# default (auto-config application) through direct-inward-dial

# default (auto-config application)

**Note** The documentation set for this product strives to use bias-free language. For purposes of this documentation set, bias-free is defined as language that does not imply discrimination based on age, disability, gender, racial identity, ethnic identity, sexual orientation, socioeconomic status, and intersectionality. Exceptions may be present in the documentation due to language that is hardcoded in the user interfaces of the product software, language used based on RFP documentation, or language that is used by a referenced third-party product.

To configure an auto-config application configuration command to its default value, use the **default** command in auto-config application configuration mode.

**default** *command*

## Syntax Description

| | |
|---|---|
| *command* | One of the auto-config application configuration commands. Valid choices are as follows:<br>• **retries**<br>• **server**<br>• **shutdown**<br>• **timeout** |

## Command Default

No default behavior or values

## Command Modes

Auto-config application configuration (auto-config-app)

## Command History

| Release | Modification |
|---|---|
| 12.3(8)XY | This command was introduced on the Communication Media Module. |
| 12.3(14)T | This command was integrated into Cisco IOS Release 12.3(14)T. |

## Examples

The following example shows the **default** command used to set the number of download retry attempts for an auto-config application to its default value:

```
Router(auto-config-app)#
default retries
```

## Related Commands

| Command | Description |
|---|---|
| **auto-config** | Enables auto-configuration or enters auto-config application configuration mode for the SCCP application. |
| **show auto-config** | Displays the current status of auto-config applications. |

# default (MGCP profile)

To configure a Media Gateway Control Protocol (MGCP profile) command to its default value, use the **default**command inMGCP profile configuration mode. To disable the default command, use the **no** form of the command for that profile parameter.

**default** *command*
**no default** *command*

## Syntax Description

| | |
|---|---|
| *command* | One of the MGCP profile commands. Valid choices are as follows:<br>• **call-agent**<br>• **description** (MGCP profile)<br>• **max1 lookup**<br>• **max1 retries**<br>• **max2 lookup**<br>• **max2 retries**<br>• **package persistent**<br>• **timeout tcrit**<br>• **timeout tdinit**<br>• **timeout tdmax**<br>• **timeout tdmin**<br>• **timeout thist**<br>• **timeout tone busy**<br>• **timeout tone cot1**<br>• **timeout tone cot2**<br>• **timeout tone dial**<br>• **timeout tone dial stutter**<br>• **timeout tone mwi**<br>• **timeout tone network congestion**<br>• **timeout tone reorder**<br>• **timeout tone ringback**<br>• **timeout tone ringback connection**<br>• **timeout tone ringing**<br>• **timeout tone ringing distinctive**<br>• **timeout tpar**<br>• **timeout tsmax**<br>• **voice-port** (MGCP profile) |

## Command Default

No default behaviors or values

## Command Modes

MGCP profile configuration (config-mgcp-profile)

### Command History

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

### Usage Guidelines

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

The **default**(MGCP profile) command instructs the MGCP profile to use the default value of the specified command whenever the profile is called. This has the same effect as using the **no** form of the specified command, but the **default** command clearly specifies which commands are using their default values.

To use the default values for more than one command, enter each command on a separate line.

### Examples

The following example shows how to configure the default values for three MGCP profile commands:

```
Router(config)# mgcp profile newyork
Router(config-mgcp-profile)# default max1 retries
Router(config-mgcp-profile)# default timeout tdinit
Router(config-mgcp-profile)# default timeout tone mwi
```

### 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. |

# default (SIP)

To reset a SIP command to its default value, use the **default** command in SIP configuration mode.

**default** *command*

## Syntax Description

| | |
|---|---|
| *command* | One of the SIP configuration commands. Valid choices are:<br>• **bind** : Configures the source address of signaling and media packets to a specific interface's IP address.<br>• **rel1xx** : Enables all SIP provisional responses (other than 100 Trying) to be sent reliably to the remote SIP endpoint.<br>• **session-transport** : Configures the underlying transport layer protocol for SIP messages to TCP or UDP.<br>• **url** : Configures URLs to either the SIP or TEL format for your voip sip calls. |

## Command Default

The default is that binding is disabled (**no bind**).

## Command Modes

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

## Command History

| Release | Modification |
|---|---|
| 12.2(2)XB | This command was introduced on the Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco AS5300, Cisco AS5350, and Cisco AS5400 platforms. |
| 12.2(2)XB2 | This command was implemented on the Cisco AS5850 platform. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T and support was added for the Cisco 3700 series. Cisco AS5300, Cisco AS5350, Cisco AS5850, and Cisco AS5400 platforms were not supported in this release. |
| 12.2(11)T | Support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms. |
| Cisco IOS XE Release 2.5 | This command was integrated into Cisco IOS XE Release 2.5. |

## Examples

The following example shows how to reset the value of the SIP **bind** command:

```
Router(config)# voice serv voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# default bind
```

## Related Commands

| Command | Description |
|---|---|
| **sip** | Enter SIP configuration mode from voice-service VoIP configuration mode. |

# default-file vfc

To specify an additional (or different) file from the ones in the default file list and stored in voice feature card (VFC) flash memory, use the **default file vfc**command in global configuration mode. To delete the file from the default file list, use the **no** form of this command.

**default-file** *filename* **vfc** *slot*
**no default-file** *filename* **vfc** *slot*

## Syntax Description

| | |
|---|---|
| *filename* | Indicates the file to be retrieved from VFC flash memory and used to boot up the system. |
| *slot* | Indicates the slot on the Cisco AS5300 in which the VFC is installed. Range is to 2. There is no default value. |

## Command Default

No default behavior or values

## Command Modes

Global configuration (config)

## Command History

| Release | Modification |
|---|---|
| 11.3(1)NA | This command was introduced on the Cisco AS5300. |
| 12.0(3)T | This command was integrated into Cisco IOS Release 12.0(3)T. |

## 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 that version of VCWare. The default file list includes the files that are used to boot up the system.

Use the **default-file vfc**command to add a specified file to the default file list, replacing the existing default for that extension type.

## Examples

The following example specifies that the bas-vfc-1.0.14.0.bin file, which is stored in VFC flash memory, be added to the default file list:

```
default-file bas-vfc-1.0.14.0.bin vfc 0
```

## Related Commands

| Command | Description |
|---|---|
| **cap-list vfc** | Adds a voice codec overlay file to the capability file list. |
| **delete vfc** | Deletes a file from VFC flash memory. |

# define

To define the transmit and receive bits for North American ear and mouth (E&M), E&M Mercury Exchange Limited Channel-Associated Signaling (MELCAS), and Land Mobile Radio (LMR) voice signaling, use the **define** command in voice-port configuration mode. To restore the default value, use the **no** form of this command.

**define** {**tx-bits** | **rx-bits**} {**seize** | **idle**} {**0000** | **0001** | **0010** | **0011** | **0100** | **0101** | **0110** | **0111** | **1000** | **1001** | **1010** | **1011** | **1100** | **1101** | **1110** | **1111**}
**no define** {**tx-bits** | **rx-bits**} {**seize** | **idle**} {**0000** | **0001** | **0010** | **0011** | **0100** | **0101** | **0110** | **0111** | **1000** | **1001** | **1010** | **1011** | **1100** | **1101** | **1110** | **1111**}

## Syntax Description

| | |
|---|---|
| **tx-bits** | The bit pattern applies to the transmit signaling bits. |
| **rx-bits** | The bit pattern applies to the receive signaling bits. |
| **seize** | The bit pattern defines the seized state. |
| **idle** | The bit pattern defines the idle state. |
| **0000** through **1111** | Specifies the bit pattern. |

## Command Default

The default is to use the preset signaling patterns as defined in American National Standards Institute (ANSI) and European Conference of Postal and Telecommunications Administrations (CEPT) standards, as follows:

- For North American E&M:
  - tx-bits idle 0000 (0001 if on E1 trunk)
  - tx-bits seize 1111
  - rx-bits idle 0000
  - rx-bits seize 1111
- For E&M MELCAS:
  - tx-bits idle 1101
  - tx-bits seize 0101
  - rx-bits idle 1101
  - rx-bits seize 0101
- For LMR:
  - tx-bits idle 0000
  - tx-bits seize 1111
  - rx-bits idle 0000
  - rx-bits seize 1111

## Command Modes

Voice-port configuration (config-voiceport)

## Command History

| Release | Modification |
|---|---|
| 11.3(1)MA3 | 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 | The command was integrated into Cisco IOS Release 12.1(2)T. |
| 12.3(4)XD | The LMR signaling type was added to the signaling types to which this command applies. |
| 12.3(7)T | This command was integrated into Cisco IOS Release 12.3(7)T. |
| 12.3(14)T | This command was implemented on the Cisco 2800 series and Cisco 3800 series. |
| 12.4(2)T | This command was integrated into Cisco IOS Release 12.4(2)T. |

## Usage Guidelines

The **define** command applies to E&M digital voice ports associated with T1/E1 controllers.

Use the **define** command to match the E&M bit patterns with the attached telephony device. Be careful not to define invalid configurations, such as all 0000 on E1, or identical seized and idle states. Use this command with the **ignore** command.

In LMR signaling, the **define** command is used to define polarity on E&M analog and digital voice ports.

## Examples

To configure a voice port on a Cisco 2600 or Cisco 3600 series router that is sending traffic in North American E&M signaling format to convert the signaling to MELCAS format, enter the following commands:

```
voice-port 1/0/0
 define rx-bits idle 1101
 define rx-bits seize 0101
 define tx-bits idle 1101
 define tx-bits seize 0101
```

In this example, reverse polarity is configured on a voice port on a Cisco 3700 series router that is sending traffic in LMR signaling format:

```
voice-port 1/0/0
 define rx-bits idle 1111
 define rx-bits seize 0000
 define tx-bits idle 1111
 define tx-bits seize 0000
```

## Related Commands

| Command | Description |
|---|---|
| **condition** | Manipulates the signaling bit-pattern for all voice signaling types. |
| **ignore** | Configures a North American E&M or E&M MELCAS voice port to ignore specific receive bits. |

# delete vfc

To delete a file from voice feature card (VFC) flash memory, use the **delete vfc**command in privileged EXEC mode.

**delete** *filename* **vfc** *slot*

## Syntax Description

| | |
|---|---|
| *filename* | Specifies the file in VFC flash memory to be deleted. |
| *slot* | Specifies the slot on the Cisco AS5300 in which the specified VFC resides. Range is from 0 to 2. |

## Command Modes

Privileged EXEC (#)

## Command History

| Release | Modification |
|---|---|
| 11.3(1)NA | This command was introduced on the Cisco AS5300. |
| 12.0(3)T | This command was integrated into Cisco IOS Release 12.0(3)T. |

## Usage Guidelines

Use the **delete vfc**command to delete a specific file from VFC flash memory and to remove the file from the default list or capability list if the specified file is included in those lists.

**Note** Deleting a file from VFC flash memory does not free the VFC flash memory space that the file occupied. To free VFC flash memory space, use the **erase vfc** command.

## Examples

The following example deletes the bas-vfc-1.0.14.0.bin file, which is stored in VFC flash memory of the VFC located in slot 0:

```
Router# delete bas-vfc-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. |
| **erase vfc** | Erases the flash memory of a specified VFC. |
| **show vfc directory** | Displays the list of all files that reside on this VFC. |

# description

To specify a description of the digital signal processor (DSP) interface, use the **description** command in voice-port or DSP farm interface configuration mode. To describe a MGCP profile that is being defined, use the **description** command in MGCP profile configuration mode. To specify the name or a brief description of a charging profile, use the **description** command in charging profile configuration mode. To delete a configured description, use the **no** form of the command in the appropriate configuration mode.

**description** *string*
**no description**

## Syntax Description

| | |
|---|---|
| *string* | Character string from 1 to 80 characters for DSP interfaces and MGCP profiles, or from 1 to 99 characters for charging profiles. |

## Command Default

Enabled with a null string. The MGCP profile has no default description. Charging profiles have no default description.

## Command Modes

Voice-port configuration (config-voiceport)
DSP farm interface configuration (config-dspfarm-profile)
MGCP profile configuration (config-mgcp-profile)
Charging profile configuration (ch-prof-conf)

## Usage Guidelines

The use of a special character such as '\'(backslash) and a three or more digit number for the character setting like **description**, results in incorrect translation.

## Command History

| Release | Modification |
|---|---|
| 11.3(1)T | This command was introduced on the Cisco 3600 series and Cisco 7200. |
| 11.3(1)MA | This command in voice-port configuration mode was implemented on the Cisco MC3810. |
| 12.0(5)XE | This command in DSP farm interface configuration mode was modified. |
| 12.1(1)T | The DSP farm interface configuration mode modification was integrated into Cisco IOS Release 12.1(1)T. |
| 12.2(2)XA | This command was implemented on the Cisco AS5300. |
| 12.2(11)T | This command was implimented on the Cisco AS5850 and integrated into Cisco IOS Release 12.2(11)T. |
| 12.3(8)XU | This command was introduced in charging profile configuration mode. |
| 12.3(11)YJ | This command in charging profile configuration mode was integrated into Cisco IOS Release 12.3(11)YJ. |
| 12.3(14)YQ | This command in charging profile configuration mode was integrated into Cisco IOS Release 12.3(14)YQ. |
| 12.4(9)T | This was integrated into Cisco IOS Release 12.4(9)T. |

| Release | Modification |
|---|---|
| 12.2(33)SXH | This command was changed to allow the description to contain spaces. |

## Usage Guidelines

Use the **description** command to describe the DSP interface connection or a defined MGCP profile. The information is displayed when a **show** command is used, and it does not affect the operation of the interface in any way.

In Release 12.2(33)SXH and later releases, you can enter spaces in the description.

## Examples

The following example identifies voice port 1/0/0 as being connected to the purchasing department:

```
voice-port 1/0/0
 description purchasing-dept
```

The following example identifies DSP farm interface 1/0 as being connected to the marketing department:

```
dspint dspfarm 1/0
 description marketing-dept
```

The following example shows a description for an MGCP profile:

```
mgcp profile newyork
 description This is the head sales office in New York.
 dot ...(socket=0)
 S:.
 R:250 NAA09092 Message accepted for delivery
 S:QUIT
 R:221 madeup@abc.com closing connection
 Freeing SMTP ctx at 0x6121D454
 returned from work-routine, context freed
```

The following example describes a charging profile as APN-level default for home users:

```
gprs charging profile
 description APN-level_default_for_home_users
```

## Related Commands

| Command | Description |
|---|---|
| **category** | Identifies the subscriber category to which a charging profile applies. |
| **cdr suppression** | Specifies that CDRs be suppressed as a charging characteristic in a charging profile. |
| **charging profile** | Associates a default charging profile to an access point. |
| **content dcca profile** | Defines a DCCA client profile in a GGSN charging profile. |
| **content postpaid time** | Specifies, as a trigger condition for postpaid users in a charging profile, the time duration limit that when exceeded causes the GGSN to collect upstream and downstream traffic byte counts and close and update the G-CDR for a particular PDP context. |

| Command | Description |
|---|---|
| **content postpaid validity** | Specifies, as a trigger condition in a charging profile, that the amount of time quota granted to a postpaid user is valid. |
| **content postpaid volume** | Specifies, as a trigger condition for postpaid users in a charging profile, the maximum number of bytes that the GGSN maintains across all containers for a particular PDP context before closing and updating the G-CDR. |
| **content rulebase** | Associates a default rule-base ID with a charging profile. |
| **gprs charging characteristics reject** | Specifies that create PDP context requests for which no charging profile can be selected be rejected by the GGSN. |
| **gprs charging container time-trigger** | Specifies a global time limit that when exceeded by a PDP context causes the GGSN to close and update the G-CDR for that particular PDP context. |
| **gprs charging profile** | Creates a new charging profile (or modifies an existing one) and enters charging profile configuration mode. |
| **limit duration** | Specifies, as a trigger condition in a charging profile, the time duration limit that when exceeded causes the GGSN to collect upstream and downstream traffic byte counts and close and update the G-CDR for a particular PDP context. |
| **limit sgsn-change** | Specifies, as a trigger condition in a charging profile, the maximum number of GGSN changes that can occur before closing and updating the G-CDR for a particular PDP context. |
| **limit volume** | Specifies, as a trigger condition in a charging profile, the maximum number of bytes that the GGSN maintains across all containers for a particular PDP context before closing and updating the G-CDR. |
| **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. |
| **tariff-time** | Specifies that a charging profile use the tariff changes configured using the **gprs charging tariff-time**global configuration command. |

# description (ctl file)

To set a description for the Cisco Certificate Trust List (CTL) file, use the **description** command in CTL file configuration mode. To remove the description for the CTL file, use the **no** form of the command.

**description** *description*

## Syntax Description

| | |
|---|---|
| *description* | Description of the CTL file. The maximum length of the description is 100 characters. |

## Command Default

No description is set.

## Command Modes

CTL file configuration mode (config-ctl-file)

## Command History

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

## Usage Guidelines

### Example

The following example shows how to set a description for the CTL file instance:

```
Device(config)# voice-ctl-file myctl
Device(config-ctl-file)# description ctlfile1
```

# description (dial peer)

To add a description to a dial peer, use the **description** command in dial peer configuration mode. To remove the description, use the **no** form of this command.

**description** *string*
**no description**

## Syntax Description

| | |
|---|---|
| *string* | Text string up to 64 alphanumeric characters. |

## Command Default

Disabled

## Command Modes

Dial peer configuration (config-dial-peer)

## Command History

| Release | Modification |
|---|---|
| 12.2(2)T | This command was introduced. |
| Cisco IOS XE Amsterdam 17.2.1r | Introduced support for YANG models. |

## Usage Guidelines

Use this command to include descriptive text about the dial peer. The description displays in **show** command output and does not affect the operation of the dial peer.

## Examples

The following example shows a description included in a dial peer:

```
dial-peer voice 1 pots
 description inbound PSTN calls
```

## Related Commands

| Command | Description |
|---|---|
| **dial-peer voice** | Defines a dial peer. |
| **show dial-peer voice** | Displays configuration information for dial peers. |

# description (DSP farm profile)

To include a description about the digital signal processor (DSP) farm profile, use the **description** command in DSP farm profile configuration mode. To remove a description, use the **no** form of this command.

**description** *text*
**no description** *text*

## Syntax Description

| | |
|---|---|
| *text* | Character string from 1 to 80 characters. |

## Command Default

No default behavior or values

## Command Modes

DSP farm profile configuration (config-dspfarm-profile)

## Command History

| Release | Modification |
|---|---|
| 12.3(8)T | This command was introduced. |

## Usage Guidelines

Use this command to include descriptive text about this DSP farm profile. This information displays in **show** commands and does not affect the operation of the interface.

## Examples

The following example identifies the DSP farm profile as being designated to the art department:

```
Router(config-dspfarm-profile)# description art dept
```

## Related Commands

| Command | Description |
|---|---|
| **codec (DSP farm profile)** | Specifies the codecs supported by a DSP farm profile. |
| **dspfarm profile** | Enters DSP farm profile configuration mode and defines a profile for DSP farm services. |
| **maximum sessions (DSP Farm profile)** | Specifies the maximum number of sessions that need to be supported by the profile. |
| **shutdown (DSP farm profile)** | Allocates DSP farm resources and associates with the application. |

# description (dspfarm)

To include a specific description about the digital signal processor (DSP) interface, use the **description** command in DSP farm interface configuration mode. To disable this feature, use the **no** form of this command.

**description** *string*
**no description** *string*

## Syntax Description

| | |
|---|---|
| *string* | Character string from 1 to 80 characters. |

## Command Default

Enabled with a null string.

## Command Modes

DSP farm interface configuration (config-dspfarm-profile)

## Command History

| Release | Modification |
|---|---|
| 11.3(1)T | This command was introduced for the Cisco 7200 series routers. |
| 12.0(5)XE | This command was modified to reduce the maximum number of allowable characters in a text string from 255 to 80. |
| 12.1(1)T | This command was integrated into Cisco IOS Release 12.1(1)T. |

## Usage Guidelines

Use the **description** command to include descriptive text about this DSP farm interface connection. This information is displayed when you issue a **show**command and does not affect the operation of the interface in any way.

## Examples

The following example identifies DSP farm interface 1/0 on the Cisco 7200 series routers router as being connected to the marketing department:

```
dspint dspfarm 1/0
description marketing dept
```

# description (media-profile)

To include a description specific to a media profile in CUBE, use the **description** command in media profile configuration mode. To remove the description, use the **no** form of this command.

**connection description** *string*
**no connection description** *string*

## Syntax Description

| | |
|---|---|
| *string* | A description specific to the media profile. |

## 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

The **description** command provides details specific to a media profile.

## Examples

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

```
router(cfg-mediaprofile)#description ?
WORD Specify hostname or IP address of proxy server

router(cfg-mediaprofile)#description <text>
```

## 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. |
| **proxy (media-profile)** | Configures IP address or hostname of proxy in 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. |

# description (phone proxy)

To specify a description for the phone proxy, use the **description** command in phone proxy configuration mode. To remove the description, use the **no** form of the command.

**description***description*
**no description**

## Syntax Description

This command has no arguments or keywords.

## Command Default

No description is specified.

## Command Modes

Phone proxy configuration mode (config-phone-proxy)

## Command History

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

## Usage Guidelines

### Example

The following example shows how to create a phone proxy instance called first-pp, enter phone-proxy configuration mode, and set the description for this instance:

```
Device(config)# voice-phone-proxy first-pp
Device(config-phone-proxy)# description cluster-test
```

# description (SCCP Cisco CallManager)

To include a description about the Cisco CallManager group, use the **description** command in SCCP Cisco CallManager configuration mode. To remove a description, use the **no** form of this command.

**description** *text*
**no description**

## Syntax Description

| | |
|---|---|
| *text* | Character string from 1 to 80 characters. |

## Command Default

No default behavior or values

## Command Modes

SCCP Cisco CallManager configuration (config-sccp-ccm)

## Command History

| Release | Modification |
|---|---|
| 12.3(8)T | This command was introduced. |

## Usage Guidelines

Use this command to include descriptive text about a Cisco CallManager group. This information is displayed in **show** commands and does not affect the operation of the interface.

## Examples

The following example identifies SCCP as being designated to the Boston office:

```
Router(config-sccp-ccm)# description boston office
```

## Related Commands

| Command | Description |
|---|---|
| **associate ccm** | Associates a Cisco CallManager with a Cisco CallManager group and establishes its priority within the group. |
| **connect retries** | Specifies the number of times that a DSP farm attempts to connect to a Cisco CallManager when the current Cisco CallManager connections fails. |
| **sccp ccm group** | Creates a Cisco CallManager group and enters SCCP Cisco CallManager configuration mode. |

# description (trunk group)

To add a description to a trunk group, use the **description** command in trunk group configuration mode. To delete the description, use the **no**form of this command.

**description** *string*
**no description** *string*

## Syntax Description

| | |
|---|---|
| *string* | Trunk group description. Maximum length is 63 alphanumeric characters. |

## Command Default

No default behavior or values

## Command Modes

Trunk group configuration (config-trunk-group)

## Command History

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

## Examples

The following example shows a description for a trunk group:

```
Router(config)# trunk group alpha1
Router(config-trunk-group)# description carrierAgroup1
```

## Related Commands

| Command | Description |
|---|---|
| **trunk group** | Initiates the definition of a trunk group. |

# description (voice class)

To provide a TLS profile group description, and associate it to a TLS profile, use the command **description** in voice class configuration mode. To delete the TLS profile group description, use **no** form of this command.

**description** *tls-profile-group-label*
**no description**

## Syntax Description

| | |
|---|---|
| *tls-profile-group-label* | Allows you to provide a description for the TLS profile group. |

## Command Default

No default behavior or values

## Command Modes

Voice class configuration (config-class)

## Command History

| Release | Modification |
|---|---|
| Cisco IOS XE Amsterdam 17.3.1a | This command was introduced under voice class configuration mode. |

## Usage Guidelines

The TLS profile group description is associated to a TLS profile through the command **voice class tls-profile** *tag*. The *tag* associates the TLS profile group description to the command **crypto signaling**.

## Examples

The following example illustrates how to create a voice class tls-profile and associate a description TLS profile group:

```
Router(config)#voice class tls-profile 2
Router(config-class)#description tlsgroupname
```

## Related Commands

| Command | Description |
|---|---|
| **voice class tls-profile** | Provides sub-options to configure the commands that are required for a TLS session. |
| **crypto signaling** | Identifies the trustpoint or the **tls-profile** *tag* that is used during the TLS handshake process. |

# description (voice source group)

To add a description to a voice source group, use the **description** command in voice source-group configuration mode. To delete the description, use the **no**form of this command.

**description** *string*
**no description** *string*

## Syntax Description

| | |
|---|---|
| *string* | Describes a voice source group, Maximum length of the voice source group description is 63 alphanumeric characters. |

## Command Default

No default behavior or values

## Command Modes

Voice source-group configuration (cfg-source-grp)

## Command History

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

## Examples

The following example shows a description for a voice source group:

```
Router(config)# voice source-group northern1
Router(cfg-source-grp)# description carrierBgroup3
```

## Related Commands

| Command | Description |
|---|---|
| **voice source-group** | Defines a source group for voice calls. |

# destination e164-pattern-map

To link an E.164 pattern map to a dial peer, use the **destination e164-pattern-map** command in dial peer configuration mode. To remove the link of an E.164 pattern map from a dial peer, use the **no** form of this command.

**destination e164-pattern-map** *tag*
**no destination e164-pattern-map**

## Syntax Description

| | |
|---|---|
| *tag* | A number that defines a destination E.164 pattern map. The range is from 1 to 10000. |

## Command Default

An E.164 pattern map is not linked to a dial peer.

## Command Modes

Dial peer configuration (config-dial-peer)

## Command History

| Release | Modification |
|---|---|
| 15.2(4)M | This command was introduced. |

## Usage Guidelines

To support dial peer with multiple destination patterns, which involve massive dial peer configuration, use an E.164 destination pattern map. You can create a destination E.164 pattern map and then link it to one or more dial peers. Based on the validation of a pattern map, you can enable or disable one or more dial peers linked to the destination E.164 pattern map. To get the status of the configured E.164 pattern map, use the **show dial-peer voice** command in dial peer configuration mode.

## Examples

The following example shows how to link an E.164 pattern map to a dial peer:

```
Device(config)# dial-peer voice 123 voip system

Device(config-dial-peer)# destination e164-pattern-map 2154
```

## Related Commands

| Command | Description |
|---|---|
| **destination-pattern** | Specifies either the prefix or the full E.164 telephone number to be used for a dial peer |
| **e164** | Configures an E.164 entry on a destination E.164 pattern map. |
| **show dial-peer voice** | Displays configuration information and call statistics for dial peers. |
| **url** | Specifies the URL of a text file that has E.164 pattern entries configured on a destination E.164 pattern map. |

# destination uri

To specify the voice class used to match a dial peer to the destination uniform resource identifier (URI) of an outgoing call, use the **destination uri**command in dial peer configuration mode. To remove the URI voice class, use the **no** form of this command.

**destination uri** *tag*
**no destination uri**

## Syntax Description

| | |
|---|---|
| *tag* | Alphanumeric label that uniquely identifies the voice class. This tag must be configured with the **voice class uri** command. |

## Command Default

No default behavior or values

## Command Modes

Dial peer configuration (config-dial-peer)

## Command History

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

## Usage Guidelines

Before you use this command, configure the voice class by using the **voice class uri** command.

This command applies new rules for dial-peer matching. The table below shows the rules and the order in which they are applied when the **destination uri** command is used. The gateway compares the dial-peer command to the call parameter in its search to match an outbound call to a dial peer. All dial peers are searched based on the first match criteria on . Only if no match is found does the gateway move on to the next criteria on.

*Table 1: Dial-Peer Matching Rules for Outbound URI*

| Match Order | Cisco IOS Command | Outgoing Call Parameter |
|---|---|---|
| 1 | **destination uri** and **carrier-id target** | Application-provided URI and target carrier ID associated with the call |
| 2 | **destination-pattern** and **carrier-id target** | Called number and target carrier ID associated with the call |
| 3 | **destination uri** | Application-provided URI |
| 4 | **destination-pattern** | Called number |
| 5 | **carrier-id target** | Target carrier ID associated with the call |

**Note** Calls whose destination is an E.164 number, rather than a URI, use the previously existing dial-peer matching rules. For information, see the *Dial Peer Configuration on Voice Gateway Routers* document, Cisco IOS Voice Library.

## Examples

The following example matches the destination URI in the outgoing call by using voice class ab100:

```
dial-peer voice 100 voip
 destination uri ab100
```

## Related Commands

| Command | Description |
|---|---|
| **answer-address** | Specifies the calling number to match for a dial peer. |
| **debug voice uri** | Displays the debugging messages related to URI voice classes. |
| **destination-pattern** | Specifies the telephone number to match for a dial peer. |
| **dial-peer voice** | Enters dial peer configuration mode to create or modify a dial peer. |
| **incoming uri** | Specifies the voice class that a VoIP dial peer uses to match the URI of an incoming call. |
| **pattern** | Matches a call based on the entire SIP or TEL URI. |
| **session protocol** | Specifies a session protocol for calls between local and remote routers using the packet network. |
| **show dialplan uri** | Displays which outbound dial peer is matched for a specific destination URI. |
| **voice class uri** | Creates or modifies a voice class for matching dial peers to calls containing a SIP or TEL URI. |

# destination-pattern

To specify either the prefix or the full E.164 telephone number to be used for a dial peer, use the **destination-pattern** command in dial peer configuration mode. To disable the configured prefix or telephone number, use the **no** form of this command.

**destination-pattern**[+]*string*[**T**]
**no destination-pattern**[+]*string*[**T**]

## Syntax Description

| | |
|---|---|
| + | (Optional) Character that indicates an E.164 standard number. |
| *string* | Series of digits that specify a pattern for the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and the following special characters:<br>• The asterisk (*) and pound sign (#) that appear on standard touch-tone dial pads.<br>• Comma (,), which inserts a pause between digits.<br>• Period (.), which matches any entered digit (this character is used as a wildcard).<br>• Percent sign (%), which indicates that the preceding digit occurred zero or more times; similar to the wildcard usage.<br>• Plus sign (+), which indicates that the preceding digit occurred one or more times.<br>**Note** The plus sign used as part of a digit string is different from the plus sign that can be used preceding a digit string to indicate that the string is an E.164 standard number.<br>• Circumflex (^), which indicates a match to the beginning of the string.<br>• Dollar sign ($), which matches the null string at the end of the input string.<br>• Backslash symbol (\), which is followed by a single character, and matches that character. Can be used with a single character with no other significance (matching that character).<br>• Question mark (?), which indicates that the preceding digit occurred zero or one time.<br>• Brackets ([ ]), which indicate a range. A range is a sequence of characters enclosed in the brackets; only numeric characters from 0 to 9 are allowed in the range.<br>• Parentheses (( )), which indicate a pattern and are the same as the regular expression rule. |
| **T** | (Optional) Control character that indicates that the **destination-pattern** value is a variable-length dial string. Using this control character enables the router to wait until all digits are received before routing the call. |

## Command Default

The command is enabled with a null string.

## Command Modes

Dial peer configuration (config-dial-peer)

## Command History

| Release | Modification |
|---|---|
| 11.3(1)T | This command was introduced on the Cisco 3600 series. |
| 11.3(1)MA | This command was implemented on the Cisco MC3810. |
| 12.0(4)XJ | This command was modified for store-and-forward fax. |
| 12.1(1) | The command was integrated into Cisco IOS Release 12.1(1). |
| 12.0(7)XR | This command was implemented on the Cisco AS5300 and modified to support the plus sign, percent sign, question mark, brackets, and parentheses symbols in the dial string. |
| 12.0(7)XK | This command was modified. Support for the plus sign, percent sign, question mark, brackets, and parentheses in the dial string was added to the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810. |
| 12.1(1)T | This command was integrated into Cisco IOS Release 12.1(1)T and implemented on the Cisco 1750, Cisco 7200 series, and Cisco 7500 series. The modifications for the Cisco MC3810 in Cisco IOS Release12.0(7)XK are not supported in this release. |
| 12.1(2)T | The modifications made in Cisco IOS Release 12.0(7)XK for the Cisco MC3810 were integrated into Cisco IOS Release 12.1(2)T. |
| 12.2(8)T | This command was implemented on the Cisco 1751, Cisco 2600 series and Cisco 3600 series, Cisco 3725, and Cisco 3745. |
| 12.2(13)T | This command was integrated into Cisco IOS Release 12.2(13)T and implemented on the Cisco 2600XM, the Cisco ICS7750, and the Cisco VG200. |
| 12.4(22)T | Support for IPv6 was added. |
| Cisco IOS XE Release 3.3S | This command was integrated into Cisco IOS XE Release 3.3S. |
| 15.2(4)M | This command was modified. With CSCub65380, behavior of dial peers with destination-patterns configured with + symbol was rectified. The + symbol is no longer dropped from the dial peers and matching occurs as expected |
| Cisco IOS XE Amsterdam 17.2.1r | Introduced support for YANG models. |

## Usage Guidelines

Use the **destination-pattern** command to define the E.164 telephone number for a dial peer.

The pattern you configure is used to match dialed digits to a dial peer. The dial peer is then used to complete the call. When a router receives voice data, it compares the called number (the full E.164 telephone number) in the packet header with the number configured as the destination pattern for the voice-telephony peer. The router then strips out the left-justified numbers that correspond to the destination pattern. If you have configured a prefix, the prefix is prepended to the remaining numbers, creating a dial string that the router then dials. If all numbers in the destination pattern are stripped out, the user receives a dial tone.

There are areas in the world (for example, certain European countries) where valid telephone numbers can vary in length. Use the optional control character **T**to indicate that a particular **destination-pattern** value is a variable-length dial string. In this case, the system does not match the dialed numbers until the interdigit timeout value has expired.

**Note** Cisco IOS software does not verify the validity of the E.164 telephone number; it accepts any series of digits as a valid number.

## Examples

The following example shows configuration of the E.164 telephone number 555-0179 for a dial peer:

```
dial-peer voice 10 pots
 destination-pattern +5550179
```

The following example shows configuration of a destination pattern in which the pattern "43" is repeated multiple times preceding the digits "555":

```
dial-peer voice 1 voip
 destination-pattern 555(43)+
```

The following example shows configuration of a destination pattern in which the preceding digit pattern is repeated multiple times:

```
dial-peer voice 2 voip
 destination-pattern 555%
```

The following example shows configuration of a destination pattern in which the possible numeric values are between 5550109 and 5550199:

```
dial-peer voice 3 vofr
 destination-pattern 55501[0-9]9
```

The following example shows configuration of a destination pattern in which the possible numeric values are between 5550439, 5553439, 5555439, 5557439, and 5559439:

```
dial-peer voice 4 voatm
 destination-pattern 555[03579]439
```

The following example shows configuration of a destination pattern in which the digit-by-digit matching is prevented and the entire string is received:

```
dial-peer voice 2 voip
 destination-pattern 555T
```

## 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. |
| **dial-peer terminator** | Designates a special character to be used as a terminator for variable-length dialed numbers. |
| **incoming called-number (dial peer)** | Specifies a digit string that can be matched by an incoming call to associate that call with a dial peer. |

| Command | Description |
|---|---|
| **prefix** | Specifies the prefix of the dialed digits for a dial peer. |
| **timeouts interdigit** | Configures the interdigit timeout value for a specified voice port. |

# destination-pattern (interface)

To specify the ISDN directory number for the telephone interface, use the **destination-pattern** command in interface configuration mode. To disable the specified ISDN directory number, use the **no** form of this command.

**destination-pattern** *isdn*
**no destination-pattern**

## Syntax Description

| | |
|---|---|
| *isdn* | Local ISDN directory number assigned by your telephone service provider. |

## Command Default

A default ISDN directory number is not defined for this interface.

## Command Modes

Interface configuration (config-if)

## Command History

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

## Usage Guidelines

This command is applicable to the Cisco 800 series routers.

You must specify this command when creating a dial peer. This command does not work if it is not specified within the context of a dial peer. For information on creating a dial peer, refer to the *Cisco 800 Series Routers Software Configuration Guide*.

Do not specify an area code with the local ISDN directory number.

## Examples

The following example specifies 555-0101 as the local ISDN directory number:

```
destination-pattern 5550101
```

## Related Commands

| Command | Description |
|---|---|
| **dial-peer voice** | Enters dial peer configuration mode, defines the type of dial peer, and defines the tag number associated with a dial peer. |
| **no call-waiting** | Disables call waiting. |
| **port (dial peer)** | Enables an interface on a PA-4R-DTR port adapter to operate as a concentrator port. |
| **ring** | Sets up a distinctive ring for telephones, fax machines, or modems connected to a Cisco 800 series router. |
| **show dial-peer voice** | Displays configuration information and call statistics for dial peers. |

# destination route-string

To configure a destination route string, use the **destination route-string** command in dial peer configuration mode. To remove the destination route string, use the **no** form of this command.

**destination route-string** *tag*
**no destination route-string**

### Syntax Description

| | |
|---|---|
| *tag* | The route string tag defined by the route string class. The range is from 1 to 10000. |

### Command Default

No destination route string is configured.

### Command Modes

Dial peer configuration (config-dial-peer)

### Command History

| Release | Modification |
|---|---|
| 15.3(3)M | This command was introduced. |
| Cisco IOS XE Release 3.10S | This command was integrated into Cisco IOS XE Release 3.10S. |

### Usage Guidelines

Use the **destination route-string** command to configure a voice class to match a destination route string. The destination route string defined in dial-peer voice configuration mode is used to match an outbound dial peer.

#### Example

The following example shows how to match the destination route string:

```
Device> enable
Device# configure terminal
Device(config)# dial-peer voice 100 voip
Device(config-dial-peer)# destination route-string 2
```

### Related Commands

| Command | Description |
|---|---|
| **voice class route-string** | Assigns a unique identifier tag to a route string. |

# detect v54 channel-group

To enable V.54 loopback detection for the command sent from the remote device, use the **detect v54 channel-group**command in controller configuration mode. To disable the V.54 loopback detection, use the **no** form of this command.

**detect v54 channel-group** *channel-number*
**no detect v54 channel-group** *channel-number*

## Syntax Description

| | |
|---|---|
| *channel-number* | Channel number from 1 to 24 (T1) or from 1 to 31 (E1). |

## Command Default

V.54 loopback detection is disabled.

## Command Modes

Controller configuration (config-controller)

## Command History

| Release | Modification |
|---|---|
| 12.1(1)T | This command was introduced on the Cisco 2600 series and Cisco 3600 series. |

## Usage Guidelines

Use the **detect v54 channel-group** controller configuration command to enable V.54 loopback detection. The remote device sends a loopup inband payload command sequence in fractional T1 (FT1).

## Examples

The following example sets the loopback detection for channel-group 1; then the loopback detection is disabled for channel-group 1:

```
detect v54 channel-group 1
no detect v54 channel-group 1
```

## Related Commands

| Command | Description |
|---|---|
| **loopback remote v54 channel-group** | Activates a remote V.54 loopback for the channel group on the far end. |

# detect-fax mode

To define fax detection and redirect as local or refer mode, use the **detect-fax** [**mode {refer | local}** ] *number* command in dial peer configuration mode. To disable the mode of fax detection and redirect, use the **no** form of this command.

**detect-fax** [ **mode {refer | local}** ] *number*
**no detect-fax** [ **mode {refer | local}** ] *number*

## Syntax Description

| | |
|---|---|
| *number* | The directory number of the fax machine. |

## Command Default

Fax mode detection is disabled.

## Command Modes

Dial Peer configuration (config-dial-peer)

## Command History

| Release | Modification |
|---|---|
| Cisco IOS XE Amsterdam 17.2.1r | This command was introduced for Unified Border Element. |

## Usage Guidelines

Use the **detect-fax** [**mode {refer | local}** ] *number* configuration command to enable detection of fax mode. Also, it refers to the directory number of the fax machine for redirect.

## Examples

The following is a sample configuration of local redirect mode for fax detection in Unified Border Element:

```
dial-peer voice 410 voip
 description "Incoming dial-peer to CUBE for fax"
 session protocol sipv2
 incoming called-number 903309
 codec g711ulaw
 detect-fax mode local 12101 12102

dial-peer voice 411 voip
 description "Outgoing dial-peer to VVB"
 destination-pattern 309903
 session protocol sipv2
 session target ipv4:9.42.25.148 //VVB IP Address
 codec g711ulaw

dial-peer voice 412 voip
 description "Incoming dial-peer for VVB"
 session protocol sipv2
 incoming called-number 309903
 codec g711ulaw
```

## Related Commands

| Command | Description |
|---|---|
| **fax-relay (dial peer)** | Enables the suppression of call menu (CM) tones or answer (ANS) tones from reaching the Super Group 3 (SG3) fax machines. |

# device-id

To identify a gateway associated with a settlement provider, use the **device-id** command in settlement configuration mode. To reset to the default value, use the **no** form of this command.

**device-id** *number*
**no device-id** *number*

## Syntax Description

| | |
|---|---|
| *number* | Device ID number as provided by the settlement server. Range is from 0 to 2147483647. |

## Command Default

The default device ID is 0.

## Command Modes

Settlement configuration (config-settlement)

## Command History

| Release | Modification |
|---|---|
| 12.0(4)XH1 | This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco AS5300. |
| 12.1(1)T | This command was integrated into Cisco IOS Release 12.1(1)T. |

## Usage Guidelines

Identifying a gateway associated with a settlement provider is optional.

## Examples

The following example sets the device ID to 1000:

```
settlement 0
 device-id 1000
```

## Related Commands

| Command | Description |
|---|---|
| **customer-id** | Identifies a carrier or Internet service provider with the settlement provider. |
| **settlement** | Enters settlement configuration mode. |

# dhcp interface

To configure an interface type for Dynamic Host Configuration Protocol (DHCP) provisioning of Session Initiation Protocol (SIP) parameters, use the **dhcp interface** command in SIP user-agent configuration mode.

**dhcp interface** *type number*

## Syntax Description

| | |
|---|---|
| *type* | Type of interface to be configured. |
| **number** | Port, connector, or interface card number.<br>**Note** The number format varies depending on the network module or line card type and the router's chassis slot it is installed in. The numbers are assigned at the factory at the time of installation or when they are added to a system; they can be displayed with the **show interfaces** command. |

## Command Default

No interface type is configured for DHCP provisioning of SIP parameters.

## Command Modes

SIP UA configuration (config-sip-ua)

## Command History

| Release | Modification |
|---|---|
| 12.4(22)YB | This command was introduced. |
| 15.0(1)M | This command was integrated in Cisco IOS Release 15.0(1)M. |

## Usage Guidelines

Multiple interfaces on the Cisco Unified Border Element can be configured with DHCP. The **dhcp interface** command specifies which one is the DHCP interface used with SIP.

This command does not have a **no** form.

The table below displays the keywords that represent the types of interfaces that can be configured with the **dhcp interface** command. Replace the *type* argument with the appropriate keyword from the table.

***Table 2: Interface Type Keywords***

| Keyword | Interface Type |
|---|---|
| **ethernet** | Ethernet IEEE 802.3 interface. |
| **fastethernet** | 100-Mbps Ethernet interface. In RITE configuration mode, specifies the outgoing (monitored) interface for exported IP traffic. |
| **gigabitethernet** | 1000-Mbps Ethernet interface. |
| **tengigabitethernet** | 10-Gigabit Ethernet interface. |

## Examples

The following example configures the Gigabit Ethernet interface of slot 0 port 0 as the DHCP interface for DHCP provisioning of SIP parameters:

```
Router> enable
Router# configure terminal
Router(config)# interface gigabitethernet 0/0
Router(config-if)# ip address dhcp
Router(config-if)# sip-ua
Router(sip-ua)# dhcp interface gigabitethernet 0/0
```

## Related Commands

| Command | Description |
|---|---|
| **show interfaces** | Displays information about interfaces. |
| **sip-ua** | Enters SIP user-agent configuration mode. |

# dial-control-mib

To specify attributes for the call history table, use the **dial-control-mib** command in global configuration mode. To restore the default maximum size or retention time of the call history table, use the **no** form of this command.

**dial-control-mib** {**max-size** *table-entries* | **retain-timer** *minutes*}
**no dial-control-mib** {**max-size** *table-entries* | **retain-timer** *minutes*}

## Syntax Description

| | |
|---|---|
| **max-size** *table-entries* | Maximum number of table entries in the call history table. Range is from 0 to 3000. **Note** Specifying a value of 0 prevents any further entries from being added to the table. Any existing table entries will be preserved for the duration specified with the **retain-timer** keyword. |
| **retain-timer** *minutes* | Duration, in minutes, for entries to remain in the call history table. Range is from 0 to 35791. **Note** Specifying a value of 0 prevents any further table entries from being retained, but does not affect any timer currently in effect. Therefore, any existing table entries will remain for the duration previously specified with the **retain-timer** keyword. |

## Command Default

The default call history table length is 500 table entries. The default retain timer is 15 minutes.

## Command Modes

Global configuration (config)

## Command History

| Release | Modification |
|---|---|
| 11.3(1)T | This command was introduced on the Cisco 3600 series routers. |
| 12.0(1)XA | This command was first applied to the CDR feature on the Cisco MC3810. |
| 12.0(2)T | The command was integrated into Cisco IOS Release 12.0(2)T. |
| 12.3T | The maximum value for the *table-entries* argument following the **max-size** keyword was increased to 1200 entries. |
| 12.3(8)T | The maximum value of the *minutes* argument following the **retain-timer** keyword was decreased to 35791 minutes. |

## Examples

The following example configures the call history table to hold 400 entries, with each entry remaining in the table for 10 minutes:

```
dial-control-mib max-size 400
dial-control-mib retain-timer 10
```

# dial-peer cor custom

To specify that named class of restrictions (COR) apply to dial peers, use the **dial-peer cor custom** command in global configuration mode.

**dial-peer cor custom**

## Syntax Description

This command has no arguments or keywords.

## Command Default

No default behavior or keywords.

## Command Modes

Global configuration (config)

## Command History

| Release | Modification |
|---|---|
| 12.1(3)T | This command was introduced. |
| Cisco IOS XE Bengaluru 17.6.1a | Introduced support for YANG models. |

## Usage Guidelines

You must use the **dial-peer cor custom**command and the **name** command to define the names of capabilities before you can specify COR rules and apply them to specific dial peers.

Examples of possible names might include the following: call1900, call527, call9, and call911.

**Note** You can define a maximum of 64 COR names.

## Examples

The following example defines two COR names:

```
dial-peer cor custom
 name samplegroup32
 name samplegroup12
```

The following example defines Webex Calling COR names:

```
dial-peer cor custom
 name wx-calling_Internal
 name wx-calling_Toll-fre
 name wx-calling_National
 name wx-calling_International
 name wx-calling_Operator_Assistance
 name wx-calling_chargeable_Directory_Assistance
 name wx-calling_Special_Sevices1
 name wx-calling_Special_Sevices2
 name wx-calling_Premium_Sevices1
 name wx-calling_Premium_Sevices2
```

### Related Commands

| Command | Description |
|---|---|
| **name (dial peer cor custom)** | Provides a name for a custom COR. |

# dpg

## Syntax Description

## Command Default

## Command Modes

## Command History

| Release | Modification |
| --- | --- |
| | |

## Usage Guidelines

**Note**

## Examples

## Related Commands

| Command | Description |
| --- | --- |
| | |

# dial-peer cor list

To define a class of restrictions (COR) list name, use the **dial-peer cor list** command in global configuration mode. To remove a previously defined COR list name, use the **no** form of this command.

**dial-peer cor list** *list-name*
**no dial-peer cor list** *list-name*

## Syntax Description

| | |
|---|---|
| *list-name* | List name that is applied to incoming or outgoing calls to specific numbers or exchanges. |

## Command Default

No default behavior or keywords.

## Command Modes

Global configuration (config)

## Command History

| Release | Modification |
|---|---|
| 12.1(3)T | This command was introduced. |
| Cisco IOS XE Bengaluru 17.6.1a | Introduced support for YANG models. |

## Usage Guidelines

A COR list defines a capability set that is used in the COR checking between incoming and outgoing dial peers.

## Examples

The following example adds two members to the COR list named list1:

```
dial-peer cor list list1
 member 900block
 member 800call
```

## Related Commands

| Command | Description |
|---|---|
| **dial-peer cor custom** | Specifies that named COR apply to dial peers. |
| **member (dial peer cor list)** | Adds a member to a dial peer COR list. |
| **name (dial peer cor custom)** | Provides a name for a custom COR. |

# dial-peer data

To create a data dial peer and to enter dial-peer configuration mode, use the **dial-peer data** command in global configuration mode. To remove a data dial peer, use the **no** form of this command.

**dial-peer data** *tag* **pots**
**no dial-peer data** *tag*

## Syntax Description

| | |
|---|---|
| *tag* | Specifies the dial-peer identifying number. Range is from 1 to 2147483647. |
| **pots** | Specifies an incoming POTS dial peer. |

## Command Default

No default behavior or values

## Command Modes

Global configuration (config)

## Command History

| Release | Modification |
|---|---|
| 12.2(13)T | This command was introduced. |
| 12.4(4)XC | This command was implemented on the Cisco 2600XM series, Cisco 2800 series, Cisco 3700 series, and Cisco 3800 series. |
| 12.4(9)T | This command was integrated into Cisco IOS Release 12.4(9)T. |

## Usage Guidelines

A data dial peer should be defined only for incoming data calls. The **incoming called-number** and **shutdown** commands on the data dial peer are allowed. However, the following POTS dial-peer commands are disabled on a data dial peer:

- **answer-address**
- **carrier-id**
- **destination-pattern**
- **information-type**
- **port**
- **trunk-group-label**

## Examples

The following example is a data dial-peer configuration:

```
dial-peer data 100 pots
 incoming called-number 100
```

The following example is a voice dial-peer configuration:

```
dial-peer voice 2001 pots
 destination-pattern 2001
```

```
no digit-strip
port 3/1:1
```

## Related Commands

| Command | Description |
|---|---|
| **dial-peer search** | Optimizes voice or data dial-peer searches. |
| **incoming called-number** | Specifies an incoming called number of an MMoIP or POTS dial peer. |
| **shutdown (dial peer)** | Changes the administrative state of a selected dial peer from up to down. |

# dial-peer hunt

To specify a hunt selection order for dial peers, use the **dial-peer hunt** command in global configuration mode. To restore the default selection order, use the **no** form of this command.

**dial-peer hunt** *hunt-order-number*
**no dial-peer hunt**

## Syntax Description

| | |
|---|---|
| *hunt-order-number* | A number from 0 to 7 that selects a predefined hunting selection order:<br>• 0--Longest match in phone number, explicit preference, random selection. This is the default hunt order number.<br>• 1--Longest match in phone number, explicit preference, least recent use.<br>• 2--Explicit preference, longest match in phone number, random selection.<br>• 3--Explicit preference, longest match in phone number, least recent use.<br>• 4--Least recent use, longest match in phone number, explicit preference.<br>• 5--Least recent use, explicit preference, longest match in phone number.<br>• 6--Random selection.<br>• 7--Least recent use. |

## Command Default

The default is the longest match in the phone number, explicit preference, random selection (hunt order number 0).

## Command Modes

Global configuration (config)

## Command History

| Release | Modification |
|---|---|
| 12.0(7)XK | This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco MC3810, and Cisco AS5300. |
| 12.1(2)T | This command was integrated into Cisco IOS Release 12.1(2)T. |

## Usage Guidelines

Use the **dial-peer hunt** dial peer configuration command if you have configured hunt groups. "Longest match in phone number" refers to the destination pattern that matches the greatest number of the dialed digits. "Explicit preference" refers to the **preference** command setting in the dial-peer configuration. "Least recent use" refers to the destination pattern that has waited the longest since being selected. "Random selection" weights all of the destination patterns equally in a random selection mode.

This command applies to POTS, VoIP, Voice over Frame Relay (VoFR), Voice over ATM (VoATM), and Multimedia Mail over Internet Protocol (MMOIP) dial peers.

## Examples

The following example configures the dial peers to hunt in the following order: (1) longest match in phone number, (2) explicit preference, (3) random selection.

```
dial-peer hunt 0
```

## Related Commands

| Command | Description |
| --- | --- |
| **destination-pattern** | Specifies the prefix or the complete telephone number for a dial peer. |
| **preference** | Specifies the preferred selection order of a dial peer within a hunt group. |
| **show dial-peer voice** | Displays configuration information for dial peers. |

# dial-peer inbound selection creation-order

To enable incoming dial-peer selection without changing the creation order when sorting the longest matched numbers, use **dial-peer inbound selection creation-order** command. To revert to the default behavior, use the **no** form of this command

**dial-peer inbound selection creation-order**
**no dial-peer inbound selection creation-order**

## Syntax Description

This command has no arguments or keywords.

## Command Default

The default behavior does not guarantee that the creation order would be retained for multiple dial-peers with the same number of matched digits due to the unstable heap sorting algorithm.

## Command Modes

Global configuration (config)

## Command History

| Release | Modification |
|---|---|
| Cisco IOS 15.6(1)T | This command was introduced. |

### Example

```
Device(config)# dial-peer inbound selection creation-order
```

# dial-peer inbound selection sip-trunk

To enable incoming SIP line-side calls to use the same dial-peer matching rules as SIP trunk-side calls, use the **dial-peer inbound selection sip-trunk** command in global configuration mode. To revert to the default behavior, use the **no** form of this command.

**dial-peer inbound selection sip-trunk**
**no dial-peer inbound selection sip-trunk**

## Syntax Description

This command has no arguments or keywords.

## Command Default

Disabled (SIP line-side and SIP trunk-side calls use different dial-peer matching rules).

## Command Modes

Global configuration (config)

## Command History

| Release | Modification |
|---|---|
| 12.4(11)T2 | This command was introduced. |

## Usage Guidelines

This command applies the same dial-peer matching rules used for calls from SIP trunks to incoming calls from SIP phones (line side). The first table below shows the rules and the order in which they are applied by default to SIP line-side calls. The second table below shows the rules and the order in which they are applied to SIP trunk-side calls and to SIP line-side calls when the **dial-peer inbound selection sip-trunk** command is used.

The router compares the dial-peer configuration to the call parameter in its search to match an inbound call to a dial peer. All dial peers are searched based on the first match criteria. The router moves on to the next criteria only if no match is found.

*Table 3: Dial-Peer Matching Rules for Inbound Calls from SIP Phones (Line Side)*

| Match Order | Cisco IOS Command | Incoming Call Parameter |
|---|---|---|
| 1 | **destination-pattern** | Calling number |
| 2 | **answer-address** | Calling number |
| 3 | **incoming called-number** | Called number |
| 4 | **incoming uri request** | Request-URI |
| 5 | **incoming uri to** | To URI |
| 6 | **incoming uri from** | From URI |
| 7 | **carrier-id source** | Carrier-is associated with the call |

*Table 4: Dial-Peer Matching Rules for Inbound Calls from SIP Trunks*

| Match Order | Cisco IOS Command | Incoming Call Parameter |
|---|---|---|
| 1 | **incoming uri request** | Request-URI |
| 2 | **incoming uri to** | To URI |
| 3 | **incoming uri from** | From URI |
| 4 | **incoming called-number** | Called number |
| 5 | **answer-address** | Calling number |
| 6 | **destination-pattern** | Calling number |
| 7 | **carrier-id source** | Carrier-is associated with the call |

## Examples

The following example shows SIP line-side calls use the same matching rules as trunk-side calls:

```
dial-peer inbound selection sip-trunk
```

## Related Commands

| Command | Description |
|---|---|
| **answer-address** | Specifies calling number to match for a dial peer. |
| **destination-pattern** | Specifies telephone number to match for a dial peer. |
| **dial-peer voice** | Defines a specific dial peer. |
| **incoming called-number** | Incoming called number matched to a dial peer. |
| **incoming uri** | Specifies the voice class used to match a VoIP dial peer to the uniform resource identifier (URI) of an incoming call. |
| **show dial-peer voice** | Displays configuration information for voice dial peers. |

# dial-peer no-match disconnect-cause

To disconnect the incoming ISDN or channel associated signaling (CAS) call when no inbound voice or modem dial peer is matched, use the **dial-peer no-match disconnect-cause**command in global configuration mode. To restore the default incoming call state (call is forwarded to the dialer), use the **no** form of this command.

**dial-peer no-match disconnect-cause** *cause-code-number*
**no dial-peer no-match disconnect-cause** *cause-code-number*

## Syntax Description

| | |
|---|---|
| *cause-code-number* | An ISDN cause code number. Range is from 1 to 127. |

## Command Default

The call is forwarded to the dialer to handle as a modem call.

## Command Modes

Global configuration (config)

## Command History

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

## Usage Guidelines

By default, calls are forwarded to the dialer to handle as a modem call when no inbound dial peer is matched. The **dial-peer no-match disconnect-cause**command changes that behavior to disconnect the incoming ISDN or CAS calls when no inbound voice or modem dial peer is matched.

Refer to the ISDN Cause Values table in the *Cisco IOS Debug Command Reference* for a list of ISDN cause codes.

## Examples

The following example shows that ISDN cause code 47 has been specified to match inbound voice or modem dial peers:

```
dial-peer no-match disconnect-cause 47
```

## Related Commands

| Command | Description |
|---|---|
| **show dial-peer voice** | Displays configuration information for dial peers. |

# dial-peer outbound status-check pots

To check the status of outbound POTS dial peers during call setup and to disallow, for that call, any dial peer whose status is down, use the **dial-peer outbound status-check pots**command in privileged EXEC mode. To disable status checking, use the **no** form of this command.

**dial-peer outbound status-check pots**
**no dial-peer outbound status-check pots**

## Syntax Description

This command has no arguments or keywords.

## Command Default

No default behavior or values.

## Command Modes

Privileged EXEC (#)

## Command History

| Release | Modification |
|---|---|
| 12.3 | This command was introduced. |

## Usage Guidelines

Use this command to disallow, during call setup, outbound POTS dial peers (except those for e-phones) whose endpoints (voice ports or trunk groups) are down.

When the **dial-peer outbound status-check pots**command is configured, if the voice-port configured under an outbound POTS dial-peer is down, that dial-peer is excluded while matching the corresponding destination-pattern. Therefore, if there are no other matching outbound POTS dial-peers for the specified destination-pattern, the gateway will disconnect the call with a cause code of 1 (Unallocated/unassigned number), which is mapped to the "404 Not Found" SIP response by default. When the **no** form of this command is configured, the outbound POTS dial-peer is matched even if the voice-port configured under is down and the gateway disconnects the call with a cause code of 34 (No circuit/channel available), which is mapped to the "503 Service Unavailable" SIP response by default.

**Note** "503 Service Unavailable" was the default behavior before the **dial-peer outbound status-check pots**command was introduced. Users who need the original behavior should configure the **no** form of this command.

The table below shows conditions under which an outbound POTS dial peer may be up or down.

*Table 5: Conditions Under Which an Outbound POTS Dial Peer Is Up or Down*

| If a Dial Peer's... | And If... | Then the Dial Peer Is... |
|---|---|---|
| Operational state is up | Its voice port is up | Up |
| | Its trunk groups and any associated trunks are up | |

| If a Dial Peer's... | And If... | Then the Dial Peer Is... |
|---|---|---|
| Operational state is down | -- | Down |
| Voice port is down | | |
| Trunk groups are down | All associated trunks are down | |

To show or verify the status (up or down) of all or selected dial peers, use the **show dial-peer voice** command.

## Examples

The following examples of output for the related **show dial-peer voice** command show the status of all or selected dial peers. You can use the **dial-peer outbound status-check pots**command to disallow the outbound POTS dial peers that are down.

The following example shows a short summary status for all dial peers. Outbound status is displayed in the OUT STAT field. POTS dial peers 31 and 42 are shown as down.

```
Router# show dial-peer voice summary
dial-peer hunt 0
             AD                                    PRE PASS                OUT
TAG    TYPE  MIN  OPER PREFIX    DEST-PATTERN      FER THRU SESS-TARGET    STAT PORT
444    voip  up   up                                0
22     voip  up   up                                0  syst
12     pots  up   up             5550123 0                      up   4/0:15
311    voip  up   up                                0  syst
31     pots  up   up             5550111            0                      down 4/1:15
421    voip  up   up             5550199 0  syst ipv4:1.8.56.2
42     pots  up   up                                0                      down
```

The following example shows the status for dial peer 12. Outbound status is displayed in the Outbound state field. The dial peer is shown as up.

```
Router# show dial-peer voice 12
VoiceEncapPeer12
        peer type = voice, information type = voice,
        description = `',
        tag = 12, destination-pattern = `5550123',
        answer-address = `', preference=0,
        CLID Restriction = None
        CLID Network Number = `'
        CLID Second Number sent
        source carrier-id = `', target carrier-id = `',
        source trunk-group-label = `',  target trunk-group-label = `',
        numbering Type = `unknown'
        group = 12, Admin state is up, Operation state is up,
        Outbound state is up,                                   <------- display status
        incoming called-number = `', connections/maximum = 0/unlimited,
        DTMF Relay = disabled,
        huntstop = disabled,
        in bound application associated: 'DEFAULT'
        out bound application associated: ''
        dnis-map =
        permission :both
        incoming COR list:maximum capability
        outgoing COR list:minimum requirement
        Translation profile (Incoming):
  .
  .
  .
```

The following example shows the status for dial peer 31. Outbound status is displayed in the Outbound state field. The dial peer is listed as down.

```
Router# show dial-peer voice 31
VoiceEncapPeer31
        peer type = voice, information type = voice,
        description = `',
        tag = 31, destination-pattern = `5550111',
        answer-address = `', preference=0,
        CLID Restriction = None
        CLID Network Number = `'
        CLID Second Number sent
        source carrier-id = `', target carrier-id = `',
        source trunk-group-label = `',  target trunk-group-label = `',
        numbering Type = `unknown'
        group = 31, Admin state is up, Operation state is up,
        Outbound state is down,                            <--------- display status
        incoming called-number = `', connections/maximum = 0/unlimited,
        DTMF Relay = disabled,
        huntstop = disabled,
        in bound application associated: 'DEFAULT'
        out bound application associated: ''
        dnis-map =
        permission :both
        incoming COR list:maximum capability
        outgoing COR list:minimum requirement
        Translation profile (Incoming):
  .
  .
  .
```

For descriptions of other significant fields shown in these outputs, see the **show dial-peer voice** command.

## Related Commands

| Command | Description |
|---|---|
| **show dial-peer voice** | Displays information for voice dial peers. |

# dial-peer search type

To optimize voice or data dial-peer searches, use the **dial-peer search type** command in global configuration mode. To disable the search parameters, use the **no** form of this command.

**dial-peer search type** {**data voice** | **data voice** | **none**}
**no dial-peer search type**

## Syntax Description

| | |
|---|---|
| **data** | Searches for data dial peers. |
| **none** | Searches for all dial peers by order of input. |
| **voice** | Searches for voice dial peers. |

## Command Default

**data** and **voice**

## Command Modes

Global configuration (confing)

## Command History

| Release | Modification |
|---|---|
| 12.2(13)T | This command was introduced. |
| 12.4(4)XC | This command was implemented on the Cisco 2600XM series, Cisco 2800 series, Cisco 3700 series, and Cisco 3800 series. |
| 12.4(9)T | This command was integrated into Cisco IOS Release 12.4(9)T. |

## Usage Guidelines

The search defines the search preference explicitly. If the **data** and **voice** keywords are specified, data dial peers are searched first. If no data dial peers are found, the voice dial peers are searched.

## Examples

The following is sample output that shows that data dial peers are searched first. Then voice dial peers are searched if no data dial peers can be matched for an incoming call:

```
dial-peer search type data voice
```

The following is sample output that shows that voice dial peers are searched first. Then data dial peers are searched if no voice dial peers can be matched for an incoming call:

```
dial-peer search type voice data
```

## Related Commands

| Command | Description |
|---|---|
| **dial-peer data** | Enable a gateway to process incoming data calls first by assigning the POTS dial peer as data. |

# dial-peer terminator

To change the character used as a terminator for variable-length dialed numbers, use the **dial-peer terminator** command in global configuration mode. To restore the default terminating character, use the **no** form of this command.

**dial-peer terminator** *character*
**no dial-peer terminator**

## Syntax Description

| | |
|---|---|
| *character* | Designates the terminating character for a variable-length dialed number. Valid numbers and characters are **#**, *****, **0**, **1**, **2**, **3**, **4**, **5**, **6**, **7**, **8**, **9**, **a**, **b**, **c**, and **d**. The default is **#**. |

## Command Default

The default terminating character is #.

## Command Modes

Global configuration (config)

## Command History

| Release | Modification |
|---|---|
| 12.0 | This command was introduced. |
| 12.0(7)XK | Usage was restricted to variable-length dialed numbers. The command was implemented on the Cisco 2600 series and Cisco 3600 series, and Cisco MC3810. |
| 12.1(2)T | The command was integrated into Cisco IOS Release 12.1(2)T. |

## Usage Guidelines

There are certain areas in the world (for example, in certain European countries) where telephone numbers can vary in length. When a dialed-number string has been identified as a variable length dialed number, the system does not place a call until the configured value for the **timeouts interdigit**command has expired or until the caller dials the terminating character. Use the **dial-peer terminator** global configuration command to change the terminating character.

## Examples

The following example shows that "9" has been specified as the terminating character for variable-length dialed numbers:

```
dial-peer terminator 9
```

## Related Commands

| Command | Description |
|---|---|
| **answer-address** | Specifies the full E.164 telephone number to be used to identify the dial peer of an incoming call. |
| **destination-pattern** | Specifies the prefix or the complete telephone number for a dial peer. |
| **timeouts interdigit** | Configures the interdigit timeout value for a specified voice port. |
| **show dial-peer voice** | Displays configuration information for dial peers. |

# dial-peer video

To define a video ATM dial peer for a local or remote video codec, to specify video-related encapsulation, and to enter dial peer configuration mode, use the **dial-peer video** command in global configuration mode. To remove the video dial peer, use the **no** form of this command.

**dial-peer video** *tag* {**videocodec** | **videoatm**}
**no dial-peer video** *tag* {**videocodec** | **videoatm**}

## Syntax Description

| | |
|---|---|
| *tag* | Digits that define a particular dial peer. Defines the dial peer and assigns the protocol type to the peer. Range is from 1 to 10000. The tag must be unique on the router. |
| **videocodec** | Specifies a local video codec connected to the router. |
| **videoatm** | Specifies a remote video codec on the ATM network. |

## Command Default

No video dial peer is configured.

## Command Modes

Global configuration (config)

## Command History

| Release | Modification |
|---|---|
| 12.0(5)XK | This command was introduced for ATM interface configuration on the Cisco MC3810. |
| 12.0(7)T | This command was integrated into Cisco IOS Release 12.0(7)T. |

## Usage Guidelines

The *tag* value must be unique to the device.

## Examples

The following example sets up a local video dial peer designated as 10:

```
dial-peer video 10 videocodec
```

## Related Commands

| Command | Description |
|---|---|
| **show dial-peer video** | Displays dial-peer video configuration. |

# dial-peer voice

To define a particular dial peer, to specify the method of voice encapsulation, and to enter dial peer configuration mode, use the **dial-peer voice**command in global configuration mode. To delete a defined dial peer, use the **no** form of this command.

**Cisco 1750 and Cisco 1751 Modular Access Routers**
**dial-peer voice** *tag* {**pots** | **vofr** | **voip system**}
**no dial-peer voice** *tag* {**pots** | **vofr** | **voip system**}

**Cisco 2600 Series, Cisco 2600XM, Cisco 3600 Series, Cisco 3700 Series, Cisco 7204VXR and Cisco 7206VXR**
**dial-peer voice** *tag* {**pots** | **voatm** | **vofr** | **voip system**}
**no dial-peer voice** *tag* {**pots** | **voatm** | **vofr** | **voip system**}

**Cisco 7200 Series**
**dial-peer voice** *tag* **vofr**
**no dial-peer voice** *tag* **vofr**

**Cisco AS5300**
**dial-peer voice** *tag* {**mmoip** | **pots** | **vofr** | **voip system**}
**no dial-peer voice** *tag* {**mmoip** | **pots** | **vofr** | **voip system**}

## Syntax Description

| | |
|---|---|
| *tag* | Digits that define a particular dial peer. Range is from 1 to 2147483647. |
| **pots** | Indicates that this is a POTS peer that uses VoIP encapsulation on the IP backbone. |
| **vofr** | Specifies that this is a Voice over Frame Relay (VoFR) dial peer that uses FRF.11 encapsulation on the Frame Relay backbone network. |
| **voip** | Indicates that this is a VoIP peer that uses voice encapsulation on the POTS network. |
| **system** | Indicates that this is a system that uses VoIP. |
| **voatm** | Specifies that this is a Voice over ATM (VoATM) dial peer that uses real-time ATM adaptation layer 5 (AAL5) voice encapsulation on the ATM backbone network. |
| **mmoip** | Indicates that this is a multimedia mail peer that uses IP encapsulation on the IP backbone. |

## Command Default

No dial peer is defined. No method of voice encapsulation is specified.

## Command Modes

Global configuration (config)

## Command History

| Release | Modification |
|---|---|
| 11.3(1)T | This command was introduced on the Cisco 3600 series. |
| 11.3(1)MA | This command was implemented on the Cisco MC3810, with support for the **pots**, **voatm**, **vofr**, and **vohdlc** keywords. |

| Release | Modification |
|---|---|
| 12.0(3)T | This command was implemented on the Cisco AS5300, with support for the **pots** and **voip** keywords. |
| 12.0(3)XG | The **vofr** keyword was added for the Cisco 2600 series and Cisco 3600 series. |
| 12.0(4)T | The **vofr** keyword was added for the Cisco 7200 series. |
| 12.0(4)XJ | The **mmoip** keyword was added for the Cisco AS5300. The **dial-peer voice**command was implemented for store-and-forward fax. |
| 12.0(7)XK | The **voip** keyword was added for the Cisco MC3810, and the **voatm** keyword was added for the Cisco 3600 series. Support for the **vohdlc** keyword on the Cisco MC3810 was removed. |
| 12.1(1) | The **mmoip** keyword addition in Cisco IOS Release 12.0(4)XJ was integrated into Cisco IOS Release 12.1(1). The **dial-peer voice** implementation for store-and-forward fax was integrated into Cisco IOS Release 12.1(1). |
| 12.1(2)T | The keyword changes in Cisco IOS Release 12.0(7)XK were integrated into Cisco IOS Release 12.1(2)T. |
| 12.1(5)T | This command was implemented on the Cisco AS5300 and integrated into Cisco IOS Release 12.1(5)T. |
| 12.2(4)T | This command was implemented on the Cisco 1750. |
| 12.2(2)XN | Support for enhanced Media Gateway Control Protocol (MGCP) voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series, Cisco 3600 series, and Cisco VG200. |
| 12.2(8)T | This command was implemented on the Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745. |
| 12.2(11)T | This command was integrated into Cisco IOS Release 12.2(11)T and Cisco CallManager Version 3.2. This command was implemented on the Cisco IAD2420 series. |
| 12.2(13)T | This command was integrated into Cisco IOS Release 12.2(13)T and implemented on the Cisco 2600XM, Cisco ICS7750, and Cisco VG200. |
| 12.4(22)T | Support for IPv6 was added. |
| Cisco IOS XE Amsterdam 17.2.1r | Introduced support for YANG models. |

## Usage Guidelines

Use the **dial-peer voice** global configuration command to switch to dial peer configuration mode from global configuration mode and to define a particular dial peer. Use the **exit**command to exit dial peer configuration mode and return to global configuration mode.

A newly created dial peer remains defined and active until you delete it with the **no** form of the **dial-peer voice** command. To disable a dial peer, use the **no shutdown** command in dial peer configuration mode.

In store-and-forward fax on the Cisco AS5300, the POTS dial peer defines the inbound faxing line characteristics from the sending fax device to the receiving Cisco AS5300 and the outbound line characteristics from the sending Cisco AS5300 to the receiving fax device. The Multimedia Mail over Internet Protocol (MMoIP) dial peer defines the inbound faxing line characteristics from the Cisco AS5300 to the receiving Simple Mail Transfer Protocol (SMTP) mail server. This command works with both on-ramp and off-ramp store-and-forward fax functions.

**Note** On the Cisco AS5300, MMoIP is available only if you have modem ISDN channel aggregation (MICA) technologies modems.

## Examples

The following example shows how to access dial peer configuration mode and configure a POTS peer identified as dial peer 10 and an MMoIP dial peer identified as dial peer 20:

```
dial-peer voice 10 pots
dial-peer voice 20 mmoip
```

The following example deletes the MMoIP peer identified as dial peer 20:

```
no dial-peer voice 20 mmoip
```

The following example shows how the **dial-peer voice** command is used to configure the extended echo canceller. In this instance, **pots** indicates that this is a POTS peer using VoIP encapsulation on the IP backbone, and it uses the unique numeric identifier tag 133001.

```
Router(config)# dial-peer voice 133001 pots
```

## Related Commands

| Command | Description |
|---|---|
| **codec (dial-peer)** | Specifies the voice coder rate of speech for a VoFR dial peer. |
| **destination-pattern** | Specifies the prefix, the full E.164 telephone number, or an ISDN directory number to be used for a dial peer. |
| **dtmf-relay (Voice over Frame Relay)** | Enables the generation of FRF.11 Annex A frames for a dial peer. |
| **preference** | Indicates the preferred order of a dial peer within a rotary hunt group. |
| **sequence-numbers** | Enables the generation of sequence numbers in each frame generated by the DSP for VoFR applications. |
| **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. |
| **shutdown** | Changes the administrative state of the selected dial peer from up to down. |

# dial-type

To specify the type of out-dialing for voice port interfaces, use the **dial-type** command in voice-port configuration mode. To disable the selected type of dialing, use the **no** form of this command.

**dial-type** {**dtmf** | **pulse** | **mf**}
**no dial-type**

## Syntax Description

| | |
|---|---|
| **dtmf** | Dual tone multifrequency (DTMF) touch-tone dialing. |
| **pulse** | Pulse (rotary) dialing. |
| **mf** | Multifrequency tone dialing. |

## Command Default

DTMF touch-tone dialing

## Command Modes

Voice-port configuration (config-voiceport)

## Command History

| Release | Modification |
|---|---|
| **11.3(1)T** | This command was introduced on the Cisco 3600 series. |
| **11.3(1)MA3** | This command was implemented on the Cisco MC3810, and the **pulse**keyword was added. |
| 12.0(7)XK | The **mf** keyword was added. |
| 12.1(2)T | This command was integrated into Cisco IOS Release 12.1(2)T. |
| 12.1(5)XM | This command was extended to the merged SGCP/MGCP software image. |
| 12.2(2)T | This command was implemented on the Cisco 7200 series and integrated into Cisco IOS Release 12.2(2)T. |
| 12.2(11)T | This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

Use the **dial-type** command to specify an out-dialing type for a Foreign Exchange Office (FXO) or E&M voice port interface. This command specifies the tone type for digit detection and out-pulsing. This command is not applicable to Foreign Exchange Station (FXS) voice ports because the ports do not generate out-dialing. This command also specifies the detection direction. Multifrequency tone dialing is not supported for FXS and FXO.

Voice ports can always detect DTMF and pulse signals. This command does not affect voice port dialing detection.

The **dial - type** command affects out-dialing as configured for the dial peer.

If you are using the **dial-type** command with E&M wink-start signaling, use the **dtmf** or **mf** option.

SGCP 1.1+ does not support pulse dialing.

## Examples

The following example shows a voice port configured to support a rotary (pulse tone) dialer:

```
Router(config)# voice-port 1/1
Router(config-voice-port)# dial-type pulse
```

The following example shows a voice port configured to support a DTMF (touch-tone) dialer:

```
Router(config)# voice-port 1/1
Router(config-voice-port)# dial-type dtmf
```

The following example shows a voice port configured to support a multifrequency tone dialer:

```
Router(config)# voice-port 1/1
Router(config-voice-port)# dial-type mf
```

## Related Commands

| Command | Description |
|---|---|
| **sgcp** | Starts and allocates resources for the SGCP daemon. |
| **sgcp call-agent** | Defines the IP address of the default SGCP call agent. |

# dialer extsig

To configure an interface to initiate and terminate calls using an external signaling protocol, use the **dialer extsig** command in interface configuration mode. To discontinue control of the interface by the external signaling protocol, use the **no** form of this command.

**dialer extsig**
**no dialer extsig**

## Syntax Description

This command has no arguments or keywords.

## Command Default

No default behavior or values

## Command Modes

Interface configuration (config-if)

## Command History

| Release | Modification |
|---|---|
| 12.2(2)XB | This command was introduced. |
| 12.2(11)T | The command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco AS5850. |

## Usage Guidelines

This command is used with the Network Access Server Package for Media Gateway Control Protocol feature. Configuring the **dialer in-band**command is a prerequisite to using this command. The configuration is blocked for profile dialers.

## Examples

The following example shows an interface to initiate and terminate calls using an external signaling protocol being configured:

```
Router(config)# interface Dialer1
Router(config-if)# dialer extsig
```

## Related Commands

| Command | Description |
|---|---|
| **debug dialer** | Provides debugging information for two types of dialer information: dial-on-demand events and dial-on-demand traffic. |
| **dialer in-band** | Specifies that DDR is to be supported. |
| **extsig mgcp** | Configures external signaling control by MGCP for a T1 or E1 trunk controller card. |
| **show dialer** | Displays dialer-related information for DNIS, interface, maps, and sessions. |

# dialer preemption level

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

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

## Syntax Description

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

## Command Default

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

## Command Modes

Map-class dialer configuration (config-map-class)

## 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 example sets a preemption level of *priority* (level 3) for the dialer map-class *dial1*.

```
Router(config)# map-class dialer dial1
Router(config-map-class)# dialer preemption level priority
```

## Related Commands

| Command | Description |
|---|---|
| **dialer map** | Configures a serial interface or ISDN interface to call one or multiple sites or to receive calls from multiple sites. |
| **dialer trunkgroup** | Defines the dial-on-demand trunk group label for the dialer interface. |
| **map-class dialer** | Defines a class of shared configuration parameters associated with the **dialer map** command for outgoing calls from an ISDN interface and for PPP callback. |
| **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** | Defines the expiry time for the preemption tone for the outgoing call being preempted by a DDR backup call. |

# dialer trunkgroup

To define the dial-on-demand trunk group label for the dialer interface, use the **dialer trunkgroup**command in map-class dialer configuration mode. To remove the trunk group label, use the **no** form of this command.

**dialer trunkgroup** *label*
**no dialer trunkgroup** *label*

## Syntax Description

| | |
|---|---|
| *label* | Unique name for the dialer interface trunk group. Valid names contain a maximum of 63 alphanumeric characters. |

## Command Default

No dialer trunk group is defined.

## Command Modes

Map-class dialer configuration (config-map-class)

## 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 example creates a trunk group named 20 for dialer map-class *dial1*.

```
Router(config)# map-class dialer dial1
Router(config-map-class)# dialer trunkgroup 20
```

## Related Commands

| Command | Description |
|---|---|
| **dialer map** | Configures a serial interface or ISDN interface to call one or multiple sites or to receive calls from multiple sites. |
| **map-class dialer** | Defines a class of shared configuration parameters associated with the **dialer map** command for outgoing calls from an ISDN interface and for PPP callback. |
| **show dialer** | Displays general diagnostic information for interfaces configured for dial-on-demand routing (DDR). |
| **trunk group** | Defines a trunk group (global configuration) and enters trunk group configuration mode. |

# digit

To designate the number of digits for SCCP telephony control (STC) application feature speed-dial codes, use the **digit** command in STC application feature speed-dial configuration mode. To reset to the default, use the **no** form of this command.

**digit** *number*
**no digit**

## Syntax Description

| | |
|---|---|
| *number* | Number of digits for speed-dial codes. Values are 1 or 2. Default is 1. |

## Command Default

The default number of digits is 1.

## Command Modes

STC application feature speed-dial configuration (stcapp-fsd

## Command History

| Release | Modification |
|---|---|
| 12.4(6)T | This command was introduced. |

## Usage Guidelines

This command is used with the STC application, which enables features on analog FXS endpoints that use Skinny Client Control Protocol (SCCP) for call control.

This command determines the number of digits that can be configured for speed-dial codes using the **speed dial** and **voicemail** commands. Use this command only if you want to change the number of digits from its default, which is 1. If you modify the value of this command, the **speed dial** and **voicemail** commands are reset to their defaults. If you set the value to 2 and then try to configure a single-digit speed-dial code, the system converts the speed-dial code into two digits.

Note that the phone numbers that are stored with various speed-dial codes are configured on the call-control device, such as Cisco CallManager or a Cisco CallManager Express router.

## Examples

The following example sets the number of digits for speed-dial codes to two. It also sets a speed-dial prefix of one pound sign (#) and a speed-dial code range from 5 to 25. After these values are configured, a phone user presses #10 on the keypad to dial the number that was stored with code 10.

```
Router(config)# stcapp feature speed-dial
Router(stcapp-fsd)# prefix #
Router(stcapp-fsd)# digit 2
Router(stcapp-fsd)# speed dial from 5 to 25
```

## Related Commands

| Command | Description |
|---|---|
| **prefix (stcapp-fsd)** | Designates a prefix to precede the dialing of an STC application feature speed-dial code. |
| **show stcapp feature codes** | Displays configured and default STC application feature access codes. |
| **speed dial** | Designates a range of STC application feature speed dial codes. |

| Command | Description |
|---|---|
| **voicemail** | Designates an STC application feature speed-dial code to dial the voice-mail number. |

# digit-strip

To enable digit stripping on a POTS dial-peer call leg, use the **digit - strip command in**dial peer configuration mode. To disable digit stripping on the dial-peer call leg, use the **no** form of this command.

**digit-strip**
**no digit-strip**

## Syntax Description

This command has no arguments or keywords.

## Command Default

Digit stripping is enabled.

## Command Modes

Dial peer configuration (config-dial-peer)

## Command History

| Release | Modification |
|---|---|
| 12.0(7)XR1 | This command was introduced for VoIP on the Cisco AS5300. |
| 12.0(7)XK | This command was supported for the following voice technologies on the following platforms:<br>• VoIP--(Cisco 2600 series, Cisco 3600 series, Cisco MC3810)<br>• Voice over Frame Relay (VoFR)--Cisco 2600 series, Cisco 3600 series, Cisco MC3810<br>• Voice over ATM (VoATM)--Cisco 3600 series and Cisco MC3810 |
| 12.1(1)T | This command was integrated in Cisco IOS Release 12.1(1)T. |
| 12.1(2)T | This command was integrated into Cisco IOS Release 12.1(2)T for the following voice technologies on the following platforms:<br>• VoIP (Cisco MC3810)<br>• VoFR (Cisco 2600 series, Cisco 3600 series, and Cisco MC3810)<br>• VoATM (Cisco 3600 series, Cisco MC3810) |

## Usage Guidelines

The **digit-strip** command is supported on POTS dial peers only.

When a called number is received and matched to a POTS dial peer, the matched digits are stripped and the remaining digits are forwarded to the voice interface.

The table below lists a series of dial peers configured with a specific destination pattern and shows the longest matched number after the digit is stripped based on the dial string 408 555-0148.

*Table 6: Dial-Peer Configurations with Longest Matched Number*

| Dial Peer | Destination Pattern | Preference | Session Target | Longest Matched Number |
|---|---|---|---|---|
| | 4085550148 | 0 (highest) | 100-voip | 10 |
| | 408[0-9]550148 | 0 | 200-voip | 9 |

| Dial Peer | Destination Pattern | Preference | Session Target | Longest Matched Number |
|---|---|---|---|---|
| | 408555 | 0 | 300-voip | 6 |
| | 408555 | 1(lower) | 400-voip | 6 |
| | 408% | 1 | 500-voip | 3 |
| | .......... | 0 | 600-voip | 0 |
| | .......... | 1 | 1:D (interface) | 0 |

The table below lists a series of dial peers configured with a specific destination pattern and shows the number after the digit strip based on the dial string 408 555-0148 and the different dial-peer symbols applied.

***Table 7: Dial-Peer Configurations with Digits Stripped***

| Dial Peer | Destination Pattern | Number After the Digit Strip |
|---|---|---|
| 1 | 408555.... | 0148 |
| 2 | 408555.% | 0148 |
| 3 | 408525.+ | 0148 |
| 4 | 408555.? | 0148 |
| 5 | 408555+ | 0148 |
| 6 | 408555% | 50148 |
| 7 | 408555? | 50148 |
| 8 | 408555[0-9].% | 30148 |
| 9 | 408555(30).% | 30148 |
| 10 | 408555(30)% | 30148 |
| 11 | 408555..48 | 30148 |

## Examples

The following example disables digit stripping on a POTS dial peer:

```
dial-peer voice 100 pots
 no digit-strip
```

## Related Commands

| Command | Description |
|---|---|
| **numbering-type** | Specifies number type for the VoIP or POTS dial peer. |
| **rule** | Applies a translation rule to a calling party number or a called party number for both incoming and outgoing calls. |

| Command | Description |
|---|---|
| **show translation-rule** | Displays the contents of all the rules that have been configured for a specific translation name. |
| **test translation-rule** | Tests the execution of the translation rules on a specific name-tag. |
| **translation-rule** | Creates a translation name and enters translation-rule configuration mode. |
| **voip-incoming translation-rule** | Captures calls that originate from H.323-compatible clients. |

# digital-filter

To specify the digital filter to be used before the voice packet is sent from the digital signal processor (DSP) to the network, use the **digital-filter** command in voice-class configuration mode. To remove the digital filter, use the **no** form of this command.

**digital-filter** {**1950hz** | **2175hz**}
**no digital-filter** {**1950hz** | **2175hz**}

## Syntax Description

| | |
|---|---|
| **1950hz** | Filter out 1950 Hz frequency. |
| **2175hz** | Filter out 2175 Hz frequency. |

## Command Default

Digital filtering is disabled.

## Command Modes

Voice-class configuration (config-voice-class)

## Command History

| Release | Modification |
|---|---|
| 12.3(4)XD | This command was introduced. |
| 12.3(7)T | This command was integrated into Cisco IOS Release 12.3(7)T. |

## Usage Guidelines

The **digital-filter** command has an effect on an ear and mouth (E&M) voice port only if the signal type for that port is Land Mobile Radio (LMR). The digital filter improves voice quality by preventing transmission of the guard tone with the voice packet from the LMR system to the VoIP network. The guard tone is configured with the **inject guard-tone**command. The digital filter can be configured to filter out either 2175 Hz or 1950 Hz. Only one of these frequencies can be filtered out at a time. Filtering is performed by the DSP.

## Examples

The following example specifies that 1950 Hz guard tone be filtered out of the voice packet before it is sent from the DSP to the network:

```
voice class tone-signal mytones
 digital-filter 1950hz
```

## Related Commands

| Command | Description |
|---|---|
| **inject guard-tone** | Plays out a guard tone with the voice packet. |

# direct-inward-dial

To enable the direct inward dialing (DID) call treatment for an incoming called number, use the **direct-inward-dial** command in dial peer configuration mode. To disable DID on the dial peer, use the **no** form of this command.

**direct-inward-dial**
**no direct-inward-dial**

## Syntax Description

This command has no arguments or keywords.

## Command Default

No default behavior or values.

## Command Modes

Dial peer configuration (config-dial-peer)

## Command History

| Release | Modification |
|---|---|
| 11.3(1)NA | This command was introduced. |
| 12.0(4)T | This command was modified for store-and-forward fax. |
| 12.1(5)T | This command was integrated into Cisco IOS Release 12.1(5)T. |
| 12.2(4)T | This command was implemented on the Cisco 1750. |
| 12.2(8)T | This command was implemented on the Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745. |

## Usage Guidelines

Use the **direct-inward-dial** command to enable the DID call treatment for an incoming called number. When this feature is enabled, the incoming call is treated as if the digits were received from the DID trunk. The called number is used to select the outgoing dial peer. No dial tone is presented to the caller.

Use the no form of this command to disable DID on the dial peer. When the command is disabled, the called number is used to select the outgoing dial peer. The caller is prompted for a called number via dial tone.

This command is applicable only to plain old telephone service (POTS) dial peers for on-ramp store-and-forward fax functions.

## Examples

The following example enables DID call treatment for the incoming called number:

```
dial-peer voice 10 pots
 direct-inward-dial
```