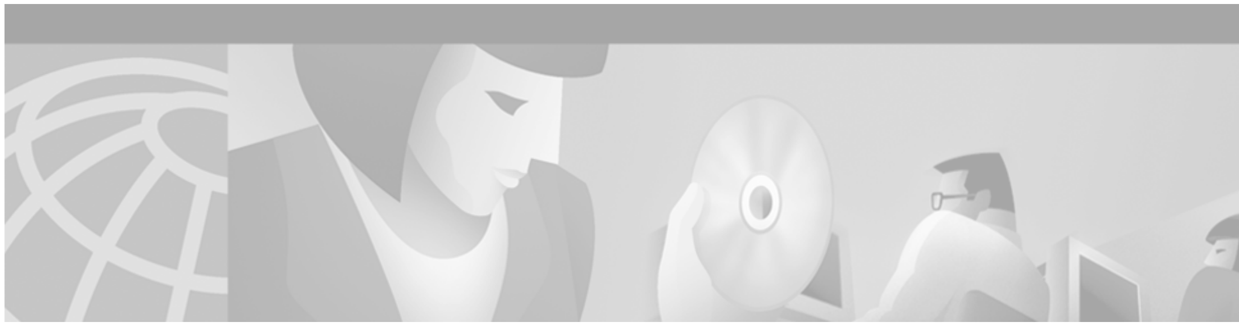

Cisco IOS Voice, Video, and Fax Commands: D Through F

This chapter presents the commands to configure and maintain Cisco IOS voice, video, and fax applications. The commands are presented in alphabetical order beginning with the letter D. Some commands required for configuring voice, video, and fax 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, Video, and Fax Configuration Guide*.

default

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

default {**inband-alerting** | **max-forwards** | **retry** {**invite** | **response** | **bye** | **cancel**} | **sip-server** | **timers** {**trying** | **connect** | **disconnect** | **expires**} | **transport**}

Syntax Description

inband-alerting	Resets inband-alerting to its default of generating the header "Require: com.cisco.inband-alerting" in outgoing INVITE messages. Tones are fed from the terminating gateway.
max-forwards	Resets max-forwards to its default of 6.
retry {**invite** | **response** | **bye** | **cancel**}	Resets the specified retry to its default (6 for invite and response; 10 for bye and cancel).
sip-server	Resets the sip-server to a null value.
timers {**trying** | **connect** | **disconnect** | **expires**}	Resets the specified timer to its default (500 for trying, connect, and disconnect; 180,000 for expires).
transport	Resets transport to the default of both User Datagram Protocol (UDP) and TCP enabled.

Defaults

No default behavior or values.

Command Modes

SIP user-agent configuration

Command History

Release	Modification
12.1(1)T	This command was introduced on the Cisco 2600 and 3600 series routers and on the CiscoAS5300 universal access server.

Examples

The following example sets inband-alerting to its default.

```
sip-ua
 default inband-alerting
```

Related Commands

Command	Description
cap-list vfc	Adds a voice codec overlay file to the capability file list.

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 universal access server in which the VFC is installed. Valid entries are from 0 to 2.

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 universal access server.
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 will be 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 multiservice concentrator.

Release	Modification
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.
12.1(2)T	The support for Cisco 2600 and 3600 series routers 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 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 multiservice concentrator 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 universal access server in which the specified VFC resides. Valid entries are from 0 to 2.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(1)NA	This command was introduced on the Cisco AS5300 universal access server.
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 include a specific description about the digital signal processor (DSP) interface, use the **description** command in voice-port configuration mode. To disable this feature, 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.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series routers router.
11.3(1)MA	This command was implemented on the Cisco MC3810 multiservice concentrator.

Usage Guidelines

Use the **description** command to include descriptive text about the DSP interface connection. This information is displayed when a **show** command is issued, 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
```

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	The 12.0(5)XE modifications were 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
```

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 indicating an E.164 standard number.
string	Series of digits that specify 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 previous digit/pattern occurred zero or multiple times, similar to the wildcard usage in the regular expression. • Plus sign (+), which matches a sequence of one or more matches of the character/pattern. **Note** The plus sign used as part of the digit string is different from the plus sign that can be used in front of the 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 matching that character or used with a single character with no other significance (matching that character). • Question mark (?), which indicates that the previous 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. This is similar to a regular expression rule. • Parentheses "( )", which indicate a pattern and is the same as the regular expression rule.
T	(Optional) Control character indicating 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 routers.
11.3(1)MA	This command was implemented on the Cisco MC3810 multiservice concentrator.
12.0(4)XJ	This command was modified for store and forward fax.
12.1(1)	The command as modified for store and forward fax was integrated into Cisco IOS Release 12.1(1).
12.0(7)XR	Support for the plus sign, percent sign, question mark, brackets, and parentheses symbols in the dial string was added to the Cisco AS5300 universal access server.
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, Cisco 3600, and Cisco MC3810 multiservice concentrator.
12.1(1)T	The modifications made in Cisco IOS Release 12.0(7)XR for the Cisco AS5300 were first supported in Cisco IOS Release 12.1(1)T and were first supported on the T train for the following additional platforms: Cisco 1750, Cisco 2600 series routers, Cisco 3600 series routers, Cisco 7200, and Cisco 7500.
12.1(2)T	The modifications made in Cisco IOS Release 12.0(7)XK for the Cisco MC3810 multiservice concentrator were first supported on the T train.

Usage Guidelines

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

This pattern 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 corresponding 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, in 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 was repeated multiple times:

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

The following example shows configuration of a destination pattern in which the digit numbers range 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 digit numbers range 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 this 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 routers.

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 and 3600 series routers.

Usage Guidelines

Use the **detect v54 channel-group** controller configuration command to enable V.54 loopback detection. The remote device will send 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. Values range is from 0 to 2,147,483,647.

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 and 3600 series routers and on the AS5300 access server.
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. Valid entries are from 0 to 500 table entries. A value of 0 prevents any history from being retained.
retain-timer *number*	Specifies the length of time, in minutes, for entries in the call history table. Valid entries are from 0 to 2,147,483,647 minutes. A value of 0 prevents any history from being retained.

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 multiservice concentrator.
12.0(2)T	The 12.0(1)XA application on the Cisco MC3810 multiservice concentrator 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
```

dial-peer hunt

To specify a hunt selection order for dial peers, use the **dial-peer hunt** command in dial-peer 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

Dial-peer configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced and supported on the Cisco 2600, 3600, and 7200 series routers, the Cisco MC3810 multiservice concentrator, and the Cisco AS5300 universal access servers.
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 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 and 3600 series routers and on the MC3810 multiservice concentrator.
12.1(2)T	The modifications made in Cisco IOS Release 12.0(7)XK were 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. Valid entries are from 1 to 10,000. 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 multiservice concentrator.
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 multiservice concentrator, 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 enter dial-peer configuration mode (and to specify the method of voice encapsulation), use the **dial-peer voice** command in global configuration mode. To disable a defined dial peer, use the **no** form of this command.

Cisco 2600 Series Routers

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

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

Cisco 3600 Series Routers

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

no dial-peer voice *tag*

Cisco 7200 Series Routers

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

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

Cisco 7204 VXR and Cisco 7206 VXR Routers

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

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

Cisco AS5300 Universal Access Server

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

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

Cisco MC3810 Multiservice Concentrator

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

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

Syntax Description

tag	Digits that define a particular dial peer. Valid entries are from 1 to 2,147,483,647.
mmoip	Indicates that this is a multimedia mail peer using IP encapsulation on the IP backbone. **Note** On the Cisco AS5300 universal access server, MMoIP is available only if you have modem ISDN channel aggregation (MICA) technologies modems.
pots	Indicates that this is a plain old telephone service (POTS) peer using Voice over IP encapsulation on the IP backbone.

voatm	(Cisco 3600 series routers, Cisco MC3810 multiservice concentrators, Cisco 7204 VXR routers, and Cisco 7206 VXR routers only) Specifies that this is a Voice over ATM dial peer using the real-time AAL5 voice encapsulation on the ATM backbone network.
vofr	Specifies that this is a Voice over Frame Relay dial peer using FRF.11 encapsulation on the Frame Relay backbone network.
voip	Indicates that this is a VoIP peer using 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 routers.
11.3(1)MA	This command was first supported on the Cisco MC3810 multiservice concentrator, with support for the **pots**, **voatm**, **vofr**, and **vohdlc** keywords.
12.0(3)T	This command was first supported on the AS5300, with support for the **pots** and **voip** keywords.
12.0(3)XG	The **vofr** keyword was added for the Cisco 2600 series routers and Cisco 3600 series platforms.
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 routers platform.
12.0(4)XJ	The **mmoip** keyword was added for the Cisco AS5300 universal access server platform. Also, 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 multiservice concentrator, and the **voatm** keyword was added for the Cisco 3600 series routers router. Support for **vohdlc** on the Cisco MC3810 multiservice concentrator was removed in this release.
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 also 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.

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 that particular dial peer. To delete a dial peer, use the **no** form of this command. To disable a dial peer, use the **shutdown** command in dial-peer configuration mode.

In store and forward fax on the Cisco AS5300 universal access server, the POTS dial peer defines the inbound faxing line characteristics from the sending fax device to the receiving Cisco AS5300 universal access server and the outbound line characteristics from the sending Cisco AS5300 universal access server to the receiving fax device. The Multimedia Mail over Internet Protocol (MMoIP) dial peer defines the inbound faxing line characteristics from the Cisco AS5300 universal access server 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 universal access server, 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
```

Related Commands

Command	Description
codec (dial-peer)	Specifies the voice coder rate of speech for a Voice over Frame Relay dial peer.
destination-pattern	Specifies the prefix, the full E.164 telephone number, or an ISDN directory number (depending on the dial plan) to be used for a dial peer.
dtmf-relay (Voice over Frame Relay)	Enables the generation of FRF.11 Annex A frames for a dial peer.
preference	Indicates the preferred order of a dial peer within a rotary hunt group.
sequence-numbers	Enables the generation of sequence numbers in each frame generated by the DSP for Voice over Frame Relay 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.
voice-port	Enters voice-port 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

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series routers.
11.3(1)MA3	The **pulse** keyword was added. This command was implemented on the Cisco MC3810 multiservice concentrator.
12.0(7)XK	The **mf** keyword was added.
12.1(2)T	The **pulse** and **mf** keyword additions and the MC3810 multiservice concentrator platform implementation were integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

Use the **dial-type** command to specify an out-dialing type for a Foreign Exchange Office (FXO) or ear and mouth (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. MF 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 multiservice concentrator. If you are using the **dial-type** command with E&M WinkStart, 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 multiservice concentrator to support a rotary (pulse tone) dialer:

```
voice-port 1/1
 dial-type pulse
```

The following example shows a voice port configured on the Cisco MC3810 multiservice concentrator to support a DTMF (touch-tone) dialer:

```
voice-port 1/1
 dial-type dtmf
```

The following example shows a voice port configured on the Cisco MC3810 multiservice concentrator to support a multifrequency tone dialer:

```
voice-port 1/1
 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 routers, Cisco 3600 series routers, Cisco MC3810 multiservice concentrator) • Voice over Frame Relay (VoFR)—Cisco 2600 series routers, Cisco 3600 series routers, Cisco MC3810 multiservice concentrator) • Voice over ATM (VoATM)—Cisco 3600 series routers and Cisco MC3810 multiservice concentrator
12.1(1)T	The modifications made in Cisco IOS Release 12.0(7)XK were first supported 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 multiservice concentrator) • VoFR (Cisco 2600 series routers, Cisco 3600 series routers, and Cisco MC3810 multiservice concentrator) • VoATM (Cisco 3600 series routers, Cisco MC3810 multiservice concentrator)

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 16 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 16 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 17 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 17 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.
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 dial (DID) call treatment for the incoming called number, use the **direct-inward-dial** command in dial-peer configuration mode. To disable DID, 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 on the Cisco AS5300 universal access server.
12.0(4)XJ	This command was implemented for store and forward fax.
12.1(1)	The implementation of the command for store and forward fax was integrated into Cisco IOS Release 12.1(1).

Usage Guidelines

Use the **direct-inward-dial** command to enable the DID call treatment for the incoming called numbers. 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. When disabled, the called number is used to select the outgoing dial peer. The caller will be 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 incoming called numbers:

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

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 Cisco 2600 series routers, 3600 series, 7200 series, 7500 series, AS5300, AS5800, and MC3810 multiservice concentrator.

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 with PI is received from the terminating switch, the Disconnect with PI is converted to a Disconnect. But if this command is enabled on the voice port of the terminating gateway, and a Disconnect with PI is received from the terminating switch, the Disconnect message is not converted to a standard Disconnect 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 the same 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 multiservice concentrator.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.
12.1(2)T	The support for the Cisco 2600 and 3600 series routers 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 multiservice concentrator:

```
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 or 3600 series router:

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

Command History

Command	Description
show voice port	Displays voice port configuration information.

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. The 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 multiservice concentrator and Cisco 2600 and 3600 series routers.
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

To specify the DS0 time slots that make up a logical voice port on a T1 or E1 controller and to specify the signaling type by which the router communicates with the PBX or Public Switched Telephone Network (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 2600 and 3600 Series and Cisco MC3810 Multiservice Concentrator—T1

ds0-group *ds0-group-no* **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**}

no ds0-group *ds0-group-no*

Cisco 2600 and 3600 Series and Cisco MC3810 Multiservice Concentrator—E1

ds0-group *ds0-group-no* **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-no*

Cisco MC3810 Multiservice Concentrator—E1

ds0-group *ds0-group-no* **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-no*

Cisco 7200 and 7500 Series T1 and E1 Voice Ports

ds0-group *ds0-group-no* **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-no*

Cisco AS5300 universal access server—T1

ds0-group *ds0-group-no* **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-no*

Cisco AS5300 universal access server—E1

ds0-group *ds0-group-no* **timeslots** *timeslot-list* **type** {**none** | **p7** | **r2-analog** | **r2-digital** | **r2-lsv181-digital** | **r2-pulse**}

no ds0-group *ds0-group-no*

Cisco AS5800 Universal Access Server—T1

ds0-group *ds0-group-no* **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-no*

Cisco AS5800 E1 Voice Ports

ds0-group *ds0-group-no* **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-no*

Syntax Description

For the Cisco 2600 and 3600 Series Routers and Cisco MC3810 Multiservice Concentrators—T1:

ds0-group-no	A value from 0 to 23 that identifies the DS0 group.
timeslots *timeslot-list*	Time slot *timeslot-list* is a single time-slot number, a single range of numbers, or multiple ranges of numbers separated by commas. For T1, allowable values are 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 ear and mouth (E&M) interface allows connection for PBX trunk lines (tie lines) and telephone equipment. The Foreign Exchange Station (FXS) interface allows connection of basic telephone equipment and PBX. The Foreign Exchange Office (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 the following: • **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-fgd**—E&M Type II Feature Group D. • **e&m-immediate-start**—E&M immediate start. • **e&m-wink-start**—E&M Mercury Exchange Limited Channel-Associated Signaling (MELCAS) wink-start signaling support. • **ext-sig**—An option available only when the **mode CCS** command is enabled on the Cisco MC3810 multiservice concentrator for FRF.11 transparent common channel signaling (CCS) support. • **fgd-eana**—Feature Group D exchange access North American. • **fxo-ground-start**—FXO ground-start signaling support. • **fxo-loop-start**—FXO loop-start signaling support. • **fxs-ground-start**—FXS ground-start signaling support. • **fxs-loop-start**—FXS loop-start signaling support.

For the Cisco 2600 and 3600 Series Routers and Cisco MC3810 Multiservice Concentrators—E1:

ds0-group-no	An identifying value from 0 to 14 and 16 to 30. 15 is reserved.
timeslots *timeslot-list*	Time slot *timeslot-list* is a single time-slot number, a single range of numbers, or multiple ranges of numbers separated by commas. For E1, allowable values are from 1 to 31. Examples are as follows: • 2 • 1-15,17-31 • 1-31 • 2,4,6-12, 17-31
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 CO to a standard PBX interface where permitted by local regulations; it is often used for OPXs. Types are the following: • **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-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 multiservice concentrator for FRF.11 transparent CCS support. • **fgd-eana**—Feature Group D exchange access North American. • **fxo-ground-start**—Specifies FXO ground-start signaling. • **fxo-loop-start**—Specifies FXO loop-start signaling. • **fxo-melcas**—MELCAS FXO signaling. • **fxs-ground-start**—FXS ground-start signaling. • **fxs-loop-start**—FXS loop-start signaling. • **fxs-melcas**—MELCAS FXS signaling. • **r2-analog**—Specifies R2 analog line signaling. • **r2-digital**—Specifies R2 digital line signaling. • **r2-pulse**—Specifies 7-pulse line signaling, a transmitted pulse that indicates a change in the line state.

For the Cisco 7200 and 7500 Series Routers—T1 and E1:

ds0-group-no	For T1, a value from 0 to 23 that identifies the DS0 group. For E1, a value from 0 to 14 and 16 to 30. 15 is reserved
timeslots *timeslot-list*	Time slot *timeslot-list* is a single time-slot number, a single range of numbers, or multiple ranges of numbers separated by commas. For T1, allowable values are from 1 to 24. For E1, allowable values are from 1 to 31. Examples are as follows: • 2 • 1-15,17-24 • 1-31 • 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 CO to a standard PBX interface where permitted by local regulations; it is often used for OPXs. Types are the following: • **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-fgd**—E&M Type II Feature Group D. • **e&m-immediate-start**—E&M immediate start. • **e&m-wink-start**—E&M MELCAS wink-start signaling support. • **fxs-ground-start**—FXO ground-start signaling support. • **fxs-loop-start**—FXS loop start. • **fxo-ground-start**—Specifies FXO ground-start signaling. • **fxo-loop-start**—FXO loop-start signaling support.

For the Cisco AS5300 Universal Access Server—T1:

ds0-group-no	A value from 0 to 23 that identifies the DS0 group.
timeslots *timeslot-list*	time slot *timeslot-list* is a single time-slot number, a single range of numbers, or multiple ranges of numbers separated by commas. Allowable values are from 1 to 24. Examples are as follows: • 2 • 1-15,17-24 • 1-23 • 2,4,6-12
service *service-type*	(Optional) Indicates the type of calls to be handled by this DS0 group—**data**, **fax**, **voice,** or **mgcp**.
type	(Optional) 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. Types are the following: • **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. • **fxs-ground-start**—FXS ground start. • **fxs-loop-start**—FXS loop start. • **fgd-eana**—Feature Group D exchange access North American. • **fgd-os**—Feature Group D operator services. • **r1-itu**—Line signaling based on international signaling standards. • **sas-ground-start**—Single attachment station (SAS) ground start. • **sas-loop-start**—SAS loop start. • **none**—Null signaling for external call control.
tone *type*	(Optional) Specifies the tone as **dtmf** or **mf**.
addr info	(Optional) Specifies the calling/called party.

For the Cisco AS5300 Universal Access Server—E1:

ds0-group-no	An identifying value from 0 to 14 and 16 to 30. 15 is reserved.
timeslots *timeslot-list*	time slot *timeslot-list* is a single time-slot number, a single range of numbers, or multiple ranges of numbers separated by commas. Allowable values are from 1 to 31. Examples are as follows: • 2 • 1-15,17-31 • 1-31 • 2,4,6-12, 24
type	The signaling method selection for the **type** keyword depends on the connection that you are making. Types are the following: • **none**—Null signaling for external call control. • **p7**—Specifies the p7 switch type. • **r2-analog**—Specifies R2 analog line signaling. • **r2-digital**—Specifies R2 digital line signaling. • **r2-lsv181-digital**—Specifies a specific R2 digital line. • **r2-pulse**—Specifies 7-pulse line signaling, a transmitted pulse that indicates a change in the line state.

For the Cisco AS5300 Universal Access Server—T1:

ds0-group-no	A value from 0 to 23 that identifies the DS0 group.
timeslots *timeslot-list*	time slot *timeslot-list* is a single time-slot number, a single range of numbers, or multiple ranges of numbers separated by commas. Allowable values are 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. Types are the following: • **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. • **fxs-ground-start**—FXS ground start. • **fxs-loop-start**—FXS loop start. • **fgd-eana**—Feature Group D exchange access North American. • **r1-itu**—A 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**—SAS ground start. • **sas-loop-start**—SAS loop start. • **none**—Null signaling for external call control.

For the Cisco AS5800 Universal Access Server—E1:

ds0-group-no	An identifying value from 0 to 14 and 16 to 30. 15 is reserved.
timeslots *timeslot-list*	time slot *timeslot-list* is a single time-slot number, a single range of numbers, or multiple ranges of numbers separated by commas. Allowable values are from 1 to 31. Examples are as follows: • 2 • 1-15,17-31 • 1-31 • 2,4,6-12, 18-31
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. Types are the following: • **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. • **fxs-ground-start**—FXS ground start. • **fxs-loop-start**—FXS loop start. • **p7**—Specifies the p7 switch type. • **r2-analog**—Specifies R2 analog line signaling. • **r2-digital**—Specifies R2 digital line signaling. • **r2-pulse**—Specifies 7-pulse line signaling, a transmitted pulse that indicates a change in the line state. • **sas-ground-start**—SAS ground start. • **sas-loop-start**—SAS loop start. • **none**—Null signaling for external call control.

Defaults

No DS0 group is defined.

Command Modes

Controller configuration

Command History

Release	Modification
11.2	This command was introduced for the Cisco AS5300 universal access server as the **cas-group** command.
11.3(1)MA	The command was introduced as the **voice-group** command for the Cisco MC3810 multiservice concentrator.
12.0(1)T	The **cas-group** command was introduced for the Cisco 3600 series routers.

Release	Modification
12.0(5)T	The command was renamed **ds0-group** on the Cisco AS5300 and Cisco 2600 and 3600 series routers. Some keyword modifications were implemented.
12.0(5)XE	This command was introduced for the Cisco 7200 series.
12.0(7)XK	Support for this command was extended to the Cisco MC3810 multiservice concentrator. When the **ds0-group** command became available on the Cisco MC3810 multiservice concentrator, 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(1)T	The **ds0-group** command was implemented for the Cisco 7200 series.
12.1(2)XH	The **e&m-fgd** and **fgd-eana** keywords were added for Feature Group D signaling.
12.1(3)T	The command was modified for the Cisco 7500 series routers. The **fgd-os** signaling type and the **voice** service type were added.

Usage Guidelines

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

- Cisco 2600, 3600, and 7200 series routers:
  - slot**/**port**:**ds0-group-no
- Cisco MC3810 multiservice concentrators:
  - slot**:**ds0-group-no

On the Cisco MC3810 multiservice concentrator, 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.

Note 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.

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 or 3600 series router:

```
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 multiservice concentrator 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
```

Related Commands

Command	Description
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 you are using.

dsn

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

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

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

Syntax Description

delay	Indicates that when the mail is sent, the next-hop mailer is requested to send a message to the FROM address if the mail message is delayed. The definition of delay is made by each mailer and is not controllable by the sender (the Cisco AS5300 universal access server). Each mailer in the path to the recipient that supports the delivery status notification (DSN) extension receives the same request.
failure	Indicates that when the mail is sent, the next-hop mailer is requested to send a message to the FROM address if the mail message failed to be delivered. Each mailer in the path to the recipient that supports the DSN extension receives the same request.
success	Indicates that when the mail is sent, the next-hop mailer is requested to send a message to the FROM address if the mail message is successfully delivered to the recipient. Each mailer in the path to the recipient that supports the DSN extension receives the same request.

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

Defaults

The default is **success** and **failure**.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.
12.1(1)	This command was integrated into Cisco IOS Release 12.1(1).

Usage Guidelines

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 in 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 can be 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, configure both the **mail from:** and **rcpt to:** commands for the DSN feature to be operational, 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, for some reason, the message was delayed while being delivered to the recipient.
- Success—Indicates that the message was successfully delivered to the recipient's mailbox.
- Failure—Indicates that, for some reason, the SMTP server was unable to deliver the message to the recipient.

Because these delivery states are not mutually exclusive, store and forward fax can be configured 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 containing a fax Tag Image File Format (TIFF) image. Use the **dsn** command to specify which notification messages will be sent to the user.

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

Note If the **failure** keyword is not included when configuring DSN, the sender will receive absolutely 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 containing the fax has been successfully delivered to the recipient or if the message containing the fax has failed, for whatever reason, to be delivered:

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

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

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

```
dsn delayed
```

Related Commands

Command	Description
mta send mail-from	Specifies the mail-from address (also called the RFC 821 envelope-from or the Return-Path address).

dspint dspfarm

To enable the digital signal processor (DSP) interface, use the **dspint dspfarm** command in global configuration mode.

dspint dspfarm *slot/port*

Syntax Description

slot	Specifies the slot number of the interface.
port	Specifies the port number of the interface.

Defaults

No default behavior or values.

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.

Usage Guidelines

To disable the DSPfarm interface, enter the **no shutdown** command.

Examples

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

```
dspint dspfarm 1/0
```

Related Commands

Command	Description
show interfaces dspfarm dsp	Displays information about the DSP interface.

dtmf-relay (Voice over IP)

To specify how an H.323 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 to 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**]

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

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 0-9, *, #, and A-D.
h245-signal	(Optional) Forwards DTMF tones by using the H.245 “signal” User Input Indication method. Supports tones 0-9, *, #, and A-D.

Defaults

No default behavior or values.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced on the Cisco AS5300 universal access server.
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 Voice over IP (VoIP) on the MC3810 multiservice concentrator.
12.1(2)T	Changes made in Cisco IOS Release 12.0(7)XK were integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

DTMF is the tone generated when you press a digit 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 a standard H.323 out-of-band method and a proprietary RTP-based mechanism.

The **dtmf-relay** command determines the outgoing format of relayed DTMF tones. The gateway automatically accepts all formats.

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

1. cisco-rtp (highest priority)
2. h245-signal
3. h245-alphanumeric
4. 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 ACD systems, and automated banking systems.

Note The **cisco-rtp** option of the **dtmf-relay** command is a proprietary Cisco implementation and operates only between two Cisco AS5800 universal access servers running Cisco IOS Release 12.0(2)XH, or between Cisco AS5800 universal access servers or Cisco 2600 or 3600 modular access 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.

Note The **h245-alphanumeric** and **h245-signal** DTMF settings on an MC3810 multiservice concentrator require a high-performance compression module (HCM) and are not supported on an MC3810 multiservice concentrator with a non-HCM voice compression module (VCM).

Examples

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

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

The next example configures DTMF relay with the **cisco-rtp** or **h245-signal** options when sending DTMF tones to dial peer 103:

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

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

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

Related Commands

Command	Description
dial-peer voice	Specifies the method of voice-related encapsulation.

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 routers, 3600 series, and MC3810 multiservice concentrator.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T, and support for the Cisco 7200 series router was added.

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 timer inter-digit

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 *milliseconds*

Syntax Description

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

Defaults

3000 milliseconds

Command Modes

T1 controller configuration

Command History

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

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 universal access server modem to answer and send an analog call.

echo-cancel comfort-noise

To specify that background noise be generated, use the **echo-cancel comfort-noise** command in controller configuration mode. To disable this feature, use the no form of this command.

echo-cancel comfort-noise

no echo-cancel comfort-noise

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Controller configuration

Command History

Release	Modification
12.1(2)T	This command was introduced on the Cisco 3600 series routers.

Usage Guidelines

Use the **echo-cancel comfort-noise** command to generate background noise to fill silent gaps during calls if voice activated dialing (VAD) is activated. If comfort noise is not enabled and VAD is enabled at the remote end of the connection, the user hears nothing or silence when the remote party is not speaking.

The configuration of comfort noise affects only the silence generated at the local interface; it does not affect the use of VAD on either end of the connection or the silence generated at the remote end of the connection.

For the OC-3/STM-1 ATM Circuit Emulation Service network module, echo cancellation must be enabled.

Examples

The following example enables comfort noise on a T1 controller:

```
controller T1 0/0
 echo-cancel enable
 echo-cancel comfort-noise
```

Related Commands

Command	Description
echo-cancel enable (controller)	Enables echo cancellation on a voice port.
voice port	Specifies which port is used for voice traffic.

echo-cancel compensation

To set attenuation for loud signals, use the **echo-cancel compensation** command in controller configuration. To disable this feature, use the **no** form of this command.

echo-cancel compensation

no echo-cancel compensation

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Controller configuration

Command History

Release	Modification
12.1(2)T	This command was introduced on the Cisco 2600 series routers.

Usage Guidelines

Use the **echo-cancel compensation** command to add attenuation control to the T1 or E1 controller. When this command is enabled, 6 decibels of attenuation are inserted if the signal level from the receive direction is loud. When loud signals are not received, the attenuation is removed.

For the OC-3/STM-1 ATM Circuit Emulation Service network module, echo cancellation must be enabled.

Examples

The following example enables attenuation control on a T1 controller:

```
controller T1 0/0
 echo-cancel enable
 echo-cancel compensation
```

Related Commands

Command	Description
echo-cancel enable (controller)	Enables echo cancellation on a voice port.
voice port	Specifies which port is used for voice traffic.

echo-cancel coverage

To adjust the maximum duration to cancel the voice echo, use the **echo-cancel coverage** command in voice-port configuration mode. To reset this command to the default value, use the **no** form of this command.

echo-cancel coverage {**8** | **16** | **24** | **32**}

no echo-cancel coverage

Syntax Description

8	8 milliseconds.
16	16 milliseconds.
24	24 milliseconds.
32	24 milliseconds.

Defaults

16 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series routers.
11.3(1)MA	This command was implemented on the Cisco MC3810 multiservice concentrator.
12.0(5)XK	The command was modified to add the 8-millisecond option.
12.0(5)XE	The command was modified to support the Cisco 7200 router platform.
12.1(1)T	The Cisco IOS Release 12.0(5)XK and 12.0(5)XE changes were integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

Use the **echo-cancel coverage** command to adjust the coverage size of the echo canceller. This command enables cancellation of voice that is sent out the interface and received back on the same interface within the configured amount of time. If the local loop (the distance from the interface to the connected equipment that is producing the echo) is longer, the configured value of this command should be extended.

If you configure a longer value for this command, it takes the echo canceller longer to converge; in this case, the user might hear a slight echo when the connection is initially set up. If the configured value for this command is too short, the user might hear some echo for the duration of the call because the echo canceller is not cancelling the longer delay echoes.

There is no echo or echo cancellation on the network side (for example, the non-POTS side) of the connection.

Note This command is valid only if the echo cancel feature has been enabled. For more information, see the **echo-cancel enable** command.

Examples

The following example adjusts the size of the echo canceller to 8 milliseconds on the Cisco 7200 series routers:

```
voice-port 1/0:0
 echo-cancel enable
 echo-cancel coverage 8
```

Related Commands

Command	Description
echo-cancel enable (controller)	Enables echo cancellation on a voice port.
echo-cancel enable	Enable echo cancellation on a voice port.

echo-cancel enable

To enable the cancellation of voice that is sent out the interface and is received back on the same interface, use the **echo-cancel enable** command in voice-port configuration mode. To disable echo cancellation, use the **no** form of this command.

echo-cancel enable

no echo-cancel enable

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled for all interface types.

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

The **echo-cancel enable** command enables cancellation of voice that is sent out the interface and is received back on the same interface; sound that is received back in this manner is perceived by the listener as an echo. Disabling echo cancellation might cause the remote side of a connection to hear an echo. Because echo cancellation is an invasive process that can minimally degrade voice quality, this command should be disabled if it is not needed.

The **echo-cancel enable** command does not affect the echo heard by the user on the analog side of the connection.

There is no echo path for a 4-wire ear and mouth (E&M) interface. The echo canceller should be disabled for this interface type.

Note This command is valid only if the **echo-cancel coverage** command has been configured. For more information, refer to the **echo-cancel coverage** command.

Examples

The following example enables the echo cancellation feature and adjusts the size of the echo canceller to 16 milliseconds on the Cisco 3600 series routers:

```
voice-port 1/0/0
 echo-cancel enable
 echo-cancel coverage 16
```

The following example enables the echo cancellation feature and adjusts the size of the echo canceller to 16 milliseconds on the Cisco MC3810 multiservice concentrator:

```
voice-port 1/1
```

```
echo-cancel enable
echo-cancel coverage 16
```

Related Commands

Command	Description
echo-cancel coverage	Specifies the amount of coverage for echo cancellation.
echo-cancel enable (controller)	Enables the echo cancellation on a voice port.
non-linear	Enables nonlinear processing in the echo canceler.

echo-cancel enable (controller)

To enable the echo cancel feature, use the **echo-cancel enable** command in controller configuration mode. To disable this feature, use the **no** form of this command.

echo-cancel enable

no echo-cancel enable

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled for all interface types.

Command Modes

Controller configuration

Command History

Release	Modification
12.1(2)T	This command was introduced for the OC-3/STM-1 ATM Circuit Emulation Service network module on the Cisco 3600 series routers.

Usage Guidelines

The **echo-cancel enable** command enables cancellation of voice that is sent out of the interface and received back on the same interface. Disabling echo cancellation might cause the remote side of a connection to hear an echo. Because echo cancellation is an invasive process that can minimally degrade voice quality, this command should be disabled if it is not needed.

The **echo-cancel enable** command does not affect the echo heard by the user on the analog side of the connection.

Note This command is valid only if the **echo-cancel coverage** command has been configured.

The following example enables the echo cancel feature on a T1 controller:

```
controller T1 0/0
 echo-cancel enable
 echo-cancel coverage 32
```

Related Commands

Command	Description
echo-cancel coverage	Specifies the amount of coverage for echo cancellation.
echo-cancel enable	Enables the echo cancellation on a voice port.
non-linear	Enables nonlinear processing in the echo canceler.
voice port	Configures the voice port.

echo-cancel loopback

To place the echo cancellation processor in loopback mode, use the **echo-cancel loopback** command in controller configuration mode. To disable loopback of the echo cancellation processor, use the **no** form of this command.

echo-cancel loopback

no echo-cancel loopback

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Controller configuration

Command History

Release	Modification
12.1(2)T	This command was introduced on the Cisco 3600 series routers.

Usage Guidelines

Use an **echo-cancel loopback** test on lines to detect and distinguish equipment malfunctions caused by either the line or the interface. If correct echo cancellation is not possible when an interface is in loopback mode, the interface is the source of the problem.

Examples

On a Cisco 3600 series routers router, the following example sets up echo cancellation loopback diagnostics:

```
controller T1 0/0
 echo-cancel enable
 echo-cancel coverage 32
 echo-cancel loopback
```

Related Commands

Command	Description
echo-cancel enable (controller)	Enables echo cancellation on a voice port.

encapsulation atm-ces

To enable circuit emulation service (CES) ATM encapsulation on the Cisco MC3810 multiservice concentrator, use the **encapsulation atm-ces** command in interface configuration mode. To disable CES ATM encapsulation, use the **no** form of this command.

encapsulation atm-ces

no encapsulation atm-ces

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Interface configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810 multiservice concentrator.
12.0	This command was integrated into Cisco IOS Release 12.0.

Usage Guidelines

This command applies to ATM configuration on the Cisco MC3810 multiservice concentrator. This command is supported only on serial ports 0 and 1.

Examples

The following example enables CES ATM encapsulation on serial port 0 on the Cisco MC3810 multiservice concentrator:

```
interface serial 0
 encapsulation atm-ces
```

Related Commands

Command	Description
ces cell-loss-integration-period	Sets the CES cell-loss integration period on the Cisco MC3810 multiservice concentrator.
ces clockmode synchronous	Configures the ATM CES synchronous clock mode on the Cisco MC3810 multiservice concentrator.
ces connect	Maps the CES service to an ATM PVC on the Cisco MC3810 multiservice concentrator.
ces initial-delay	Configures the size of the receive buffer of a CES circuit on the Cisco MC3810 multiservice concentrator.

Command	Description
ces max-buf-size	Configures the send buffer of a CES circuit on the Cisco MC3810 multiservice concentrator.
ces partial-fill	Configures the number of user octets per cell for the ATM CES on the Cisco MC3810 multiservice concentrator.
ces service	Configures the ATM CES type on the Cisco MC3810 multiservice concentrator.

encapsulation ftc-trunk

This command was removed in Cisco IOS Release 12.1(2)T and is no longer supported in Cisco IOS Release 12.2.

encryption

To set the algorithm to be negotiated with the provider, use the **encryption** command in settlement configuration mode. To reset to the default encryption method, use the **no** form of this command.

encryption {**des-cbc-sha** | **des40-cbc-sha** | **dh-des-cbc-sha** | **dh-des40-cbc-sha** | **null-md5** | **null-sha** | **all**}

no encryption {**des-cbc-sha** | **des40-cbc-sha** | **dh-des-cbc-sha** | **dh-des40-cbc-sha** | **null-md5** | **null-sha** | **all**}

Syntax Description

des-cbc-sha	Encryption type ssl_rsa_with_des_cbc_sha cipher suite.
des40-cbc-sha	Encryption type ssl_rsa_export_with_des40_cbc_sha cipher suite.
dh-des-cbc-sha	Encryption type ssl_dh_rsa_with_des_cbc_sha cipher suite.
dh-des40-cbc-sha	Encryption type ssl_dh_rsa_export_with_des40_cbc_sha cipher suite.
null-md5	Encryption type ssl_rsa_with_null_md5 cipher suite.
null-sha	Encryption type ssl_rsa_with_null_sha cipher suite.
all	All encryption methods are used in the Secure Socket Layer (SSL).

Defaults

The default encryption method is **all**. If none of the encryption methods is configured, the system uses all of the encryption methods in the SSL session negotiation.

Command Modes

Settlement configuration

Command History

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

Usage Guidelines

For Cisco IOS Release 12.0(4)XH1, only one encryption method is allowed for each provider.

Examples

The following example sets the algorithm to be negotiated with the provider, using the **encryption** command:

```
settlement 0
 encryption des-cbc-sha
```

Related Commands

Command	Description
connection-timeout	Sets the connection timeout.
customer-id	Sets the customer identification.
device-id	Sets the device identification.
max-connection	Sets the maximum number of simultaneous connections.
response-timeout	Sets the response timeout.
retry-delay	Sets the retry delay.
retry-limit	Sets the connection retry limit.
session-timeout	Sets the session timeout.
settlement	Enters settlement configuration mode.
show settlement	Displays the configuration for all settlement server transactions.
shutdown	Disables the settlement provider.
type	Specifies the provider type.
url	Specifies the ISP address.

erase vfc

To erase the Flash memory of a specified voice feature card (VFC), use the **erase vfc** command in privileged EXEC mode.

erase vfc *slot*

Syntax Description

slot	Specifies the slot on the Cisco AS5300 universal access server in which the specified VFC resides. Valid entries are from 0 to 2.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco AS5300 universal access server.

Usage Guidelines

Use the **erase vfc** command to erase the contents of Flash memory for a specified VFC (thereby freeing space in VFC Flash memory) including the default file list and the capability file list.

Examples

The following example erases the Flash memory on the VFC located in slot 0:

```
Router# erase vfc 0
```

Related Commands

Command	Description
delete vfc	Deletes a file from VFC Flash memory.

expect-factor

To specify when the router generates an alarm to the network manager, indicating that the expected quality of voice has dropped, use the **expect-factor** command in dial-peer configuration mode. To reset the default value, use the **no** form of this command.

expect-factor *value*

no expect-factor *value*

Syntax Description

value	Integers that represent the International Telecommunication Union (ITU) specification for quality of voice as described in G.113. Valid entries are from 0 to 20, with 0 representing toll quality.

Defaults

0

Command Modes

Dial-peer configuration

Command History

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

Usage Guidelines

This command applies to Voice over IP (VoIP) dial peers.

VoIP monitors the quality of voice received over the network. Use the **expect-factor** command to specify when the router generates a Simple Network Management Protocol (SNMP) trap to the network manager.

Examples

The following example configures toll quality of voice when connecting to a dial peer:

```
dial-peer voice 10 voip
 expect-factor 0
```

fax interface-type

To specify the voice feature card (VFC) to be used for a fax call, use the **fax interface-type** command in global configuration mode. To return to the default fax protocol, use the **no** form of this command.

fax interface-type {**modem** | **vfc**}

no fax interface-type {**modem** | **vfc**}

Syntax Description

modem	Specifies modem fax calls.
vfc	Specifies the VFC fax calls.

Defaults

None

Command Modes

Global configuration

Command History

Release	Modification
12.1(3)XI	This command was introduced on the Cisco AS5300 access server.
12.1(5)T	This command was integrated into the Cisco IOS Release 12.1(5)T.

Usage Guidelines

When using this command to change the interface type for fax calls, you must reload (reboot or reset) the router.

On the Cisco AS5300 access server, the keyword **vfc** maps to the **fax-mail** keyword. If you enter the **show run** command, the **fax-mail** keyword will display. The voice gateway defaults are as follows:

- If the gateway has modem cards only, the default is the **modem** keyword.
- If the gateway has voice cards only, the default is the **fax-mail** keyword. The **modem** keyword is unavailable. This applies to all platforms except the Cisco AS5300 access server.
- If the gateway has both modems and voice cards, the default is the **modem** keyword.

Examples

The following example specifies the use of a VFC interface for fax calls:

```
configure terminal
  fax interface-type vfc
```

fax protocol (dial-peer)

To specify the fax protocol for a specific Voice over IP (VoIP) dial peer, use the **fax protocol** command in dial-peer configuration mode. To return to the default fax protocol, use the **system** keyword. To disable the T.38 fax protocol for a specific dial peer, use the **no** form of this command.

fax protocol {**cisco** | **t38** [**ls_redundancy** *value*] [**hs_redundancy** *value*] | **system**}

no fax protocol

Syntax Description

cisco	Cisco proprietary fax protocol.
t38	ITU-T T.38 standard fax protocol.
ls_redundancy *value*	(Optional) Low-speed redundancy for the T.38 fax protocol. The *value* can be from 0 to 5. The default is 0. The **ls_redundancy** parameter refers to data redundancy in the low-speed V.21-based T.30 fax machine protocol.
hs_redundancy *value*	(Optional) High-speed redundancy for the T.38 fax protocol. The *value* can be from 0 to 2. The default is 0. The **hs_redundancy** parameter refers to data redundancy in the high-speed V.17, V.27, and V.29 T.4 or T.6 fax machine image data.
system	Global fax protocol when neither **cisco** nor **t38** is specified. The value is taken from the global configuration by default.

Defaults

The default protocol is system.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco 2600 series routers, Cisco 3600 series routers, and Cisco MC3810 multiservice concentrators.

Usage Guidelines

Use the **fax protocol t38** command to configure T.38 Fax Relay for a specific dial peer. The **t38** keyword enables the T.38 Fax Relay protocol. The **cisco** keyword selects the original Cisco proprietary fax protocol. When the **system** keyword is selected in the dial peer, it specifies the global default fax protocol used by a dial peer, set by the **fax protocol t.38** command. The optional **ls_redundancy** and **hs_redundancy** parameters are used to send redundant T.38 fax packets.

Note The **ls_redundancy** and **hs_redundancy** parameters are applicable only to the T.38 Fax Relay protocol.

The **ls_redundancy** parameter refers to data redundancy in the low-speed, V.21-based T.30 fax machine protocol. For the **ls_redundancy** parameter, the *value* can be from 0 to 5. The default is 0 (no redundancy). The parameter *value* sets the redundancy factor for T.38 Fax Relay.

The **hs_redundancy** parameter refers to data redundancy in the high-speed V.17, V.27, and V.29 T.4 or T.6 fax machine image data. For the **hs_redundancy** parameter, the *value* can be from 0 to 2. The default is 0 (no redundancy). The parameter *value* sets the redundancy factor for T.38 Fax Relay.

Note Setting the **hs_redundancy** parameter to a value greater than 0 causes a significant increase in the network bandwidth consumed by the fax call.

Examples

The following example configures T.38 Fax Relay for VoIP, beginning in global configuration mode:

```
dial-peer voice 99 voip
 fax protocol t38
```

Related Commands

Command	Description
fax rate	Establishes the rate at which a fax is sent to the specified dial peer.

fax protocol (voice-service)

To specify the global default fax protocol for all Voice over IP (VoIP) dial peers, use the **fax protocol** command in voice-service configuration mode. To return to the default fax protocol, use the **no** form of this command.

fax protocol {**cisco** | **t38** [**ls_redundancy** *value*] [**hs_redundancy** *value*]}

no fax protocol

Syntax Description

cisco	Cisco proprietary fax protocol.
t38	ITU-T T.38 standard fax protocol.
ls_redundancy *value*	(Optional) Low-speed redundancy for the T.38 fax protocol. The *value* can be from 0 to 5. The default is 0. The **ls_redundancy** parameter refers to data redundancy in the low-speed V.21-based T.30 fax machine protocol.
hs_redundancy *value*	(Optional) High-speed redundancy for the T.38 fax protocol. The *value* can be from 0 to 2. The default is 0. The **hs_redundancy** parameter refers to data redundancy in the high-speed V.17, V.27, and V.29 T.4 or T.6 fax machine image data.

Defaults

Cisco fax protocol

Command Modes

Voice-service configuration

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco 2600 series routers, Cisco 3600 series routers, and Cisco MC3810 multiservice concentrators.

Usage Guidelines

Use the **fax protocol t38** command to configure T.38 Fax Relay for VoIP. The **t38** keyword enables the T.38 Fax Relay protocol. The **cisco** keyword selects the original Cisco proprietary fax protocol. When the **system** keyword is selected in the dial peer, it specifies the global default fax protocol used by a dial peer, set by the **fax protocol t.38** command. The optional **ls_redundancy** and **hs_redundancy** parameters are used to send redundant T.38 fax packets.

Note The **ls_redundancy** and **hs_redundancy** parameters are applicable only to the T.38 Fax Relay protocol.

The **ls_redundancy** parameter refers to data redundancy in the low-speed V.21-based T.30 fax machine protocol. For the **ls_redundancy** parameter, the *value* can be from 0 to 5. The default is 0 (no redundancy). The parameter *value* sets the redundancy factor for T.38 Fax Relay.

The **hs_redundancy** parameter refers to data redundancy in the high-speed V.17, V.27, and V.29 T.4 or T.6 fax machine image data. For the **hs_redundancy** parameter, the *value* can be from 0 to 2. The default is 0 (no redundancy). The parameter *value* sets the redundancy factor for T.38 Fax Relay.

Note Setting the **hs_redundancy** parameter to a value greater than 0 causes a significant increase in the network bandwidth consumed by the fax call.

Examples

The following example configures the T.38 fax protocol for VoIP, beginning in global configuration mode:

```
voice service voip
 fax protocol t38
```

Related Commands

Command	Description
fax protocol (dial-peer)	Specifies the fax protocol for a specific VoIP dial peer.
fax protocol t.38	

fax rate

To establish the rate at which a fax is sent to the specified dial peer, use the **fax rate** command in dial-peer configuration mode. To reset the dial peer for voice calls, use the **no** form of this command.

fax rate {**2400** | **4800** | **7200** | **9600** | **12000** | **14400**} {**disable** | **voice**} [**bytes** *rate*]

no fax rate

Syntax Description

2400	Specifies a fax transmission speed of 2400 bits per second (bps).
4800	Specifies a fax transmission speed of 4800 bps.
7200	Specifies a fax transmission speed of 7200 bps.
9600	Specifies a fax transmission speed of 9600 bps.
12000	Specifies a fax transmission speed of 12,000 bps.
14400	Specifies a fax transmission speed of 14,400 bps.
disable	Disables Fax Relay transmission capability.
voice	Specifies the highest possible transmission speed allowed by the voice rate. For example, if the voice codec is G.711, fax transmission may occur up to 14,400 bps because 14,400 bps is less than the 64-k voice rate. If the voice codec is G.729 (8k), the fax transmission speed will be 7200 bps.
bytes *rate*	(Optional) Specifies fax packetization rate, in milliseconds. Range is 20 to 48. Default is 20. • For Cisco fax relay, this keyword-argument pair is valid only on Cisco 2600 series, Cisco 3600 series, Cisco 5300, and Cisco 7200 series routers. • For T.38 fax relay, this keyword-argument pair is valid only on Cisco 5350, Cisco 5400, and Cisco 5850 routers. For other routers, the packetization rate for T.38 fax relay is fixed at 40 ms and cannot be changed.

Defaults

Voice calls

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced as the **fax-rate** command on the Cisco 3600 series routers.
12.0(2)XH	The fax transmission rate of 12000 was added.
12.0(4)T	This command was supported on the Cisco MC3810 multiservice concentrator.
12.1(3)T	The command name changed from **fax-rate** to **fax rate** (nonhyphenated).

Release	Modification
12.1(3)XI	This command was implemented on the Cisco AS5300 universal access server.
12.1(5)T	The modifications made in Cisco IOS Release 12.1(3)XI were integrated into Cisco IOS Release 12.1(5)T.

Usage Guidelines

Use the **fax rate** command to specify the fax transmission rate to the specified dial peer.

The values for this command apply only to the fax transmission speed and do not affect the quality of the fax itself. The higher transmission speed values (14,400 bps) provide a faster transmission speed but monopolize a significantly large portion of the available bandwidth. The lower transmission speed values (2400 bps) provide a slower transmission speed and use a relatively smaller portion of the available bandwidth.

Note The fax call will not get compressed using the **ip rtp header-compression** command because User Datagram Protocol (UDP) is being used and not Real-Time Transport Protocol (RTP). For example, a 9600 bps fax call will take approximately 24 kbps.

If the fax rate transmission speed is set higher than the codec rate in the same dial peer, the data sent over the network for fax transmission will be above the bandwidth reserved for Resource Reservation Protocol (RSVP).

Tips Because a large portion of the available network bandwidth will be monopolized by the fax transmission, Cisco does not recommend setting the fax rate value higher than the value of the selected codec. If the fax rate value is set lower than the codec value, faxes will take longer to send but will use less bandwidth.

The **voice** keyword specifies the highest possible transmission speed allowed by the voice rate. For example, if the voice codec is G.711, the fax transmission may occur up to 14,400 bps because 14,400 bps is less than the 64-k voice rate. If the voice codec is G.729 (8k), the fax transmission speed will be 7200 bps.

Examples

The following example configures a fax rate transmission speed of 9600 bps for faxes sent using a dial peer:

```
dial-peer voice 100 voip
 fax rate 9600 voice
```

The following example sets a fax rate transmission speed at 12,000 bps and the packetization rate at 20 milliseconds:

```
fax rate 12000 bytes 20
```

Related Commands

Command	Description
codec (dial-peer)	Specifies the voice coder rate of speech for a dial peer.
fax protocol (dial-peer)	Specifies the fax protocol for a specific VoIP dial peer.

fax receive called-subscriber

To define the called subscriber identifier (CSI), use the **fax receive called-subscriber** command in global configuration mode. To disable the configured number, use the **no** form of this command.

fax receive called-subscriber {**d** | *string*}

no fax receive called-subscriber {**d** | *string*}

Syntax Description

d	Wildcard that specifies that the information displayed is captured from the configured destination pattern.
string	Destination telephone number. Valid entries are the plus sign (+), numbers 0 through 9, and the space character. This string can specify an E.164 telephone number; if you choose to configure an E.164 telephone number, use the plus sign as the first character.

Defaults

Enabled with a null string

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1)	This command was integrated into Cisco IOS Release 12.1(1).

Usage Guidelines

Use the **fax receive called-subscriber** command to define the number displayed in the liquid crystal display (LCD) of the sending fax device when you are sending a fax to a recipient. Typically, with a standard Group 3 fax device, this is the telephone number associated with the receiving fax device. The command defines the CSI.

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

Examples

The following example configures 555-1234 as the called-subscriber number:

```
fax receive called-subscriber 5551234
```

fax-relay ecm disable

To disable fax-relay Error Correction Mode (ECM) on the Voice over IP (VoIP) dial peer, use the **fax-relay ecm disable** command in dial-peer configuration mode. To enable ECM, use the **no** form of this command.

fax-relay ecm disable

no fax-relay ecm disable

Syntax Description

This command has no arguments or keywords.

Defaults

Fax-relay ECM is enabled.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.1(3)T	This command was introduced.

Usage Guidelines

When this command is entered, the digital signal processor (DSP) fax-relay firmware disables ECM by modifying the Digital Information Signal (DIS) T.30 message. This is performed on DIS signals in both directions so that ECM is disabled in both directions even if only one gateway is configured with ECM disabled.

This setting is provisioned when the DSP channel starts fax relay and cannot be changed during the fax relay session.

Examples

The following dial-peer configuration disables ECM on the voice dial peer:

```
fax-relay ecm disable
```

The following dial-peer configuration enables ECM on the voice dial peer:

```
no fax-relay ecm disable
```

fax send center-header

To specify the data that will appear in the center position of the fax header information, use the **fax send center-header** command in global configuration mode. To disable the selected options, use the **no** form of this command.

fax send center-header {**a** | **d** | **p** | **s** | **t** | *string*}

no fax send center-header {**a** | **d** | **p** | **s** | **t** | *string*}

Syntax Description

a	Wildcard that inserts the date in the selected position.
d	Wildcard that inserts the destination address in the selected position.
p	Wildcard that inserts the page count in the selected position.
s	Wildcard that inserts the sender address in the selected position.
t	Wildcard that inserts the transmission time in the selected position.
string	Text string providing information (in addition to any preset wildcard values) included in the text header of the fax message. Valid characters include any text string by itself or a string of text in combination with command-specific wildcards—for example, a text string displaying the sender's company name or a combinational string like Time:t.

Defaults

No center header information is displayed by default.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1)	This command was integrated into Cisco IOS Release 12.1(1).

Usage Guidelines

Mail messages that contain only text or that contain text attachments (Multipurpose Internet Mail Extensions [MIME] media-type text) can be converted by the off-ramp Cisco AS5300 into a format understood by fax machines using the Cisco AS5300 universal access server text-to-fax converter. When this conversion is performed, this **fax send center-header** command is used to indicate what header information should be added to the center top of pages.

Mail messages with Tag Image File Format (TIFF) attachments (MIME media image and subtype of TIFF) are expected to include their own per-page headers. Cisco AS5300 software does not modify TIFF attachments.

Note Because the Cisco AS5300 universal access server does not alter fax TIFF attachments, you cannot configure faxed header information for faxes being converted from TIFF files to standard fax transmissions.

This command lets you configure multiple options at once—meaning that you can combine one or more wildcards with text string information to personalize your fax header information.

Note If the information selected for the **fax send center-header** command exceeds the space allocated for the center fax header, the information is truncated.

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

Examples

The following example selects the transmission time of the fax as the center fax header information:

```
fax send center-header $t$
```

The following example configures the company name abc and its address as the center fax header information:

```
fax send center-header abc $s$
```

Related Commands

Command	Description
fax send left-header	Specifies the data that will appear on the left in the fax header information.
fax send right-header	Specifies the data that will appear on the right in the fax header information.

fax send coverpage comment

To define personalized text for the title field of a fax cover sheet, use the **fax send coverpage comment** command in global configuration mode. To disable the defined comment, use the **no** form of this command.

fax send coverpage comment *string*

no fax send coverpage comment *string*

Syntax Description

string	Text string that adds personalized text in the title field of the fax cover sheet. Valid characters are any ASCII characters.

Defaults

No coverpage comment is displayed by default.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1)	This command was integrated into Cisco IOS Release 12.1(1).

Usage Guidelines

The **fax send coverpage comment** command can be overridden by the **fax send coverpage e-mail controllable** command.

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

Examples

The following example configures an individualized title comment of ABC Fax Services for generated fax cover sheets:

```
fax send coverpage enable
fax send coverpage comment ABC Fax Services
```

Related Commands

Command	Description
fax send coverpage e-mail-controllable	Defers to the cover page setting in the e-mail header to generate a standard fax cover sheet.
fax send coverpage enable	Enables the Cisco AS5300 universal access server to generate fax cover sheets for faxes that originate from e-mail messages.
fax send coverpage show-detail	Prints all of the e-mail header information as part of the fax cover sheet.

fax send coverpage e-mail-controllable

To defer to the cover page setting in the e-mail header to generate a standard fax cover sheet, use the **fax send coverpage e-mail-controllable** command in global configuration mode. To disable standard fax sheet generation, use the **no** form of this command.

fax send coverpage e-mail-controllable

no fax send coverpage e-mail-controllable

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1)	This command was integrated into Cisco IOS Release 12.1(1).

Usage Guidelines

Also use the destination address of an e-mail message to control the cover page generation on a per-recipient basis. Use the **fax send coverpage e-mail-controllable** command to configure the router to defer to the cover page setting in the e-mail header.

In essence, the off-ramp router defers to the setting configured in the e-mail address itself. For example, if the address has a parameter set to cover=no, this parameter will override the setting for the **fax send coverpage enable** command and the off-ramp gateway will not generate and send a fax cover page. If the address has a parameter set to cover=yes, the off-ramp gateway will defer to this parameter and generate and send a fax cover page.

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

Table 18 shows examples of what the user would enter in the To: field of the e-mail message.

Table 18 Example Entries for the To: Field

To: Field Entry	Description
FAX=**+1-312-555-3260@fax.com**	Fax sent to an E.164-compliant long distance telephone number in the United States. If the **fax send coverpage enable** command has been configured, store and forward fax will generate a fax cover page.
FAX=**+1-312-555-3260/cover=no@fax.com**	Fax sent to an E.164-compliant long distance telephone number in the United States. In this example, the **fax send coverpage enable** command is superseded by the cover=no statement. No cover page will be generated.
FAX=**+1-312-555-3260/cover=yes@fax.com**	Fax sent to an E.164-compliant long distance telephone number in the United States. In this example, the **fax send coverpage enable** command is superseded by the cover=yes statement. Store and forward fax will generate a fax coverpage.

Examples

The following example enables standard generated fax cover sheets:

```
fax send coverpage enable
fax send coverpage e-mail-controllable
```

Related Commands

Command	Description
fax send coverpage comment	Defines personalized text for the title field of a fax cover sheet.
fax send coverpage enable	Enables the Cisco AS5300 universal access server to generate fax cover sheets for faxes that originate from e-mail messages.
fax send coverpage show-detail	Prints all of the e-mail header information as part of the fax cover sheet.

fax send coverpage enable

To enable the Cisco AS5300 universal access server to generate fax cover sheets for faxes that originate from e-mail messages, use the **fax send coverpage enable** command in global configuration mode. To disable the generation of fax cover sheets, use the **no** form of this command.

fax send coverpage enable

no fax send coverpage enable

Syntax Description

This command has no arguments or keywords.

Defaults

This feature is disabled by default.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1)	This command was integrated into Cisco IOS Release 12.1(1).

Usage Guidelines

This command is applicable only for faxes that originate as e-mail messages. The Cisco AS5300 does not alter fax Tag Image File Format (TIFF) attachments. Therefore, this command cannot be used to enable the AS5300 to generate fax cover pages for faxes that are being converted from TIFF files to standard fax transmissions.

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

Examples

The following example enables the Cisco AS5300 universal access server to generate fax cover sheets:

```
fax send coverpage enable
```

Related Commands

Command	Description
fax send coverpage comment	Defines personalized text for the title field of a fax cover sheet.
fax send coverpage e-mail-controllable	Defers to the cover page setting in the e-mail header to generate a standard fax cover sheet.
fax send coverpage show-detail	Prints all of the e-mail header information as part of the fax cover sheet.

fax send coverpage show-detail

To print all of the e-mail header information as part of the fax cover sheet, use the **fax send coverpage show-detail** command in global configuration mode. To disable the e-mail header information being displayed, use the **no** form of this command.

fax send coverpage show-detail

no fax send coverpage show-detail

Syntax Description

This command has no arguments or keywords.

Defaults

No coverpage details are displayed by default.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1)	This command was integrated into Cisco IOS Release 12.1(1).

Usage Guidelines

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

This command is applicable only for faxes that originate as e-mail messages. The Cisco AS5300 does not alter fax Tag Image File Format (TIFF) attachments. Therefore, this command cannot be used to enable the AS5300 to display additional fax cover page information for faxes that are being converted from TIFF files to standard fax transmissions.

Examples

The following example configures an individualized generated fax cover sheet that contains the e-mail header text:

```
fax send coverpage enable
no fax send coverpage e-mail-controllable
fax send coverpage show-detail
```

Related Commands

Command	Description
fax send coverpage comment	Defines personalized text for the title field of a fax cover sheet.
fax send coverpage e-mail-controllable	Defers to the cover page setting in the e-mail header to generate a standard fax cover sheet.
fax send coverpage enable	Enables the Cisco AS5300 universal access server to generate fax cover sheets for faxes that originate from e-mail messages.

fax send left-header

To specify the data that will appear on the left in the fax header information, use the **fax send left-header** command in global configuration mode. To disable the selected options, use the **no** form of this command.

fax send left-header {**a** | **d** | **p** | **s** | **t** | *string*}

no fax send left-header {**a** | **d** | **p** | **s** | **t** | *string*}

Syntax Description

a	Wildcard that inserts the date in the selected position.
d	Wildcard that inserts the destination address in the selected position.
p	Wildcard that inserts the page count in the selected position.
s	Wildcard that inserts the sender address in the selected position.
t	Wildcard that inserts the transmission time in the selected position.
string	Text string providing information (in addition to any preset wildcard values) included in the text header of the fax message. Valid characters include any text string by itself or a string of text in combination with command-specific wildcards—for example, a text string displaying the sender's company name or a combinational string like Time:t.

Defaults

No left header information is displayed by default.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1)	This command was integrated into Cisco IOS Release 12.1(1).

Usage Guidelines

Mail messages that contain only text or that contain text attachments (Multipurpose Internet Mail Extensions [MIME] media text) can be converted by the off-ramp Cisco AS5300 into a format understood by fax machines using the Cisco AS5300 text-to-fax converter. When this conversion is performed, this **fax send left-header** command is used to indicate what header information should be added to the left top of pages.

Mail messages with Tag Image File Format (TIFF) attachments (MIME media image and subtype of TIFF) are expected to include their own per-page headers. Cisco AS5300 software does not modify TIFF attachments.

This command lets you configure multiple options at once—meaning that you can combine one or more wildcards with text string information to personalize your fax header information.

Note If the information selected for the **fax send left-header** command exceeds the space allocated for the left fax header, the information is truncated.

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

Examples

The following example selects the transmission time of the fax as the left fax header information:

```
fax send left-header $t$
```

The following example configures the company name Widget and its address as the left fax header information:

```
fax send left-header widget $s$
```

Related Commands

Command	Description
fax send center-header	Specifies the data that will appear in the center position of the fax header information.
fax send right-header	Specifies the data that will appear on the right in the fax header information.

fax send max-speed

To specify the maximum speed at which an outbound fax will be sent, use the **fax send max-speed** command in global configuration mode. To disable the selected speed, use the **no** form of this command.

fax send max-speed {**2400** | **4800** | **7200** | **9600** | **12000** | **14400**}

no fax send max-speed {**2400** | **4800** | **7200** | **9600** | **12000** | **14400**}

Syntax Description

2400	Indicates a transmission speed of 2400 bits per second (bps).
4800	Indicates a transmission speed of 4800 bps.
7200	Indicates a transmission speed of 7200 bps.
9600	Indicates a transmission speed of 9600 bps.
12000	Indicates a transmission speed of 12,000 bps.
14400	Indicates a transmission speed of 14,400 bps.

Defaults

14400 bps

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1)	This command was integrated into Cisco IOS Release 12.1(1).

Usage Guidelines

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

Examples

The following example sets the outbound fax transmission rate at 2400 bps:

```
fax send max-speed 2400
```

fax send right-header

To specify the data that will appear on the right in the fax header information, use the **fax send right-header** command in global configuration mode. To disable the selected options, use the **no** form of this command.

fax send right-header {**a** | **d** | **p** | **s** | **t** | *string*}

no fax send right-header {**a** | **d** | **p** | **s** | **t** | *string*}

Syntax Description

a	Wildcard that inserts the date in the selected position.
d	Wildcard that inserts the destination address in the selected position.
p	Wildcard that inserts the page count in the selected position.
s	Wildcard that inserts the sender address in the selected position.
t	Wildcard that inserts the transmission time in the selected position.
string	Text string providing information (in addition to any preset wildcard values) included in the text header of the fax message. Valid characters include any text string by itself or a string of text in combination with command-specific wildcards—for example, a text string displaying the sender's company name or a combinational string like Time:t.

Defaults

No right header information is displayed by default.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1)	This command was integrated into Cisco IOS Release 12.1(1).

Usage Guidelines

Mail messages that contain only text or contain text attachments (Multipurpose Internet Mail Extensions [MIME] media text) can be converted by the off-ramp Cisco AS5300 into a format understood by fax machines using the Cisco AS5300 text-to-fax converter. When this conversion is performed, this **fax send right-header** command is used to indicate what header information should be added to right top of pages.

Mail messages with Tag Image File Format (TIFF) attachments (MIME media image and subtype of TIFF) are expected to include their own per-page headers. Cisco AS5300 software does not modify TIFF attachments.

This command lets you configure multiple options at once—meaning that you can combine one or more wildcards with text string information to personalize your fax header information.

Note If the information selected for the **fax send right-header** command exceeds the space allocated for the right fax header, the information is truncated.

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

Examples

The following example selects the date of the fax as the right fax header information:

```
fax send right-header $a$
```

The following example configures the company name Widget and its address as the right fax header information:

```
fax send right-header widget $s$
```

Related Commands

Command	Description
fax send center-header	Specifies the data that will appear in the center position of the fax header information.
fax send left-header	Specifies the data that will appear on the left in the fax header information.

fax send transmitting-subscriber

To define the transmitting subscriber identifier (TSI), use the **fax send transmitting-subscriber** command in global configuration mode. To disable the configured value, use the **no** form of this command.

fax send transmitting-subscriber {**s** | *string*}

no fax send transmitting-subscriber {**s** | *string*}

Syntax Description

s	Wildcard that inserts the sender name from the RFC 822 header (captured by the on-ramp from the sending fax machine) in the selected position.
string	Originating telephone number. Valid entries are the plus sign (+), numbers 0 through 9, and the space character. This string can specify an E.164 telephone number; if you choose to configure an E.164 telephone number, use the plus sign as the first character.

Defaults

No TSI information is displayed by default.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced on the Cisco AS5300 universal access server.
12.1(1)	This command was integrated into Cisco IOS Release 12.1(1).

Usage Guidelines

The transmitting subscriber number is the number displayed in the liquid crystal display (LCD) of the receiving fax device. Typically, with a standard Group 3 fax device, this is the telephone number associated with the transmitting or sending fax device. This command defines the TSI.

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

Examples

The following example configures the company name (as captured by the on-ramp from the sending fax machine as +18005551234) as the TSI:

```
fax send transmitting-subscriber +18005551234
```

forward-alarms

To turn on alarm forwarding so that alarms arriving on one T1/E1 port are sent to the other port on dual-mode multiflex trunk interface cards, use the **forward-alarms** command in controller configuration mode on the one port. To restore the default value so that no alarms are forwarded, use the **no** form of this command.

forward-alarms

no forward-alarms

Syntax Description

This command has no arguments or keywords.

Defaults

Alarm forwarding is disabled.

Command Modes

Controller configuration

Command History

Release	Modification
12.0(7)XR	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

When you enter this command, physical-layer alarms on the configured port are forwarded to the other port on dual-port cards, simulating a one-way repeater operation. The system forwards RAIs (remote alarm indications, Yellow Alarms), alarm indication signals (AIS, or Blue Alarms), losses of frame (LOF alarms or Red Alarms), and losses of signaling (LOS alarms or Red Alarms).

Examples

The following example shows how to turn on alarm forwarding on controller E1 0/0 of a Cisco 2600 series router:

```
controller e1 0/0
 forward-alarms
```

forward-digits

To specify which digits to forward for voice calls, use the **forward-digits** command in dial-peer configuration mode. To specify that any digits not matching the destination-pattern are not to be forwarded, use the **no** form of this command. To restore the default state, use the **default** form of this command.

forward-digits {*num-digit* | **all** | **extra**}

no forward-digits

default forward-digits

Syntax Description

num-digit	The number of digits to be forwarded. If the number of digits is greater than the length of a destination phone number, the length of the destination number is used. The valid range is from 0 to 32. Setting the value to 0 is equivalent to entering the **no forward-digits** command.
all	Forwards all digits. If **all** is entered, the full length of the destination pattern is used.
extra	If the length of the dialed digit string is greater than the length of the dial-peer destination pattern, the extra right-justified digits are forwarded. However, if the dial-peer destination pattern is variable length ending with the character “T” (for example: T, 123T, 123...T), extra digits are not forwarded.

Defaults

Dialed digits not matching the destination pattern are forwarded.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810 multiservice concentrator.
12.0(2)T	The **implicit** option was added.
12.0(4)T	This command was modified to support ISDNBF PRI QSIG signaling calls.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers, the **implicit** keyword was removed, and the **extra** keyword was added.
12.1(2)T	The modifications made in Cisco IOS Release 12.0(7)XK were integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

This command applies only to plain old telephone service (POTS) dial peers. Forwarded digits are always right justified so that extra leading digits are stripped. The destination pattern includes both explicit digits and wildcards if present. Use the default form of this command if a nondefault digit-forwarding scheme was entered previously and you wish to restore the default.

For QSIG ISDN connections, entering the **forward-digits all** command implies that all the digits of the called party number are sent to the ISDN connection. When the **forward-digits** *num-digit* command and a number from 1 to 32 are entered, the number of digits of the called party number specified (right justified) are sent to the ISDN connection.

Examples

The following example shows that all digits in the destination pattern of a POTS dial peer are forwarded:

```
dial-peer voice 1 pots
 destination-pattern 8...
 forward-digits all
```

The following example shows that four of the digits in the destination pattern of a POTS dial peer are forwarded:

```
dial-peer voice 1 pots
 destination-pattern 555....
 forward-digits 4
```

The following example shows that the extra right-justified digits that exceed the length of the destination pattern of a POTS dial peer are forwarded:

```
dial-peer voice 1 pots
 destination-pattern 555....
 forward-digits extra
```

Related Commands

Command	Description
destination-pattern	Defines the prefix or the full E.164 telephone number to be used for a dial peer.
show dial-peer voice	Displays configuration information for dial peers.

frame-relay voice bandwidth

To specify how much bandwidth should be reserved for voice traffic on a specific data-link connection identifier (DLCI), use the **frame-relay voice bandwidth** command in map-class configuration mode. To release the bandwidth previously reserved for voice traffic, use the **no** form of this command.

frame-relay voice bandwidth *bps-reserved*

no frame-relay voice bandwidth *bps-reserved*

Syntax Description

bps-reserved	The bandwidth, in bits per second (bps), reserved for voice traffic for the specified map class. The range is from 8000 to 45,000,000 bps; the default is 0, which disables voice calls.

Defaults

Disabled (zero)

Command Modes

Map-class configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco 2600 series routers, 3600 series, 7200 series, and on the MC3810 multiservice concentrator.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.0(5)T	The **queue** *depth* keyword and argument option was added.
12.2(1)	The **queue** *depth* keyword and argument option was deleted.

Usage Guidelines

To use this command, you must first associate a Frame Relay map class with a specific DLCI and then enter map-class configuration mode and set the amount of bandwidth to be reserved for voice traffic for that map class.

If a call is attempted and there is not enough remaining bandwidth reserved for voice to handle the additional call, the call will be rejected. For example, if 64 kbps is reserved for voice traffic and a codec and payload size is being used that requires 10 kbps of bandwidth for each call, the first six calls attempted will be accepted, but the seventh call will be rejected.

Reserve queues are not required for Voice over Frame Relay (VoFR).

Note Cisco strongly recommends that you set voice bandwidth to a value less than the committed information rate (CIR) if Frame Relay traffic shaping is configured. Cisco also strongly recommends that you set the minimum CIR (using the **frame-relay mincir** command) to be at least equal to or greater than the voice bandwidth.

Calculating Required Bandwidth

The bandwidth required for a voice call depends on the bandwidth of the codec, the voice packetization overhead, and the voice frame payload size. The smaller the voice frame payload size, the higher the bandwidth required for the call. To make the calculation, use the following formula:

required_bandwidth = codec_bandwidth x (1 + overhead / payload_size)

As an example, the overhead for a VoFR voice packet is between 6 and 8 bytes: a 2-byte Frame Relay header, a 1- or 2-byte FRF.11 header (depending on the CID value), a 2-byte cyclic redundancy check (CRC), and a 1-byte trailing flag. If voice sequence numbers are enabled in the voice packets, there is an additional 1-byte sequence number. Table 19 shows the required voice bandwidth for the G.729 8000-bps speech coder for various payload sizes.

Table 19 Required Voice Bandwidth Calculations for G.729

Codec	Codec Bandwidth	Voice Frame Payload Size	Required Bandwidth per Call (6-Byte OH)	Required Bandwidth per Call (8-Byte OH)
G.729	8000 bps	120 bytes	8400 bps	8534 bps
G.729	8000 bps	80 bytes	8600 bps	8800 bps
G.729	8000 bps	40 bytes	9200 bps	9600 bps
G.729	8000 bps	30 bytes	9600 bps	10134 bps
G.729	8000 bps	20 bytes	10400 bps	11200 bps

To configure the payload size for the voice frames, use the **codec** command from dial-peer configuration mode.

Examples

The following example shows how to reserve 64 kbps for voice traffic for the “vofr” Frame Relay map class on a Cisco 2600 series routers, 3600 series, or 7200 series router or on an MC3810 multiservice concentrator:

```
interface serial 1/1
 frame-relay interface-dlci 100
  class vofr
  exit
map-class frame-relay vofr
 frame-relay voice bandwidth 64000
```

Related Commands

Command	Description
codec (dial-peer)	Specifies the voice coder rate of speech for a VoFR dial peer.
frame-relay fair-queue	Enables weighted fair queueing for one or more Frame Relay PVCs.
frame-relay fragment	Enables fragmentation for a Frame Relay map class.
frame-relay interface-dlci	Assigns a DLCI to a specified Frame Relay subinterface on the router or access server.
frame-relay mincir	Assigns the minimum CIR for Frame Relay traffic shaping.
map-class frame-relay	Specifies a map class to define QoS values for an SVC.

frag-pre-queuing

This command was removed in Cisco IOS Release 12.1(2)T and is no longer supported in this release.

freq-max-delay

This command is not supported in Cisco IOS Release 12.2. This command was added in Cisco IOS Release 12.2(2)T. For information about this command, refer to the *Cisco IOS Voice, Video, and Fax Command Reference, Release 12.2 T*, at the following URL:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122newft/122tcr/122tvr/index.htm

freq-max-deviation

This command is not supported in Cisco IOS Release 12.2. This command was added in Cisco IOS Release 12.2(2)T. For information about this command, refer to the *Cisco IOS Voice, Video, and Fax Command Reference, Release 12.2 T*, at the following URL:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122newft/122tcr/122tvr/index.htm

freq-max-power

This command is not supported in Cisco IOS Release 12.2. This command was added in Cisco IOS Release 12.2(2)T. For information about this command, refer to the *Cisco IOS Voice, Video, and Fax Command Reference, Release 12.2 T*, at the following URL:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122newft/122tcr/122tvr/index.htm

freq-min-power

This command is not supported in Cisco IOS Release 12.2. This command was added in Cisco IOS Release 12.2(2)T. For information about this command, refer to the *Cisco IOS Voice, Video, and Fax Command Reference, Release 12.2 T*, at the following URL:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122newft/122tcr/122tvr/index.htm

freq-pair

To specify the frequency components of a tone to be detected, use the **freq-pair** command in voice-class configuration mode. To cancel detection of a tone, use the **no** form of this command.

freq-pair *tone-id frequency-1 frequency-2*

no freq-pair *tone-id*

Syntax Description

tone-id	A tag identifier for a tone to be detected. The range is from 1 to 16. There is no default.
frequency-1	One frequency component of the tone to be detected, in Hz. The range is from 300 to 3600. There is no default.
frequency-2	A second frequency component of the tone to be detected, in Hz. The range is from 300 to 3600, or you can specify 0. There is no default.

Defaults

No tone is specified for detection.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco 2600 and 3600 series routers and on the MC3810 multiservice concentrator.

Usage Guidelines

To detect a tone with two frequency components (a dualtone), configure frequencies for *frequency-1* and *frequency-2*.

To detect a tone with only one frequency component, configure a frequency for *frequency-1* and enter 0 for *frequency-2*.

You can configure a router to detect up to 16 tones.

Examples

The following example configures tone number 1 (tone-id 1) with frequency components of 480 Hz and 2400 Hz:

```
voice class dualtone 100
 freq-pair 1 480 2400
 exit
```

The following example configures tone number 1 (tone-id 1) with frequency components of 480 Hz and 2400 Hz and tone number 2 (tone-id 2) with frequency components of 560 Hz and 880 Hz:

```
voice class dualtone 50
 freq-pair 1 480 2400
 freq-pair 2 560 880
 exit
```

Related Commands

Command	Description
voice class dualtone	Creates a voice class for FXO tone detection parameters.

freq-power-twist

This command is not supported in Cisco IOS Release 12.2. This command was added in Cisco IOS Release 12.2(2)T. For information about this command, refer to the *Cisco IOS Voice, Video, and Fax Command Reference, Release 12.2 T*, at the following URL:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122newft/122tcr/122tvr/index.htm

ftc-trunk frame-relay-dlci

This command was removed in Cisco IOS Release 12.1(2)T and is no longer supported in this release.

ftc-trunk management-dlci

This command was removed in Cisco IOS Release 12.1(2)T and is no longer supported in this release.

ftc-trunk management-protocol

This command was removed in Cisco IOS Release 12.1(2)T and is no longer supported in this release.