Cisco IOS Voice Commands: D

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

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

date-format (cm-fallback)

To set the date display format on all the Cisco IP phones attached to the router, use the **date-format** command in call-manager-fallback configuration mode. To display the date in default format, use the **no** form of this command.

date-format {**mm-dd-yy** | **dd-mm-yy**}

no date-format {**mm-dd-yy** | **dd-mm-yy**}

Syntax Description

mm-dd-yy	Sets to month, day, and year. Each slot needs a two-digit number. This format is the default setting.
dd-mm-yy	Sets to day, month, and year. Each slot needs a two-digit number.

Defaults

The default is **mm-dd-yy**.

Command Modes

Call-manager-fallback configuration

Command History

Release	Modification
12.2(4)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series IADs.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	This command was implemented on the Cisco 1760.

Usage Guidelines

The **date-format** command sets the date display format on all the Cisco IP phones attached to the router.

Examples

The following example sets the date format to date, month, and year for all affected Cisco IP phones:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# date-format dd-mm-yy
```

Related Commands

Command	Description
call-manager-fallback	Enables SRS Telephony feature support and enters call-manager-fallback configuration mode.

date-format (telephony-service)

To set the date display format on all the Cisco IP phones attached to a router, use the **date-format** command in telephony-service configuration mode. To display the date in the default format, use the **no** form of this command.

date-format {**mm-dd-yy** | **dd-mm-yy**}

no date-format {**mm-dd-yy** | **dd-mm-yy**}

Syntax Description

mm-dd-yy	Sets to month, day, and year. Each slot needs a two-digit number. This format is the default setting.
dd-mm-yy	Sets to day, month, and year. Each slot needs a two-digit number.

Defaults

The default is set to **mm-dd-yy**.

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

The **date-format** command sets the date display format on all Cisco IP phones attached to the router.

Examples

The following example sets the date format to date, month, and year for all affected Cisco IP phones:

```
Router(config)# telephony-service
Router(config-telephony-service)# date-format dd-mm-yy
```

Related Commands

Command	Description
telephony-service	Enables Cisco IOS Telephony Service and enters telephony-service configuration mode.

default (MGCP profile)

To configure an MGCP profile command to its default value, use the **default** command in MGCP 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: • **call-agent** • **description (MGCP profile)** • **max1 lookup** • **max1 retries** • **max2 lookup** • **max2 retries** • **package persistent** • **timeout tcrit** • **timeout tdinit** • **timeout tdmax** • **timeout tdmin** • **timeout thist** • **timeout tone busy** • **timeout tone cot1** • **timeout tone cot2** • **timeout tone dial** • **timeout tone dial stutter** • **timeout tone mwi** • **timeout tone network congestion** • **timeout tone reorder** • **timeout tone ringback** • **timeout tone ringback connection** • **timeout tone ringing** • **timeout tone ringing distinctive** • **timeout tpar** • **timeout tsmax** • **voice-port (MGCP profile)**

Defaults No default behaviors or values

Command Modes MGCP profile configuration

Command History

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

Usage Guidelines This command is used when configuring values for a Media Gateway Control Protocol (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-destination (cm-fallback)

To assign a default destination number for incoming telephone calls on the Survivable Remote Site (SRS) Telephony router, use the **default-destination** command in call-manager-fallback configuration mode. To disable the default destination number on the SRS Telephony router, use the **no** form of this command.

default-destination *telephone-number*

no default-destination *telephone-number*

Syntax Description

telephone-number	Telephone number of the default destination.

Defaults

No default behavior or values.

Command Modes

Call-manager-fallback configuration

Command History

Release	Modification
12.1(5)YD	This command was introduced on the Cisco 2600 series and Cisco 3600 series routers and Cisco IAD2420 series IADs.
12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	This command was implemented on the Cisco 1760.

Usage Guidelines

The **default-destination** command assigns default destination numbers when a call arrives at a foreign exchange office (FXO) port without called number information. A default directory number is required to route the call.

If a default destination number is set, calls arriving on an FXO port are routed to the default destination number that is provided. If a default destination number is not set, calls arriving on the FXO port receive a secondary dial tone. The caller must then enter an extension number.

Examples

The following example shows how to set the default destination as 40802:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# default-destination 40802
```

Related Commands

Command	Description
alias	Provides a mechanism for servicing calls to telephone numbers that are unavailable during CallManager fallback.
call-manager-fallback	Enables SRS Telephony feature support and enters call-manager-fallback 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 (as the default file) 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.

Defaults

No default behavior or values

Command Modes

Global configuration

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 is 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) and E&M Mercury Exchange Limited Channel-Associated Signaling (MELCAS) 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.

Defaults

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

Command Modes

Voice-port configuration

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.

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.

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
```

To configure a voice port on a Cisco MC3810 that is sending traffic in North American E&M signaling format to convert the signaling to MELCAS format, enter the following commands:

```
voice-port 0/8
 define rx-bits idle 1101
 define rx-bits seize 0101
 define tx-bits idle 1101
 define tx-bits seize 0101
```

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 disable the interface description, use the **no** form of this command.

description *string*

no description

Syntax Description

string	Character string from 1 to 80 characters.

Defaults

Enabled with a null string.
The MGCP profile has no default value.

Command Modes

Voice-port configuration
DSP farm interface configuration
MGCP profile configuration

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 dspfarm configuration mode was modified.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.
12.2(2)XA	This command was implemented on the Cisco AS5300.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T.
12.2(11)T	This command is supported on the Cisco AS5300, and Cisco AS5850 in this release.

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.

Examples

The following example identifies voice port 1/0/0 on the Cisco 3600 series routers as being connected to the purchasing department:

```
voice-port 1/0/0
 description purchasing_dept
```

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
```

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
```

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.

description (dspfarm)

To include a specific description about the digital signal processor (DSP) interface, use the **description** command in DSPfarm 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.

Defaults

Enabled with a null string.

Command Modes

DSPfarm interface configuration

Command History

Release	Modification
11.3(1)T	This command was introduced for the Cisco 7200 series routers.
12.0(5)XE	The 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 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 DSPfarm 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 (ephone-dn)

To specify a directory number description to appear in the header bar of a Cisco IP Phone 7940 or Cisco IP Phone 7960 using Cisco IOS Telephony Service (ITS), use the **description** command in ephone-dn configuration mode. To return to the default, use the **no** form of this command.

description *string*

no description

Syntax Description

string	Alphanumeric characters to be displayed in the header bar of the phone display. If spaces appear in the string, enclose it in quotation marks. The maximum string length is 40, but the string is truncated to 14 characters in the display of a Cisco IP Phone 7940 or Cisco IP Phone 7960.

Defaults

The directory number of the first line on the phone appears in the header bar.

Command Modes

Ephone-dn configuration

Command History

Release	Modification
12.2(11)T	This command was introduced.
12.2(11)YT	The number of characters in the string was modified.
12.2(15)T	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

Use this command with Cisco ITS V2.1 or a later version.

To change the top-line black header bar display, use the **description** command in the ephone-dn configuration of the first line button.

A typical use for the **description** command is to display the entire E.164 telephone number in the header bar rather than the extension number that is displayed next to the first line button, which is the default.

Examples

The following example defines a header bar display for a phone on which the first line button is the ephone-dn 51355:

```
Router(config)# ephone-dn 4
Router(config-ephone-dn)# number 51355
Router(config-ephone-dn)# description 888-555-1355
```

Related Commands

Command	Description
ephone-dn	Enters ephone directory number configuration mode and configures directory numbers for the Cisco IP phone lines.
number	Configures a valid number for a Cisco IP phone.

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 *text*

no description *text*

Syntax Description

text	Trunk group description. Maximum length is 63 alphanumeric characters.

Defaults

No default behavior or values

Command Modes

Trunk group configuration

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 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 *text*

no description *text*

Syntax Description

text	Describes a voice source group, Maximum length of the voice source group description is 63 alphanumeric characters.

Defaults

No default behavior or values

Command Modes

Voice source-group configuration

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-pattern

To specify either the prefix or the full E.164 telephone number (depending on your dial plan) 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: • The asterisk (*) and pound sign (#) that appear on standard touch-tone dial pads. • Comma (,), which inserts a pause between digits. • Period (.), which matches any entered digit (this character is used as a wildcard). • Percent sign (%), which indicates that the preceding digit occurred zero or more times; similar to the wildcard usage. • Plus sign (+), which indicates that the preceding digit occurred one or more times. **Note** The plus sign used as part of a digit string is different from the plus sign that can be used in front of a digit string to indicate that the string is an E.164 standard number. • Circumflex (^), which indicates a match to the beginning of the string. • Dollar sign ($), which matches the null string at the end of the input string. • 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). • Question mark (?), which indicates that the preceding digit occurred zero or one time. • 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. • 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.

Defaults Enabled with a null string

Command Modes Dial-peer configuration

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	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.

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 appended to the front of the remaining numbers, creating a dial string, which the router then dials. If all numbers in the destination pattern are stripped out, the user receives a dial tone.

There are certain 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 check 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 555-7922 for a dial peer:

```
dial-peer voice 10 pots
 destination-pattern +5557922
```

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 5553409 and 5559499:

```
dial-peer voice 3 vofr
 destination-pattern 555[3-9]4[0-9]9
```

The following example shows configuration of a destination pattern in which the possible numeric values are between 5551439, 5553439, 5555439, 5557439, and 5559439:

```
dial-peer voice 4 voatm
 destination-pattern 555[13579]439
```

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.
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.

Defaults

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

Command Modes

Interface configuration

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 555-1111 as the local ISDN directory number:

```
destination-pattern 5551111
```

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.

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).

Defaults

V.54 loopback detection is disabled.

Command Modes

Controller configuration

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.

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.

Defaults

The default device ID is 0

Command Modes

Settlement configuration

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

It is optional to identify a gateway associated with a settlement provider.

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.

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** *number* | **retain-timer** *number*}

no dial-control-mib {**max-size** *number* | **retain-timer** *number*}

Syntax Description

max-size *number*	Specifies the maximum size of the call history table. Range is from 0 to 500 table entries. **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 *number*	Specifies the duration, in minutes, for entries to remain in the call history table. Range is from 0 to 2147483647. **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.

Defaults

The default call history table length is 50 table entries. The default retain timer is 15 minutes.

Command Modes

Global configuration

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.

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
```

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.

Defaults

No default behavior or values

Command Modes

Interface configuration

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 output from the **dialer extsig** command:

```
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.

dial-peer cor custom

See the *Cisco IOS Dial Technologies Command Reference*, Release 12.3 for a description of the **dial-peer cor custom** command.

dial-peer cor list

See the *Cisco IOS Dial Technologies Command Reference*, Release 12.3 for a description of the **dial-peer cor list** command.

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 number. Range is from 1 to 2147483647.
pots	Specifies the incoming POTS dial peer.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

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

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 to minimize the run-time impact:

- **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: 0—Longest match in phone number, explicit preference, random selection. This is the default hunt order number. 1—Longest match in phone number, explicit preference, least recent use. 2—Explicit preference, longest match in phone number, random selection. 3—Explicit preference, longest match in phone number, least recent use. 4—Least recent use, longest match in phone number, explicit preference. 5—Least recent use, explicit preference, longest match in phone number. 6—Random selection. 7—Least recent use.

Defaults

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

Command Modes

Global configuration

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** 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, Voice over IP (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 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.

Defaults

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

Command Modes

Global configuration

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 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**}|{**voice data**}|{**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.

Defaults

data and **voice**

Command Modes

Global configuration

Command History

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

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 and then voice dial peers are searched only 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 and then data dial peers are searched only 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 #.

Defaults

The default terminating character is #

Command Modes

Global configuration

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 interdigits** 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 preferred selection order of a dial peer within a hunt group.
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.

Defaults

No video dial peer is configured

Command Modes

Global configuration

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

On a Cisco MC3810, 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 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. Alternately, to disable a dial peer, use the **no shutdown** command in dial-peer configuration mode.

Cisco 1750 and Cisco 1751 Modular Access Routers and Cisco 2600 Series

dial-peer voice *tag* {**pots** | **vofr** | **voip**}

no dial-peer voice *tag* {**pots** | **vofr** | **voip**}

Cisco 2600 Series, Cisco 2600XM, Cisco 3600 Series, Cisco 3700 Series, Cisco IAD2420 Series, and Cisco VG200

dial-peer voice *tag* {**pots** | **voatm** | **vofr** | **voip**}

no dial-peer voice *tag* {**pots** | **voatm** | **vofr** | **voip**}

Cisco 7200 Series

dial-peer voice *tag* {**vofr**}

no dial-peer voice *tag* {**vofr**}

Cisco 7204VXR and Cisco 7206VXR

dial-peer voice *tag* {**pots** | **voatm** | **vofr** | **voip**}

no dial-peer voice *tag* {**pots** | **voatm** | **vofr** | **voip**}

Cisco AS5300

dial-peer voice *tag* {**mmoip** | **pots** | **vofr** | **voip**}

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

Cisco MC3810

dial-peer voice *tag* {**pots** | **voatm** | **vofr**}

no dial-peer voice *tag* {**pots** | **voatm** | **vofr**}

Syntax Description

tag	Digits that define a particular dial peer. Range is from 1 to 2147483647.
mmoip	Indicates that this is a multimedia mail peer that uses IP encapsulation on the IP backbone.
pots	Indicates that this is a POTS peer that uses VoIP encapsulation on the IP backbone.
voatm	Specifies that this is a VoATM dial peer that uses real-time AAL5 voice encapsulation on the ATM backbone network.

vofr	Specifies that this is a 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.

Defaults

No default behavior or values

Command Modes

Global configuration

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.
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 integrated into Cisco IOS Release 12.0(4)T. The **vofr** keyword was added to 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 **vohdlc** 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 this mainline release.
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 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.

Release	Modification
12.2(11)T	This command was integrated into the 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, the Cisco ICS7750, and the Cisco VG200.

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.

After you have created a dial peer, that dial peer remains defined and active until you delete it. To delete a dial peer, use the **no** form of this 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 applies to 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 plain old telephone service (POTS) peer using Voice over IP 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.

Command	Description
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.

dialplan-pattern (cm-fallback)

To create a global prefix that can be used to expand the abbreviated extension numbers (automatically obtained from the Cisco IP phones) into fully qualified E.164 numbers, use the **dialplan-pattern** command in call-manager-fallback configuration mode. To disable a global prefix, use the **no** form of this command.

dialplan-pattern *tag pattern* **extension-length** *length* [**no-reg**]

no dialplan-pattern *tag* [*pattern* **extension-length** *length*]

Syntax Description

tag	Dial-plan string tag used before a ten-digit telephone number. Range is from 1 to 5. There is no default
pattern	Dial-plan pattern, such as the area code, the prefix, and the first one or two digits of the extension number plus wildcards.
extension-length	Number of digits in an extension number.
length	Number of extension digits. Range is from 1 to 32. There is no default.
no-reg	(Optional) Prevents the E.164 numbers in the dial-peer from registering to the gatekeeper.

Defaults

No default behavior or values.

Command Modes

Call-manager-fallback configuration

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series and Cisco IAD2420 series IADs.
12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751. The **no-reg** keyword was added.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was implemented on the Cisco 1760.

Usage Guidelines

Directory numbers or virtual voice ports for Cisco IP phones are entered in extension-number format. The extension number should be greater than or equal to the extension length. Otherwise, the extension number cannot be converted to a qualified E.164 number. The **dialplan-pattern** command creates a global prefix that can be used to expand the abbreviated extension numbers to fully qualified E.164 numbers. The dial-plan pattern is also required in order to register the Cisco IP phone lines with a gatekeeper. The **dialplan-pattern** command can resolve an incoming call into a fully qualified E.164 number.

The **extension-length** keyword enables the system to convert a full E.164 telephone number back to an extension number for the purposes of caller ID display, received-call, and missed-call lists. For example, a company uses extension number range 5000–5099 across several sites, with only the extensions from 5000 to 5009 present on the local router. An incoming call from 5066 arrives from the company's internal ISDN network and this call includes the calling number as 5553335066 in its full E.164 format.

The **no-reg** keyword provides dialing flexibility. You have the option not to register some specific numbers to the gatekeeper so that those numbers can be used for other telephony services.

When the called number matches the dial-plan pattern, the call is considered a local call and has a distinctive ring that identifies the call as internal. Any call that does not match the dial-plan pattern is considered an external call and has a ring different from the internal ringing. The valid dial-plan pattern with the lowest tag is used as a prefix to all local Cisco IP phones.

Examples

The following example shows how to create dialplan-pattern 1 for extension numbers from 5001 to 5099, following a telephone prefix that starts with 408555. If the following example is set, the router sees that 5553335066 matches dial-plan pattern 1. It uses the **extension-length** keyword to extract the last four digits of the number 5066 and to present this as the caller ID for the incoming call.

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# dialplan-pattern 1 55533350.. extension-length 4 no-reg
```

Related Commands

Command	Description
call-manager-fallback	Enables SRS Telephony feature support and enters call-manager-fallback configuration mode.

dialplan-pattern (telephony-service)

To create a global prefix that can be used to expand the abbreviated extension numbers into fully qualified E.164 numbers, use the **dialplan-pattern** command in telephony-service configuration mode. To disable a **dialplan-pattern** command setting, use the **no** form of this command.

dialplan-pattern *tag pattern* **extension-length** *length* [**extension-pattern** *extension-pattern*] [**no-reg**]

no dialplan-pattern *tag* [*pattern* **extension-length** *length* **extension-pattern** *extension-pattern*]

Syntax Description

tag	Dial-plan string tag used before a ten-digit telephone number. Range is from 1 to 5. There is no default value.
pattern	Dial-plan pattern, such as the area code, the prefix, and the first one or two digits of the extension number, plus wild card markers or dots (.) for the remainder of the extension number digits.
extension-length	Number of extension digits.
length	Number of extension digits. Range is from 1 to 32. There is no default value.
extension-pattern *extension-pattern*	(Optional) Sets the extension number leading digit pattern when the leading digits of the extension number are different than the E.164 telephone number leading digits defined by the *pattern* argument. Legal characters for this argument are one or more digits and wildcard markers or dots (.). For example, 5.. would include extensions 500 to 599 and 5... would include extensions 5000 to 5999.
no-reg	(Optional) Prevents the E.164 numbers in the dial-peer from registering with the gatekeeper.

Defaults

No default behavior or values

Command Modes

Telephony-service configuration

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series and Cisco IAD2420 series IADs.
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was implemented on the Cisco 1760.
12.2(11)YT	The **extension-pattern** keyword was added.
12.2(15)T	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

Directory numbers for the Cisco IP phones are entered in extension-number format. The extension number should be greater than or equal to the extension length. Otherwise, the extension number cannot be converted to a qualified E.164 number. The **dialplan-pattern** command creates a global prefix that can be used to expand the abbreviated extension numbers to fully qualified E.164 numbers. The dial plan pattern is also required in order to register the Cisco IP phone lines with a gatekeeper. The **dialplan-pattern** command can resolve an incoming call with a full E.164 number to a Cisco IP phone extension number.

The **extension-length** keyword enables the system to convert a full E.164 telephone number back to an extension number for the purposes of caller ID display, received-call, and missed-call lists. For example, a company uses extension number range 5000–5099 across several sites, with only the extensions from 5000 to 5009 present on the local router.An incoming call from 5044 arrives from the company's internal VoIP H.323 network and this call includes the calling number as 4085555044 in its full E.164 format.

The **no-reg** keyword provides dialing flexibility. You have the option not to register some specific numbers to the gatekeeper so that those numbers can be used for other telephony services.

When the called number matches the dial-plan pattern, the call is considered a local call and has a distinctive ring that identifies the call as internal. Any call that does not match the dial-plan pattern is considered an external call and has a ring different from the internal ringing. The valid dial-plan pattern with the lowest tag is used as a prefix to all local Cisco IP phones.

The number of *extension-pattern* characters must match the extension length that is specified in this command. For example, if the extension *length* is three, then the *extension-pattern* can be 8.., 1.., 5.., and so forth.

Examples

The following example shows how to create dialplan-pattern 1 for extension numbers 5001 to 5099 with the telephone prefix starting with 555333. If the following example is set, the routers sees that the 5553335066 matches dialplan-pattern 1, and uses the **extension-length** keyword to extract the last four digits of the number 5066 and present this as the caller ID for the incoming call.

```
Router(config)# telephony-service
Router(config-telephony-service)# dialplan-pattern 1 55533350.. extension-length 4 no-reg
```

In the following example the **dialplan-pattern** command creates dial-plan pattern 1 for extensions 801 to 899 with the telephone prefix starting with 4085559. As each number in the extension pattern is declared with the **number** command, two POTs dial peers are created. In the following example, they are 801 (an internal office number) and 408-555-9001 (an external number).

```
Router(config)# telephony-service
Router(config-telephony-service)# dialplan-pattern 1 4085559... extension-length 3
extension-pattern 8..
Router(config-telephony-service)# ephone-dn 1
Router(config-ephone-dn)# number 801
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode.
ephone-dn	Enters ephone directory number configuration mode.
telephony-service	Enables Cisco IOS Telephony Service and enters telephony-service configuration mode.

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.

Defaults

DTMF touch-tone dialing

Command Modes

Voice-port configuration

Command History

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

The **dial-type** command is not supported on FXO voice port interfaces on the Cisco MC3810. 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 on the Cisco MC3810 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 on the Cisco MC3810 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 on the Cisco MC3810 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.

digit-strip

To enable digit stripping on a plain old telephone service (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.

Defaults

Digit stripping is enabled.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(7)XR1	This command was introduced for Voice over IP (VoIP) on the Cisco AS5300.
12.0(7)XK	This command was first supported for the following voice technologies on the following platforms: • VoIP (Cisco 2600 series, Cisco 3600 series, Cisco MC3810) • Voice over Frame Relay (VoFR)—Cisco 2600 series, Cisco 3600 series, Cisco MC3810) • 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 first implemented in Cisco IOS Release 12.1(2)T for the following voice technologies on the following platforms: • VoIP (Cisco MC3810) • VoFR (Cisco 2600 series, Cisco 3600 series, and Cisco MC3810) • 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.

Table 17 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 408 555-3048.

Table 17 Dial Peer Configurations with Longest Matched Number

Dial Peer	Destination Pattern	Preference	Session Target	Longest Matched Number
1	4085553048	0 (highest)	100-voip	10
2	408[0-9]553048	0	200-voip	9
3	408555	0	300-voip	6
4	408555	1(lower)	400-voip	6
5	408%	1	500-voip	3
6	.....	0	600-voip	0
7	.....	1	1:D (interface)	0

Table 18 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 408 555-3048 and the different dial peer symbols applied.

Table 18 Dial Peer Configurations with Digits Stripped

Dial Peer	Destination Pattern	Number After the Digit Strip
1	408555....	3048
2	408555.%	3048
3	408525.+	3048
4	408555.?	3048
5	408555+	3048
6	408555%	53048
7	408555?	53048
8	408555[0-9].%	3048
9	408555(30).%	3048
10	408555(30)%	3048
11	408555..48	3048

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.
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.

Command	Description
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.

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.

Defaults

Disabled

Command Modes

Dial-peer configuration

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 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. This command applies to 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
```

directory (telephony-service)

To define the naming order for the local directory all the Cisco IP phone users, use the **directory** command in telephony-service configuration mode. To delete the naming order of the local directory, use the **no** form of this command.

directory {**first-name-first** | **last-name-first**}

no directory {**first-name-first** | **last-name-first**}

Syntax Description

first-name-first	First name is entered first in the Cisco IP phone directory name field.
last-name-first	Last name is entered first in the Cisco IP phone directory name field.

Defaults

first-name-first

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

The **directory** command defines the naming order in the local directory. The actual directory of names and phone numbers is built using the **name** command and the **number** command in ephone directory number configuration mode.

When the command is configured with the **first-name-first** keyword, you see the directory information displayed on the phone, for example, Jane E. Smith. When the command is configured with the **last-name-first** keyword, you see the directory information displayed on the phone, for example, Smith, Jane E.

Note The local directory is accessed by using the url command by specifying the directory keyword and the url string.

Examples

The following example shows how to configure the local directory with the first name first:

```
Router(config)# telephony-service
Router(config-telephony-service)# directory first-name-first
```

The following example shows how to configure the local directory with the last name first:

```
Router(config)# telephony-service
Router(config-telephony-service)# directory last-name-first
```

Related Commands

Command	Description
name	Configures a username associated with a directory number.
number	Configures a valid number for a Cisco IP phone.
telephony-service	Enables Cisco IOS Telephony Service and enters telephony-service configuration mode.
url	Provisions URLs for use by the Cisco IP phones connected to the Cisco IOS Telephony Service router.

disable-early-media 180

To specify which call treatment, early media or local ringback, is provided for 180 responses with 180 responses with Session Description Protocol (SDP), use the **disable-early-media 180** command in sip-ua configuration mode. To enable early media cut-through for 180 messages with SDP, use the **no** form of this command.

disable-early-media 180

no disable-early-media 180

Syntax Description

This command has no arguments or keywords.

Defaults

Early media cut-through for 180 responses with SDP is enabled.

Command Modes

SIP-UA configuration

Command History

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

Usage Guidelines

This command provides the ability to enable or disable early media cut-through on Cisco IOS gateways for Session Initiation Protocol (SIP) 180 responses with SDP. Use the **disable-early-media 180** command to configure the gateway to ignore the SDP message and provide local ringback. To restore the default treatment, early media cut-through, use the **no disable-early-media 180** command.

Examples

The following example disables early media cut-through for SIP 180 responses with SDP:

```
Router(config-sip-ua)# disable-early-media 180
```

Related Commands

Command	Description
show sip-ua retry	Displays SIP retry statistics.
show sip-ua statistics	Displays response, traffic, and retry SIP statistics.
show sip-ua timers	Displays the current settings for SIP-UA timers.
sip-ua	Enables the SIP-UA configuration commands.

disc_pi_off

To enable an H.323 gateway to disconnect a call when it receives a disconnect message with a progress indicator (PI) value, use the **disc_pi_off** command in voice-port configuration mode. To restore the default state, use the **no** form of this command.

disc_pi_off

no disc_pi_off

Syntax Description

This command has no arguments or keywords.

Defaults

The gateway does not disconnect a call when it receives a disconnect message with a PI value.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(5)T	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco 7500 series, Cisco AS5300, Cisco AS5800, and Cisco MC3810.
12.2(2)XA	This command was implemented on the Cisco AS5400 and Cisco AS5350.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into the Cisco IOS Release 12.2(11)T.

Usage Guidelines

The **disc_pi_off** voice-port command is valid only if the disconnect with PI is received on the inbound call leg. For example, if this command is enabled on the voice port of the originating gateway, and a disconnect message with PI is received from the terminating switch, the disconnect message is converted to a disconnect message. But if this command is enabled on the voice port of the terminating gateway, and a disconnect message with PI is received from the terminating switch, the disconnect message is not converted to a standard disconnect message because the disconnect message is received on the outbound call leg.

Note The **disc_pi_off** voice-port configuration command is valid only for the default session application; it does not work for interactive voice response (IVR) applications.

Examples

The following example handles a disconnect message with a PI value in the same way as a standard disconnect message for voice port 0:23:

```
voice-port 0:D
 disc_pi_off
```

Related Commands

Command	Description
isdn t306	Sets a timer for Disconnect messages.

disconnect-ack

To configure a Foreign Exchange Station (FXS) voice port to return an acknowledgment upon receipt of a disconnect signal, use the **disconnect-ack** command in voice-port configuration mode. To disable the acknowledgment, use the **no** form of this command.

disconnect-ack

no disconnect-ack

Syntax Description

This command has no arguments or keywords.

Defaults

FXS voice ports return an acknowledgment upon receipt of a disconnect signal

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

The **disconnect-ack** command configures an FXS voice port to remove line power if the equipment on an FXS loop-start trunk disconnects first.

Examples

The following example, which begins in global configuration mode, turns off the disconnect acknowledgment signal on voice port 1/1 on a Cisco MC3810:

```
voice-port 1/1
 no disconnect-ack
```

The following example, which begins in global configuration mode, turns off the disconnect acknowledgment signal on voice port 1/1/0 on a Cisco 2600 series or Cisco 3600 series router:

```
voice-port 1/0/0
 no disconnect-ack
```

Command History

Command	Description
show voice port	Displays voice port configuration information.

dnis (DNIS group)

To add a dialed number identification service (DNIS) number to a DNIS map, use the **dnis** command in DNIS-map configuration mode. To delete a DNIS number, use the **no** form of this command.

dnis *number* [**url** *url*]

no dnis

Syntax Description

number	Adds a user-selected DNIS number to a DNIS map.
url *url*	(Optional) URL that links a DNIS number to a specific VoiceXML document. If a URL is not entered, the DNIS number is linked to the VoiceXML application in the dial peer, which must be configured using the **application** command. This keyword is not valid for Tool Command Language (TCL) applications.

Defaults

If no URL is entered, the DNIS number links to the VoiceXML application that is configured in the dial peer with the **application** command.

Command Modes

DNIS-map configuration

Command History

Release	Modification
12.2(2)XB	This command was introduced on the Cisco AS5300, Cisco AS5350, and Cisco AS5400.
12.2(11)T	This command was implemented on the Cisco 3640 and Cisco 3660.

Usage Guidelines

To enter DNIS-map configuration mode for the **dnis** command, use the **voice dnis-map** command.

Enter the **dnis** command once for each telephone number that you want to map to a voice application. A separate entry must be made for each telephone number in a DNIS map. Wildcards are not supported.

URLs in DNIS entries are used only by VoiceXML applications. When an incoming called number matches a DNIS entry, it loads the VoiceXML document that is specified by the URL, provided that a VoiceXML application is configured in the dial peer using the **application** command.

Non-VoiceXML applications, such as TCL applications, ignore the URLs in DNIS maps and link a call to the TCL application that is configured in the dial peer using the **application** command.

For a DNIS map to be applied to an outbound dial peer, a VoiceXML application must be configured by using the **application** command with the **out-bound** keyword. Otherwise, the call is not handed off to the application that is specified in the URL of the DNIS map.

The number of allowable DNIS entries is limited by the amount of available configuration memory on the gateway. As a guideline, DNIS maps that contain more than several hundred DNIS entries should be maintained in an external text file.

To associate a DNIS map with a dial peer, use the **dnis-map** command.

Examples

The first line in the following example shows how the **voice dnis-map** command is used to create a DNIS map named "dmap1". The last two lines show how the **dnis** command is used to enter DNIS entries.

The first DNIS entry specifies the location of a VoiceXML document. The second DNIS entry does not specify a URL. A DNIS number without a URL is, by default, matched to the URL of the application that is configured in the dial peer by using the **application** command.

```
voice dnis-map dmap1
 dnis 5553305 url tftp://blue/sky/test.vxml
 dnis 5558888
```

Related Commands

Command	Description
dnis-map	Associates a DNIS map with a dial peer.
show voice dnis-map	Displays configuration information about DNIS maps.
voice dnis-map	Enters DNIS map configuration mode to create a DNIS map.
voice dnis-map load	Reloads a DNIS map that has changed since the previous load.

dnis-map

To associate a dialed number identification service (DNIS) map with a dial peer, use the **dnis-map** command in dial-peer configuration mode. To remove a DNIS map from the dial peer, use the **no** form of this command.

dnis-map *map-name*

no dnis-map

Syntax Description

map-name	Name of the configured DNIS map.

Defaults

No default behavior or values

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.2(2)XB	This command was introduced on the Cisco AS5300, Cisco AS5350, and Cisco AS5400.
12.2(11)T	This command was implemented on the Cisco 3640 and Cisco 3660.

Usage Guidelines

A DNIS map is a table of destination numbers with optional URLs that link to specific VoiceXML documents. When configured in a dial peer, a DNIS map enables you to link multiple called numbers to a single Tool Command Language (TCL) application or to individual VoiceXML documents.

The **dnis-map** command must be used with the **application** command.

Only one DNIS map can be configured in each dial peer.

To create a DNIS map, use the **voice dnis-map** command to enter DNIS-map configuration mode, and then use the **dnis** command to add entries to the DNIS map. Or you can create an external text file of DNIS entries and link to its URL by using the **voice dnis-map** command.

To view the configuration information for DNIS maps, use the **show voice dnis-map** command.

A URL configured for a DNIS number is ignored by a TCL application; the TCL script that is configured for the application is used instead.

Note For a DNIS map to be applied to an outbound dial peer, the call application must be configured as an outbound application. That is, a VoiceXML application must be configured by using the **application** command with the **out-bound** keyword. Otherwise, the call is not handed off to the application that is specified in the URL of the DNIS map.

Examples

In the following example the DNIS map named “dmap1” is associated with the VoIP dial peer 3. The outbound application “vapptest1” is associated through this dial peer with DNIS map “dmap1”.

```
dial-peer voice 3 voip
 dnis-map dmap1
 application vapptest1 outbound
```

Related Commands

Command	Description
dnis	Adds a DNIS number to a DNIS map.
show voice dnis-map	Displays configuration information about DNIS maps.
voice dnis-map	Enters DNIS map configuration mode to create a DNIS map.
voice dnis-map load	Reloads a DNIS map that has changed since the previous load.

dn-webedit (telephony-service)

To enable adding of directory numbers through a web interface, use the **dn-webedit** command in telephony-service configuration mode. To disable this feature, use the **no** form of this command.

dn-webedit

no dn-edit

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

The **dn-webedit** command enables adding of the directory through the Cisco IOS Telephony Service web-based graphical user interface (GUI). If the **dn-webedit** command is enabled, the GUI administrator can modify and assign the phone numbers associated with the Cisco IOS Telephony Service router.

If the set of numbers used by the Cisco IOS Telephony Service router are part of a larger telephone network, limitations on modification might be needed to ensure network integrity. Disabling dn-webedit prevents the administrator from allocating phone numbers and also prevents assignment of numbers that may already be used elsewhere in the network.

Examples

The following example shows how to enable editing of directory numbers through the web-based GUI interface:

```
Router(config)# telephony-service
Router(config-telephony-service)# dn-webedit
```

Related Commands

Command	Description
ephone-dn	Configures directory numbers for Cisco IP phone lines and enters ephone directory number configuration mode.
telephony-service	Enables Cisco IOS Telephony Service and enters telephony-service configuration mode.
time-webedit	Enables time setting through the web interface.

domain-name (annex G)

To set the domain name that is reported in service relationships, use the **domain-name** command in annex G neighbor configuration mode. To remove the domain name, use the **no** form of this command.

domain-name *id*

no domain-name *id*

Syntax Description

id	Domain name that is reported in service relationships.

Command Modes

Annex G neighbor configuration mode

Defaults

No default behavior or values

Command History

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

Usage Guidelines

Use this command to set the domain name reported that is reported in service relationships.

Examples

The following example shows how to set a domain name to “boston1”:

```
Router(config-annexg-neigh)# domain-name boston1
```

Related Commands

Command	Description
access-policy	Requires that a neighbor be explicitly configured.

ds0 busyout (voice)

To force a DS0 time slot on a controller into the busyout state, use the **ds0 busyout** command in controller configuration mode. To remove the DS0 time slot from the busyout state, use the **no** form of this command.

ds0 busyout *ds0-time-slot*

no ds0 busyout *ds0-time-slot*

Syntax Description

ds0-time-slot	DS0 time slots to be forced into the busyout state. Range is from 1 to 24 and can include any combination of time slots.

Defaults

DS0 time slots are not in busyout state.

Command Modes

Controller configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced on Cisco MC3810 and Cisco 2600 series and Cisco 3600 series.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

The **ds0 busyout** command affects only DS0 time slots that are configured into a DS0 group and that function as part of a digital voice port. If multiple DS0 groups are configured on a controller, any combination of DS0 time slots can be busied out, provided that each DS0 time slot to be busied out is part of a DS0 group.

If a DS0 time slot is in the busyout state, only the **no ds0 busyout** command can restore the DS0 time slot to service.

To avoid conflicting command-line interface (CLI) commands, do not use the **ds0 busyout** command and the **busyout forced** command on the same controller.

Examples

The following example configures DS0 time slot 6 on controller T1 0 to be forced into the busyout state:

```
controller t1 0
 ds0 busyout 6
```

The following example configures DS0 time slots 1, 3, 4, 5, 6, and 24 on controller E1 1 to be forced into the busyout state:

```
controller e1 1
 ds0 busyout 1,3-6,24
```

Related Commands

Command	Description
busyout seize	Changes the busyout seize procedure for a voice port.
show running configuration	Determines which DS0 time slots have been forced into the busyout state.

ds0-group (E1)

To specify the DS0 time slots that make up a logical voice port on an E1 controller, specify the signaling type by which the router communicates with the PBX or PSTN, and define E1 channels for compressed voice calls and the channel-associated signaling (CAS) method by which the router connects to the PBX or PSTN, use the **ds0-group** command in controller configuration mode. To remove the group and signaling setting, use the **no** form of this command.

Cisco 1750 and Cisco 1751

ds0-group *ds0-group-number* **timeslots** *timeslot-list* [**service** *service-type*] [**type** {**e&m-fgb** | **e&m-fgd** | **e&m-immediate-start** | **fgd-eana** | **fgd-os** | **fxs-ground-start** | **fxs-loop-start** | **none** | **r1-itu** | **r1-modified** | **r1-turkey** | **sas-ground-start** | **sas-loop-start**}]

no ds0-group *ds0-group-number*

Cisco 2600 Series, Cisco 3600, Cisco 3725, and Cisco 3745

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | **e&m-immediate-start** | **e&m-melcas-delay** | **e&m-melcas-immed** | **e&m-melcas-wink** | **e&m-wink-start** | **ext-sig** | **fgd-eana** | **fxo-ground-start** | **fxo-loop-start** | **fxo-melcas** | **fxs-ground-start** | **fxs-loop-start** | **fxs-melcas** | **r2-analog** | **r2-digital** | **r2-pulse**}

no ds0-group *ds0-group-number*

Cisco MC3810

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | **e&m-immediate-start** | **e&m-melcas-delay** | **e&m-melcas-immed** | **e&m-melcas-wink** | **e&m-wink-start** | **ext-sig** | **fgd-eana** | **fxo-ground-start** | **fxo-loop-start** | **fxo-melcas** | **fxs-ground-start** | **fxs-loop-start** | **fxs-melcas**}

no ds0-group *ds0-group-number*

Cisco 7200 Series and Cisco 7500 Series Voice Ports

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | **e&m-fgd** | **e&m-immediate-start** | **e&m-wink-start** | **fxs-ground-start** | **fxs-loop-start** | **fxo-ground-start** | **fxo-loop-start**}

no dso-group *ds0-group-number*

Cisco 7700 Series Voice Ports

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | **e&m-immediate-start** | **e&m-wink-start** | **fxs-ground-start** | **fxs-loop-start** | **fxo-ground-start** | **fxo-loop-start**}

no ds0-group *ds0-group-number*

Cisco AS5300 and the Cisco AS5400

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**none** | **p7** | **r2-analog** | **r2-digital** | **r2-lsv181-digital** | **r2-pulse**}

no ds0-group *ds0-group-number*

Cisco AS5800 Voice Ports

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-fgb** | **e&m-fgd** | **e&m-immediate-start** | **fxs-ground-start** | **fxs-loop-start** | **p7** | **r2-analog** | **r2-digital** | **r2-pulse** | **sas-ground-start** | **sas-loop-start** | **none**}

no ds0-group *ds0-group-number*

Note This command does not support the extended echo canceller (EC) feature on the Cisco AS5x00 series.

Syntax Description

ds0-group-number	A value that identifies the DS0 group. Range is from 0 to 14 and 16 to 30; 15 is reserved.
timeslots *timeslot-list*	The *timeslot-list* argument is a single time-slot number, a single range of numbers, or multiple ranges of numbers separated by commas. Range is from 1 through 31. Examples are as follows: • 2 • 1-15,17-24 • 1-23 • 2,4,6-12
type	The signaling method selection for the **type** keyword depends on the connection that you are making. The E&M interface allows connection for PBX trunk lines (tie lines) and telephone equipment. The FXS interface allows connection of basic telephone equipment and PBX. The FXO interface is for connecting the central office (CO) to a standard PBX interface where permitted by local regulations; it is often used for off-premise extensions (OPXs). Types are as follows: • **e&m-delay-dial**—The originating endpoint sends an off-hook signal and then waits for an off-hook signal followed by an on-hook signal from the destination. • **e&m-fgb**—E&M Type II Feature Group B. • **e&m-fgd**—E&M Type II Feature Group D. • **e&m-immediate-start**—E&M immediate start. • **e&m-melcas-delay**—E&M MELCAS delay-start signaling support. • **e&m-melcas-immed**—E&M MELCAS immediate-start signaling support. • **e&m-melcas-wink**—E&M MELCAS wink-start signaling support. • **e&m-wink-start**—The originating endpoint sends an off-hook signal and waits for a wink-start from the destination. • **ext-sig**—An option available only when the **mode CCS** command is enabled on the Cisco MC3810 for FRF.11 transparent CCS support. • **fgd-eana**—Feature Group D exchange access North American. • **fgd-os**—Feature Group D operator services. • **fxo-ground-start**—FXO ground-start signaling. • **fxo-loop-start**—FXO loop-start signaling. • **fxo-melcas**—FXO MELCAS signaling. • **fxs-ground-start**—FXS ground-start signaling. • **fxs-loop-start**—FXS loop-start signaling. • **fxs-melcas**—FXS MELCAS signaling. • **none**—Null signaling for external call control. • **p7**—Specifies the p7 switch type. • **r1-itu**—Line signaling based on international signaling standards.

	• **r1-modified**—An international signaling standard that is common to channelized T1/E1 networks. • **r1-turkey**—A signaling standard used in Turkey. • **r2-analog**—R2 analog line signaling. • **r2-digital**—R2 digital line signaling. • **r2-lsv181-digital**—Specifies a specific R2 digital line. • **r2-pulse**—7-pulse line signaling, a transmitted pulse that indicates a change in the line state. • **sas-ground-start**—Single attachment station (SAS) ground-start. • **sas-loop-start**—SAS loop-start.
service *service-type*	(Optional) Specifies the type of service. • **data**—data service • **fax**— store-and-forward fax service • **voice**—voice service (for FGD-OS service) • **mgcp**—MGCP mode
tone *type*	(Optional) Specifies the tone as dual tone multifrequency (DTMF) or multifrequency tones (MF).
addr info	(Optional) Specifies the calling or called party.

Defaults

There is no DS0 group. Calls are allowed in both directions.

Command Modes

Controller configuration

Command History

Release	Modification
11.2	This command was introduced for the Cisco AS5300 as the **cas-group** command.
11.3(1)MA	The command was introduced as the **voice-group** command for the Cisco MC3810.
12.0(1)T	This command was integrated into Cisco IOS Release 12.0(1)T, and the **cas-group** command was implemented on the Cisco 3600 series routers.
12.0(5)T	The command was renamed **ds0-group** on the Cisco AS5300 and Cisco 2600 series and Cisco 3600 series routers. Some keyword modifications were implemented.
12.0(5)XE	This command was implemented on the Cisco 7200 series.
12.0(7)XK	Support for this command was implemented on the Cisco MC3810. When the **ds0-group** command became available on the Cisco MC3810, the **voice-group** command was removed and no longer supported. The **ext-sig** keyword replaced the **ext-sig-master** and **ext-sig-slave** keywords that were available with the **voice-group** command.
12.0(7)XR	The **mgcp** service type was added.

Release	Modification
12.1(2)XH	The **e&m-fgd** and **fgd-eana** keywords were added for Feature Group D signaling.
12.1(3)T	This command was modified for Cisco 7500 series routers. The **fgd-os** signaling type and the **voice** service type were added.
12.1(5)XM	The **sgcp** keyword was removed.
12.2(2)XA	This command was implemented on the Cisco AS5300.
12.2(2)T	This command was integrated into Cisco IOS Release 12.2(2)T and implemented on the Cisco 7200 series.
12.2(4)T	Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 is not included in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on Cisco 1750 and Cisco 1751 routers. Support for other Cisco platforms is not included in this release.
12.2(2)XN	Support for the **mgcp** keyword 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 integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command was supported with Cisco IOS Release 12.2(11)T and Cisco CallManager Version 3.2. This command is supported on the Cisco IAD2420 series, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5850 in this release.
12.2(13)T	This command was integrated into Cisco IOS Release 12.2(13)T. The Cisco 1750 and Cisco 1751 do not support T1 and E1 voice and data cards in Cisco IOS Release 12.2(13)T. The Cisco 17xx platforms can support only HC DSP firmware images in this release.
12.2(15)T	This command was implemented on the Cisco 2600XM, Cisco 3725, and Cisco 3745.

Usage Guidelines

The **ds0-group** command automatically creates a logical voice port that is numbered as follows:

- Cisco 2600 series, Cisco 2600XM, Cisco 3660, Cisco 3725, and Cisco 3745, and Cisco 7200 series:
  - *slot*/*port*:*ds0-group-number*
- Cisco MC3810:
  - *slot*:*ds0-group-number*

Note On the Cisco MC3810, the *slot* number is the controller number.

Although only one voice port is created for each group, applicable calls are routed to any channel in the group.

Be sure you take the following into account when you are configuring DS0 groups:

- Channel groups, CAS voice groups, DS0 groups, and TDM groups all use group numbers. All group numbers configured for channel groups, CAS voice groups, DS0 groups, and TDM groups must be unique on the local router. For example, you cannot use the same group number for a channel group and for a TDM group.
- The **ds0-group** command keywords are dependent upon the Cisco IOS software release that you are using. For the most current information, go to the Cisco Feature Navigator home page at the following URL:
  http://www.cisco.com/go/fn
- When you are using command-line interface (CLI) help, the keywords for the **ds0-group** command are configuration specific. For example, if MGCP is configured, you see the **mgcp** keyword. If you are not using MGCP, you do not see the **mgcp** keyword.

Examples

The following example shows ranges of E1 controller time slots configured for FXS ground-start and FXO loop-start signaling on a Cisco 2600 series, Cisco 2600XM, Cisco 3660, Cisco 3725, or Cisco 3745:

```
E1 1/0
 framing esf
 linecode b8zs
 ds0-group 1 timeslots 1-14 type fxs-ground-start
 ds0-group 2 timeslots 16-30 type fxo-loop-start
```

The following example shows DS0 groups 1 and 2 on controller E1 1 configured on the Cisco MC3810 to support transparent CCS:

```
controller E1 1
 mode ccs cross-connect
 ds0-group 1 timeslots 1-14 type ext-sig
 ds0-group 2 timeslots 16-30 type ext-sig
```

The following example shows how to configure ranges of E1 controller time slots for FXS ground-start signaling:

```
controller E1 1/0
 ds0-group 1 timeslots 1-4 type fxs-ground-start
```

The following example illustrates setting the E1 channels for SS7 service on any trunking gateway in **mgcp** mode:

```
Router(config-controller)# ds0-group 0 timeslots 1-24 type none service mgcp
```

In the following example, the time slot maximum is 12 and the time slot is 1, so two voice-ports are created successfully.

```
controller E1 0/0
 ds0-group 0 timeslots 1-4 type e&m-immediate-start
 ds0-group 1 timeslots 6-12 type e&m-immediate-start
```

If a third DS0 group is added, the voice-port is rejected even though the total number of voice channels is less than 16.

```
 ds0-group 2 timeslots 17-18 type e&m-immediate-start
```

Related Commands

Command	Description
cas-group	Configures channelized T1 time slots with robbed bit signaling.
codec	Specifies the voice coder rate of speech for a dial peer.
codec complexity	Specifies call density and codec complexity based on the codec standard that you are using.

ds0-group (T1)

To specify the DS0 time slots that make up a logical voice port on a T1 controller, specify the signaling type by which the router communicates with the PBX or PSTN, and define T1 channels for compressed voice calls and the channel-associated signaling (CAS) method by which the router connects to the PBX or PSTN, use the **ds0-group** command in controller configuration mode. To remove the group and signaling setting, use the **no** form of this command.

Cisco 1750 and Cisco 1751

ds0-group *ds0-group-number* **timeslots** *timeslot-list* [**service** *service-type*] [**type** {**e&m-fgb** | **e&m-fgd** | **e&m-immediate-start** | **fgd-eana** | **fgd-os** | **fxs-ground-start** | **fxs-loop-start** | **none** | **r1-itu** | **r1-modified** | **r1-turkey** | **sas-ground-start** | **sas-loop-start**}]

no ds0-group *ds0-group-number*

Cisco 2600 Series, Cisco 3600, Cisco 3725, Cisco 3745, and the Cisco MC3810

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | **e&m-fgd** | **e&m-immediate-start** | **e&m-wink-start** | **ext-sig** | **fgd-eana** | **fxo-ground-start** | **fxo-loop-start** | **fxs-ground-start** | **fxs-loop-start**}

Cisco 7200 Series and Cisco 7500 Series Voice Ports

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | **e&m-fgd** | **e&m-immediate-start** | **e&m-wink-start** | **fxs-ground-start** | **fxs-loop-start** | **fxo-ground-start** | **fxo-loop-start**}

no dso-group *ds0-group-number*

Cisco 7700 Series Voice Ports

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | **e&m-immediate-start** | **e&m-wink-start** | **fxs-ground-start** | **fxs-loop-start** | **fxo-ground-start** | **fxo-loop-start**}

no ds0-group *ds0-group-number*

Cisco AS5300 and Cisco AS5400

ds0-group *ds0-group-number* **timeslots** *timeslot-list* [**service** *service-type*] [**type** {**e&m-fgb** | **e&m-fgd** | **e&m-immediate-start** | **fxs-ground-start** | **fxs-loop-start** | **fgd-eana** | **fgd-os** | **r1-itu** | **sas-ground-start** | **sas-loop-start** | **none**}] [**tone** *type*] [**addr info**]

no ds0-group *ds0-group-number*

Cisco AS5800

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-fgb** | **e&m-fgd** | **e&m-immediate-start** | **fxs-ground-start** | **fxs-loop-start** | **fgd-eana** | **r1-itu** | **r1-modified** | **r1-turkey** | **sas-ground-start** | **sas-loop-start** | **none**}

no ds0-group *ds0-group-number*

Note This command does not support the extended echo canceller (EC) feature on the Cisco AS5x00 series.

Syntax Description

ds0-group-number	A value that identifies the DS0 group. Range is from 0 to 23.
timeslots *timeslot-list*	The *timeslot-list* argument is a single time-slot number, a single range of numbers, or multiple ranges of numbers separated by commas. Range is from 1 to 24. Examples are as follows: • 2 • 1-15,17-24 • 1-23 • 2,4,6-12
type	The signaling method selection for the **type** keyword depends on the connection that you are making. The E&M interface allows connection for PBX trunk lines (tie lines) and telephone equipment. The FXS interface allows connection of basic telephone equipment and PBX. The FXO interface is for connecting the central office (CO) to a standard PBX interface where permitted by local regulations; it is often used for off-premise extensions (OPXs). Types are as follows: • **e&m-delay-dial**—The originating endpoint sends an off-hook signal and then waits for an off-hook signal followed by an on-hook signal from the destination. • **e&m-fgb**—E&M Type II Feature Group B. • **e&m-fgd**—E&M Type II Feature Group D. • **e&m-immediate-start**—E&M immediate start. • **e&m-wink-start**—The originating endpoint sends an off-hook signal and waits for a wink-start from the destination. • **ext-sig**—An option available only when the **mode CCS** command is enabled on the Cisco MC3810 for FRF.11 transparent CCS support. • **fgd-eana**—Feature Group D exchange access North American. • **fgd-os**—Feature Group D operator services. • **fxo-ground-start**—FXO ground-start signaling. • **fxo-loop-start**—FXO loop-start signaling. • **fxs-ground-start**—FXS ground-start signaling. • **fxs-loop-start**—FXS loop-start signaling. • **none**—Null signaling for external call control. • **r1-itu**—Line signaling based on international signaling standards. • **r1-modified**—An international signaling standard that is common to channelized T1/E1 networks. • **r1-turkey**—A signaling standard used in Turkey. • **sas-ground-start**—Single attachment station (SAS) ground-start. • **sas-loop-start**—SAS loop-start.

service *service-type*	(Optional) Specifies the type of service. • **data**—data service • **fax**— store-and-forward fax service • **voice**—voice service (for FGD-OS service) • **mgcp**—MGCP mode
tone *type*	(Optional) Specifies the tone as dual tone multifrequency (DTMF) or multifrequency tones (MF).
addr info	(Optional) Specifies the calling or called party.

Defaults

There is no DS0 group. Calls are allowed in both directions.

Command Modes

Controller configuration

Command History

Release	Modification
11.2	This command was introduced for the Cisco AS5300 as the **cas-group** command.
11.3(1)MA	The command was introduced as the **voice-group** command for the Cisco MC3810.
12.0(1)T	This command was integrated into Cisco IOS Release 12.0(1)T, and the **cas-group** command was implemented on the Cisco 3600 series routers.
12.0(5)T	The command was renamed **ds0-group** on the Cisco AS5300 and Cisco 2600 series and Cisco 3600 series routers. Some keyword modifications were implemented.
12.0(5)XE	This command was implemented on the Cisco 7200 series.
12.0(7)XK	Support for this command was implemented on the Cisco MC3810. When the **ds0-group** command became available on the Cisco MC3810, the **voice-group** command was removed and no longer supported. The **ext-sig** keyword replaced the **ext-sig-master** and **ext-sig-slave** keywords that were available with the **voice-group** command.
12.0(7)XR	The **mgcp** service type was added.
12.1(2)XH	The **e&m-fgd** and **fgd-eana** keywords were added for Feature Group D signaling.
12.1(3)T	This command was modified for Cisco 7500 series routers. The **fgd-os** signaling type and the **voice** service type were added.
12.1(5)XM	The **sgcp** keyword was removed.
12.2(2)XA	This command was implemented on the Cisco AS5300.
12.2(2)T	This command was integrated into Cisco IOS Release 12.2(2)T and implemented on the Cisco 7200 series.
12.2(4)T	Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 is not included in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.

Release	Modification
12.2(4)XM	This command was implemented on Cisco 1750 and Cisco 1751 routers. Support for other Cisco platforms is not included in this release.
12.2(2)XN	Support for the **mgcp** keyword 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 integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command was supported with Cisco IOS Release 12.2(11)T and Cisco CallManager Version 3.2. This command is supported on the Cisco IAD2420 series, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5850 in this release.
12.2(13)T	This command was integrated into Cisco IOS Release 12.2(13)T. The Cisco 1750 and Cisco 1751 do not support T1 and E1 voice and data cards in Cisco IOS Release 12.2(13)T. The Cisco 17xx platforms can support only HC DSP firmware images in this release.
12.2(15)T	This command was implemented on the Cisco 2600XM, Cisco 3725, and Cisco 3745.

Usage Guidelines

The **ds0-group** command automatically creates a logical voice port that is numbered as follows:

- Cisco 2600 series, Cisco 2600XM, Cisco 3660, Cisco 3725, and Cisco 3745, and Cisco 7200 series:
  - *slot*/*port*:*ds0-group-number*
- Cisco MC3810:
  - *slot*:*ds0-group-number*

Note On the Cisco MC3810, the *slot* number is the controller number.

- Cisco AS5300:
  - *slot*/*port*

Although only one voice port is created for each group, applicable calls are routed to any channel in the group.

Be sure you take the following into account when you are configuring DS0 groups:

- Channel groups, CAS voice groups, DS0 groups, and TDM groups all use group numbers. All group numbers configured for channel groups, CAS voice groups, DS0 groups, and TDM groups must be unique on the local router. For example, you cannot use the same group number for a channel group and for a TDM group.
- The **ds0-group** command keywords are dependent upon the Cisco IOS software release that you are using. For the most current information, go to the Cisco Feature Navigator home page at the following URL:
  http://www.cisco.com/go/fn
- When you are using command-line interface (CLI) help, the keywords for the **ds0-group** command are configuration specific. For example, if MGCP is configured, you see the **mgcp** keyword. If you are not using MGCP, you do not see the **mgcp** keyword.

Examples

The following example shows ranges of T1 controller time slots configured for FXS ground-start and FXO loop-start signaling on a Cisco 2600 series, Cisco 2600XM, Cisco 3660, Cisco 3725, or Cisco 3745:

```
T1 1/0
 framing esf
 linecode b8zs
 ds0-group 1 timeslots 1-10 type fxs-ground-start
 ds0-group 2 timeslots 11-24 type fxo-loop-start
```

The following example shows DS0 groups 1 and 2 on controller T1 1 configured on the Cisco MC3810 to support transparent CCS:

```
controller T1 1
 mode ccs cross-connect
 ds0-group 1 timeslots 1-10 type ext-sig
 ds0-group 2 timeslots 11-24 type ext-sig
```

The following example shows how to configure ranges of T1 controller time slots for FXS ground-start signaling:

```
controller T1 1/0
 ds0-group 1 timeslots 1-4 type fxs-ground-start
```

The following example illustrates setting the T1 channels for SS7 service on any trunking gateway in **mgcp** mode:

```
Router(config-controller)# ds0-group 0 timeslots 1-24 type none service mgcp
```

In the following example, the time slot maximum is 12 and the time slot is 1, so two voice-ports are created successfully.

```
controller T1 0/0
 ds0-group 0 timeslots 1-4 type e&m-immediate-start
 ds0-group 1 timeslots 6-12 type e&m-immediate-start
```

If a third DS0 group is added, the voice-port is rejected even though the total number of voice channels is less than 16.

```
 ds0-group 2 timeslots 17-18 type e&m-immediate-start
```

Related Commands

Command	Description
cas-group	Configures channelized T1 time slots with robbed bit signaling.
codec	Specifies the voice coder rate of speech for a dial peer.
codec complexity	Specifies call density and codec complexity based on the codec standard that you are using.

dsn

To specify that a delivery status notice (DSN) be delivered to the sender, use the **dsn** command in dial-peer configuration mode. To cancel a specific DSN option, use the **no** form of this command.

dsn {**delay** | **failure** | **success**}

no dsn {**delay** | **failure** | **success**}

Syntax Description

delay	Defines the delay for each mailer.
failure	Requests that a failed message be sent to the FROM address. This is a default.
success	Requests that message be sent to the FROM address that the message was delivered successfully to the recipient.

Note In the absence of any other DSN settings (for example, **no dsn**, or a mailer in the path that does not support the DSN extension), a failure to deliver message always causes a nondelivery message to be generated. This nondelivery message is called a *bounce*.

Defaults

The default is to send a nondelivery message in the event of a failure

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.
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 and Cisco 3600 series, Cisco 3725, and Cisco 3745.

Usage Guidelines

When the **delay** keyword is selected, the next-hop mailer sends a message to the FROM address saying that the mail message was delayed. The definition of the **delay** keyword is made by each mailer and is not controlled by the sender. Each mailer in the path to the recipient that supports the DSN extension receives the same request.

When the **failur**e keyword is selected, the next-hop mailer sends a message to the FROM address that the mail message delivery failed. Each mailer in the path to the recipient that supports the DSN extension receives the same request.

When the **success** keyword is selected, the next-hop mailer sends a message to the FROM address saying that the mail message was successfully delivered to the recipient. Each mailer in the path to the recipient that supports the DSN extension receives the same request.

This command is applicable to Multimedia Mail over Internet Protocol (MMoIP) dial peers.

DSNs are messages or responses that are automatically generated and sent to the sender or originator of an e-mail message by the Simple Mail Transfer Protocol (SMTP) server, notifying the sender of the status of the e-mail message. Specifications for DSN are described in RFC 1891, RFC 1892, RFC 1893, and RFC 1894.

The on-ramp DSN request is included as part of the fax-mail message sent by the on-ramp gateway when the matching MMoIP dial peer has been configured. The on-ramp DSN response is generated by the SMTP server when the fax-mail message is accepted. The DSN is sent back to the user defined by the **mta send mail-from** command. The off-ramp DSN is requested by the e-mail client. The DSN response is generated by the SMTP server when it receives a request as part of the fax-mail message.

Note DSNs are generated only if the mail client on the SMTP server is capable of responding to a DSN request.

Because the SMTP server generates the DSNs, you need to configure both **mail from:** and **rcpt to:** on the server for the DSN feature to work. For example:

```
mail from: <user@mail-server.company.com>
rcpt to: <fax=555-1212@company.com> NOTIFY=SUCCESS,FAILURE,DELAY
```

There are three different states that can be reported back to the sender:

- Delay—Indicates that the message was delayed in being delivered to the recipient or mailbox.
- Success—Indicates that the message was successfully delivered to the recipient or mailbox.
- Failure—Indicates that the SMTP server was unable to deliver the message to the recipient or mailbox.

Because these delivery states are not mutually exclusive, you can configure store-and-forward fax to generate these messages for all or any combination of these events.

DSN messages notify the sender of the status of a particular e-mail message that contains a fax TIFF image. Use the **dsn** command to specify which notification messages are sent to the user.

The **dsn** command allows you to select more than one notification option by reissuing the command and specifying a different notification option each time. To discontinue a specific notification option, use the **no** form of the command for that specific keyword.

If the **failure** keyword is not included when DSN is configured, the sender receives no notification of message delivery failure. Because a failure is usually significant, care should be taken to always include the **failure** keyword as part of the **dsn** command configuration.

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

Examples

The following example specifies that a DSN message be returned to the sender when the e-mail message that contains the fax has been successfully delivered to the recipient or if the message that contains the fax has failed to be delivered:

```
dial-peer voice 10 mmoip
 dsn success
 dsn failure
```

Related Commands

Command	Description
mta send mail-from hostname	Specifies the originator (host-name portion) of the e-mail fax message.
mta send mail-from username	Specifies the originator (username portion) of the e-mail fax message.

dsp services dspfarm

To configure digital-signal-processor (DSP) farm services for a particular digital T1/E1 packet voice trunk network module (NM-HDV) or high-density voice (HDV) transcoding/conferencing DSP farm (NM-HDV-FARM), use the **dsp services dspfarm** command in interface configuration mode. To disable services, use the **no** form of this command.

dsp services dspfarm

no dsp services dspfarm

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values

Command Modes

Voice-card configuration

Command History

Release	Modification
12.2(13)T	This command was introduced on the Cisco VG200, Cisco 2600 series, Cisco 3620, Cisco 3640, Cisco 3660, and Cisco 3700 series.

Usage Guidelines

The router on which this command is used must be equipped with one or more NM-HDVs or NM-HDV-FARMs to provide DSP resources.

The number of NM-HDVs or NM-HDV-FARMs that need to be enabled for DSP-farm services depends on the number of DSPs on the NM-HDV or NM-HDV-FARM and on the maximum number of transcoding and conferencing sessions configured on the DSP farm. DSP resources on an NM-HDV or NM-HDV-FARM are used only if this command is configured under that particular card.

Use this command before enabling DSP-farm services with the **dspfarm** command.

Examples

The following example configures an NM-HDV or NM-HDV-FARM, specifies the maximum number of transcoding sessions, and enables DSP-farm services:

```
Router# configure terminal
Router(config)# no dspfarm
Router(config)# voice-card 2
Router(config-voicecard)# dsp services dspfarm
Router(config-voicecard)# exit
Router(config)# dspfarm transcoder maximum sessions 15
Router(config)# dspfarm
```

Related Commands

Command	Description
dspfarm (DSP farm)	Enables DSP-farm service.
dspfarm transcoder maximum sessions	Specifies the maximum number of transcoding sessions to be supported by a DSP farm.
show dspfarm	Displays summary information about DSP resources.

dspfarm (DSP farm)

To enable digital-signal-processor (DSP) farm service, use the **dspfarm** command in global configuration mode. To disable the service, use the **no** form of this command.

dspfarm

no dspfarm

Syntax Description

This command has no arguments or keywords.

Defaults

DSP-farm service is disabled.

Command Modes

Global configuration

Command History

Release	Modification
12.1(5)YH	This command was introduced on the Cisco VG200.
12.2(13)T	This command was implemented on the Cisco 2600 series, Cisco 3620, Cisco 3640, Cisco 3660, and Cisco 3700 series.

Usage Guidelines

The router on which this command is used must be equipped with one or more digital T1/E1 packet voice trunk network modules (NM-HDVs) or high-density voice (HDV) transcoding/conferencing DSP farms (NM-HDV-FARMs) to provide DSP resources.

Before enabling DSP-farm services, you must configure the NM-HDV or NM-HDV-FARM on which DSP-farm services are to be enabled using the **dsp service dspfarm** command. You must also specify the maximum number of transcoding sessions to be supported by the DSP farm using the **dspfarm transcoder maximum sessions** command.

This command causes the system to download new firmware into the DSPs, start up the required subsystems, and wait for a service request from the transcoding and conferencing applications.

Examples

The following example configures an NM-HDV or NM-HDV-FARM, specifies the maximum number of transcoding sessions, and enables DSP-farm services:

```
Router# configure terminal
Router(config)# no dspfarm
Router(config)# voice-card 2
Router(config-voicecard)# dsp services dspfarm
Router(config-voicecard)# exit
Router(config)# dspfarm transcoder maximum sessions 15
Router(config)# dspfarm
```

Related Commands

Command	Description
dsp services dspfarm	Specifies the NM-HDV or NM-HDV-FARM on which DSP-farm services are to be enabled.
dspfarm transcoder maximum sessions	Specifies the maximum number of transcoding sessions to be supported by a DSP farm.
show dspfarm	Displays summary information about DSP resources.

dspfarm (voice-card)

To add a specified voice card to those participating in a digital signal processor (DSP) resource pool, use the **dspfarm** command in voice-card configuration mode. To remove the specified card from participation in the DSP resource pool, use the **no** form of this command.

dspfarm

no dspfarm

Syntax Description

This command has no arguments or keywords.

Defaults

A card participates in the DSP resource pool

Command Modes

Voice-card configuration

Command History

Release	Modification
12.1(5)XM	This command was introduced for the Cisco 3660.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T.
12.2(2)XB	This command was implemented on the Cisco 2600 series routers.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(15)T	This command was implemented on the Cisco 2600XM, Cisco 3725, and Cisco 3745.

Usage Guidelines

DSP mapping occurs when DSP resources on one AIM or network module are available for processing of voice time-division multiplexing (TDM) streams on a different network module or on a voice/WAN interface card (VWIC). This command is used on Cisco 3660 routers with multiservice interchange (MIX) modules installed or on Cisco 2600 series routers with AIMs installed.

To reach voice-card configuration mode for a particular voice card, from global configuration mode enter the **voice-card** command and the slot number for the AIM or network module that you want to add to the pool. See the **voice-card** command reference for details on slot numbering.

The assignment of DSP pool resources to particular TDM streams is based on the order in which the streams are configured with the **ds0-group** command for T1/E1 channel-associated signaling (CAS) or with the **pri-group** command for ISDN PRI.

The assignment of DSP pool resources does not occur dynamically during call signaling.

Examples

The following example adds to the DSP resource map the DSP resources on the network module in slot 5 on a Cisco 3660 with a MIX module:

```
voice-card 5
 dspfarm
```

The following example makes available the DSP resources on an AIM on a Cisco 2600 series router:

```
voice-card 0
 dspfarm
```

Related Commands

Command	Description
ds0-group	Specifies the DS0 time slots that make up a logical voice port on a T1 or E1 controller, to specify the signaling type by which the router communicates with the PBX or PSTN, and to define T1or E1 channels for compressed voice calls and the channel-associated signaling (CAS) method by which the router connects to the PBX or PSTN.
pri-group	Specifies ISDN Primary Rate Interface (PRI) on a channelized T1 or E1 controller.
voice-card	Enters voice-card configuration mode.

dspfarm confbridge maximum sessions

To specify the maximum number of concurrent conference sessions for which digital-signal-processor (DSP) farm resources should be allocated, use the **dspfarm confbridge maximum sessions** command in global configuration mode. To reset to the default, use the **no** form of this command.

dspfarm confbridge maximum sessions *number*

no dspfarm confbridge maximum sessions

Syntax Description

number	Number of conference sessions. A single DSP supports 1 conference session with up to 6 participants.

Defaults

0 sessions

Command Modes

Global configuration

Command History

Release	Modification
12.1(5)YH	This command was introduced on the Cisco VG200.
12.2(13)T	This command was implemented on the Cisco 2600 series, Cisco 3620, Cisco 3640, Cisco 3660, and Cisco 3700 series.

Usage Guidelines

The router on which this command is used must be equipped with one or more digital T1/E1 packet voice trunk network modules (NM-HDVs) or high-density voice (HDV) transcoding/conferencing DSP farms (NM-HDV-FARMs) to provide DSP resources.

Before using this command, you must disable DSP-farm service using the **no dspfarm** command.

The maximum number of conference sessions depends upon DSP availability in the DSP farm. A single DSP supports one conference session with up to six participants. However, you may need to allocate additional DSP resources for transcoding to support conferences. If all participants use G.711 or G.729 codecs, you need not allocate any additional DSP resources because transcoding is done in the conferencing DSP.

When you use this command, take into consideration the number of DSPs allocated for transcoding services with the **dspfarm transcoder maximum sessions** command.

Examples

The following example sets the maximum number of conferencing sessions to 8:

```
Router# dspfarm confbridge maximum sessions 8
```

Related CommandsT

Command	Description
dspfarm (DSP farm)	Enables DSP-farm service.
dspfarm transcoder maximum sessions	Specifies the maximum number of transcoding sessions to be supported by a DSP farm.
show dspfarm	Displays summary information about DSP resources.

dspfarm connection interval

To specify the time interval during which to monitor Real-Time Transport Protocol (RTP) inactivity before deleting an RTP stream, use the **dspfarm connection interval** command in global configuration mode. To reset to the default, use the **no** form of this command.

dspfarm connection interval *seconds*

no dspfarm connection interval

Syntax Description

seconds	Interval, in seconds, during which to monitor RTP inactivity. Range is from 60 to 10800. Default is 600.

Defaults

600 seconds

Command Modes

Global configuration

Command History

Release	Modification
12.1(5)YH	This command was introduced on the Cisco VG200.
12.2(13)T	This command was implemented on the Cisco 2600 series, Cisco 3620, Cisco 3640, Cisco 3660, and Cisco 3700 series.

Usage Guidelines

The router on which this command is used must be equipped with one or more digital T1/E1 packet voice trunk network modules (NM-HDVs) or high-density voice (HDV) transcoding/conferencing DSP farms (NM-HDV-FARMs) to provide digital-signal-processor (DSP) resources.

After each interval, RTP streams are checked for inactivity. If all RTP streams for a particular call are inactive, the RTP timer, as set with the **dspfarm rtp timeout** command, is started. When the RTP timer expires, the call is deleted.

Examples

The following example sets the connection interval to 60 seconds:

```
Router(config)# dspfarm connection interval 60
```

Related Commands

dspfarm rtp timeout	Specifies the RTP timeout interval used to clear hanging connections.

dspfarm rtp timeout

To specify the Real-Time Transport Protocol (RTP) timeout interval used to clear hanging connections, use the **dspfarm rtp timeout** command in global configuration mode. To reset to the default, use the **no** form of this command.

dspfarm rtp timeout *seconds*

no dspfarm rtp timeout

Syntax Description

seconds	RTP timeout interval, in seconds. Range is from 10 to 7200. Default is 1200.

Defaults

1200 seconds (20 minutes)

Command Modes

Global configuration

Command History

Release	Modification
12.1(5)YH	This command was introduced on the Cisco VG200.
12.2(13)T	This command was implemented on the Cisco 2600 series, Cisco 3620, Cisco 3640, Cisco 3660, and Cisco 3700 series.

Usage Guidelines

The router on which this command is used must be equipped with one or more digital T1/E1 packet voice trunk network modules (NM-HDVs) or high-density voice (HDV) transcoding/conferencing DSP farms (NM-HDV-FARMs) to provide digital-signal-processor (DSP) resources.

Use this command to set the RTP timeout interval for when the error condition “RTP port unreachable” occurs.

Examples

The following example sets the RTP timeout value to 600 seconds (10 minutes):

```
Router# dspfarm rtp timeout 600
```

Related Commands

Command	Description
dspfarm (DSP farm)	Enables DSP-farm service.
dspfarm connection interval	Specifies the time interval during which to monitor RTP inactivity before deleting an RTP stream.
show dspfarm	Displays summary information about DSP resources.

dspfarm transcoder maximum sessions

To specify the maximum number of transcoding sessions to be supported by the digital-signal-processor (DSP) farm, use the **dspfarm transcoder maximum sessions** command in global configuration mode. To reset to the default, use the **no** form of this command.

dspfarm transcoder maximum sessions *number*

no dspfarm transcoder maximum sessions

Syntax Description

number	Number of transcoding sessions.

Defaults

0 sessions

Command Modes

Global configuration

Command History

Release	Modification
12.1(5)YH	This command was introduced on the Cisco VG200.
12.2(13)T	This command was implemented on the Cisco 2600 series, Cisco 3620, Cisco 3640, Cisco 3660, and Cisco 3700 series.

Usage Guidelines

The router on which this command is used must be equipped with one or more digital T1/E1 packet voice trunk network modules (NM-HDVs) or high-density voice (HDV) transcoding/conferencing DSP farms (NM-HDV-FARMs) to provide DSP resources.

Before using this command, you must disable DSP-farm service using the **no dspfarm** command.

Use this command in conjunction with the **dspfarm confbridge maximum sessions** commands.

The maximum number of transcoding sessions depends upon DSP availability in the DSP farm. A single DSP supports four transcoding sessions transmission to and from G.711 and G.729 codecs.

Examples

The following example configures an NM-HDV or NM-HDV-FARM, specifies the maximum number of transcoding sessions, and enables DSP-farm services:

```
Router# configure terminal
Router(config)# no dspfarm
Router(config)# voice-card 2
Router(config-voicecard)# dsp services dspfarm
Router(config-voicecard)# exit
Router(config)# dspfarm transcoder maximum sessions 15
Router(config)# dspfarm
```

Related Commands

Command	Description
dspfarm (DSP farm)	Enables DSP-farm service.
dspfarm confbridge maximum sessions	Specifies the maximum number of conferencing sessions to be supported by a DSP farm.
dsp services dspfarm	Specifies the NM-HDV or NM-HDV-FARM on which DSP-farm services are to be enabled.
show dspfarm	Displays summary information about DSP resources.

dspint dspfarm

To enable the digital signal processor (DSP) interface, use the **dspint dspfarm** command in global configuration mode. This command does not have a no form.

dspint dspfarm *slot*/*port*

Syntax Description

slot	Slot number of the interface.
port	Port number of the interface.

Defaults

Enabled

Command Modes

Global configuration

Command History

Release	Modification
12.0(5)XE	This command was introduced on the Cisco 7200 series routers.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.
12.2(13)T	This command was implemented on the Cisco 7200 series.

Usage Guidelines

DSP mapping occurs when DSP resources on one advanced interface module (AIM) or network module are available for processing of voice time-division multiplexing (TDM) streams on a different network module or on a voice/WAN interface card (VWIC). This command is used on Cisco 3660 routers with multiservice interchange (MIX) modules installed or on Cisco 2600 series routers with AIMs installed.

To reach voice-card configuration mode for a particular voice card, from global configuration mode enter the **voice-card** command and the slot number for the AIM or network module that you want to add to the pool. See the **voice-card** command reference for details on slot numbering.

The assignment of DSP pool resources to particular TDM streams is based on the order in which the streams are configured using the **ds0-group** command for T1/E1 channel-associated signaling (CAS) or using the **pri-group** command for ISDN PRI.

The assignment of DSP pool resources does not occur dynamically during call signaling.

To disable the interface use the **no shutdown** command.

Examples

The following example creates a DSP farm interface with a slot number of 1 and a port number of 0.

```
dspint dspfarm 1/0
```

To change codec complexity on the Cisco 7200 series, you must enter the following commands:

```
Router# configure terminal
Router(config)# dspint dspfarm 2/0
Router(config-dspfarm)# codec medium | high ecan-extended
```

Related Commands

Command	Description
ds0-group	Specifies the DS0 time slots that make up a logical voice port on a T1 or E1 controller
no shutdown	Disables the interface.
pri-group	Specifies an ISDN PRI on a channelized T1 or E1 controller
show interfaces dspfarm dsp	Displays information about the DSP interface.
voice-card	Enters voice-card configuration mode.

dtmf timer inter-digit

Command	Description
dial-peer voice	Specifies the method of voice-related encapsulation.
rtp payload-type	Chooses the type of payload in the RTP NTE packet.

To configure the dual tone multifrequency (DTMF) interdigit timer for a DS0 group, use the **dtmf timer inter-digit** command in T1 controller configuration mode. To restore the timer to its default value, use the **no** form of this command.

dtmf timer inter-digit *milliseconds*

no dtmf timer inter-digit

Syntax Description

milliseconds	DTMF interdigit timer duration, in milliseconds. Range is from 250 to 3000. The default is 3000.

Defaults

3000 milliseconds

Command Modes

T1 controller configuration

Command History

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

Usage Guidelines

Use the **dtmf timer inter-digit** command to specify the duration in milliseconds the router waits to detect the end of DTMF digits. After this period, the router expects no more digits to arrive and establishes the call.

Examples

The following example, beginning in global configuration mode, sets the DTMF interdigit timer value to 250 milliseconds:

```
controller T1 2
 ds0-group 2 timeslots 4-10 type e&m-fgb dtmf dnis
 cas-custom 2
 dtmf timer inter-digit 250
```

Related Commands

Command	Description
cas-custom	Customizes E1 R2 signaling parameters for a particular E1 channel group on a channelized E1 line.
ds0-group	Configures channelized T1 time slots, which enables a Cisco AS5300 modem to answer and send an analog call.

dtmf-relay (Voice over Frame Relay)

To enable the generation of FRF.11 Annex A frames for a dial peer, use the **dtmf-relay** command in dial-peer configuration mode. To disable the generation of FRF.11 Annex A frames and return to the default handling of dial digits, use the **no** form of this command.

dtmf-relay

no dtmf-relay

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and MC3810.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T, and implemented on the Cisco 7200 series router.

Usage Guidelines

This command applies to all Voice over Frame Relay (VoFR) and Voice over ATM (VoATM) applications on the Cisco MC3810 multiservice concentrator and to VoFR applications on the Cisco 2600 series routers and 3600 series routers.

Cisco recommends that this command be used with low bit-rate codecs.

When **dtmf-relay** (VoFR) is enabled, the digital signal processor (DSP) generates Annex A frames instead of passing a dual tone multifrequency (DTMF) tone through the network as a voice sample. For information about the payload format of FRF.11 Annex A frames, refer to the *Cisco IOS Wide-Area Networking Configuration Guide* and *Cisco IOS Wide-Area Networking Command Reference, Release 12.2*.

Examples

The following example shows how to enable FRF.11 Annex A frames on a Cisco 2600 series routers or 3600 series router or on an MC3810 multiservice concentrator for VoFR dial peer 200, starting from global configuration mode:

```
dial-peer voice 200 vofr
 dtmf-relay
```

Related Commands

Command	Description
called-number (dial-peer)	Enables an incoming VoFR call leg to get bridged to the correct POTS call leg when using a static FRF.11 trunk connection.
codec (dial-peer)	Specifies the voice coder rate of speech for a VoFR dial peer.
connection	Specifies a connection mode for a voice port.
cptone	Specifies a regional analog voice interface-related tone, ring, and cadence setting.
destination-pattern	Specifies the prefix, the full E.164 telephone number, or an ISDN directory number (depending on the dial plan) to be used for a dial peer.
preference	Indicates the preferred order of a dial peer within a rotary hunt group.
session protocol	Establishes a session protocol for calls between the local and remote routers via the packet network.
session target	Specifies a network-specific address for a specified dial peer or destination gatekeeper.
signal-type	Sets the signaling type to be used when connecting to a dial peer.

dtmf-relay (Voice over IP)

To specify how an H.323 or Session Initiation Protocol (SIP) gateway relays dual tone multifrequency (DTMF) tones between telephony interfaces and an IP network, use the **dtmf-relay** command in dial-peer configuration mode. To remove all signaling options and send the DTMF tones as part of the audio stream, use the **no** form of this command.

dtmf-relay [**cisco-rtp**] [**h245-alphanumeric**] [**h245-signal**] [**rtp-nte**]

no dtmf-relay [cisco-rtp] [**h245-alphanumeric**] [**h245-signal**] [**rtp-nte**]

Syntax Description

cisco-rtp	(Optional) Forwards DTMF tones by using Real-Time Transport Protocol (RTP) with a Cisco proprietary payload type.
h245-alphanumeric	(Optional) Forwards DTMF tones by using the H.245 “alphanumeric” User Input Indication method. Supports tones from 0 to 9, *, #, and from A to D.
h245-signal	(Optional) Forwards DTMF tones by using the H.245 “signal” User Input Indication method. Supports tones are from 0 to 9, *, #, and from A to D.
rtp-nte	(Optional) Forwards DTMF tones by using Real-Time Transport Protocol (RTP) with the Named Telephone Event (NTE) payload type.

Defaults

Disables DTMF tones are sent inband that is, left in the audio stream.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced on the Cisco AS5300.
12.0(2)XH	The **cisco-rtp**, **h245-alphanumeric**, and **h245-signal** keywords were added.
12.0(5)T	This command was integrated into Cisco IOS Release 12.0(5)T.
12.0(7)XK	This command was first supported for VoIP on the MC3810.
12.1(2)T	Changes made in Cisco IOS Release 12.0(7)XK 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.
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(4)T	This command does not support the Cisco AS5300, Cisco AS5350, and Cisco AS5400 in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850 platform.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

DTMF is the tone generated when you press a button on a touch-tone phone. This tone is compressed at one end of a call; when the tone is decompressed at the other end, it can become distorted, depending on the codec used. The DTMF relay feature transports DTMF tones generated after call establishment out of band using either a standard H.323 out-of-band method and a proprietary RTP-based mechanism or for Session Initiation Protocol (SIP) calls, a Named Telephony Events (NTE) Real-time Transport Protocol (RTP) packet.

The gateway only sends DTMF tones in the format you specify if the remote device supports it. If the remote device supports multiple formats, the gateway chooses the format according to the following priority:

1. cisco-rtp (highest priority)
2. h245-signal
3. h245-alphanumeric
4. rtp-nte
5. None—DTMF sent in-band

The principal advantage of the **dtmf-relay** command is that it sends DTMF tones with greater fidelity than is possible in-band for most low-bandwidth codecs, such as G.729 and G.723. Without the use of DTMF relay, calls established with low-bandwidth codecs may have trouble accessing automated DTMF-based systems, such as voice mail, menu-based Automatic Call Distributor (ACD) systems, and automated banking systems.

Note

- The **cisco-rtp** keyword of the **dtmf-relay** command is a proprietary Cisco implementation and operates only between two Cisco AS5800 access concentrators running Cisco IOS Release 12.0(2)XH or between Cisco AS5800 access concentrators or Cisco 2600 series or Cisco 3600 series routers running Cisco IOS Release 12.0(2)XH or later releases. Otherwise, the DTMF relay feature does not function, and the gateway sends DTMF tones in-band.
- The **cisco-rtp** keyword of the **dtmf-relay** command is supported on Cisco 7200 series routers.
- The h245-alphanumeric and h245-signal DTMF settings on a Cisco MC3810 multiservice access concentrator require a high-performance compression module (HCM) and are not supported on a Cisco MC3810 with a non-HCM voice compression module (VCM).

Examples

The following example configures DTMF relay with the **cisco-rtp** keyword when sending DTMF tones to dial peer 103:

```
dial-peer voice 103 voip
 dtmf-relay cisco-rtp
```

The following example configures DTMF relay with the **cisco-rtp** or **h245-signal** keywords when DTMF tones are sent to dial peer 103:

```
dial-peer voice 103 voip
 dtmf-relay cisco-rtp h245-signal
```

The following example configures the gateway to send DTMF in-band (the default) when DTMF tones to are sent dial peer 103:

```
dial-peer voice 103 voip
 no dtmf-relay
```

The following example shows use of the **dtmf-relay** command with the SIP NTE DTMF relay feature:

```
Router(config-dial-peer)# dtmf-relay rtp-nte
```

dualtone

To enter the cp-dualtone configuration mode for specifying a custom call-progress tone, use the **dualtone** command in custom-cptone voice-class configuration mode. To configure the custom-cptone voice class not to detect a call-progress tone, use the **no** form of this command.

dualtone {**ringback** | **busy** | **reorder** | **out-of-service** | **number-unobtainable** | **disconnect**}

no dualtone {**ringback** | **busy** | **reorder** | **out-of-service** | **number-unobtainable** | **disconnect**}

Syntax Description

ringback	Ringback tone.
busy	Busy tone.
reorder	Reorder tone.
out-of-service	Out-of-service tone.
number-unobtainable	Number-unavailable tone.
disconnect	Disconnect tone.

Defaults

No call-progress tones are defined within the custom-cptone voice class

Command Modes

Custom-cptone voice-class configuration

Command History

Release	Modification
12.1(5)XM	This command was introduced on the Cisco 2600 and Cisco 3600 series and on the Cisco MC3810.
12.2(2)T	This command was implemented on the Cisco 1750 router and integrated into Cisco IOS Release 12.2(2)T.

Usage Guidelines

The **dualtone** command enters the cp-dualtone configuration mode and specifies a call-progress tone to be detected. You can specify additional call-progress tones without exiting cp-dualtone configuration mode.

Any call-progress tones that are not specified are not detected.

To delete a call-progress tone from this custom-cptone voice class, use the **no** form of this command and the keyword for the tone that should not be detected; for example, **no dualtone busy**.

You need to associate the class of custom call-progress tones with a voice port for this command to affect tone detection.

Examples

The following example enters cp-dualtone configuration mode and specifies busy tone and ringback tone in the custom-cptone voice class country-x.

```
voice class custom-cptone country-x
 dualtone busy
 frequency 440 480
 cadence 500 500
 dualtone ringback
 frequency 400 440
 cadence 2000 4000
```

The following example deletes ringback tone from the custom-cptone voice class country-x.

```
voice class custom-cptone country-x
 no dualtone ringback
```

Related Commands

Command	Description
cadence	Defines the tone on and off durations for a call-progress tone.
frequency	Defines the frequency components for a call-progress tone.
supervisory custom-cptone	Associates a class of custom call-progress tones with a voice port.
voice class custom-cptone	Creates a voice class for defining custom call-progress tones.