# clid

To preauthenticate calls on the basis of the Calling Line Identification (CLID) number, use the **clid** command in AAA preauthentication configuration mode. To remove the **clid** command from your configuration, use the **no** form of this command.

**clid** [**if-avail** | **required**] [**accept-stop**] [**password** *password*]

**no clid** [**if-avail** | **required**] [**accept-stop**] [**password** *password*]

## Syntax Description

| | |
|---|---|
| **if-avail** | (Optional) Implies that if the switch provides the data, RADIUS must be reachable and must accept the string in order for preauthentication to pass. If the switch does not provide the data, preauthentication passes. |
| **required** | (Optional) Implies that the switch must provide the associated data, that RADIUS must be reachable, and that RADIUS must accept the string in order for preauthentication to pass. If these three conditions are not met, preauthentication fails. |
| **accept-stop** | (Optional) Prevents subsequent preauthentication elements such as **ctype** or **dnis** from being tried once preauthentication has succeeded for a call element. |
| **password** *password* | (Optional) Defines the password for the preauthentication element. |

## Command Default

The **if-avail** and **required** keywords are mutually exclusive. If the **if-avail** keyword is not configured, the preauthentication setting defaults to **required**.

The default password string is cisco.

## Command Modes

AAA preauthentication configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(2)T | This command was introduced. |

## Usage Guidelines

You may configure more than one of the authentication, authorization and accounting (AAA) preauthentication commands (**clid**, **ctype**, **dnis**) to set conditions for preauthentication. The sequence of the command configuration decides the sequence of the preauthentication conditions. For example, if you configure **dnis**, then **clid**, then **ctype**, in this order, then this is the order of the conditions considered in the preauthentication process.

In addition to using the preauthentication commands to configure preauthentication on the Cisco router, you must set up the preauthentication profiles on the RADIUS server.

## Examples

The following example specifies that incoming calls be preauthenticated on the basis of the CLID number:

```
aaa preauth
 group radius
 clid required
```

## Related Commands

| Command | Description |
|---|---|
| **ctype** | Preauthenticates calls on the basis of the call type. |
| **dnis (RADIUS)** | Preauthenticates calls on the basis of the DNIS number. |
| **dnis bypass (AAA preauthentication configuration)** | Specifies a group of DNIS numbers that will be bypassed for preauthentication. |
| **group (RADIUS)** | Specifies the AAA RADIUS server group to use for preauthentication. |

# clid (dial peer)

To control the presentation and use of calling-line ID (CLID) information, use the **clid** command in dial peer configuration mode. To remove CLID controls, use the **no** form of this command.

**clid** {**network-number** *number* [**second-number** strip] | **network-provided** | **override rdnis** | **restrict** | **strip** [**name** | **pi-restrict** [**all**]] | **substitute name**}

**no clid** {**network-number** *number* [**second-number** strip] | **network-provided** | **override rdnis** | **restrict** | **strip** [**name** | **pi-restrict** [**all**]] | **substitute name**}

## Syntax Description

| | |
|---|---|
| **network-number** *number* | Network number. Establishes the calling-party network number in the CLID for this router. |
| **network-provided** | Allows you to set the screening indicator to reflect the number that was provided by the network. |
| **override rdnis** | Supported for POTS dial peers only Overrides the CLID with the redirected dialed number identification service (RDNIS) if available. |
| **pi-restrict** | Restricted progress indicator (PI). Causes removal of the calling-party number from the CLID when the PI is restricted. |
| **restrict** | Restricts presentation of the caller ID in the CLID. |
| **second-number strip** | (Optional) Removes a previously configured second network number from the CLID. |
| **strip** | Strips the calling-party number from the CLID.<br>• **name**—(Optional) Calling-party name. Causes removal of the calling-party name from the CLID.<br>• **pi-restrict** [**all**]—(Optional) Restricted PI. Causes removal of all calling-party names and numbers from the CLID when the PI is restricted. |
| **substitute name** | Copies the calling number into the display name if PI allows it (and the calling name is empty). |

## Command Default

No default behavior or values

## Command Modes

Dial Peer configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(11)T | This command was introduced. |
| 12.2(13)T | The **override rdnis** keywords were added. |
| 12.4(4)T | The following keywords were added: **network-provided**, **pi-restrict all**, and **substitute name**. |

## Usage Guidelines

The **override rdnis** keywords are supported only for POTS dial peers.

CLID is the collection of information about the billing telephone number from which a call originated. The CLID value might be the entire phone number, the area code, or the area code plus the local exchange. It is also known as caller ID. The various keywords to this command manage the presentation, restriction, or stripping of the various CLID elements.

The **clid network-number** command sets the presentation indicator to "y" and the screening indicator to "network-provided." The **second-number strip** keyword strips from the H.225 source-address field the original calling-party number, and is valid only if a network number was previously configured.

The **clid override rdnis** command overrides the CLID with the RDNIS if it is available.

The **clid restrict** command causes the calling-party number to be present in the information element, but the presentation indicator is set to "n" to prevent its presentation to the called party.

The **clid strip** command causes the calling-party number to be null in the information element, and the presentation indicator is set to "n" to prevent its presentation to the called party.

## Examples

The following example sets the calling-party network number to 98765 for POTS dial peer 4321:

```
Router(config)# dial-peer voice 4321 pots
Router(config-dial-peer)# clid network-number 98765
```

An alternative method of accomplishing this result is to enter the **second-number strip** keywords as part of the **clid network-number** command. The following example sets the calling-party network number to 56789 for VoIP dial peer 1234 and also prevents the second network number from being sent:

```
Router(config)# dial-peer voice 1234 voip
Router(config-dial-peer)# clid network-number 56789 second-number strip
```

The following example overrides the calling-party number with RDNIS if available:

```
Router(config-dial-peer)# clid override rdnis
```

The following example prevents the calling-party number from being presented:

```
Router(config-dial-peer)# clid restrict
```

The following example removes the calling-party number from the CLID information and prevents the calling-party number from being presented:

```
Router(config-dial-peer)# clid strip
```

The following example strips the name from the CLID information and prevents the name from being presented:

```
Router(config-dial-peer)# clid strip name
```

The following example strips the calling party number when PI is set to restrict clid strip from the CLID information and prevents the calling party number from being presented:

```
Router(config-dial-peer)# clid strip pi-restrict
```

The following example strips calling party name and number when the PI is set to the restrict all clid strip from the CLID information and prevents the calling party name and number from being presented:

```
Router(config-dial-peer)# clid strip pi-restrict all
```

The following example substitutes the calling party number into the display name:

```
Router(config-dial-peer)# clid substitute name
```

The following example allows you to set the screening indicator to reflect that the number was provided by the network:

```
Router(config-dial-peer)# clid network-provided
```

## Related Commands

| Command | Description |
|---|---|
| **clid (voice-service-voip)** | Passes the network provided ISDN numbers in an ISDN calling party information element screening indicator field, removes the calling party name and number from the calling-line identifier in voice service voip configuration mode, or allows a presentation of the calling number by substituting for the missing Display Name field in the Remote-Party-ID and From headers. |

# clid (voice-service-voip)

To pass the network-provided ISDN numbers in an ISDN calling party information element screening indicator field, and remove the calling party name and number from the calling-line identifier in voice service voip configuration mode, or allow a presentation of the calling number by substituting for the missing Display Name field in the Remote-Party-ID and From headers use the **clid** command in voice service voip configuration mode. To return to the default configuration, use the **no** form of this command.

**clid** {**network-provided** | **strip pi-restrict all** | **substitute name**}

**no clid** {**network-provided** | **strip pi-restrict all** | **substitute name**}

## Syntax Description

| | |
|---|---|
| **network-provided** | Sets the screen indicator as network-provided. |
| **strip pi-restrict all** | Removes the CLID when the progress indicator (PI) is restricted for PSTN to SIP operations and removes the calling party name and number when the PI is restricted for PSTN to SIP operations. |
| **substitute name** | Copies the calling number to the display name if unavailable for PSTN to SIP operations. |

## Command Default

The **clid** command passes along user-provided ISDN numbers in an ISDN calling party information element screening indicator field.

## Command Modes

Voice-service-VoIP configuration

## Command History

| Release | Modification |
|---|---|
| 12.4(4)T | This command was introduced. |

## Usage Guidelines

Use the **clid network-provided** keyword to pass along network-provided ISDN numbers in an ISDN calling party information element screening indicator field.

Use the **clid strip pi-restrict all** keyword to remove the Calling Party Name and Calling Party Number from the CLID.

Use the **clid substitute name** keyword to allow a presentation of the Display Name field in the Remote-Party-ID and From headers. The Calling Number is substituted for the Display Name field.

## Examples

The following example passes along network-provided ISDN numbers in an ISDN calling party information element screening indicator field:

```
Router(conf-voi-serv)# clid network-provided
```

The following example passes along user-provided ISDN numbers in an ISDN calling party information element screening indicator field:

```
Router(conf-voi-serv)# no clid network-provided
```

The following example removes the calling party name and number from the calling-line identifier (CLID):

```
Router(conf-voi-serv)# clid strip pi-restrict all
```

The following example does not remove the calling party name and number from the CLID:

```
Router(conf-voi-serv)# no clid strip pi-restrict all
```

The following example allows the presentation of the calling number to be substituted for the missing Display Name field in the Remote-Party-ID and From headers:

```
Router(conf-voi-serv)# clid substitute name
```

The following example disallows the presentation of the calling number to be substituted for the missing Display Name field in the Remote-Party-ID and From headers:

```
Router(conf-voi-serv)# no clid substitute name
```

## Related Commands

| Command | Description |
|---|---|
| **clid (dial-peer)** | Controls the presentation and use of CLID information in dial peer configuration mode. |

# clid strip

To remove the calling-party number from calling-line-ID (CLID) information and to prevent the calling-party number from being presented to the called party, use the **clid strip** command in dial peer configuration mode. To remove the restriction, use the **no** form of this command.

**clid strip** [**name**]

**no clid strip** [**name**]

## Syntax Description

| | |
|---|---|
| **name** | (Optional) Removes the calling-party name for both incoming and outgoing calls. |

## Command Default

Calling-party number and name are included in the CLID information.

## Command Modes

Dial peer configuration

## Command History

| Cisco IOS Release | Cisco CME Version | Modification |
|---|---|---|
| 12.2(11)T | 2.01 | This command was introduced. |
| 12.2(15)ZJ1 | 3.0 | The **name** keyword was added. |
| 12.3(4)T | 3.0 | This command was integrated into Cisco IOS Release 12.3(4)T. |

## Usage Guidelines

If the **clid strip** command is issued, the calling-party number is null in the information element, and the presentation indicator is set to "n" to prevent the presentation of the number to the called party.

If you want to remove both the number and the name, you must issue the command twice, once with the **name** keyword.

## Examples

The following example removes the calling-party number from the CLID information and prevents the calling-party number from being presented:

```
Router(config-dial-peer)# clid strip
```

The following example removes both the calling-party number and the calling-party name from the caller-ID display:

```
Router(config-dial-peer)# clid strip
Router(config-dial-peer)# clid strip name
```

## Related Commands

| Command | Description |
|---|---|
| **clid network-number** | Configures a network number in the router for CLID and uses it as the calling-party number. |
| **clid restrict** | Prevents the calling-party number from being presented by CLID. |
| **clid second-number strip** | Prevents the second network number from being sent in the CLID information. |

# clid strip reason

To remove the calling-line ID (CLID) reason code and to prevent it from being displayed on the phone, use the **clid strip reason** command in dial peer voice configuration mode. To disable the configuration, use the **no** form of this command.

**clid strip reason**

**no clid strip reason**

## Syntax Description

This command has no arguments or keywords.

## Command Default

The CLID reason code is not removed.

## Command Modes

Dial peer voice configuration (config-dial-peer)

## Command History

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

## Usage Guidelines

When the **caller-id enable** command is enabled on the gateway so that the gateway forwards information depending on the preference of the caller, client layer interface port (CLIP), or calling line identification restriction (CLIR), an "unavailable" message is displayed on the terminating phone. An "unavailable" message is a standard message that indicates the reason for the absence of calling party name.

You can use the **clid strip reason** command to remove the message and have only the call parameters forwarded.

## Examples

The following example shows how to remove the CLID reason code:

```
Router# configure terminal
Router(config)# dial-peer voice 88 voip
Router(config-dial-peer)# clid strip reason
```

## Related Commands

| Command | Description |
|---|---|
| **caller-id enable** | Allows the sending or receiving of caller-ID information. |
| **clid strip** | Removes the calling-party number from CLID information and prevents the calling-party number from being presented to the called party. |
| **dial-peer voice** | Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial peer configuration mode. |

# clock-rate (codec-profile)

To set the clock rate, in Hz, for the codec, use the **clock-rate** command in codec-profile configuration mode. To return to the default value, use the **no** form of this command.

**clock-rate** *clock-rate*

**no clock-rate**

## Syntax Description

| | |
|---|---|
| *clock-rate* | Number in the range of 1 to 1000000. |

## Command Default

The default clock rate is 0.

## Command Modes

Codec-profile configuration (config-codec-profile)

## Command History

| Release | Modification |
|---|---|
| 12.4(22)T | This command was introduced. |

## Usage Guidelines

The clock-rate must be set to 90000 for H.263/H.264.

## Examples

The following example shows:

```
codec profile 116 h263
   clock-rate 500000
   fmtp "fmtp "fmtp:120 SQCIF=1;QCIF=1;CIF=1;CIF4=2;MAXBR=3840;I=1""
!
```

## Related Commands

| Command | Description |
|---|---|
| **codec profile** | Defines video capabilities needed for video endpoints. |

# clock-select

To establish the sources and priorities of the requisite clocking signals for the OC-3/STM-1 ATM Circuit Emulation Service network module, use the **clock-select** command in CES configuration mode.

**clock-select** *priority-number interface slot/port*

## Syntax Description

| | |
|---|---|
| *priority-number* | Priority of the clock source. Range is from 1 (high priority) to 4 (low priority). There is no default value. |
| *interface* | Specifies the interface to supply the clock source. |
| *slot/port* | Backplane slot number and port number on the interface. |

## Command Default

No default behavior or values

## Command Modes

CES configuration

## Command History

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

## Usage Guidelines

This command is used on Cisco 3600 series routers that have OC-3/STM-1 ATM CES network modules.

To support synchronous or synchronous residual time stamp (SRTS) clocking modes, you must specify a primary reference source to synchronize the flow of constant bit rate (CBR) data from its source to its destination.

You can specify up to four clock priorities. The highest priority active interface in the router supplies primary reference source to all other interfaces that require network clock synchronization services. The fifth priority is the local oscillator on the network module.

Use the **show ces clock-select** command to display the currently configured clock priorities on the router.

## Examples

The following example defines two clock priorities on the router:

```
clock-select 1 cbr 2/0
clock-select 2 atm 2/0
```

## Related Commands

| Command | Description |
|---|---|
| **channel-group** | Configures the timing recovery clock for the CES interface. |
| **clock source** | Configures a transmit clock source for the CES interface. |
| **show ces clock** | Displays which ports are designated as network clock sources. |

# codec (dial peer)

To specify the voice coder rate of speech for a dial peer, use the **codec** command in dial peer voice configuration mode. To reset command settings to the default value, use the **no** form of this command.

### Cisco 1750 and Cisco 1751 Modular Access Routers, Cisco AS5300 and AS5800 Universal Access Servers, and Cisco MC3810 Multiservice Concentrators

**codec** *codec* [**bytes** *payload-size*] [**fixed-bytes**] [**mode** {**independent** | **adaptive**} [**bit-rate** *value*] [**framesize** {**30** | **60**} [**fixed**]]

**no codec** *codec* [**bytes** *payload-size*] [**fixed-bytes**] [**mode** {**independent** | **adaptive**} [**bit-rate** *value*] [**framesize** {**30** | **60**} [**fixed**]]

### Cisco 2600, 3600, 7200, and 7500 Series Routers

**codec** {*codec* [**bytes** *payload-size*] | **transparent**} [**fixed-bytes**] [**mode** {**independent** | **adaptive**} [**bit-rate** *value*] [**framesize** {**30** | **60**} [**fixed**]]

**no codec** {*codec* [**bytes** *payload-size*] | **transparent**} [**fixed-bytes**] [**mode** {**independent** | **adaptive**} [**bit-rate** *value*] [**framesize** {**30** | **60**} [**fixed**]]

## Syntax Description

| | |
|---|---|
| *codec* | Specifies the voice coder rate for speech. Codec options available for various platforms are described in Table 11. |
| **bytes** | (Optional) Precedes the argument that specifies the number of bytes in the voice payload of each frame. |
| *payload-size* | (Optional) Number of bytes in the voice payload of each frame. See Table 12 for valid entries and default values. |
| **transparent** | Enables codec capabilities to be passed transparently between endpoints in a Cisco Unified Border Element.<br>**Note** The **transparent** keyword is available only on the Cisco 2600, 3600, 7200, and 7500 series router platforms. |
| **fixed-bytes** | (Optional) Indicates that the codec byte size is fixed and non-negotiable. |
| **mode** | (Optional) For iSAC codec only. Specifies the iSAC operating frame mode that is encapsulated in each packet. |
| **independent** \| **adaptive** | (Optional) For iSAC codec only. Determines whether configuration mode (VBR) is **independent** (value 1) or **adaptive** (value 0). |
| **bit rate** *value* | (Optional) For iSAC codec only. Configures the target bit rate. The range is 10 to 32 kbps. |
| **frame-size** | (Optional) For iSAC codec only. Specifies the operating frame in milliseconds (ms). Valid entries are:<br>• **30**—30-ms frames<br>• **60**—60-ms frames<br>• **fixed**—This keyword is applicable only for adaptive mode. |

## Command Default

g729r8, 30-byte payload for VoFR and VoATM.
g729r8, 20-byte payload for VoIP.
See Table 12 for valid entries and default values for codecs.

## Command Modes

Dial peer configuration (config-dialpeer)

## Command History

| Release | Modification |
|---|---|
| 11.3(1)T | This command was introduced on the Cisco 3600 series. |
| 11.3(3)T | This command was implemented on the Cisco 2600 series. |
| 12.0(3)T | This command was implemented on the Cisco AS5300. This release does not support the **clear-channel** keyword. |
| 12.0(4)T | This command was implemented on the Cisco 3600 series, Cisco 7200 series, and Cisco MC3810, and the command was modified for VoFR dial peers. |
| 12.0(5)XE | Additional *codec* choices and other options were implemented. |
| 12.0(5)XK | The **g729br8** and **pre-ietf** codec choices were added for the Cisco 2600 and Cisco 3600 series. |
| 12.0(7)T | This command was integrated into Cisco IOS Release 12.0.(7)T and implemented on the Cisco AS5800. Voice coder rates of speech were added. This release does not support the **clear-channel** keyword were added on this platform. |
| 12.0(7)XK | **g729abr8** and **g729ar8** codec choices were for the Cisco MC3810, and the keyword **pre-ietf** was deleted. |
| 12.1(1)T | This command was integrated in Cisco IOS Release 12.1(1)T. |
| 12.1(5)T | **gsmefr** and **gsmfr** codec keywords were added. |
| 12.2(8)T | The command was implemented on the Cisco 1750 and Cisco 1751. |
| 12.2(13)T3 | The **transparent** keyword was added for use with H.323 to H.323 connections. This keyword is available only in js2 images. |
| 12.4(11)XJ2 | **gsmefr** and **gsmfr** keywords were removed as configurable codec options for all platforms with the exception of the **gsmfr** codec on the Cisco AS5400 and AS5350 with MSAv6 DSPs. The **transparent** keyword now supports H.323 to SIP connections. |
| 12.4(15)T | This command was integrated into Cisco IOS Release 12.4(15)T. |
| 12.4(15)XY | The **g722-64** keyword was added. |
| 12.4(20)T | This command was integrated into Cisco IOS Release 12.4(20)T. |
| 15.0(1)M | The **fixed-bytes** keyword was added. |
| 15.1(1)T | This command was modified. The **isac** keyword was added as a codec type, and the **mode**, **independent**, **adaptive**, **bit rate**, and **fixed** keywords were added as configurable parameters. |

## Usage Guidelines

Use this command to define a specific voice coder rate of speech and payload size for a VoIP or VoFR dial peer. This command is also used for VoATM.

*Table 11 Codec Support by Platform*

| Codec | Cisco 1750 and Cisco 1751 Modular Access Routers | Cisco 2600 and 3600 Series Routers and Cisco 7200 and 7500 Series Routers | Cisco AS5300 and AS5800 Universal Access Servers | Cisco MC3810 Multiservice Concentrators |
|---|---|---|---|---|
| **clear-channel**—Clear channel at 64,000 bits per second (bps) | Yes | Yes | — | Yes |
| **g711alaw**—G.711 A-Law at 64,000 bps | Yes | Yes | Yes | Yes |
| **g711ulaw**—G.711 mu-Law at 64,000 bps | Yes | Yes | Yes | Yes |
| **g722-64**—G.722-64 at 64,000 bps | Yes | Yes | Yes | — |
| **g723ar53**—G.723.1 Annex A at 5300 bps | — | Yes | Yes | Yes |
| **g723ar63**—G.723.1 Annex A at 6300 bps | — | Yes | Yes | Yes |
| **g723r53**—G.723.1 at 5300 bps | — | Yes | Yes | Yes |
| **g723r63**—G.723.1 at 6300 bps | — | Yes | Yes | Yes |
| **g726r16**—G.726 at 16,000 bps | Yes | Yes | Yes | Yes |
| **g726r24**—G.726 at 24,000 bps | Yes | Yes | Yes | Yes |
| **g726r32**—G.726 at 32,000 bps | Yes | Yes | Yes | Yes |
| **g726r53**—G.726 at 53,000 bps | Yes | Yes | Yes | — |
| **g726r63**—G.726 at 63,000 bps | Yes | Yes | Yes | — |
| **g728**—G.728 at 16,000 bps | — | Yes | Yes | Yes |
| **g729abr8**—G.729 Annex A and B at 8000 bps | Yes | Yes | Yes | Yes |
| **g729ar8**—G.729 Annex A at 8000 bps | Yes | Yes | Yes | Yes |
| **g729br8**—G.729 Annex B at 8000 bps | Yes | Yes | Yes | Yes |
| **g729r8**—G.729 at 8000 bps. This is the default codec. | Yes | Yes | Yes | Yes |
| **isac**—Cisco internet Speech Audio Codec (iSAC) codec. | Yes | Yes | Yes | Yes |

A specific codec type can be configured on the dial peer as long as the codec is supported by the setting used with the **codec complexity** voice-card configuration command. The **codec complexity** command is voice-card specific and platform specific. The **codec complexity** voice-card configuration command is set to either high or medium.

If the **codec complexity** command is set to high, the following keywords are available: **g711alaw**, **g711ulaw**, **g722-64**, **g723ar53**, **g723ar63**, **g723r53**, **g723r63**, **g726r16**, **g726r24**, **g726r32**, **g728**, **g729r8**, and **g729br8**.

If the **codec complexity** command is set to medium, the following keywords are available: **g711alaw**, **g711ulaw**, **g726r16**, **g726r24**, **g726r32**, **g729r8**, and **g729br8**.

The **codec** dial peer configuration command is particularly useful when you must change to a small-bandwidth codec. Large-bandwidth codecs, such as G.711, do not fit in a small-bandwidth link. However, the g711alaw and g711ulaw codecs provide higher quality voice transmission than other codecs. The g729r8 codec provides near-toll quality with considerable bandwidth savings.

The **transparent** keyword is available with H.323 to H.323 call connections beginning in Cisco IOS Release 12.2(13)T3. Support for the keyword in H.32 to SIP call connections begins in Cisco IOS Release 12.4(11)XJ2.

If codec values for the dial peers of a connection do not match, the call fails.

You can change the payload of each VoIP frame by using the **bytes** keyword; you can change the payload of each VoFR frame by using the **bytes** keyword with the *payload-size* argument. However, increasing the payload size can add processing delay for each voice packet.

Table 12 describes the voice payload options and default values for the codecs and packet voice protocols.

***Table 12 Voice Payload-per-Frame Options and Defaults***

| Codec | Protocol | Voice Payload Options (in Bytes) | Default Voice Payload (in Bytes) |
|---|---|---|---|
| **g711alaw**<br>**g711ulaw** | VoIP<br>VoFR<br>VoATM | 80, 160<br>40 to 240 in multiples of 40<br>40 to 240 in multiples of 40 | 160<br>240<br>240 |
| **g722-64** | VoIP | 80, 160, 240 | 160 |
| **g723ar53**<br>**g723r53** | VoIP<br>VoFR<br>VoATM | 20 to 220 in multiples of 20<br>20 to 240 in multiples of 20<br>20 to 240 in multiples of 20 | 20<br>20<br>20 |
| **g723ar63**<br>**g723r63** | VoIP<br>VoFR<br>VoATM | 24 to 216 in multiples of 24<br>24 to 240 in multiples of 24<br>24 to 240 in multiples of 24 | 24<br>24<br>24 |
| **g726r16** | VoIP<br>VoFR<br>VoATM | 20 to 220 in multiples of 20<br>10 to 240 in multiples of 10<br>10 to 240 in multiples of 10 | 40<br>60<br>60 |
| **g726r24** | VoIP<br>VoFR<br>VoATM | 30 to 210 in multiples of 30<br>15 to 240 in multiples of 15<br>30 to 240 in multiples of 15 | 60<br>90<br>90 |
| **g726r32** | VoIP<br>VoFR<br>VoATM | 40 to 200 in multiples of 40<br>20 to 240 in multiples of 20<br>40 to 240 in multiples of 20 | 80<br>120<br>120 |
| **g728** | VoIP<br>VoFR<br>VoATM | 10 to 230 in multiples of 10<br>10 to 240 in multiples of 10<br>10 to 240 in multiples of 10 | 40<br>60<br>60 |
| **g729abr8**<br>**g729ar8**<br>**g729br8**<br>**g729r8** | VoIP<br>VoFR<br>VoATM | 10 to 230 in multiples of 10<br>10 to 240 in multiples of 10<br>10 to 240 in multiples of 10 | 20<br>30<br>30 |
| **isac** | VoIP | 10 to 230 in multiples of 10 | 30<br>60 |

**Note** If you are configuring G.729r8 or G.723 as the *codec-type*, the maximum value for the *payload-size* argument is 60 bytes.

For toll quality, use the **g711alaw** or **g711ulaw** keyword. These values provide high-quality voice transmission but use a significant amount of bandwidth. For nearly toll quality (and a significant savings in bandwidth), use the **g729r8** keyword.

**Note** The G.723 and G.728 codecs are not supported on the Cisco 1700 platform for Cisco Hoot and Holler applications.

**Note** The **clear-channel** keyword is not supported on the Cisco AS5300.

**Note** The G.722-64 codec is  supported only for H.323 and SIP.

## Examples

The following example shows how to configure a voice coder rate that provides toll quality voice with a payload of 120 bytes per voice frame on a router that acts as a terminating node. The sample configuration begins in global configuration mode and is for VoFR dial peer 200.

```
dial-peer voice 200 vofr
 codec g711ulaw bytes 240
```

The following example shows how to configure a voice coder rate for VoIP dial peer 10 that provides toll quality but uses a relatively high amount of bandwidth:

```
dial-peer voice 10 voip
 codec g711alaw
```

The following example shows how to configure the transparent codec used by the Cisco Unified Border Element:

```
dial-peer voice 1 voip
 incoming called-number .T
 destination-pattern .T
 session target ras
 codec transparent
```

## Related Commands

| Command | Description |
|---|---|
| **codec (DSP interface dsp farm)** | Specifies call density and codec complexity. |
| **codec (voice port)** | Specifies voice compression. |
| **codec complexity** | Specifies call density and codec complexity based on the codec used. |
| **show dial peer voice** | Displays the codec setting for dial peers. |

# codec (dsp)

To specify call density and codec complexity based on a particular codec standard, use the **codec** command in DSP interface DSP farm configuration mode. To reset the card type to the default, use the **no** form of the command.

**codec** {**high** | **med**}

**no codec** {**high** | **med**}

## Syntax Description

| | |
|---|---|
| *high* | Specifies high complexity: two channels of any mix of codec. |
| *med* | Specifies medium complexity: four channels of g711/g726/g729a/fax. |

## Command Default

Medium complexity

## Command Modes

DSP interface DSP farm

## Command History

| Release | Modification |
|---|---|
| 12.0(5)XE | This command was introduced on the Cisco 7200 series. |
| 12.1(1)T | This command was integrated into Cisco Release 12.1(1)T. |
| 12.1(3)T | This command was implemented on the Cisco 7500 series. |

## Usage Guidelines

This command is supported on only the Cisco 7200 series and Cisco 7500 series routers.

Codec complexity refers to the amount of processing required to perform compression. Codec complexity affects the number of calls, referred to as call density, that can take place on the DSPfarm interfaces. The greater the codec complexity, the fewer the calls that are handled. For example, G.711 requires less DSP processing than G.728, so as long as the bandwidth is available, more calls can be handled simultaneously by using the G.711 standard than by using G.728.

The DSPinterface dspfarm **codec** complexity setting affects the options available for the **codec** dial peer configuration command.

To change codec complexity, you must first remove any configured channel associated signaling (CAS) or DS0 groups and then reinstate them after the change.

**Note** On the Cisco 2600 series routers, and 3600 series codec complexity is configured using the **codec complexity** command in voice-card configuration mode.

## Examples

The following example configures the DSPfarm interface 1/0 on the Cisco 7200 series routers to support high compression:

```
dspint DSPFarm 1/0
 codec high
```

## Related Commands

| Command | Description |
|---|---|
| **codec (dial peer)** | Specifies the voice codec rate of speech for a dial peer. |
| **codec complexity** | Specifies call density and codec complexity based on the codec standard you are using. |

# codec (DSP farm profile)

To specify the codecs that are supported by a digital signal processor (DSP) farm profile, use the **codec** command in DSP farm profile configuration mode. To remove the codec, use the **no** form of this command.

**codec** {*codec-type* [*resolution*] | [**frame-rate** *framerate*] | [**bitrate** *bitrate*] | [**rfc-2190**] | **pass-through**}

**no codec** {*codec-type* [*resolution*] | [**frame-rate** *framerate*] | [**bitrate** *bitrate*] | [**rfc-2190**] | **pass-through**}

## Syntax Description

| | |
|---|---|
| *codec-type* | Specifies the codec preferred.<br>• **g711alaw**—G.711 a-law 64,000 bits per second (bps)<br>• **g711ulaw**—G.711 mu-law 64,000 bps<br>• **g722r-64**—G.722-64 at 64,000 bps<br>• **g729abr8**—G.729 ANNEX A and B 8000 bps<br>• **g729ar8**—G.729 ANNEX A 8000 bps<br>• **g729br8**—G.729 ANNEX B 8000 bps<br>• **g729r8**—G.729 8000 bps<br>• **h263**—H.263 video codec<br>• **h264**—H.264 video codec<br>• **ilbc**—Internet Low Bitrate Codec (iLBC)<br>• **isac**—Cisco internet Speech Audio Codec (iSAC) codec |
| *resolution* | Specifies the supported video resolution. The valid entries are:<br>• For H.263—**qcif** and **cif**<br>• For H.264—**qcif**, **cif, vga, w360p**, **w448p, 4cif,** and **720p**<br>**Note** 720p option applies only to homogeneous video conferences. |
| **frame-rate** *framerate* | Specifies the frame rate. The valid entries are 15 fps or 30 fps.<br>This option applies to homogeneous conferences only. |
| **bitrate** *bitrate* | Specifies the bitrate.<br>This option applies to homogeneous conferences only. |
| **rfc-2190** | Specifies the payload format follow RFC-2190. |
| **pass-through** | Enables codec pass-through. Supported for transcoding and media termination point (MTP) profiles. |

## Command Default

The following transcoding default apply when you are configuring audio profiles only. When you configure video transcoding, you must specify the audio codecs.

### Transcoding

- **g711alaw**

- **g711ulaw**
- **g729abr8**
- **g729ar8**

**Conferencing**

- **g711alaw**
- **g711ulaw**
- **g729abr8**
- **g729ar8**
- **g729br8**
- **g729r8**

**MTP**

- **g711ulaw**

## Command Modes

DSP farm profile configuration (config-dspfarm-profile)

## Command History

| Release | Modification |
|---|---|
| 12.3(8)T | This command was introduced. |
| 12.4(4)T | The **pass-through** keyword was added. |
| 12.4(11)XJ2 | The **gsmefr** and **gsmfr** keywords were removed as configurable codec options for all platforms. |
| 12.4(15)T | This command was integrated into Cisco IOS Release 12.4(15)T. |
| 12.4(15)XY | The **g722r-64** keyword was added. |
| 12.4(20)T | This command was integrated into Cisco IOS Release 12.4(20)T. |
| 12.4(22)T | Support for IPv6 was added. |
| 15.1(1))T | This command was modified. The **isac** keyword was added. |
| 15.1(4)M | This command was modified. The **frame-rate, bitrate, rfc-2190, and pass-through** keywords were added and codec support was added for **ilbc**, **h.263** and **h.264**. |

## Usage Guidelines

Only one codec is supported for each MTP profile. To support multiple codecs, you must define a separate MTP profile for each codec.

For homogeneous video profiles, only one video format is supported

For heterogeneous and heterogeneous guaranteed-audio video profiles, multiple video formats and audio codecs are supported.

To change the configured codec in the profile, you must first enter a **no maximum session** command.

Table 13 shows the relationship between DSP farm functions and codecs.

*Table 13 DSP Farm Functions and Codec Relationships*

| DSP Farm Function | Supported Codec |
|---|---|
| Transcoding | • **g711alaw**<br>• **g711ulaw**<br>• **g729abr8**<br>• **g729ar8**<br>• **iSAC**<br>• **h263**<br>• **h264** |
| Conferencing | • **g711alaw**<br>• **g711ulaw**<br>• **g722r-64**<br>• **g729abr8**<br>• **g729ar8**<br>• **g729br8**<br>• **g729r8**<br>• **h263**<br>• **h264**<br>• **ilbc** |
| MTP | • **g711ulaw**<br>• **iSAC** |

Hardware MTPs support only G.711 a-law and G.711 mu-law. If you configure a profile as a hardware MTP and you want to change the codec to other than G.711, you must first remove the hardware MTP by using the **no maximum sessions hardware** command.

The **pass-through** keyword is supported for transcoding and MTP profiles only; the keyword is not supported for conferencing profiles. To support the Resource Reservation Protocol (RSVP) agent on a Skinny Client Control Protocol (SCCP) device, you must use the **codec pass-through** command. In the pass-through mode, the SCCP device processes the media stream by using a pure software MTP, regardless of the nature of the stream, which enables video and data streams to be processed in addition to audio streams. When the pass-through mode is set in a transcoding profile, no transcoding is done for the session; the transcoding device performs a pure software MTP function. The pass-through mode can be used for secure Real-Time Transport Protocol (RTP) sessions.

## Examples

The following example shows how to set the call density and codec complexity to g729abr8:

```
Router(config)# dspfarm profile 123 transcode
Router(config-dspfarm-profile)# codec g729abr8
```

The following example shows how to set up a video conference with guaranteed-audio.

```
Router(config)# dspfarm profile 99 conference video guaranteed-audio
Router(config-dspfarm-profile)# codec h264 4cif
Router(config-dspfarm-profile)# codec h264 cif
Router(config-dspfarm-profile)# maximum conference-participants 8
```

## Related Commands

| Command | Description |
|---|---|
| **associate application** | Associates the SCCP protocol to the DSP farm profile. |
| **dspfarm profile** | Enters DSP farm profile configuration mode and defines a profile for DSP farm services. |
| **maximum sessions (DSP Farm profile)** | Specifies the maximum number of sessions that are supported by the profile. |
| **rsvp** | Enables RSVP support on a transcoding or MTP device. |
| **maximum conference-participants (DSP Farm profile)** | Specifies the maximum number of conference participants that are supported by this profile. |
| **shutdown (DSP Farm profile)** | Disables a DSP farm profile. |

# codec (voice-card)

To specify call density and codec complexity according to the codec standard that is being used or to increase processing frequency for the G.711 codec, use the **codec** command in voice-card configuration mode. To reset the flex complexity default or to disable configured values, use the **no** form of this command.

**codec** {**complexity** {**flex** [**reservation-fixed** {**high** | **medium**}] | **high** | **medium** | **secure**} | **sub-sample**}

**no codec complexity**

## Syntax Description

| | |
|---|---|
| **complexity** | Manages the complexity and density of codecs used in voice processing. |
| **flex** | When the **flex** keyword is used, up to 16 calls can be completed per digital signal processor (DSP). The number of supported calls varies from 6 to 16, depending on the codec used for a call. In this mode, reservation for analog voice interface cards (VICs) may be needed for certain applications such as Central Automatic Message Accounting (CAMA) E-911 calls because oversubscription of DSPs is possible. If this is true, enable the **reservation-fixed** keyword. There is no reservation by default. |
| **reservation-fixed** | (Optional) If you have specified the **flex** keyword, the **reservation-fixed** keyword ensures that sufficient DSP resources are available to handle a call. If you enter the **reservation-fixed** keyword, set the complexity for **high** or **medium**. (See the guidelines following to understand the effects of the keywords.) This option appears only when there is an analog VIC present. |
| **high** | If you specify the **high** keyword to define the complexity, each DSP supports two voice channels encoded in any of the following formats:<br>• g711alaw—G.711 a-law 64,000 bps.<br>• g711ulaw—G.711 mu-law 64,000 bps.<br>• g723ar53—G.723.1 Annex A 5300 bps.<br>• g723ar63—G.723.1 Annex A 6300 bps.<br>• g723r53—G.723.1 5300 bps.<br>• g723r63—G.723.1 6300 bps.<br>• g726r16—G.726 16,000 bps.<br>• g726r24—G726 24,000 bps.<br>• g726r32—G.726 32,000 bps.<br>• g728—G.728 16,000 bps.<br>• g729r8—G.729 8000 bps. This is the default.<br>• g729br8—G.729 Annex B 8000 bps.<br>• fax relay—2400 bps, 4800 bps, 7200 bps, 9600 bps, 12 kbps, and 14.4 kbps.<br>**Note** Codecs G.723.1 and G.728 are not supported on Cisco 1750 and Cisco 1751 modular access routers for Cisco Hoot and Holler over IP applications. |

| | |
|---|---|
| **medium** | If you specify the **medium** keyword to define the complexity, each DSP supports four voice channels encoded in any of the following formats:<br>• g711alaw—G.711 a-law 64,000 bps.<br>• g711ulaw—G.711 mu-law 64,000 bps.<br>• g726r16—G.726 16,000 bps.<br>• g726r24—G.726 24,000 bps.<br>• g726r32—G.726 32,000 bps.<br>• g729r8—G.729 Annex A 8000 bps.<br>• g729br8—G.729 Annex B with Annex A 8000 bps.<br>• fax relay—2400 bps, 4800 bps, 7200 bps, 9600 bps, 12 kbps, and 14.4 kbps. Fax relay is the default. |
| **secure** | If you specify the **secure** keyword to define complexity, each DSP on an NM-HDV network module supports two voice channels encoded in any of the following formats:<br>• g711alaw—G.711 a-law 64,000 bps.<br>• g711ulaw—G.711 mu-law 64,000 bps.<br>• g729—G.729 8000 bps.<br>• g729A—G.729 8000 bps. |
| **sub-sample** | Increases the processing frequency for the G.711 codec with reduced 5510 DSP density. |

## Defaults

The default type of codec complexity is **flex**. The default value for the G.711 codec is 10 milliseconds (ms).

## Command Modes

Voice-card configuration (config-voice-card)

## Command History

| Release | Modification |
|---|---|
| 12.0(5)XK | This command was introduced as the codec complexity on the Cisco 2600 and Cisco 3600 series. |
| 12.0(7)T | This command was integrated into Cisco IOS Release 12.0(7)T. |
| 12.0(7)XK | This command was implemented on the Cisco MC3810 for use with the high-performance compression module (HCM). |
| 12.1(2)T | This command was integrated into Cisco IOS Release 12.1(2)T. |
| 12.2(8)T | This command was implemented on the Cisco 1750 and Cisco 1751. |
| 12.2(13)T | The **ecan-extended** keyword was added. |
| 12.2(15)T | This command was integrated into Cisco IOS Release 12.2(15)T with support for the Cisco 2600 series, Cisco 2600XM, Cisco 3660, Cisco 3725, and Cisco 3745 routers. High codec complexity is supported for DSP processing on these platforms. |

| Release | Modification |
|---|---|
| 12.2(15)ZJ | This command was integrated into Cisco IOS Release 12.2(15)ZJ and the **flex** keyword was added. The **ecan-extended** keyword was removed and G.168 echo-cancellation compliance became the default. |
| 12.3(4)T | This command was integrated into Cisco IOS Release 12.3(4)T. |
| 12.3(7)T | This command was integrated into Cisco IOS Release 12.3(7)T and the **reservation-fixed** keyword was added. |
| 12.3(14)T | This command was integrated into Cisco IOS Release 12.3(14)T and the **secure** keyword was added to provide secure codec complexity for TI-549 DSP processing on the NM-HDV network module. |
| 12.4(22)T1 | The **codec complexity** command was changed to the **codec** (voice-card) command and the **sub-sample** keyword was added for the 5510 DSP. |

## Usage Guidelines

Codec complexity refers to the amount of processing required to perform voice compression. Codec complexity affects the call density—the number of calls reconciled on the DSPs. With higher codec complexity, fewer calls can be handled. Select a higher codec complexity if that is required to support a particular codec or combination of codecs. Select a lower codec complexity to support the greatest number of voice channels, provided that the lower complexity is compatible with the particular codecs in use.

For codec complexity to change, all of the DSP voice channels must be in the idle state.

When you have specified the **flex** keyword, you can connect (or configure in the case of DS0 groups and PRI groups) more voice channels to the module than the DSPs can accommodate. If all voice channels should go active simultaneously, the DSPs become oversubscribed, and calls that are unable to allocate a DSP resource fail to connect. The **flex** keyword allows the DSP to process up to 16 channels. In addition to continuing support for configuring a fixed number of channels per DSP, the **flex** keyword enables the DSP to handle a flexible number of channels. The total number of supported channels varies from 6 to 16, depending on which codec is used for a call. Therefore, the channel density varies from 6 per DSP (high-complexity codec) to 16 per DSP (g.711 codec).

The **high** keyword selects a higher codec complexity if that is required to support a particular codec or combination of codecs. When you use the **codec complexity high** command to change codec complexity, the system prompts you to remove all existing DS0 or PRI groups using the specified voice card, then all DSPs are reset, loaded with the specified firmware image, and released.

The **medium** keyword selects a lower codec complexity to support the greatest number of voice channels, provided that the lower complexity is compatible with the particular codecs in use.

The **secure** keyword restricts the number of TI-549 DSP channels to 2, which is the lower codec complexity required to support Secure Real-Time Transport Protocol (SRTP) package capability on the NM-HDV and enable media authentication and encryption. If the **secure** command is not configured then the gateway will not advertise secure capability to Cisco CallManager, resulting in nonsecure calls. You do not need to use any command to specify secure codec complexity for TI-5510 DSPs, which support SRTP capability in all modes. Use the **mgcp package-capability** *srtp-package* command to enable MGCP gateway capability to process SRTP packages. Use the **show voice dsp** command to display codec complexity status.

Voice quality issues may occur when there are more than 15 G.711 channels on one 5510 DSP. To resolve the voice-quality issue, change the processing period (or segment size) of the G.711 codec from 5 ms to 10 ms. (The segment size of most voice codecs is 10 ms.) However, a voice call with 10-ms segment size has longer end-to-end delay (+ 5ms to 10 ms) than a call with 5-ms segment size.

Beginning in Cisco IOS Release 12.4(22)T1, the **sub-sample** keyword is added for applications that have strict requirements for round-trip delay times for VoIP. You can now accept the default G.711 (10 ms with lower MIPS) or enter the **codec sub-sample** command to select 5-ms G.711 (lower delay with higher MIPS). The **sub-sample** keyword is enabled only for the 5510 DSP.

The **codec sub-sample** command enables 5-ms processing for the G.711 codec inside the DSP to reduce the delay. However, this reduces the channel density of G.711 channels from 16 to 14. There is no difference in secure channel density when this mode is enabled.

## Examples

The following example sets the codec complexity to high on voice card 1 installed on a router, and configures local calls to bypass the DSP:

```
voice-card 1
 codec complexity high
local-bypass
```

The following example sets the codec complexity to secure on voice card 1 installed on the NM-HDV, and configures it to support SRTP package processing, media authentication, and encryption:

```
voice-card 1
 codec complexity secure
```

The following example shows how to enable 5-ms processing for the G.711 codec inside the 5510 DSP:

```
voice-card 1
 codec sub-sample
```

## Related Commands

| Command | Description |
|---|---|
| **ds0-group** | Defines T1/E1 channels for compressed voice calls and the CAS method by which the router connects to the PBX or PSTN. |
| **mgcp package-capability** | Enables MGCP gateway capability to process SRTP packages. |
| **show voice dsp** | Displays the current status of all DSP voice channels. |

# codec aal2-profile

To set the codec profile for a digital signal processor (DSP) on a per-call basis, use the **codec aal2-profile** command in dial peer configuration mode. To restore the default codec profile, use the **no** form of this command.

**codec aal2-profile** {**itut** | **custom** | **atmf**} *profile-number codec*

**no codec aal2-profile**

## Syntax Description

| | |
|---|---|
| **itut** | The *profile-number* as an ITU-T type. |
| **custom** | The *profile-number* as a custom type. |
| **atmf** | The *profile-number* as an Asynchronous Transfer Mode Forum (ATMF) type. |
| *profile-number* | The available *profile-number* selections depend on the profile type.<br>For ITU-T:<br>• **1** = G.711 mu-law<br>• **2** = G.711 mu-law with silence insertion descriptor (SID)<br>• **7** = G.711 mu-law and G.729ar8<br>For ATMF:<br>**9** = Broadband Loop Emulation Services (BLES) support for VoAAL2<br>For custom:<br>• **100** = G.711 mu-law and G.726r32<br>• **110** = G.711 mu-law, G.726r32, and G.729ar8 |
| *codec* | Enter one codec for the DSP. The possible *codec* entries depend on the *profile-number* value. The valid entries are as follows:<br>• For ITU 1—**g711 mu-law**<br>• For ITU 2—**g711 mu-law**<br>• For ITU 7—**g711 mu-law** or **g729ar8**<br>• For ATMF—**g711 mu-law**<br>• For custom 100—**g711 mu-law** or **g726r32**<br>• For custom 110—**g711 mu-law** or **g726r32** or **g729ar8**<br>• For lossless compression—**llcc** |

## Command Default

ITU-T profile 1 (G.711 mu-law)

## Command Modes

Dial peer configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(1)XA | This command was introduced on the Cisco MC3810. |
| 12.1(2)T | This command was integrated into Cisco IOS Release 12.1(2)T. |
| 12.2(2)T | This command was implemented on the Cisco 7200 series. |
| 12.2(11)T | This command was implemented on the Cisco IAD2420 series. |
| 12.3(4)XD | The lossless compression codec (**llcc**) keyword was added. |
| 12.3(7)T | This command was integrated into Cisco IOS Release 12.3(7)T. |

## Usage Guidelines

Use this command to configure the DSP to operate with a specified profile type and codecs.

You must enter the **session protocol aal2-trunk** command before configuring the codec ATM adaptation Layer 2 (AAL2) profile.

This command is used instead of the **codec (dial peer)** command for AAL2 trunk applications.

## Examples

The following example sets the codec AAL2 profile type to ITU-T and configures a profile number of 7, enabling codec G.729ar8:

```
dial-peer voice 100 voatm
 session protocol aal2-trunk
 codec aal2-profile itut 7 g729ar8
```

The following example sets the codec AAL2 profile type to custom and configures a profile number of 100, enabling codec G.726r32:

```
dial-peer voice 200 voatm
 session protocol aal2-trunk
 codec aal2-profile custom 100 g726r32
```

## Related Commands

| Command | Description |
|---|---|
| **session protocol (dial peer)** | Establishes a session protocol for calls between the local and remote routers via the packet network. |

# codec gsmamr-nb

To specify the Global System for Mobile Adaptive Multi-Rate Narrow Band (GSMAMR-NB) codec for a dial peer, use the **codec gsmamr-nb** command in dial peer voice configuration mode. To disable the GSMAMR-NB codec, use the **no** form of this command.

**codec gsmamr-nb** [**packetization-period 20**] [**encap rfc3267**] [**frame-format** {**bandwidth-efficient** | **octet-aligned** [**crc** | **no-crc**]}] [**modes** *modes-value*]

**no codec gsmamr-nb**

## Syntax Description

| | |
|---|---|
| **packetization-period 20** | (Optional) Sets the packetization period at 20 ms. |
| **encap rfc3267** | (Optional) Sets the encapsulation value to comply with RFC 3267. |
| **frame-format** | (Optional) Specifies a frame format. Supported values are octet-aligned and bandwidth-efficient. The default is octet-aligned. |
| **crc** \| **no-crc** | (Optional) CRC is applicable only for octet-aligned frame format. If you enter bandwidth-efficient frame format, the crc \| no-crc options will not be available because they are inapplicable. |
| **modes** | (Optional) The eight speech-encoding modes (bit rates between 4.75 and 12.2 kbps) available in the GSMAMR-NB codec. |
| *modes-value* | (Optional) Valid values are from 0 to 7. You can specify modes as a range (for example, 0-2), or individual modes separated by commas (for example, 2,4,6), or a combination of the two (for example, 0-2,4,6-7). |

## Command Default

Packetization period is **20** ms.
Encapsulation is **rfc3267**.
Frame format is **octet-aligned**.
CRC is **no-crc**.
Modes value is **0-7**.

## Command Modes

Dial peer voice configuration

## Command History

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

## Usage Guidelines

The **codec gsmamr-nb** command configures the GSMAMR-NB codec and its parameters on the Cisco AS5350XM and Cisco AS5400XM platforms.

## Examples

The following example sets the codec to **gsmamr-nb** and sets parameters:

```
Router(config-dial-peer)# codec gsmamr-nb packetization-period 20 encap rfc3267
frame-format octet-aligned crc
```

## Related Commands

| Command | Description |
|---|---|
| **codec complexity** | Specifies call density and codec complexity based on the codec used. |
| **show dial peer voice** | Displays the codec setting for dial peers. |

# codec ilbc

To specify the voice coder rate of speech for a dial peer using the internet Low Bandwidth Codec (iLBC), use the **codec ilbc** command in dial peer configuration mode. To reset the default value, use the **no** form of this command.

**codec ilbc** [**mode** *frame_size* [**bytes** *payload_size*]]

**no codec ilbc** [**mode** *frame_size* [**bytes** *payload_size*]]

## Syntax Description

| | |
|---|---|
| **mode** | (Optional) Specifies the iLBC operating frame mode that is encapsulated in each packet. |
| *frame_size* | (Optional) iLBC operating frame in milliseconds (ms). Valid entries are:<br>• 20—20ms frames for 15.2kbps bit rate<br>• 30—30ms frames for 13.33 kbps bit rate<br>Default is 20. |
| **bytes** | (Optional) Specifies the number of bytes in the voice payload of each frame. |
| *payload_size* | (Optional) Number of bytes in the voice payload of each frame. Valid entries are:<br>• For **mode 20**—38, 76, 114, 152, 190, 228. Default is 38.<br>• For **mode 30**—50, 100, 150, 200. Default is 50. |

## Command Default

20ms frames with a 15.2kbps bit rate.

## Command Modes

Dial peer configuration

## Command History

| Release | Modification |
|---|---|
| 12.4(11)T | This command was introduced. |
| IOS Release XE 2.5 | This command was integrated into Cisco IOS XE Release 2.5. |

## Usage Guidelines

Use this command to define a specific voice coder rate of speech and payload size for a VoIP dial peer using an iLBC codec.

If codec values for the dial peers of a connection do not match, the call fails.

You can change the payload of each VoIP frame by using the **bytes** keyword. However, increasing the payload size can add processing delay for each voice packet.

## Examples

The following example shows how to configure the iLBC codec on an IP-to-IP Gateway:

```
dial-peer voice 1 voip
 rtp payload-type cisco-codec-ilbc 100
 codec ilbc mode 30 bytes 200
```

## Related Commands

| Command | Description |
|---|---|
| **show dial peer voice** | Displays the codec setting for dial peers. |

# codec preference

To specify a list of preferred codecs to use on a dial peer, use the **codec preference** command in voice-class configuration mode. To disable this functionality, use the **no** form of this command.

**codec preference** *value codec-type* [**mode** {**independent** | **adaptive**}] [**frame-size** {**20** | **30** | **60** | **fixed**] [**bit rate** *value*] [**bytes** *payload-size*] [**packetization-period 20**] [**encap rfc3267**] [**frame-format** {**bandwidth-efficient** | **octet-aligned** [**crc** | **no-crc**]}] [**modes** *modes-value*]

**no codec preference** *value codec-type*

## Syntax Description

| | |
|---|---|
| *value* | The order of preference; 1 is the most preferred and 14 is the least preferred. |
| *codec-type* | The codec preferred. Values are as follows:<br>• **clear-channel—**Clear Channel 64,000 bps.<br>• **g711alaw**—G.711 a-law 64,000 bps.<br>• **g711ulaw**—G.711 mu-law 64,000 bps.<br>• **g722r-64**—G.722-64 at 64,000 bps.<br>• **g723ar53**—G.723.1 Annex-A 5300 bps.<br>• **g723ar63**—G.723.1 Annex-A 6300 bps.<br>• **g723r53**—G.723.1 5300 bps.<br>• **g723r63**—G.723.1 6300 bps.<br>• **g726r16**—G.726 16,000 bps<br>• **g726r24**—G.726 24,000 bps<br>• **g726r32**—G.726 32,000 bps.<br>• **g728**—G.728 16,000 bps.<br>• **g729abr8**—G.729 ANNEX-A and B 8000 bps.<br>• **g729br8**—G.729 ANNEX-B 8000 bps.<br>• **g729r8**—G.729 8000 bps.<br>• **gsmamr-nb—**Enables GSMAMR-NB codec capability.<br>• **gsmfr—**Global System for Mobile Communications Full Rate (GSMFR) 13,200 bps.<br>**Note** The **gsmfr** keyword is configurable only on the Cisco AS5350 and AS5400 with MSAv6 digital signal processors (DSPs).<br>• **ilbc**—internet Low Bitrate Codec (iLBC) at 13,330 bps or 15,200 bps.<br>• **isac**—Cisco internet Speech Audio Codec (iSAC) codec.<br>• **transparent**—Enables codec capabilities to be passed transparently between endpoints.<br>**Note** The **transparent** keyword is not supported when the **call-start** command is configured. |
| **mode** | (Optional) For iLBC and iSAC codecs only. Specifies the iLBC or iSAC operating frame mode that is encapsulated in each packet. |

| | |
|---|---|
| **independent** | (Optional) For iSAC codec only. Specifies that the configuration mode variable bit rate (VBR) is independent (value 1). |
| **adaptive** | (Optional) For iSAC codec only. Specifies that the configuration mode VBR is adaptive (value 0). |
| **frame-size** | (Optional) For iLBC and iSAC codecs only. Specifies the operating frame in milliseconds (ms). Valid entries are:<br>• **20**—20-ms frames (iLBC only)<br>• **30**—30-ms frames (iLBC or iSAC)<br>• **60**—60-ms frames (iLBC or iSAC)<br>• **fixed**—This keyword is applicable only for adaptive mode. |
| **bit rate** *value* | (Optional) Configures the target bit rate in kilobits per second. The range is 10 to 32. |
| **bytes** | (Optional) Specifies that the size of the voice frame is in bytes. |
| *payload-size* | (Optional) Number of bytes that you specify as the voice payload of each frame. Values depend on the codec type and the packet voice protocol. |
| **packetization-period 20** | (Optional) Sets the packetization period at 20 ms. This keyword is applicable only to GSMAMR-NB codec support. |
| **encap rfc3267** | (Optional) Sets the encapsulation value to comply with RFC 3267. This keyword is applicable only to GSMAMR-NB codec support. |
| **frame-format** | (Optional) Specifies a frame format. Supported values are **octet-aligned** and **bandwidth-efficient**. The default is **octet-aligned**. This keyword is applicable only to GSMAMR-NB codec support. |
| **crc** \| **no-crc** | (Optional) Cyclic Redundancy Check (CRC) is applicable only for octet-aligned frame format. If you enter bandwidth-efficient frame format, the **crc** \| **no-crc** options are not available because they are inapplicable. This keyword is applicable only to GSMAMR-NB codec support. |
| **modes** *modes-values* | (Optional) Valid values are from 0 to 7. You can specify modes as a range (for example, 0-2), or individual modes separated by commas (for example, 2,4,6), or a combination of the two (for example, 0-2,4,6-7). This argument is applicable only to GSMAMR-NB codec support. |

## Command Default

If this command is not entered, no specific types of codecs are identified with preference.

If you enter the **gsmamr-nb** keyword, the default values are as follows:

Packetization period is 20 ms.
Encap is **rfc3267**.
Frame format is **octet-aligned**.
CRC is **no-crc**.
Modes value is **0-7**.

If you enter the **isac** keyword, the default values are as follows:

Mode is **independent**.
Target bit-rate is **32000 bps**.
Framesize is **30ms**.

## Command Modes

Voice-class configuration (config-voice-class)

## Command History

| Release | Modification |
|---|---|
| 12.0(2)XH | This command was introduced on the Cisco AS5300. |
| 12.0(7)T | This command was implemented on the Cisco 2600 series and Cisco 3600 series. |
| 12.0(7)XK | This command was implemented on the Cisco MC3810. |
| 12.1(2)T | This command was integrated into Cisco Release IOS Release 12.1(2)T. |
| 12.1(5)T | This command was modified. The **gsmefr** and **gsmfr** keywords were added. |
| 12.2(13)T3 | This command was modified.The **transparent** keyword was added. |
| 12.4(4)XC | This command was extended to include GSMAMR-NB codec parameters on the Cisco AS5350XM and Cisco AS5400XM platforms. |
| 12.4(9)T | This command was integrated into Cisco IOS Release 12.4(9)T. |
| 12.4(11)T | This command was modified. The **ilbc** and **mode** keywords were added. |
| 12.4(11)XJ2 | This command was modified. The **gsmefr** and **gsmfr** keywords were removed as configurable codec options for all platforms with the exception of the **gsmfr** codec on the Cisco AS5400 and AS5350 with MSAv6 dsps. |
| 12.4(15)T | This command was integrated into Cisco IOS Release 12.4(15)T. |
| 12.4(15)XY | This command was modified. The **g722r-64** keyword was added. |
| 12.4(20)T | This command was integrated into Cisco IOS Release 12.4(20)T. |
| IOS Release XE 2.5 | This command was integrated into Cisco IOS XE Release 2.5. |
| 15.1(1)T | This command was modified. The **isac** keyword was added as a codec type, and the **independent**, **adaptive**, **bit rate**, and **fixed** keywords were added as configurable parameters. |

## Usage Guidelines

The routers at opposite ends of the WAN may have to negotiate the codec selection for the network dial peers. The **codec preference** command specifies the order of preference for selecting a negotiated codec for the connection. Table 14 describes the voice payload options and default values for the codecs and packet voice protocols.

**Note** The **transparent** keyword is not supported when the **call start** command is configured.

*Table 14 Voice Payload-per-Frame Options and Defaults*

| Codec | Protocol | Voice Payload Options (in Bytes) | Default Voice Payload (in Bytes) |
|---|---|---|---|
| **g711alaw**<br>**g711ulaw** | VoIP<br>VoFR<br>VoATM | 80, 160<br>40 to 240 in multiples of 40<br>40 to 240 in multiples of 40 | 160<br>240<br>240 |
| **g722r-64** | VoIP | 80, 160, 240 | 160 |

*Table 14 Voice Payload-per-Frame Options and Defaults (continued)*

| Codec | Protocol | Voice Payload Options (in Bytes) | Default Voice Payload (in Bytes) |
|---|---|---|---|
| **g723ar53**<br>**g723r53** | VoIP<br>VoFR<br>VoATM | 20 to 220 in multiples of 20<br>20 to 240 in multiples of 20<br>20 to 240 in multiples of 20 | 20<br>20<br>20 |
| **g723ar63**<br>**g723r63** | VoIP<br>VoFR<br>VoATM | 24 to 216 in multiples of 24<br>24 to 240 in multiples of 24<br>24 to 240 in multiples of 24 | 24<br>24<br>24 |
| **g726r16** | VoIP<br>VoFR<br>VoATM | 20 to 220 in multiples of 20<br>10 to 240 in multiples of 10<br>10 to 240 in multiples of 10 | 40<br>60<br>60 |
| **g726r24** | VoIP<br>VoFR<br>VoATM | 30 to 210 in multiples of 30<br>15 to 240 in multiples of 15<br>30 to 240 in multiples of 15 | 60<br>90<br>90 |
| **g726r32** | VoIP<br>VoFR<br>VoATM | 40 to 200 in multiples of 40<br>20 to 240 in multiples of 20<br>40 to 240 in multiples of 20 | 80<br>120<br>120 |
| **g728** | VoIP<br>VoFR<br>VoATM | 10 to 230 in multiples of 10<br>10 to 240 in multiples of 10<br>10 to 240 in multiples of 10 | 40<br>60<br>60 |
| **g729abr8**<br>**g729ar8**<br>**g729br8**<br>**g729r8** | VoIP<br>VoFR<br>VoATM | 10 to 230 in multiples of 10<br>10 to 240 in multiples of 10<br>10 to 240 in multiples of 10 | 20<br>30<br>30 |
| **ilbc** | VoIP | For the **mode 20** keyword, 38, 76, 114, 152, 190, 228<br>For the **mode 30** keyword, 50, 100, 150, 200 | 38<br>50 |
| **iSAC** | VoIP | — | — |

## Examples

The following example show how to set the codec preference to the GSMAMR-NB codec and specify parameters:

```
Router(config-voice-class)# codec preference 1 gsmamr-nb packetization-period 20 encap
rfc3267 frame-format octet-aligned crc
```

The following example shows how to create codec preference list 99 and applies it to dial peer 1919:

```
voice class codec 99
codec preference 1 g711alaw
codec preference 2 g711ulaw bytes 80
codec preference 3 g723ar53
codec preference 4 g723ar63 bytes 144
codec preference 5 g723r53
codec preference 6 g723r63 bytes 120
codec preference 7 g726r16
codec preference 8 g726r24
codec preference 9 g726r32 bytes 80
codec preference 10 g729br8
codec preference 11 g729r8 bytes 50
end
```

```
dial-peer voice 1919 voip
 voice-class codec 99
```

The following example shows how to configure the transparent codec used by the Cisco Unified Border Element:

```
voice class codec 99
codec preference 1 transparent
```

**Note** You can assign a preference value of 1 only to the transparent codec. Additional codecs assigned to other preference values are ignored if the transparent codec is used.

The following example shows how to configure the iLBC codec used by the Cisco Unified Border Element:

```
voice class codec 99
codec preference 1 ilbc mode 30 bytes 200
```

## Related Commands

| Command | Description |
|---|---|
| **call-start** | Forces an H.323 Version 2 gateway to use fast connect or slow connect procedures for a dial peer. |
| **voice class codec** | Enters voice-class configuration mode and assigns an identification tag number to a codec voice class. |
| **voice-class codec (dial peer)** | Assigns a previously configured codec selection preference list to a dial peer. |

# codec profile

To define video capabilities needed for video endpoints, use the **codec profile** command in telephony-service configuration mode. To disable the codec profile, use the **no** form of this command.

**codec profile** *tag profile*

**no codec profile**

## Syntax Description

| | |
|---|---|
| *tag* | A number in the range of 1 to 1000000. |
| *profile* | The name of the audio or video codec profile:<br>• aacld<br>• h263<br>• h263+<br>• h264 |

## Command Default

No codec profile is configured.

## Command Modes

Global configuration (config)

## Command History

| Release | Modification |
|---|---|
| 12.4(22)T | This command was introduced. |

## Usage Guidelines

For the Cisco Unified Customer Voice Portal solution, only h263 and h263+ are supported profile options.

## Examples

The following example shows the codec tagged 116 assigned to the H263 profile.

```
codec profile 116 H263
    clockrate 90000
    fmtp "fmtp:120 SQCIF=1;QCIF=1;CIF=1;CIF4=2;MAXBR=3840;I=1"
```

The codec profile can then be added to a voice class codec list, or the VoIP dial peer:

```
voice class codec 998
 codec preference 1 g711ulaw
 video codec h263 profile 116
```

## Related Commands

| Command | Description |
|---|---|
| **clockrate** | Sets the clock rate for the codec. |
| **fmtp** | Defines a string for video endpoints. |

# comfort-noise

To generate background noise to fill silent gaps during calls if voice activity detection (VAD) is activated, use the **comfort-noise** command in voice-port configuration mode. To provide silence when the remote party is not speaking and VAD is enabled at the remote end of the connection, use the **no** form of this command.

**comfort-noise**

**no comfort-noise**

## Syntax Description

This command has no arguments or keywords.

## Command Default

Background noise is generated by default.

## Command Modes

Voice-port configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(1)T | This command was introduced on the Cisco 3600 series. |
| 12.2(13)T | This command was integrated into Cisco IOS Release 12.2(13)T and was implemented on the Cisco 2600 series, the Cisco 7200 series, and the Cisco 7500 series using the extended echo canceller. |

## Usage Guidelines

Use the **comfort-noise** command to generate background noise to fill silent gaps during calls if VAD is activated. If the **comfort-noise** command is not enabled, and VAD is enabled at the remote end of the connection, the user hears dead silence when the remote party is not speaking.

The configuration of the **comfort-noise** command 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.

## Examples

The following example enables background noise on voice port 1/0/0:

```
voice-port 1/0/0
 comfort-noise
```

## Related Commands

| Command | Description |
|---|---|
| **vad (dial peer configuration)** | Enables VAD for the calls using a particular dial peer. |
| **vad (voice-port configuration)** | Enables VAD for the calls using a particular voice port. |

# compand-type

To specify the companding standard used to convert between analog and digital signals in pulse code modulation (PCM) systems, use the **compand-type** command in voice-port configuration mode. To disable the compand type, use the **no** form of this command.

**compand-type** {**u-law** | **a-law**}

**no compand-type** {**u-law** | **a-law**}

## Syntax Description

| | |
|---|---|
| **u-law** | Specifies the North American mu-law ITU-T PCM encoding standard. |
| **a-law** | Specifies the European a-law ITU-T PCM encoding standard. |

## Command Default

**mu-law** (T1 digital)
**a-law** (E1 digital)

## Command Modes

Voice-port configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(1)MA | This command was introduced. |

## Usage Guidelines

The Cisco 2660 and the Cisco 3640 routers do not require configuration of the **compand-type a-law** command. However, if you request a list of commands, the **compand-type a-law** command displays.

**Note** On the Cisco 3600 series routers router, the mu-law and a-law settings are configured using the **codec** dial peer configuration command.

**Note** This command is not supported on the Cisco AS 5300/5350/5400 and 5850 Universal Gateway series which use the Nextport DSP.

## Examples

The following example configures a-law encoding on voice port 1/1:

```
voice-port 1/1
 compand-type a-law
```

## Related Commands

| Command | Description |
|---|---|
| **codec (voice-port configuration)** | Configures voice compression. |

# conference

To define a Feature Access Code (FAC) to initiate a three-party conference in feature mode on analog phones connected to FXS ports, use the **conference** command in STC application feature-mode call-control configuration mode. To return the code to its default, use the **no** form of this command.

**conference** *keypad-character*

**no conference**

## Syntax Description

| | |
|---|---|
| *keypad-character* | Character string of one to four characters that can be dialed on a telephone keypad (0—9, *, #). Default is #3. |

## Command Default

The default value is #3.

## Command Modes

STC application feature-mode call-control configuration (config-stcapp-fmcode)

## Command History

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

## Usage Guidelines

This command changes the value of the FAC for the Call Conference feature from the default (#3) to the specified value.

If you attempt to configure this command with a value that is already configured for another FAC in feature mode, you receive a message. This message will not prevent you from configuring the feature code. If you configure a duplicate FAC, the system implements the first feature it matches in the order of precedence as determined by the value for each FAC (#1 to #5).

If you attempt to configure this command with a value that precludes or is precluded by another FAC in feature mode, you receive a message. If you configure a FAC to a value that precludes or is precluded by another FAC in feature mode, the system always executes the call feature with the shortest code and ignores the longer code. For example, 1 will always preclude 12 and 123. These messages will not prevent you from configuring the feature code. You must configure a new value for the precluded code in order to enable phone user access to that feature.

## Examples

The following example shows how to change the value of the feature code for Call Conference from the default (#3). With this configuration, a phone user presses hook flash to get the first dial tone, then dials an extension number to connect to a second call. When the second call is established, the user presses hook flash to get the feature tone and then dials 33 to initiate a three-party conference.

```
Router(config)# stcapp call-control mode feature
Router(config-stcapp-fmcode)# conference 33
Router(config-stcapp-fmcode)# exit
```

## Related Commands

| Command | Description |
|---|---|
| **drop-last-conferee** | Defines FAC in feature mode to use to drop last active call during a three-party conference. |
| **hangup-last-active-call** | Defines FAC in feature mode to drop last active call during a three-party conferencee. |
| **toggle-between-two-calls** | Defines FAC in feature mode to toggle between two active calls. |
| **transfer** | Defines FAC in feature mode to connect a call to a third party that the phone user dials. |

# conference-join custom-cptone

To associate a custom call-progress tone to indicate joining a conference with a DSP farm profile, use the **conference-join custom-cptone** command in DSP farm profile configuration mode. To remove the custom call-progress tone association and disable the tone for the conference profile, use the **no** form of this command.

**conference-join custom-cptone** *cptone-name*

**no conference-join custom-cptone** *cptone-name*

## Syntax Description

| | |
|---|---|
| *cptone-name* | Descriptive identifier for this custom call-progress tone that indicates joining a conference. |

## Command Default

No custom call-progress tone to indicate joining a conference is associated with the DSP farm profile.

## Command Modes

DSP farm profile configuration

## Command History

| Cisco IOS Release | Version | Modification |
|---|---|---|
| 12.4(11)XJ2 | Cisco Unified CME 4.1 | This command was introduced. |
| 12.4(15)T | Cisco Unified CME 4.1 | This command was integrated into Cisco IOS Release 12.4(15)T |

## Usage Guidelines

To have a tone played when a party joins a conference, define the join tone, then associate it with the DSP farm profile for that conference.

- Use the **voice class custom-cptone** command to create a voice class for defining custom call-progress tones to indicate joining a conference.
- Use the **cadence** and **frequency** commands to define the characteristics of the join tone.
- Use the **conference-join custom-cptone** command to associate the join tone to the DSP farm profile for that conference. Use the **show dspfarm profile command** to display the DSP farm profile.

## Examples

The following example defines a custom call-progress tone to indicate joining a conference and associates that join tone to a DSP farm profile defined for conferencing. Note that the custom call-progress tone names in the **voice class custom-cptone** and **conference-join custom-cptone** commands must be the same.

```
Router(config)# voice class custom-cptone jointone
Router(cfg-cptone)# dualtone conference
Router(cfg-cp-dualtone)# frequency 500 500
Router(cfg-cp-dualtone)# cadence 100 100 100 100 100
!
Router(config)# dspfarm profile 1 conference
Router(config-dspfarm-profile)# conference-join custom-cptone jointone
```

## Related Commands

| Command | Description |
|---|---|
| **cadence** | Defines the tone-on and tone-off durations for a call-progress tone. |
| **conference-leave** | Associates a custom call-progress tone to indicate leaving a conference with a DSP farm profile. |
| **daultone conference** | Enters cp-dualtone configuration mode for specifying a custom call-progress tone. |
| **frequency** | Defines the frequency components for a call-progress tone. |
| **show dspfarm profile** | Display configured digital signal processor (DSP) farm profile information. |
| **voice class custom-cptone** | Creates a voice class for defining custom call-progress tones to be detected. |

# conference-leave custom-cptone

To associate a custom call-progress tone to indicate leaving a conference with a DSP farm profile, use the **conference-leave custom-cptone** command in DSP farm profile configuration mode. To remove the custom call-progress tone association and disable the tone for the conference profile, use the **no** form of this command.

**conference-leave custom-cptone** *cptone-name*

**no conference-leave custom-cptone** *cptone-name*

## Syntax Description

| | |
|---|---|
| *cptone-name* | Descriptive identifier for this custom call-progress tone that indicates leaving a conference. |

## Command Default

No custom call-progress tone to indicate leaving a conference is is associated with the DSP farm profile.

## Command Modes

DSP farm profile configuration

## Command History

| Cisco IOS Release | Version | Modification |
|---|---|---|
| 12.4(11)XJ2 | Cisco Unified CME 4.1 | This command was introduced. |
| 12.4(15)T | Cisco Unified CME 4.1 | This command was integrated into Cisco IOS Release 12.4(15)T |

## Usage Guidelines

For a tone to be played when a party leaves a conference, define the leave tone, then associate it with the DSP farm profile for that conference.

Use the **voice class custom-cptone** command to create a voice class for defining custom call-progress tones to indicate leaving a conference.

Use the **cadence** and **frequency** commands to define the characteristics of the leave tone.

Use the **conference-join custom-cptone** command to associate the leave tone to the DSP farm profile for that conference. Use the **show dspfarm profile command** to display the DSP farm profile.

## Examples

The following example defines a custom call-progress tone to indicate leaving a conference and associates that leave tone to a DSP farm profile defined for conferencing. Note that the custom call-progress tone names in the **voice class custom-cptone** and **conference-join custom-cptone** commands must be the same.

```
Router(config)# voice class custom-cptone leavetone
Router(cfg-cptone)# dualtone conference
Router(cfg-cp-dualtone)# frequency 500 500
Router(cfg-cp-dualtone)# cadence 100 100 100 100 100
!
Router(config)# dspfarm profile 1 conference
Router(config-dspfarm-profile)# conference-join custom-cptone leavetone
```

## Related Commands

| Command | Description |
|---|---|
| **cadence** | Defines the tone-on and tone-off durations for a call-progress tone. |
| **conference-join** | Associates a custom call-progress tone to indicate joining a conference with a DSP farm profile. |
| **dualtone conference** | Enters cp-dualtone configuration mode for specifying a custom call-progress tone. |
| **frequency** | Defines the frequency components for a call-progress tone. |
| **show dspfarm profile** | Display configured digital signal processor (DSP) farm profile information. |
| **voice class custom-cptone** | Creates a voice class for defining custom call-progress tones to be detected. |

# condition

To manipulate the signaling format bit-pattern for all voice signaling types, use the **condition** command in voice-port configuration mode. To turn off conditioning on the voice port, use the **no** form of this command.

**condition** {**tx-a-bit** | **tx-b-bit**| **tx-c-bit**| **tx-d-bit**} {**rx-a-bit** | **rx-b-bit**| **rx-c-bit**| **rx-d-bit**} {**on** | **off** | **invert**}

**no condition** {**tx-a-bit** | **tx-b-bit**| **tx-c-bit**| **tx-d-bit**} {**rx-a-bit** | **rx-b-bit**| **rx-c-bit**| **rx-d-bit**} {**on** | **off** | **invert**}

## Syntax Description

| | |
|---|---|
| **tx-a-bit** | Sends A bit. |
| **tx-b-bit** | Sends B bit. |
| **tx-c-bit** | Sends C bit. |
| **tx-d-bit** | Sends D bit. |
| **rx-a-bit** | Receives A bit. |
| **rx-b-bit** | Receives B bit. |
| **rx-c-bit** | Receives C bit. |
| **rx-d-bit** | Receives D bit. |
| **on** | Forces the bit state to 1. |
| **off** | Forces the bit state to 0. |
| **invert** | Inverts the bit state. |

## Command Default

The signaling format is not manipulated (for all sent or received A, B, C, and D bits).

## Command Modes

Voice-port configuration

## Command History

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

## Usage Guidelines

Use the **condition** command to manipulate the sent or received bit patterns to match expected patterns on a connected device. Be careful not to destroy the information content of the bit pattern. For example, forcing the a-bit on or off prevents Foreign Exchange Office (FXO) interfaces from being able to generate both an on-hook and off-hook state.

The **condition** command is applicable to digital voice ports only.

## Examples

The following example manipulates the signaling format bit pattern on digital voice port 0:5:

```
voice-port 0:5
 condition tx-a-bit invert
 condition rx-a-bit invert
```

The following example manipulates the signaling format bit pattern on voice port 1/0:0:

```
voice-port 1/0:0
 condition tx-a-bit invert
 condition rx-a-bit invert
```

## Related Commands

| Command | Description |
|---|---|
| **define** | Defines the transmit and receive bits for North American E&M and E&M MELCAS voice signaling. |
| **ignore** | Configures the North American E&M or E&M MELCAS voice port to ignore specific receive bits. |

# connect (channel bank)

To define connections between T1 or E1 controller ports for the channel bank feature, use the **connect** command in global configuration mode. To restore default values, use the **no** form of this command.

**connect** *connection-id* **voice-port** *voice-port-number* {**t1** | **e1**} *controller-number ds0-group-number*

**no connect** *connection-id* **voice-port** *voice-port-number* {**t1** | **e1**} *controller-number ds0-group-number*

## Syntax Description

| | |
|---|---|
| *connection-id* | A name for this connection. |
| **voice-port** | Specifies that a voice port is used in the connection. |
| *voice-port-number* | The voice port slot number and port number. |
| **t1** | Specifies a T1 port. |
| **e1** | Specifies an E1 port. |
| *controller-number* | The location of the first T1 or E1 controller to be connected. Valid values for the slot and port are 0 and 1. |
| *ds0-group-number* | The number identifier of the DS0 group associated with the first T1 or E1 controller port. The number is created by using the **ds0-group** command. Valid values are from 0 to 23 for T1 and from 0 to 30 for E1. |

## Command Default

There is no drop-and-insert connection between the ports.

## Command Modes

Global configuration

## Command History

| Release | Modification |
|---|---|
| 12.0(5)XK | This command was introduced. |
| 12.0(7)T | This command was integrated into Cisco IOS Release 12.0(7)T. |
| 12.2(15)ZJ | The **voice-port** keyword was added. |
| 12.3(4)T | This command was integrated into Cisco IOS Release 12.3(4)T. |

## Usage Guidelines

The **connect** command creates a named connection between two DS0 groups associated with voice ports on T1 or E1 interfaces where the groups have been defined by the **ds0-group** command.

## Examples

The following example shows how to configure a channel bank connection for FXS loop-start signaling:

```
Router(config)# controller t1 1/0
Router(config-controller)# ds0-group 1 timeslot 0 type fxo-loop-start
Router(config-controller)# exit
Router(config)# voice-port 1/1/0
Router(config-voiceport)# signal-type fxs-loop-start
Router(config-voiceport)# exit

Router(config)# connect connection1 voice-port 1/1/0 t1 1/0 0
```

## Related Commands

| Command | Description |
|---|---|
| **ds0-group** | Specifies the DS0 time slots that make up a logical voice port on a T1 or E1 controller and the signaling type by which the router communicates with the PBX or PSTN. |
| **show connect** | Displays configuration information about drop-and-insert connections that have been configured on a router. |

# connect (drop-and-insert)

To define connections among T1 or E1 controller ports for drop-and-insert (also called TDM cross-connect), use the **connect** command in global configuration mode. To restore default values, use the **no** form of this command.

**connect** *connection-id* {**t1** | **e1**} *slot*/*port-1 tdm-group-no-1* {**t1** | **e1**} *slot*/*port-2 tdm-group-no-2*

**no connect** *connection-id* {**t1** | **e1**} *slot*/*port-1 tdm-group-no-1* {**t1** | **e1**} *slot*/*port-2 tdm-group-no-2*

## Syntax Description

| | |
|---|---|
| *connection-id* | A name for this connection. |
| **t1** | Specifies a T1 port. |
| **e1** | Specifies an E1 port. |
| *slot*/*port-1* | The location of the first T1 or E1 controller to be connected. Range for *slot* and *port* is 0 and 1. |
| *tdm-group-no-1* | The number identifier of the TDM) group associated with the first T1 or E1 controller port and created by using the **tdm-group** command. Range is from 0 to 23 for T1 and from 0 to 30 for E1. |
| *slot*/*port-2* | The location of the second T1 or E1 controller port to be connected. Range for *slot* is from 0 to 5, depending on the platform. Range for *port* is from 0 to 3, depending on the platform and the presence of a network module. |
| *tdm-group-no-2* | The number identifier of the TDM group associated with the second T1 or E1 controller and created by using the **tdm-group** command. Range is from 0 to 23 for T1 and from 0 to 30 for E1. |

## Command Default

There is no drop-and-insert connection between the ports.

## Command Modes

Global configuration

## Command History

| Release | Modification |
|---|---|
| 12.0(5)XK | The command was introduced on the Cisco 2600 series and Cisco 3600 series. |
| 12.0(7)T | This command was integrated into Cisco IOS Release 12.0(7)T. |
| 12.1(1)T | The command was modified to accommodate two channel groups on a port for 1- and 2-port T1/E1 multiflex voice/WAN interface cards (VWICs) on the Cisco 3600 series. |

## Usage Guidelines

The **connect** command creates a named connection between two TDM groups associated with drop-and-insert ports on T1 or E1 interfaces where you have already defined the groups by using the **tdm-group** command.

Once TDM groups are created on two different physical ports, use the **connect** command to start the passage of data between the ports. If a crosspoint switch is provided in the AIM slot, the connections can extend between ports on different cards. Otherwise, the connection is restricted to ports on the same VWIC.

The VWIC can make a connection only if the number of time slots at the source and destination are the same. For the connection to be error-free, the two ports must be driven by the same clock source; otherwise, slips occur.

## Examples

The following example shows a fractional T1 terminated on port 0 using time slots 1 through 8, a fractional T1 is terminated on port 1 using time slots 2 through 12, and time slots 13 through 20 from port 0 are connected to time slots 14 through 21 on port 1 by using the **connect** command:

```
controller t1 0/0
 channel-group 1 timeslots 1-8
 tdm-group 1 timeslots 13-20
 exit
controller t1 0/1
 channel-group 1 timeslots 2-12
 tdm-group 2 timeslot 14-21
 exit
 connect exampleconnection t1 0/0 1 t1 0/1 2
```

## Related Commands

| Command | Description |
|---|---|
| **show connect** | Displays configuration information about drop-and-insert connections that have been configured on a router. |
| **tdm-group** | Configures a list of time slots for creating clear channel groups (pass-through) for TDM cross-connect. |

# connect atm

To define connections between T1 or E1 controller ports and the ATM interface, enter the **connect atm** command in global configuration mode. Use the **no** form of this command to restore the default values.

**connect** *connection-id* **atm** *slot/port-1 virtual-circuit-name* | *vpi/vci* {**atm** | **T1** | **E1**} *slot/port-2 TDM-group-number* | {*virtual-circuit-name* | *vpi/vci*}

**no connect** *connection-id* **atm** *slot/port-1 virtual-circuit-name* | *vpi/vci* {**atm** | **T1** | **E1**} *slot/port-2 TDM-group-number* | {*virtual-circuit-name* | *vpi/vci*}

## Syntax Description

| | |
|---|---|
| *connection-id* | A name for this connection. |
| **atm** | Specifies the first ATM interface. |
| *slot/port-1* | The location of the ATM controller to be connected. |
| *virtual-circuit- name* | Specifies the permanent virtual circuit (PVC) or switched virtual circuit (SVC). |
| *vpi/vci* | Specifies a virtual path identifier (VPI) and virtual channel identifier (VCI). |
| **atm** | Specifies the second ATM interface. |
| **T1** | Specifies a T1 port. |
| **E1** | Specifies an E1 port. |
| *slot/port-2* | The location of the T1 or E1 controller to be connected. |
| *TDM-group-number* | The number identifier of the time-division multiplexing (TDM) group associated with the T1 or E1 controller port and created by using the **tdm-group** command. Range is 0 to 23 for T1 and 0 to 30 for E1. |

## Command Default

No default behavior or values

## Command Modes

Global configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(2)T | This command was introduced for ATM interfaces on the Cisco 2600 series and Cisco 3600 series. |
| 12.3(4)XD | ATM-to-ATM connections are allowed. |
| 12.3(7)T | Support for ATM-to-ATM connections was integrated into Cisco IOS Release 12.3(7)T. |

## Usage Guidelines

This command is used on Cisco 2600, Cisco 3600, and Cisco 3700 series routers to provide connections between T1/E1 and ATM interfaces. This command is used after all interfaces are configured.

After TDM groups are created on two different physical ports, you can use the **connect atm** command to start the passage of data between the ports. If a crosspoint switch is provided in the advanced integration module (AIM) slot, the connections can extend between ports on different cards. Otherwise, the connection is restricted to ports on the same VWIC card.

The VWIC can make a connection only if the number of time slots at the source and destination are the same. For the connection to be error free, the two ports must be driven by the same clock source; otherwise, slips occur.

## Examples

The following example shows how the ATM permanent virtual circuit (PVC) and T1 TDM group are set up and then connected:

```
interface atm 1/0
 pvc pvc1 10/100 ces
 exit
controller T1 1/1
 tdm-group 3 timeslots 13-24 type e&m
 exit
connect tdm1 atm 1/0 pvc1 10/100 T1 1/1 3
```

## Related Commands

| Command | Description |
|---|---|
| **tdm-group** | Creates TDM groups that can be connected. |
| **pvc** | Creates a private virtual circuit. |

# connect interval

To specify the amount of time that a given digital signal processor (DSP) farm profile waits before attempting to connect to a Cisco Unified CallManager when the current Cisco Unified CallManager fails to connect, use the **connect interval** command in SCCP Cisco Unified CallManager configuration mode. To reset to the default value, use the **no** form of this command.

**connect interval** *seconds*

**no connect interval**

## Syntax Description

| | |
|---|---|
| *seconds* | Timer value, in seconds. Range is 1 to 3600. Default is 60. |

## Command Default

60 seconds

## Command Modes

SCCP Cisco Unified CallManager configuration (config-sccp-ccm)

## Command History

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

## Usage Guidelines

The optimum setting for this command depends on the platform and your individual network characteristics. Adjust the connect interval value to meet your needs.

## Examples

The following example specifies that the profile attempts to connect to another Cisco Unified CallManager after 1200 seconds (20 minutes) when the current Cisco Unified CallManager connection fails:

```
Router(config-sccp-ccm)# connect interval 1200
```

## Related Commands

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

# connect retries

To specify the number of times that a digital signal processor (DSP) farm attempts to connect to a Cisco Unified CallManager when the current Cisco Unified CallManager connections fails, use the **connect retries** command in SCCP Cisco CallManager configuration mode. To reset this number to the default value, use the **no** form of this command.

**connect retries** *number*

**no connect retries**

## Syntax Description

| | |
|---|---|
| *number* | Number of connection attempts. Range is 1 to 32. Default is 3. |

## Command Default

3 connection attempts

## Command Modes

SCCP Cisco CallManager configuration

## Command History

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

## Usage Guidelines

The value of this command specifies the number of times that the given DSP farm attempts to connect to the higher-priority Cisco Unified CallManager before it gives up and attempts to connect to the next Cisco Unified CallManager.

**Note** The optimum setting for this command depends on the platform and your individual network characteristics. Adjust the connect retries value to meet your needs.

## Examples

The following example allows a DSP farm to make 5 attempts to connect to the Cisco Unified CallManager before giving up and attempting to connect to the next Cisco Unified CallManager specified in the group:

```
Router(config-sccp-ccm)# connect retries 5
```

## Related Commands

| Command | Description |
|---|---|
| **associate ccm** | Associates a Cisco Unified CallManager with a Cisco CallManager group and establishes its priority within the group. |
| **associate profile** | Associates a DSP farm profile with a Cisco CallManager group. |
| **bind interface** | Binds an interface to a Cisco CallManager group. |

| Command | Description |
|---|---|
| **connect interval** | Specifies how many times a given profile attempts to connect to the specific Cisco Unified CallManager. |
| **sccp ccm group** | Creates a Cisco CallManger group and enters SCCP Cisco CallManager configuration mode. |

# connection

To specify a connection mode for a voice port, use the **connection** command in voice-port configuration mode. To disable the selected connection mode, use the **no** form of this command.

**connection** {**plar** | **tie-line** | **plar opx** [**cut-through-wait** | **immediate**]} *phone-number* | {**trunk** *phone-number* [**answer-mode**]}

**connection** {**plar** | **tie-line** | **plar opx** [**cut-through-wait** | **immediate**]} *phone-number* | {**trunk** *phone-number* [**answer-mode**]}

## Syntax Description

| | |
|---|---|
| **plar** | Specifies a private line automatic ringdown (PLAR) connection. PLAR is an autodialing mechanism that permanently associates a voice interface with a far-end voice interface, allowing call completion to a specific telephone number or PBX without dialing. When the calling telephone goes off-hook, a predefined network dial peer is automatically matched, which sets up a call to the destination telephone or PBX. |
| **tie-line** | Specifies a connection that emulates a temporary tie-line trunk to a private branch exchange (PBX). A tie-line connection is automatically set up for each call and torn down when the call ends. |
| **plar opx** | Specifies a PLAR off-premises extension (OPX) connection. Using this option, the local voice port provides a local response before the remote voice port receives an answer. On Foreign Exchange Office (FXO) interfaces, the voice port does not answer until the remote side has answered. |
| **cut-through-wait** | (Optional) Specifies that the router waits for the off-hook signal before cutting through the audio path.<br>**Note** This keyword suppresses the subtle clicking sound that is heard when a phone goes off-hook. Users may have difficulty perceiving when the local FXO port has gone off-hook. |
| **immediate** | (Optional) Configures the FXO port to set up calls immediately (without waiting for Caller ID information) so the ring-cycle perception is identical for the caller and the called party. When the Caller ID is available, it is forwarded to the called number if the called party has not already answered the call.<br>**Note** This option cannot be configured on an FXO port that is configured as a Centralized Automatic Message Accounting (CAMA) port. |
| *phone-number* | Specifies the destination telephone number. Valid entries are any series of digits that specify the E.164 telephone number. |
| **trunk** | Specifies a connection that emulates a permanent trunk connection to a PBX. A trunk connection remains permanent in the absence of any active calls. |
| **answer-mode** | (Optional) Specifies that the router does not initiate a trunk connection but waits for an incoming call before establishing the trunk. Use only with the **trunk** keyword. |

## Command Default

No connection mode is specified, and the standard session application outputs a dial tone when the interface goes off-hook until enough digits are collected to match a dial peer and complete the call.

## Command Modes

Voice-port configuration (config-voiceport)

## Command History

| Release | Modification |
|---|---|
| 11.3(1)T | This command was introduced on the Cisco 3600 series. |
| 11.3(1)MA1 | This command was implemented on the Cisco MC3810, and the **tie-line** keyword added. |
| 11.3(1)MA5 | The **plar opx** keyword was implemented on the Cisco MC3810 as the **plar-opx-ringrelay** keyword. The keyword was shortened in a subsequent release. |
| 12.0(2)T | This command was integrated into Cisco IOS Release 12.0(2)T. |
| 12.0(3)XG | The **trunk** keyword was implemented on the Cisco MC3810. The **trunk answer-mode** option was added. |
| 12.0(4)T | This command was integrated in Cisco IOS Release 12.0(4)T. |
| 12.0(7)XK | This command was unified across the Cisco 2600, Cisco 3600, and Cisco MC3810. |
| 12.1(2)T | This command was integrated into Cisco IOS Release 12.1(2)T. |
| 12.3(8)T | The **cut-through-wait** keyword was added. |
| 12.4(11)XW | The **immediate** keyword was added. |
| 12.4(20)T | This command was integrated into Cisco IOS Release 12.4(20)T. |

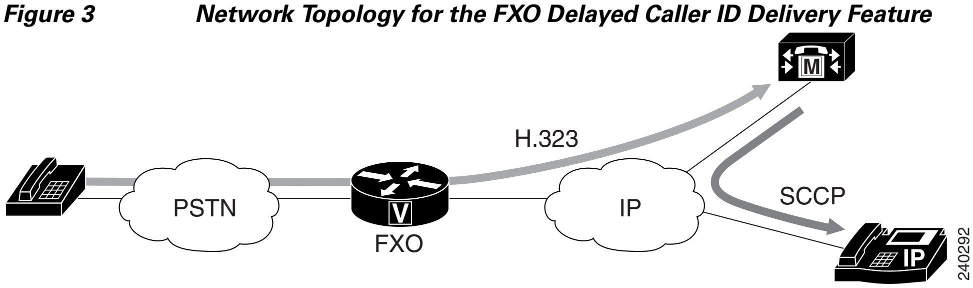
## Usage Guidelines

Use the **connection** command to specify a connection mode for a specific interface. For example, use the **connection plar** command to specify a PLAR interface. The string you configure for this command is used as the called number for all incoming calls over this connection. The destination peer is determined by the called number.

The **connection plar opx immediate** option enables FXO ports to set up calls with no ring discrepancy for Caller ID between the caller and the called party. To implement the FXO Delayed Caller ID Delivery feature, you must have a configured network with a Cisco 2800 or Cisco 3800 series integrated services router running Cisco IOS Release 12.4(11)XW. The integrated services router must have at least one voice interface card. Cisco CallManager Release 4.2.3 SR1 or later releases must be installed on the network to support this feature.

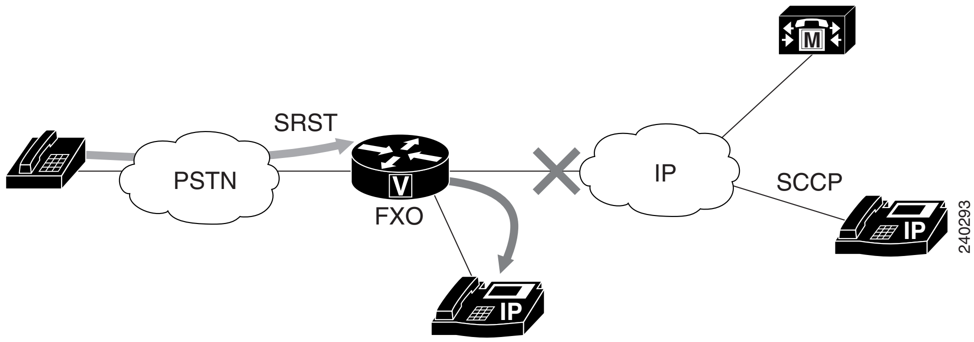
Figure 3 and Figure 4 show the network topology and call flow for the FXO Delayed Caller ID feature. The caller is in the PSTN, and the call arrives via an FXO port at the gateway. In Figure 3, the gateway is connected via H.323 to Cisco CallManager. Cisco CallManager extends the call to the called party which is a SCCP-based IP phone (Cisco 7941).

*Figure 3 Network Topology for the FXO Delayed Caller ID Delivery Feature*



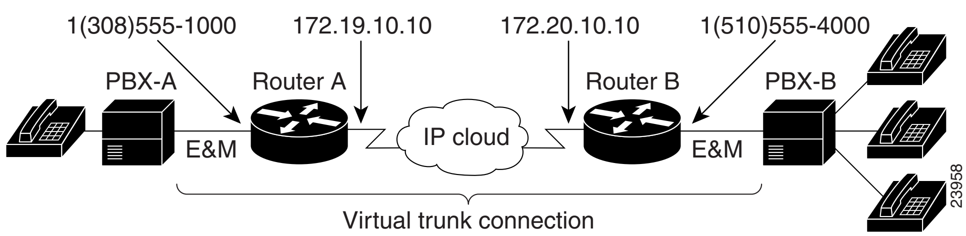
In Figure 4, the gateway is on the same router, and Survivable Remote Site Telephony (SRST) is active. SRST extends the call to the called party, which is a Skinny Client Control Protocol (SCCP)-based IP phone (Cisco 7941).

***Figure 4 Network Topology for the FXO Delayed Caller ID Delivery Feature (using SRST)***




Use the **connection trunk** command to specify a permanent tie-line connection to a PBX. VoIP simulates a trunk connection by creating virtual trunk tie lines between PBXs connected to Cisco devices on each side of a VoIP connection (see Figure 5). In this example, two PBXs are connected using a virtual trunk. PBX-A is connected to Router A via an E&M voice port; PBX-B is connected to Router B via an E&M voice port. The Cisco routers spoof the connected PBXs into believing that a permanent trunk tie line exists between them.

***Figure 5 Virtual Trunk Connection***




When configuring virtual trunk connections in VoIP, the following restrictions apply:

- You can use the following voice port combinations:
  - E&M to E&M (same type)
  - Foreign Exchange Station (FXS) to Foreign Exchange Office (FXO)
  - FXS to FXS (with no signaling)
- Do not perform number expansion on the destination pattern telephone numbers configured for trunk connection.
- Configure both end routers for trunk connections.

**Note** Because virtual trunk connections do not support number expansion, the destination patterns on each side of the trunk connection must match exactly.

To configure one of the devices in the trunk connection to act as slave and only receive calls, use the **answer-mode** option with the **connection trunk** command when configuring that device.

**Note** When using the **connection trunk** command, you must enter the **shutdown** command followed by the **no shutdown** command on the voice port.

VoIP establishes the trunk connection immediately after configuration. Both ports on either end of the connection are dedicated until you disable trunking for that connection. If for some reason the link between the two switching systems goes down, the virtual trunk reestablishes itself after the link comes back up.

Use the **connection tie-line** command when the dial plan requires you to add digits in front of any digits dialed by the PBX, and the combined set of digits is used to route the call onto the network. The operation is similar to the **connection plar** command operation, but in this case, the tie-line port waits to collect thedigits from the PBX. Tie-line digits are automatically stripped by a terminating port.

## Examples

The following example shows PLAR as the connection mode with a destination telephone number of 555-0100:

```
voice-port 1/0/0
 connection trunk 5550100
```

The following example shows the tie-line as the connection mode with a destination telephone number of 555-0100:

```
voice-port 1/1
 connection tie-line 5550100
```

The following example shows a PLAR off-premises extension connection with a destination telephone number of 555-0100:

```
voice-port 1/0/0
 connection plar-opx 5550100
```

The following example shows a trunk connection configuration that is established only when the trunk receives an incoming call:

```
voice-port 1/0/0
 connection trunk 5550100 answer-mode
```

The following example shows a PLAR off-premises extension connection with a destination telephone number of 0199. The router waits for the off-hook signal before cutting through the audio path:

```
voice-port 2/0/0
 connection plar opx 0199 cut-through-wait
```

The following examples show configuration of the routers on both sides of a VoIP connection (as illustrated in Figure 5) to support trunk connections.

### Router A

```
voice-port 1/0/0
 connection trunk +15105550190
dial-peer voice 10 pots
 destination-pattern +13085550181
 port 1/0/0
dial-peer voice 100 voip
 session-target ipv4:172.20.10.10
 destination-pattern +15105550190
```

#### Router B

```
voice-port 1/0/0
 connection trunk +13085550180
dial-peer voice 20 pots
 destination-pattern +15105550191
 port 1/0/0
dial-peer voice 200 voip
 session-target ipv4:172.19.10.10
 destination-pattern +13085550180
```

## Related Commands

| Command | Description |
|---|---|
| **destination-pattern** | Specifies the prefix or the full E.164 telephone number for a dial peer. |
| **dial peer voice** | Enters dial peer configuration mode and specifies the voice encapsulation type. |
| **session-protocol** | Establishes a session protocol for calls between the local and remote routers via the packet network. |
| **session-target** | Configures a network-specific address for a dial peer. |
| **shutdown** | Takes a specific voice port or voice interface card offline. |
| **voice-port** | Enters voice-port configuration mode. |

# connection-timeout

To configure the time in seconds for which a connection is maintained after completion of a communication exchange, use the **connection-timeout** command in settlement configuration mode. To return to the default value, use the **no** form of this command.

**connection-timeout** *seconds*

**no connection-timeout** *seconds*

## Syntax Description

| | |
|---|---|
| *seconds* | Time, in seconds, for which a connection is maintained after the communication exchange is completed. Range is from 0 to 86400; 0 means that the connection does not time out. The default is 3600 (1 hour). |

## Command Default

3600 seconds (1 hour)

## Command Modes

Settlement configuration

## Command History

| Release | Modification |
|---|---|
| 12.0(4)XH1 | This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco AS5300. |
| 12.0(7)T | This command was integrated into Cisco IOS Release 12.0(7)T. |

## Usage Guidelines

The router maintains the connection for the configured period in anticipation of future communication exchanges to the same server.

## Examples

The following example shows a connection configured to be maintained for 3600 seconds after completion of a communications exchange:

```
settlement 0
 connection-timeout 3600
```

## Related Commands

| Command | Description |
|---|---|
| **customer-id** | Sets the customer identification. |
| **device-id** | Sets the device identification. |
| **encryption** | Specifies the encryption method. |
| **max-connection** | Sets the maximum simultaneous connections. |
| **response-timeout** | Sets the response timeout. |
| **retry-delay** | Sets the retry delay. |
| **retry-limit** | Sets the connection retry limit. |

| Command | Description |
|---|---|
| **session-timeout** | Sets the session timeout. |
| **settlement** | Enters settlement configuration mode. |
| **show settlement** | Displays the configuration for all settlement server transactions. |
| **shutdown** | Brings up or shuts down the settlement provider. |
| **type** | Specifies the provider type. |
| **url** | Specifies the Internet service provider address. |

# copy flash vfc

To copy a new version of VCWare from the Cisco AS5300 universal access server motherboard to voice feature card (VFC) flash memory, use the **copy flash vfc** command in privileged EXEC mode.

**copy flash vfc** *slot-number*

## Syntax Description

| | |
|---|---|
| *slot-number* | Slot on the Cisco AS5300 in which the VFC is installed. Range is from 0 to 2. |

## Command Modes

Privileged EXEC

## Command History

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

## Usage Guidelines

Use the **copy flash vfc** command to use the standard copy user interface in order to copy a new version of VCWare from the Cisco AS5300 universal access server motherboard to VFC flash memory. The VFC is a plug-in feature card for the Cisco AS5300 universal access server and has its own Flash memory storage for embedded firmware. For more information about VFCs, refer to *Voice-over-IP Card.*

Once the VCWare file has been copied, use the **unbundle vfc** command to uncompress and install VCWare.

## Examples

The following example copies a new version of VCWare from the Cisco AS5300 universal access server motherboard to VFC flash memory:

```
Router# copy flash vfc 0
```

## Related Commands

| Command | Description |
|---|---|
| **copy tftp vfc** | Copies a new version of VCWare from a TFTP server to VFC flash memory. |
| **unbundle vfc** | Unbundles the current running image of VCWare or DSPWare into separate files. |

# copy tftp vfc

To copy a new version of VCWare from a TFTP server to voice feature card (VFC) flash memory, use the **copy tftp vfc** command in privileged EXEC mode.

**copy tftp vfc** *slot-number*

## Syntax Description

| | |
|---|---|
| *slot-number* | Slot on the Cisco AS5300 in which the VFC is installed. Range is from 0 to 2. There is no default. |

## Command Default

No default behavior or values

## Command Modes

Privileged EXEC

## Command History

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

## Usage Guidelines

Use the **copy tftp vfc** command to copy a new version of VCWare from a TFTP server to VFC flash memory. The VFC is a plug-in feature card for the Cisco AS5300 universal access server and has its own flash storage for embedded firmware. For more information about VFCs, refer to *Voice-over-IP Card*.

Once the VCWare file has been copied, use the **unbundle vfc** command to uncompress and install VCWare.

## Examples

The following example copies a file from the TFTP server to VFC flash memory:

```
Router# copy tftp vfc 0
```

## Related Commands

| Command | Description |
|---|---|
| **copy flash vfc** | Copies a new version of VCWare from the Cisco AS5300 motherboard to VFC flash memory. |
| **unbundle vfc** | Unbundles the current running image of VCWare or DSPWare into separate files. |

# corlist incoming

To specify the class of restrictions (COR) list to be used when a specified dial peer acts as the incoming dial peer, use the **corlist incoming** command in dial peer configuration mode. To clear the previously defined incoming COR list in preparation for redefining the incoming COR list, use the **no** form of this command.

**corlist incoming** *cor-list-name*

**no corlist incoming** *cor-list-name*

## Syntax Description

| | |
|---|---|
| *cor-list-name* | Name of the dial peer COR list that defines the capabilities that the specified dial peer has when it is used as an incoming dial peer. |

## Command Default

No default behavior or values.

## Command Modes

Dial peer configuration

## Command History

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

## Usage Guidelines

The **dial-peer cor list** and **member** commands define a set of capabilities (a COR list). These lists are used in dial peers to indicate the capability set that a dial peer has when it is used as an incoming dial peer (the **corlist incoming** command) or to indicate the capability set that is required for an incoming dial peer to make an outgoing call through the dial peer (the **corlist outgoing** command). For example, if dial peer 100 is the incoming dial peer and its incoming COR list name is list100, dial peer 200 has list200 as the outgoing COR list name. If list100 does not include all the members of list200 (that is, if list100 is not a superset of list200), it is not possible to have a call from dial peer 100 that uses dial peer 200 as the outgoing dial peer.

## Examples

In the following example, incoming calls from 526.... are blocked from being switched to outgoing calls to 1900.... because the COR list for the incoming dial peer (list2) is not a superset of the COR list for the outgoing dial peer (list1):

```
dial-peer list list1
 member 900call

dial-peer list list2
 member 800call
 member othercall

dial-peer voice 526 pots
 answer-address 408555....
 corlist incoming list2
 direct-inward-dial
```

```
dial-peer voice 900 pots
 destination pattern 1900.......
 direct-inward-dial
 trunkgroup 101
 prefix 333
 corlist outgoing list1
```

## Related Commands

| Command | Description |
|---|---|
| **corlist outgoing** | Specifies the COR list to be used by outgoing dial peers. |
| **dial-peer cor list** | Defines a COR list name. |
| **member** | Adds a member to a dial peer COR list. |

# corlist outgoing

To specify the class of restrictions (COR) list to be used by outgoing dial peers, use the **corlist outgoing** command in dial peer configuration mode. To clear the previously defined outgoing COR list in preparation for redefining the outgoing COR list, use the **no** form of this command.

**corlist outgoing** *cor-list-name*

**no corlist outgoing** *cor-list-name*

## Syntax Description

| | |
|---|---|
| *cor-list-name* | Required name of the dial peer COR list for outgoing calls to the configured number using this dial peer. |

## Command Default

No default behavior or values.

## Command Modes

Dial peer configuration

## Command History

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

## Usage Guidelines

If the COR list for the incoming dial peer is not a superset of the COR list for the outgoing dial peer, calls from the incoming dial peer cannot use that outgoing dial peer.

## Examples

In the following example, incoming calls from 526.... are blocked from being switched to outgoing calls to 1900.... because the COR list for the incoming dial peer (list2) is not a superset of the COR list for the outgoing dial peer (list1):

```
dial-peer list list1
member 900call

dial-peer list list2
 member 800call
 member othercall

dial-peer voice 526 pots
 answer-address 408555....
 corlist incoming list2
 direct-inward-dial

dial-peer voice 900 pots
 destination pattern 1900.......
 direct-inward-dial
 trunk group 101
 prefix 333
 corlist outgoing list1
```

# cptone

To specify a regional analog voice-interface-related tone, ring, and cadence setting for a voice port, use the **cptone** command in voice-port configuration mode. To disable the selected tone, use the **no** form of this command.

**cptone** *locale*

**no cptone** *locale*

## Syntax Description

| | |
|---|---|
| *locale* | Country-specific voice-interface-related default tone, ring, and cadence setting (for ISDN PRI and E1 R2 signaling). Keywords are shown in Table 15. The default keyword is **us** in Cisco IOS Release 12.0(4)T and later releases. |

## Command Default

The default keyword is **us** for all supported gateways and interfaces in Cisco IOS Release 12.0(4)T and later releases.

## Command Modes

Voice-port configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(1)T | This command was introduced on the Cisco 3600 series. |
| 11.3(1)MA | This command was modified. The full keyword names for the countries were first added on the Cisco MC3810. |
| 12.0(4)T | This command was modified. ISO 3166 two-letter country codes were added on the Cisco MC3810. |
| 12.1(5)XM | This command was modified. The following keywords were added: **eg**, **gh**, **jo**, **ke**, **lb**, **ng, np**, **pa, pk**, **sa**, and **zw**. |
| 12.2(2)T | This command was implemented on the Cisco 1750 and integrated into Cisco IOS Release 12.2(2)T. |
| 12.2(15)ZJ | This command was modified. The **c1** and **c2** keywords were added for the following platforms: Cisco 2610XM, Cisco 2611XM, Cisco 2620XM, Cisco 2621XM, Cisco 2650XM, Cisco 2651XM, Cisco 2691, Cisco 3640A, Cisco 3660, Cisco 3725, and Cisco 3745. |
| 12.3(4)T | This command was integrated into Cisco IOS Release 12.3(4)T. |
| 12.4(15)T | This command was modified. The following keywords were added: **ae**, **kw**, and **om**. |
| 15.0(1)M | This command was modified. The **cl** keyword was added. |
| 15.1(3)T | This command was modified. The **mt** keyword was added. |

## Usage Guidelines

This command defines the detection of call-progress tones generated at the local interface. It does not affect any information passed to the remote end of a connection, and it does not define the detection of tones generated at the remote end of a connection. Use the **cptone** command to specify a regional analog voice interface-related default tone, ring, and cadence setting for a specified voice port.

If your device is configured to support E1 R2 signaling, the E1 R2 signaling type (whether ITU, ITU variant, or local variant as defined by the **cas-custom** command) must match the appropriate pulse code modulation (PCM) encoding type as defined by the **cptone** command. For countries for which a **cptone** value has not yet been defined, you can try the following:

- If the country uses a-law E1 R2 signaling, use the **gb** value for the **cptone** command.
- If the country uses mu-law E1 R2 signaling, use the **us** value for the **cptone** command.

Table 15 lists valid entries for the *locale* argument.

***Table 15 Valid Command Entries for* locale *Argument***

| Country | cptone *locale* Command Entry | Country | cptone *locale* Command Entry |
|---|---|---|---|
| Argentina | **ar** | Lebanon | **lb** |
| Australia | **au** | Luxembourg | **lu** |
| Austria | **at** | Malaysia | **my** |
| Belgium | **be** | Malta | **mt** |
| Brazil | **br** | Mexico | **mx** |
| Canada | **ca** | Nepal | **np** |
| Chile | **cl** | Netherlands | **nl** |
| China | **cn** | New Zealand | **nz** |
| Colombia | **co** | Nigeria | **ng** |
| Custom 1[1] | **c1** | Norway | **no** |
| Custom 2[1] | **c2** | Oman | **om** |
| Czech Republic | **cz** | Pakistan | **pk** |
| Denmark | **dk** | Panama | **pa** |
| Egypt | **eg** | Peru | **pe** |
| Finland | **fi** | Philippines | **ph** |
| France | **fr** | Poland | **pl** |
| Germany | **de** | Portugal | **pt** |
| Ghana | **gh** | Russian Federation | **ru** |
| Great Britain | **gb** | Saudi Arabia | **sa** |
| Greece | **gr** | Singapore | **sg** |
| Hong Kong | **hk** | Slovakia | **sk** |
| Hungary | **hu** | Slovenia | **si** |
| Iceland | **is** | South Africa | **za** |
| India | **in** | Spain | **es** |
| Indonesia | **id** | Sweden | **se** |

*Table 15 Valid Command Entries for* locale *Argument (continued)*

| Country | cptone *locale* Command Entry | Country | cptone *locale* Command Entry |
|---|---|---|---|
| Ireland | **ie** | Switzerland | **ch** |
| Israel | **il** | Taiwan | **tw** |
| Italy | **it** | Thailand | **th** |
| Japan | **jp** | Turkey | **tr** |
| Jordan | **jo** | United Arab Emirates | **ae** |
| Kenya | **ke** | United States | **us** |
| Korea Republic | **kr** | Venezuela | **ve** |
| Kuwait | **kw** | Zimbabwe | **zw** |

1. Automatically configured the first time the XML file is downloaded to the gateway.

## Examples

The following example configures United States as the call-progress tone locale:

```
voice-port 1/0/0
 cptone us
```

The following example configures Brazil as the call-progress tone locale on a Cisco universal access server:

```
voice-port 1:0
 cptone br
 description Brasil Tone
```

## Related Commands

| Command | Description |
|---|---|
| **voice-port** | Enters voice-port configuration mode. |
| **cas-custom** | Customizes signaling parameters for a particular E1 or T1 channel group on a channelized line. |

# cptone call-waiting repetition interval

To set the call-waiting alert pattern on analog endpoints that are connected to Foreign Exchange Station (FXS) ports, use the **cptone call-waiting repetition interval** command in supplementary-service voice-port configuration mode. To return to the default behavior, use the **no** form of this command.

**cptone call-waiting repetition interval** *second*

**no cptone call-waiting repetition interval**

## Syntax Description

| | |
|---|---|
| *second* | Length of time, in seconds for the tone repetition interval. Range: 0 to 30. Default: 0. |

## Command Default

A single-beep tone is the default behavior.

## Command Modes

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

## Command History

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

## Usage Guidelines

Use the **cptone call-waiting repetition interval** command to set the call-waiting alert pattern on analog endpoints that are connected to FXS ports on a Cisco IOS voice gateway, such as a Cisco Integrated Services Router (ISR) or Cisco VG224 Analog Phone Gateway.

When configured, the ringtone periodically repeats with configured interval until either the user switches to the new call or the calling party hangs up.

## Examples

The following example shows how to set the call-waiting alert pattern on analog endpoints connected to port 2/0 on a Cisco VG224:

```
Router(config)# stcapp supplementary-services
Router(config-stcapp-suppl-serv)# port 2/0
Router(config-stcapp-suppl-serv-port)# cptone call-waiting repetition interval 20
Router(config-stcapp-suppl-serv-port)# end
```

## Related Commands

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

# credential load

To reload a credential file into flash memory, use the **credential load** command in privileged EXEC mode.

**credential load** *tag*

## Syntax Description

| | |
|---|---|
| *tag* | Number that identifies the credential (.csv) file to load. Range: 1 to 5. This is the number that was defined with the **authenticate credential** command. |

## Command Default

The credential file is not reloaded.

## Command Modes

Privileged EXEC

## Command History

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

## Usage Guidelines

This command provides a shortcut to reload credential files that were defined with the **authenticate credential** command.

Up to five .csv files can be configured and loaded into the system. The contents of these five files are mutually exclusive, that is, the username/password pairs must be unique across all the files. For Cisco Unified CME, these username/password pairs cannot be the same ones defined for SCCP or SIP phones with the **username** command.

## Examples

The following example shows how to reload credential file 3:

```
credential load 3
```

## Related Commands

| Command | Description |
|---|---|
| **authenticate (voice register global)** | Defines the authenticate mode for SIP phones in a Cisco Unified CME or Cisco Unified SRST system. |
| **username (ephone)** | Defines a username and password for SCCP phones. |
| **username (voice register pool)** | Defines a username and password for authenticating SIP phones. |

# credentials (SIP UA)

To configure a Cisco IOS Session Initiation Protocol (SIP) time-division multiplexing (TDM) gateway, a Cisco Unified Border Element (Cisco UBE), or Cisco Unified Communications Manager Express (Cisco Unified CME) to send a SIP registration message when in the UP state, use the **credentials** command in SIP UA configuration mode. To disable SIP digest credentials, use the **no** form of this command.

**credentials** {**dhcp** | **number** *number* **username** *username*} **password** [**0** | **7**] *password* **realm** *realm*

**no credentials** {**dhcp** | **number** *number* **username** *username*} **password** [**0** | **7**] *password* **realm** *realm*

## Syntax Description

| | |
|---|---|
| **dhcp** | (Optional) Specifies the Dynamic Host Configuration Protocol (DHCP) is to be used to send the SIP message. |
| **number** *number* | (Optional) A string representing the registrar with which the SIP trunk will register (must be at least four characters). |
| **username** *username* | A string representing the username for the user who is providing authentication (must be at least four characters). This option is only valid when configuring a specific registrar using the **number** keyword. |
| **password** | Specifies password settings for authentication. |
| **0** | (Optional) Specifies the encryption type as cleartext (no encryption). This is the default. |
| **7** | (Optional) Specifies the encryption type as encrypted. |
| *password* | A string representing the password for authentication. If no encryption type is specified, the password will be cleartext format. The string must be between 4 and 128 characters. |
| **realm** *realm* | (Optional) A string representing the domain where the credentials are applicable. |

## Command Default

SIP digest credentials are disabled.

## Command Modes

SIP UA configuration (sip-ua)

## Command History

| Release | Modification |
|---|---|
| 12.3(8)T | This command was introduced. |
| 12.4(22)T | This command was integrated into Cisco IOS Release 12.4(22)T. |
| 12.4(22)YB | This command was modified. The **dhcp** keyword was added and the **username** keyword and *username* argument were removed. |
| 15.0(1)M | This command was integrated into Cisco IOS Release 15.0(1)M. |
| 15.0(1)XA | This command was modified. The **number** keyword and *number* argument were added and the **username** keyword and *username* argument reintroduced to configure credentials for a given registrar when multiple registrars are configured. |
| 15.1(1)T | This command was integrated into Cisco IOS Release 15.1(1)T. |

## Usage Guidelines

The following configuration rules are applicable when credentials are enabled:

- Only one password is valid for all domain names. A new configured password overwrites any previously configured password.
- The password will always be displayed in encrypted format when the **credentials** command is configured and the **show running-config** command is used.

The **dhcp** keyword in the command signifies that the primary number is obtained via DHCP and the Cisco IOS SIP TDM gateway, Cisco UBE, or Cisco Unified CME on which the command is enabled uses this number to register or unregister the received primary number.

## Examples

The following example shows how to configure SIP digest credentials without specifying the password encryption type:

```
Router> enable
Router# configure terminal
Router(config)# sip-ua
Router(config-sip-ua)# credentials dhcp password MyPassword realm example.com
```

The following example shows how to configure SIP digest credentials using the encrypted format:

```
Router> enable
Router# configure terminal
Router(config)# sip-ua
Router(config-sip-ua)# credentials dhcp password 7 095FB01AA000401 realm example.com
```

The following example shows how to disable SIP digest credentials where the encryption type was specified:

```
Router> enable
Router# configure terminal
Router(config)# sip-ua
Router(config-sip-ua)# no credentials dhcp password 7 095FB01AA000401 realm example.com
```

The following example shows how to configure SIP digest credentials for two different realms without specifying the encryption type:

```
Router> enable
Router# configure terminal
Router(config)# sip-ua
Router(config-sip-ua)# credentials number 1111 username MyUser password MyPassword realm
MyLocation1.example.com
Router(config-sip-ua)# credentials number 1111 username MyUser password MyPassword realm
MyLocation2.example.com
```

## Related Commands

| Command | Description |
|---|---|
| **authentication (dial peer)** | Enables SIP digest authentication on an individual dial peer. |
| **authentication (SIP UA)** | Enables SIP digest authentication. |
| **localhost** | Configures global settings for substituting a DNS localhost name in place of the physical IP address in the From, Call-ID, and Remote-Party-ID headers of outgoing messages. |
| **registrar** | Enables Cisco IOS SIP TDM gateways to register E.164 numbers for FXS, EFXS, and SCCP phones on an external SIP proxy or SIP registrar. |
| **voice-class sip localhost** | Configures settings for substituting a DNS localhost name in place of the physical IP address in the From, Call-ID, and Remote-Party-ID headers of outgoing messages on an individual dial peer, overriding the global setting. |