server**]

Syntax Description

dialpeer-prefix *prefix*	(Optional) Specifies the prefix to use for autogenerated dial peers. Range is 1 to 2147483647. The default is 999. **Note** When manually adding a dial peers prefix, select a prefix number other than the default.
server {*ip-address* | *name*}	(Optional) Specifies the IP address or logical name of the TFTP server from which the XML configuration files are downloaded. The arguments are as follows: • *ip-address*—IP address of the TFTP server from which to download the XML configuration files to the local MGCP voice gateway. • *name*—Logical (symbolic) name of the TFTP server from which to download XML configuration files to the local MGCP voice gateway.

Command Default

The configuration download feature is disabled.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(2)XN	This command was introduced and implemented on the Cisco 2600 series, Cisco 3600 series, and the Cisco VG200.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco IAD2420 series.

Usage Guidelines

The **ccm-manager config** command is required to enable the download of Cisco UCM XML configuration files. If you separate the MGCP and H.323 dial peers under different dial-peer tags, ensure that the MGCP dial peers are configured before the H.323 dial peers. Direct-inward-dial (DID) is required for E1 PRI dial peers.

Note To keep manually added dial peers from being deleted from the running configuration when Cisco UCM downloads the configuration to the gateway, use a dial peer-prefix value other than the default (999).

Do not delete the POTS dial peer created by the automatic download process. However, if a dial peer *has* been deleted, you can restore the deleted dial peer by entering the following commands to repeat the download of the configuration file:

```
no mgcp
no ccm-manager config
ccm-manager config
mgcp
```

After you enter these commands, use the **show ccm-manager config-download** command to display the the configuration file downloaded from the TFTP server via the interface specified. The following is an example of the output:

```
Loading sample.cnf.xml from 9.13.22.100 (via GigabitEthernet0/0): !
[OK - 12759 bytes]
```

Examples

The following example shows how to enable the automatic download of configuration files:

```
ccm-manager config
```

In the following example, the IP address of the TFTP server from which a configuration file is downloaded is identified:

```
ccm-manager config server 10.10.0.21
```

Related Commands

Command	Description
debug ccm-manager config-download	Displays dialog during configuration download from the Cisco UCM to the gateway.
show ccm-manager config-download	Displays whether the Cisco UCM configuration is enabled.

ccm-manager download-tones

To configure a Cisco IOS gateway to download a XML configuration file that contains custom tone information from a TFTP server at the time of gateway registration, use the **ccm-manager download-tones** command in global configuration mode. To disable this functionality, use the **no** form of this command.

ccm-manager download-tones

no ccm-manager download-tones

Syntax Description

This command has no arguments or keywords.

Command Default

Cisco CallManager download tones are disabled.

Command Modes

Global configuration

Command History

Release	Modification
12.2(15)ZJ	This command was introduced.
12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T.

Examples

The following example shows a Cisco IOS gateway being configured to download an XML configuration file that contains custom tone information from a TFTP server:

```
Router(config)# ccm-manager download-tones
```

Related Commands

Command	Description
cptone	Specifies a regional voice-interface-related tone, ring, and cadence setting.
debug ccm-manager	Displays debugging of Cisco CallManager.
show ccm-manager	Displays a list of Cisco CallManager servers and their current status and availability.

ccm-manager fallback-mgcp

To enable the gateway fallback feature and allow a Media Gateway Control Protocol (MGCP) voice gateway to provide call processing services when Cisco CallManager is unavailable, use the **ccm-manager fallback-mgcp** command in global configuration mode. To disable fallback on the MGCP voice gateway, use the **no** form of this command.

ccm-manager fallback-mgcp

no ccm-manager fallback-mgcp

Syntax Description

This command has no arguments or keywords.

Command Default

The gateway fallback feature is enabled

Command Modes

Global configuration

Command History

Release	Modification
12.2(2)XN	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and the Cisco VG200.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and Cisco CallManager Version 3.2 and implemented on Cisco IAD2420 series.
12.2(15)ZJ	This command was integrated into Cisco IOS Release 12.2(15)ZJ.
12.3(2)T	This command was implemented on the Cisco 26xxXM, Cisco 2691, Cisco 3640, Cisco 3640A, Cisco 3660, and Cisco 37xx.

Usage Guidelines

This command causes the gateway to fall back and provide call processing services if connectivity is lost between the gateway and all Cisco CallManager servers. The mode and timing are set by default.

Examples

The following example enables fallback:

```
Router(config)# ccm-manager fallback-mgcp
```

Related Commands

Related Command	Purpose
ccm-manager config	Supplies the local MGCP voice gateway with the IP address or logical name of the TFTP server from which to download XML configuration files and enable the download of the configuration.
debug ccm-manager	Displays debugging information about the Cisco CallManager.
show ccm-manager fallback-mgcp	Displays the status of the MGCP gateway fallback feature.

ccm-manager fax protocol

To enable fax-relay protocol for endpoints on a gateway, use the **ccm-manager fax protocol** command in global configuration mode. To disable fax-relay protocol, use the **no** form of this command.

ccm-manager fax protocol cisco

no ccm-manager fax protocol cisco

Syntax Description

protocol cisco	Cisco-proprietary fax-relay protocol. This is the only choice.

Command Default

Cisco-proprietary fax-relay protocol is enabled by default.

Command Default

protocol cisco

Command Modes

Global configuration

Command History

Release	Modification
12.2(9)T	This command was introduced.

Usage Guidelines

Use the **no** form of this command to disable fax relay.

Because fax relay is enabled by default, the **show running-config** command does not explicitly show it to be enabled.

Fax over IP enables interoperability of traditional analog fax machines with IP telephony networks. In its original form, fax data is digital. For transmission across a traditional public switched telephone network (PSTN), it is converted to analog form. For transmission across the IP (packet) network, it is reconverted to digital form, and then, at the destination fax machine, converted again to analog form.

Most Cisco voice gateways support two methods of transmitting fax traffic across the IP network:

- Cisco fax relay—The gateway terminates the T.30 fax signaling. This is the preferred method.
- Fax pass-through—The gateway does not distinguish a fax call from a voice call. All Cisco voice gateways support fax pass-through.

Examples

The following example configures a Media Gateway Control Protocol (MGCP) gateway for Cisco fax relay:

```
Router(config)# ccm-manager fax protocol cisco
Router(config)# mgcp fax t38 inhibit
```

The following example configures an MGCP gateway for fax pass-through:

```
Router(config)# ccm-manager fax protocol cisco
Router(config)# mgcp modem passthrough voip mode nse
Router(config)# mgcp modem passthrough voip codec g711ulaw
```

Related Commands

Command	Description
debug ccm-manager	Displays debugging of Cisco CallManager.
show ccm-manager	Displays a list of Cisco CallManager servers and their current status and availability.
show running-config	Displays the contents of the currently running configuration file.

ccm-manager mgcp

To enable the gateway to communicate with Cisco CallManager through the Media Gateway Control Protocol (MGCP) and to supply redundant control agent services, use the **ccm-manager mgcp** command in global configuration mode. To disable communication with Cisco CallManager and redundant control agent services, use the **no** form of this command.

ccm-manager mgcp

no ccm-manager mgcp

Syntax Description

This command has no arguments or keywords.

Command Default

Cisco CallManager does not communicate with the gateway through MGCP.

Command Modes

Global configuration

Command History

Release	Modification
12.1(3)T	This command was introduced with Cisco CallManager Version 3.0 on the Cisco VG200.
12.2(2)XA	The command was implemented on the Cisco 2600 series and Cisco 3600 series.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series, 3600 series, and Cisco VG200.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T.
12.2(11)T	This command was integrated into the Cisco IOS Release 12.2(11)T and Cisco CallManager Version 3.2 and was implemented on the Cisco IAD2420 series routers.
12.2(11)YU	This command was implemented on the Cisco 1760 gateway.

Usage Guidelines

This command enables the gateway to communicate with Cisco CallManager through MGCP. This command also enables control agent redundancy when a backup Cisco CallManager server is available.

Examples

In the following example, support for Cisco CallManager and redundancy is enabled within MGCP:

```
ccm-manager mgcp
```

Related Commands

Command	Description
ccm-manager redundant-host	Configures the IP address or the DNS name of up to two backup Cisco CallManagers.
ccm-manager switchback	Configures the switchback mode that determines when the primary Cisco CallManager is used if it becomes available again while a backup Cisco CallManager is being used.
mgcp	Enables Media Gateway Control Protocol mode.

ccm-manager music-on-hold

To enable the multicast music-on-hold (MOH) feature on a voice gateway, use the **ccm-manager music-on-hold** command in global configuration mode. To disable the MOH feature, use the **no** form of this command.

ccm-manager music-on-hold

no ccm-manager music-on-hold

Syntax Description

This command has no arguments or keywords.

Command Default

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.2(2)XN	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco VG200.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and Cisco CallManager Version 3.2 and implemented on the Cisco IAD2420 series routers.

Examples

The following example shows multicast MOH configured for a MGCP voice gateway:

```
mgcp call-agent 10.0.0.21 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000
mgcp modem passthrough voip mode cisco
mgcp package-capability rtp-package
mgcp package-capability sst-package
no mgcp timer receive-rtcp
call rsvp-sync
!
ccm-manager redundant-host 10.0.0.21
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server 10.0.0.21
!
```

Related Commands

Command	Description
ccm-manager music-on-hold bind	Enables the multicast MOH feature on a voice gateways.

Command	Description
debug ccm-manager music-on-hold	Displays debugging information for MOH.
show ccm-manager music-on-hold	Displays MOH information.

ccm-manager music-on-hold bind

To bind the multicast music-on-hold (MOH) feature to a designated interface, use the **ccm-manager music-on-hold bind** command in global configuration mode. To unbind the MOH feature on the voice gateway, use the **no** form of this command.

ccm-manager music-on-hold bind *type slot*/*port*

no ccm-manager music-on-hold bind *type slot*/*port*

Syntax Description

type	Interface type to which the MOH feature is bound. The options follow: • **async**—Asynchronous interface • **bvi**—Bridge-Group Virtual Interface • **ctunnel**—CTunnel interface • **dialer**—Dialer interface • **ethernet**—IEEE 802.3 • **lex**—Lex interface • **loopback**—Loopback interface • **mfr**—Multilink Frame Relay bundle interface • **multilink**—Multilink interface • **null**—Null interface • **serial**—Serial interface • **tunnel**—Tunnel interface • **vif**—PGM Multicast Host interface • **virtual-FrameRelay**—Virtual Frame Relay interface • **virtual-Template**—Virtual template interface • **virtual-TokenRing**—Virtual Token Ring
slot/*port*	Number of the slot being configured. See the appropriate hardware manual for slot and port information.

Command Default

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.2(11)T	This command was introduced.

Examples

The following example shows multicast MOH bound to serial interface 0/0:

```
ccm-manager music-on-hold bind serial 0/0
```

Related Commands

Command	Description
ccm-manager music-on-hold	Enables the MOH feature.
debug ccm-manager music-on-hold	Displays debugging information for MOH.
show ccm-manager music-on-hold	Displays MOH information.

ccm-manager redundant-host

To configure the IP address or the Domain Name System (DNS) name of one or two backup Cisco CallManager servers, use the **ccm-manager redundant-host** command in global configuration mode. To disable the use of backup Cisco CallManager servers as call agents, use the **no** form of this command.

ccm-manager redundant-host {*ip-address* | *dns-name*} [*ip-address* | *dns-name*]

no ccm-manager redundant-host {*ip-address* | *dns-name*} [*ip-address* | *dns-name*]

Syntax Description

ip-address	IP address of the backup Cisco CallManager server.
dns-name	DNS name of the backup Cisco CallManager server.

Command Default

If you do not configure a backup Cisco CallManager, the redundancy is disabled.

Command Modes

Global configuration

Command History

Release	Modification
12.1(3)T	This command was introduced with Cisco CallManager Version 3.0 on the Cisco Voice Gateway 200 (VG200).
12.2(2)XA	The command was implemented on the Cisco 2600 series and Cisco 3600 series. The *dns-name* argument was added.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series, 3600 series, and the Cisco VG200.
12.2(11)T	This command was integrated into the Cisco IOS Release 12.2(11)T and Cisco CallManager Version 3.2 and implemented on the Cisco IAD2420 series routers.

Usage Guidelines

The list of IP addresses or DNS names is an ordered and prioritized list. The Cisco CallManager server that was defined with the **mgcp call-agent** command has the highest priority—it is the primary Cisco CallManager server. The gateway selects a Cisco CallManager server on the basis of the order of its appearance in this list.

Examples

In the following example, the IP address 10.0.0.50 is configured as the backup Cisco CallManager :

```
ccm-manager redundant-host 10.0.0.50
```

Related Commands

Command	Description
ccm-manager application	Configures the port number for the redundant link application.
ccm-manager switchback	Configures the switchback mode that determines when the primary Cisco CallManager is used if it becomes available again while a backup Cisco CallManager is being used.
ccm-manager switchover-to-backup	Redirects (manually and immediately) a Cisco 2600 series router or Cisco 3600 series router to the backup Cisco CallManager server.
mgcp call-agent	Defines the Cisco CallManager server as the highest priority.

ccm-manager sccp

To enable Cisco CallManager autoconfiguration of the Cisco IOS gateway, use the **ccm manager sccp** command in global configuration mode. To disable autoconfiguration, use the **no** form of this command.

ccm-manager sccp

no ccm-manager sccp

Syntax Description

This command has no arguments or keywords.

Command Default

Autoconfiguration is disabled.

Command Modes

Global configuration

Command History

Release	Modification
12.3(14)T	This command was introduced.

Usage Guidelines

Use this command to trigger TFTP download of the eXtensible Markup Language (XML) configuration file. Issuing this command immediately triggers the download, and also enables the Skinny Client Control Protocol (SCCP) and SCCP Telephony Control Application (STCAPP), applications that enable Cisco CallManager control of gateway-connected telephony endpoints.

Examples

The following example enables autoconfiguration of gateway-connected endpoints:

```
Router(config)# ccm-manager sccp
```

Related Commands

Command	Description
ccm-manager config	Specifies the TFTP server from which the Cisco IOS gateway downloads Cisco CallManager XML configuration files.
ccm-manager sccp local	Selects the local interface for SCCP application use for Cisco CallManager registration.
show ccm-manager config-download	Displays information about the status of the Cisco IOS gateway configuration download.

ccm-manager sccp local

To select the local interface that the Skinny Client Control Protocol (SCCP) application uses to register with Cisco CallManager, use the **ccm-manager sccp local** command in global configuration mode. To deselect the interface, use the **no** form of this command.

ccm-manager sccp local *interface-type interface-number*

no ccm-manager sccp local *interface-type interface-number*

Syntax Description

interface-type	Interface type that the SCCP application uses for Cisco CallManager registration.
interface-number	Interface number that the SCCP application uses for Cisco CallManager registration.

Command Default

No local interface is selected.

Command Modes

Global configuration

Command History

Release	Modification
12.3(14)T	This command was introduced.

Usage Guidelines

You must specify this interface before enabling the Cisco CallManager autoconfiguration process. The MAC address of this interface is used to identify gateway endpoints.

Examples

The following example configures a FastEthernet interface for SCCP application use for Cisco CallManager registration:

```
Router(config)# ccm-manager sccp local fastethernet 0/0
```

Related Commands

Command	Description
show ccm-manager	Displays a list of Cisco CallManager servers and their current status and availability.

ccm-manager shut-backhaul-interfaces

To disable ISDN Layer 2 connectivity on a Cisco Call Manager Media Gateway Control Protocol (MGCP) PRI or BRI backhauled trunk when communication is lost between the Cisco Call Manager and the MGCP gateway, use the **ccm-manager shut-backhaul-interfaces** command in global configuration mode. To restore the default behavior, where ISDN Layer 2 is maintained between the MGCP gateway and the ISDN switch even when no connectivity exists between the MGCP gateway and any Cisco Call Manager, use the **no** form of this command.

ccm-manager shut-backhaul-interfaces

no ccm-manager shut-backhaul-interfaces

Syntax Description

This command has no arguments or keywords.

Command Default

The default behavior is for the ISDN Layer 2 connection to be maintained (to make the Cisco Call Manager MGCP PRI or BRI backhaul continue to function) between the MGCP gateway and the ISDN switch even if no connectivity exists between the MGCP gateway and any Cisco Call Manager.

Command Modes

Global configuration

Command History

Release	Modification
12.4(8)	This command was introduced.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(3f)	This command was integrated into Cisco IOS Release 12.4(3f).
12.4(5c)	This command was integrated into Cisco IOS Release 12.4(5c).
12.4(7c)	This command was integrated into Cisco IOS Release 12.4(7c).
12.4(4)T5	This command was integrated into Cisco IOS Release 12.4(4)T5.
12.4(6)T4	This command was integrated into Cisco IOS Release 12.4(6)T4.

Usage Guidelines

Use this command on Cisco IOS voice routers configured for Cisco Call Manager MGCP PRI or BRI backhaul.

Prior to the introduction of the **ccm-manager shut-backhaul-interfaces** command, a Cisco Call Manager MGCP PRI or BRI backhaul trunk would maintain ISDN Layer 2 connectivity between the MGCP gateway and the ISDN switch in a MULTIPLE_FRAMES_ESTABLISHED state even if Layer 3 Q.931 backhaul connectivity between the Cisco Call Manager and the MGCP gateway was unavailable. This causes problems because the ISDN switch interprets the PRI or BRI trunk as being active and continues to place calls to the MGCP gateway, even though all of the calls fail. After you enter the **ccm-manager shut-backhaul-interfaces** command, Layer 2 is disabled when connectivity between the Cisco Call Manager and the MGCP gateway is unavailable.

Examples

The following example disables ISDN Layer 2 connectivity on a Cisco Call Manager MGCP PRI or BRI backhauled trunk when communication is lost between Cisco Call Manager and the MGCP gateway:

```
ccm-manager shut-backhaul-interfaces
```

The following example restores the default behavior (functionality of the **ccm-manager shut-backhaul-interfaces** command is disabled) so that the ISDN Layer 2 connection is maintained between the MGCP gateway and the ISDN switch, even when no connectivity exists between the MGCP gateway and any Cisco Call Manager:

```
no ccm-manager mgcp
no ccm-manager shut-backhaul-interfaces
ccm-manager mgcp
```

Related Commands

Command	Description
ccm-manager mgcp	Enables the gateway to communicate with the Cisco Call Manager through the MGCP and to supply redundant control agent services.

ccm-manager shut-interfaces-tftp-fails

To configure the number of TFTP download failures allowed before the gateway shuts down ports, use the **ccm-manager shut-interfaces-tftp-fails** command in global configuration mode. To return to the default configuration, use the **no** form of this command.

ccm-manager shut-interfaces-tftp-fails *retries*

no ccm-manager shut-interfaces-tftp-fails

Syntax Description

retries	Number or TFTP retries. Range is from 2 to 10. The default is 2.

Command Default

Ports shut down after the second TFTP retry. However TFTP download attempts continue.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.4(15)T2	This command was introduced.
12.4(20)T	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use the **ccm-manager shut-interfaces-tftp-fails** command to configure the number of TFTP download failures allowed before the gateway put the port in a shutdown state.

Examples

The following example shows a gateway being configured to put the port in a shutdown state after four TFTP download failures:

```
Router(config)# ccm-manager shut-interfaces-tftp-fails 4
```

Related Commands

Command	Description
show ccm-manager	Displays a list of Cisco Unified Communications Manager servers and their current status and availability.

ccm-manager switchback

To specify the time when control is to be returned to the primary Cisco CallManager server once it becomes available, use the **ccm-manager switchback** command in global configuration mode. To reset to the default, use the **no** form of this command.

ccm-manager switchback {**graceful** | **immediate** | **schedule-time** *hh:mm* | **uptime-delay** *minutes*}

no ccm-manager switchback

Syntax Description

graceful	Specifies that control is returned to the primary Cisco CallManager server after the last active call ends (when there is no voice call in active setup mode on the gateway). Default value.
immediate	Specifies an immediate switchback to the primary Cisco CallManager server when the TCP link to the primary Cisco CallManager server is established, regardless of current call conditions.
schedule-time *hh:mm*	Specifies an hour and minute, based on a 24-hour clock, when control is returned to the primary Cisco CallManager server. If the specified time is earlier than the current time, the switchback occurs at the specified time on the following day.
uptime-delay *minutes*	Specifies the number, of minutes the primary Cisco CallManager server must run after the TCP link to is reestablished and control is returned to that primary call agent. Valid values are from 1 to 1440 (1 minute to 24 hours).

Command Default

Graceful switchback

Command Modes

Global configuration

Command History

Release	Modification
12.1(3)T	This command was introduced with Cisco CallManager Version 3.0 on the Cisco VG200.
12.2(2)XA	The command was implemented on the Cisco 2600 series and Cisco 3600 series.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series, 3600 series, and the Cisco VG200.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T.
12.2(11)T	This command was implemented on the Cisco IAD2420 series routers.

Usage Guidelines

This command allows you to configure switchback to the higher priority Cisco CallManager when it becomes available. Switchback allows call control to revert to the original (primary) Cisco CallManager once service has been restored.

Examples

In the following example, the primary Cisco CallManager is configured to be used as soon as it becomes available:

```
ccm-manager switchback immediate
```

Related Commands

Command	Description
ccm-manager application	Configures the port number for the redundant link application.
ccm-manager redundant-host	Configures the IP address or the DNS name of up to two backup Cisco CallManagers.
ccm-manager switchover-to-backup	Redirects a Cisco 2600 series or Cisco 3600 series router to the backup Cisco CallManager.

ccm-manager switchover-to-backup

To manually redirect a gateway to the backup Cisco CallManager server, use the **ccm-manager switchover-to-backup** command in privileged EXEC mode.

ccm-manager switchover-to-backup

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(2)XN	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco VG200.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and Cisco CallManager Version 3.2 and implemented on the Cisco IAD2420 series.

Usage Guidelines

Switchover to the backup Cisco CallManager server occurs immediately. This command does not switch the gateway to the backup Cisco CallManager server if you have the **ccm-manager switchback** command option set to "immediate" and the primary Cisco CallManager server is still running.

Examples

In the following example, the backup Cisco CallManager server is configured to be used as soon as it becomes available:

```
ccm-manager switchover-to-backup
```

Related Commands

Command	Description
ccm-manager application redundant-link	Configures the port number for the redundant link application (that is, for the secondary Cisco CallManager server).
ccm-manager redundant-host	Configures the IP address or the DNS name of up to two backup Cisco CallManager servers.
ccm-manager switchback	Specifies the time at which control is returned to the primary Cisco CallManager server once the server is available.