

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 <i>password</i>	(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 <i>number</i>	(Optional) Network number. Establishes the calling-party network number in the CLID for this router.
second-number strip	(Optional) Removes a previously configured second network number from the CLID.
network-provided	(Optional) Allows you to set the screening indicator to reflect the number that was provided by the network.
override rdnis	(Optional; supported for POTS dial peers only) Overrides the CLID with the redirected dialed number identification service (RDNIS) if available.
pi-restrict	(Optional) Restricted progress indicator (PI). Causes removal of the calling-party number from the CLID when the PI is restricted.
restrict	(Optional) Restricts presentation of the caller ID in the CLID.
strip	(Optional) Strips the calling-party number from the CLID. <ul style="list-style-type: none"> name—(Optional) Calling-party name. Causes removal of the calling-party name from the CLID. 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	(Optional) 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)

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 { **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 (voice-service-voip)` 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 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 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 removes the calling party name and number from the calling-line identifier (CLID):

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

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

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

The following 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 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.

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		
<i>priority-number</i>	Priority of the clock source. Range is from 1 (high priority) to 4 (low priority). There is no default value.	
<i>interface</i>	Specifies the interface to supply the clock source.	
<i>slot/port</i>	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 configuration mode. To reset 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*]

no codec *codec* [**bytes** *payload-size*]

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

codec { *codec* [**bytes** *payload-size*] | **transparent** }

no codec { *codec* [**bytes** *payload-size*] | **transparent** }

Syntax Description		
codec		codec options available for various platforms are described in Table 11 .
bytes		(Optional) Specifies the number of bytes in the voice payload of each frame.
<i>payload-size</i>		(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 Multiservice IP-to-IP Gateway.
	Note	The transparent keyword is only available on the Cisco 2600 and 3600 Series Router and Cisco 7200 and 7500 Series Router platforms.

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 u-Law at 64,000 bps	Yes	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

Table 11 **Codec Support by Platform (continued)**

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
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
gsmfr —Global System for Mobile Communications Full Rate (GSMFR) at 13,200 bps. Note The gsmfr codec is configurable only on the Cisco AS5400 and AS5350 with MSAv6 dsps.	—	—	—	—

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.

Command Modes dial peer configuration

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 the Cisco MC3810, and the command was modified for VoFR dial peers.
	12.0(5)XE	Additional <i>codec</i> 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.

Release	Modification
12.0(7)T	This command was integrated into Cisco IOS Release 12.0.(7)T and implemented on the Cisco AS5800. Additional voice coder rates of speech were added. This release does not support the clear-channel keyword 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 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.

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.

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**, **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	VoIP	80, 160	160
g711ulaw	VoFR	40 to 240 in multiples of 40	240
	VoATM	40 to 240 in multiples of 40	240
g723ar53	VoIP	20 to 220 in multiples of 20	20
	VoFR	20 to 240 in multiples of 20	20
g723r53	VoATM	20 to 240 in multiples of 20	20
g723ar63	VoIP	24 to 216 in multiples of 24	24
	VoFR	24 to 240 in multiples of 24	24
g723r63	VoATM	24 to 240 in multiples of 24	24
g726r16	VoIP	20 to 220 in multiples of 20	40
	VoFR	10 to 240 in multiples of 10	60
	VoATM	10 to 240 in multiples of 10	60
g726r24	VoIP	30 to 210 in multiples of 30	60
	VoFR	15 to 240 in multiples of 15	90
	VoATM	30 to 240 in multiples of 15	90
g726r32	VoIP	40 to 200 in multiples of 40	80
	VoFR	20 to 240 in multiples of 20	120
	VoATM	40 to 240 in multiples of 20	120
g728	VoIP	10 to 230 in multiples of 10	40
	VoFR	10 to 240 in multiples of 10	60
	VoATM	10 to 240 in multiples of 10	60
g729abr8	VoIP	10 to 230 in multiples of 10	20
g729ar8	VoFR	10 to 240 in multiples of 10	30
g729br8	VoATM	10 to 240 in multiples of 10	30
g729r8			



Note

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

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

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 Multiservice IP-to-IP Gateway:

```
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

<i>high</i>	Specifies high complexity: two channels of any mix of codec.
<i>med</i>	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* | **pass-through**}

no codec {*codec-type* | **pass-through**}

Syntax Description

<i>codec-type</i>	Specifies the codec preferred. <ul style="list-style-type: none"> • g711alaw—G.711 a-law 64,000 bps. • g711ulaw—G.711 u-law 64,000 bps. • g729abr8—G.729 ANNEX A and B 8000 bps. • g729br8—G.729 ANNEX B 8,000 bps. • g729ar8—G.729 ANNEX A and R 8000 bps. • g729r8—G.729 8000 bps.
pass-through	Enables codec pass-through. Supported for transcoding and MTP profiles.

Command Default

Transcoding

- **g711alaw**
- **g711ulaw**
- **g729abr8**
- **g729ar8**

Conferencing

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

MTP

- **g711ulaw**

Command Modes

DSP farm profile configuration

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.

Usage Guidelines

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

Hardware MTPs support only G.711 a-law and G.711 u-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 RSVP agent on a 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. This enables video and data streams to be processed in addition to audio. 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 RTP sessions.

Examples

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

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

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.
shutdown (DSP Farm profile)	Disables a DSP farm profile.

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 <i>profile-number</i> as an ITU-T type.
custom	The <i>profile-number</i> as a custom type.
atmf	The <i>profile-number</i> as an Asynchronous Transfer Mode Forum (ATMF) type.
<i>profile-number</i>	The available <i>profile-number</i> selections depend on the profile type. For ITU-T: <ul style="list-style-type: none"> • 1 = G.711 u-law • 2 = G.711 u-law with silence insertion descriptor (SID) • 7 = G.711 u-law and G.729ar8 For ATMF: <ul style="list-style-type: none"> • 9 = Broadband Loop Emulation Services (BLES) support for VoAAL2 For custom: <ul style="list-style-type: none"> • 100 = G.711 u-law and G.726r32 • 110 = G.711 u-law, G.726r32, and G.729ar8
<i>codec</i>	Enter one codec for the DSP. The possible <i>codec</i> entries depend on the <i>profile-number</i> . The valid entries are as follows: <ul style="list-style-type: none"> • For ITU 1—g711 u-law • For ITU 2—g711 u-law • For ITU 7—g711 u-law or g729ar8 • For ATMF—g711 u-law • For custom 100—g711 u-law or g726r32 • For custom 110—g711 u-law or g726r32 or g729ar8 • For lossless compression—llcc

Command Default

ITU-T profile 1 (G.711 u-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 (Ilcc) 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 complexity

To specify call density and codec complexity according to the codec standard that is being used, use the **codec complexity** command in voice-card configuration mode. To reset the flex complexity default, use the **no** form of this command.

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

no codec complexity

Syntax Description

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 may be enabled. 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	<p>If you specify the high keyword to define the complexity, each DSP supports two voice channels encoded in any of the following formats:</p> <ul style="list-style-type: none"> • g711alaw—G.711 A-law 64,000 bps. • g711ulaw—G.711 u-law 64,000 bps. • g723ar53—G.723.1 Annex A 5300 bps. • g723ar63—G.723.1 Annex A 6300 bps. • g723r53—G.723.1 5300 bps. • g723r63—G.723.1 6300 bps. • g723r16—G.726 16,000 bps. • g726r24—G.726 24,000 bps. • g726r32—G.726 32,000 bps. • g728—G.728 16,000 bps. • g729r8—G.729 8000 bps. This is the default. • g729br8—G.729 Annex B 8000 bps. • fax relay—2400 bps, 4800 bps, 7200 bps, 9600 bps, 12 kbps, and 14.4 kbps. <p>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.</p>

medium	<p>If you specify the medium keyword to define the complexity, each DSP supports four voice channels encoded in any of the following formats:</p> <ul style="list-style-type: none"> • g711alaw—G.711 A-law 64,000 bps. • g711ulaw—G.711 u-law 64,000 bps. • g726r16—G.726 16,000 bps. • g726r24—G.726 24,000 bps. • g726r32—G.726 32,000 bps. • g729r8—G.729 Annex A 8000 bps. • g729br8—G.729 Annex B with Annex A 8000 bps. • fax relay—2400 bps, 4800 bps, 7200 bps, 9600 bps, 12 kbps, and 14.4 kbps. Fax relay is the default.
secure	<p>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:</p> <ul style="list-style-type: none"> • g711alaw—G.711 a-law 64,000 bps. • g711ulaw—G.711 u-law 64,000 bps. • g729—G.729 8000 bps. • g729A—G.729 8000 bps.

Command Default flex complexity

Command Modes Voice-card configuration

Command History	Release	Modification
	12.0(5)XK	This command was introduced 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.
	12.2(15)ZJ	This command was integrated into Cisco IOS Release 12.2(15)ZJ and the flex keyword was added.
	12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T.

Release	Modification
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. Secure codec complexity is supported for TI-549 DSP processing on the NM-HDV network module.

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 view codec complexity status.

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
```

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 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.
<i>frame_size</i>	(Optional) iLBC operating frame in milliseconds (ms). Valid entries are: <ul style="list-style-type: none"> • 20—20ms frames for 15.2kbps bit rate • 30—30ms frames for 13.33 kbps bit rate Default is 20.
bytes	(Optional) Specifies the number of bytes in the voice payload of each frame.
<i>payload_size</i>	(Optional) Number of bytes in the voice payload of each frame. Valid entries are: <ul style="list-style-type: none"> • For mode 20—38, 76, 114, 152, 190, 228. Default is 38. • 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.

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 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.
<i>modes-value</i>	(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 encaps 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 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 frame-size][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:

- **clear-channel**—Clear Channel 64,000 bps.
- **g711alaw**—G.711 a law 64,000 bps.
- **g711ulaw**—G.711 mu-law 64,000 bps.
- **g723ar53**—G.723.1 ANNEX-A 5300 bps.
- **g723ar63**—G.723.1 ANNEX-A 6300 bps.
- **g723r53**—G.723.1 5300 bps.
- **g723r63**—G.723.1 6300 bps.
- **g726r16**—G.726 16,000 bps
- **g726r24**—G.726 24,000 bps
- **g726r32**—G.726 32,000 bps.
- **g728**—G.728 16,000 bps.
- **g729abr8**—G.729 ANNEX-A and B 8000 bps.
- **g729br8**—G.729 ANNEX-B 8000 bps.
- **g729r8**—G.729 8000 bps.
- **gsmamr-nb**—Enables GSMAMR codec capability.
- **gsmfr**—Global System for Mobile Communications (GSM) Full Rate (GSMFR) 13,200 bps.



Note The **gsmfr** keyword is configurable only on the Cisco AS5350 and AS5400 with MSAv6 dsp.

- **ilbc**—internet Low Bitrate Codec (iLBC) at 13,330 bps or 15,200 bps.
- **transparent**—Enables codec capabilities to be passed transparently between endpoints.



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

mode	(Optional) For iLBC codecs only. Specifies the iLBC operating frame mode that is encapsulated in each packet.
<i>frame-size</i>	(Optional) For iLBC codecs only. iLBC operating frame in milliseconds (ms). Valid entries are the following: <ul style="list-style-type: none"> • 20—20ms frames for the following 15.2kbps bit rate • 30—30ms frames for the following 13.33 kbps bit rate Default is 20.
bytes	(Optional) Specifies that the size of the voice frame is in bytes.
<i>payload-size</i>	(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. Applicable only to GSMAMR-NB codec support.
encap rfc3267	(Optional) Sets the encapsulation value to comply with RFC 3267. 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 . Applicable only to GSMAMR-NB codec support.
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. 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). Applicable only to GSMAMR-NB codec support.

Command Default

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**.

Command Modes

Voice-class configuration

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	The codecs gsmefr and gsmfr were added.

Release	Modification
12.2(13)T3	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	The ilbc codec and mode keyword were added.
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.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.

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 13](#) describes the voice payload options and default values for the codecs and packet voice protocols.



Note

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

Table 13 Voice Payload-per-Frame Options and Defaults

Codec	Protocol	Voice Payload Options (in Bytes)	Default Voice Payload (in Bytes)
g711alaw g711ulaw	VoIP	80, 160	160
	VoFR	40 to 240 in multiples of 40	240
	VoATM	40 to 240 in multiples of 40	240
g723ar53 g723r53	VoIP	20 to 220 in multiples of 20	20
	VoFR	20 to 240 in multiples of 20	20
	VoATM	20 to 240 in multiples of 20	20
g723ar63 g723r63	VoIP	24 to 216 in multiples of 24	24
	VoFR	24 to 240 in multiples of 24	24
	VoATM	24 to 240 in multiples of 24	24
g726r16	VoIP	20 to 220 in multiples of 20	40
	VoFR	10 to 240 in multiples of 10	60
	VoATM	10 to 240 in multiples of 10	60
g726r24	VoIP	30 to 210 in multiples of 30	60
	VoFR	15 to 240 in multiples of 15	90
	VoATM	30 to 240 in multiples of 15	90
g726r32	VoIP	40 to 200 in multiples of 40	80
	VoFR	20 to 240 in multiples of 20	120
	VoATM	40 to 240 in multiples of 20	120
g728	VoIP	10 to 230 in multiples of 10	40
	VoFR	10 to 240 in multiples of 10	60
	VoATM	10 to 240 in multiples of 10	60

Table 13 Voice Payload-per-Frame Options and Defaults (continued)

Codec	Protocol	Voice Payload Options (in Bytes)	Default Voice Payload (in Bytes)
g729abr8	VoIP	10 to 230 in multiples of 10	20
g729ar8	VoFR	10 to 240 in multiples of 10	30
g729br8 g729r8	VoATM	10 to 240 in multiples of 10	30
ilbc	VoIP	For mode 20, 38, 76, 114, 152, 190, 228. For mode 30, 50, 100, 150, 200	38 50

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 encapsulation 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 Multiservice IP-to-IP gateway:

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



Note

You can only assign a preference value of 1 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 Multiservice IP-to-IP Gateway:

```
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.

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 u-law ITU-T PCM encoding standard.
a-law	Specifies the European a-law ITU-T PCM encoding standard.

Command Default

u-law (T1 digital)
a-law (E1 digital)

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810.

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 u-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-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	<i>cptone-name</i>	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	<p>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.</p> <ul style="list-style-type: none"> 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	<p>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.</p>
-----------------	---

```
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	<i>cptone-name</i>	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 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 (atm)

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

```
connect id atm slot/port-1 name-of-PVC_SVC | vpi/vci { atm | T1 | E1 } slot/port-2
TDM-group-number | { name-of-PVC | vpi/vci }
```

```
no connect id atm slot/port-1 name-of-PVC_SVC | vpi/vci { atm | T1 | E1 } slot/port-2
TDM-group-number | { name-of-PVC | vpi/vci }
```

Syntax Description

<i>id</i>	A name for this connection.
atm	Specifies the first ATM interface.
<i>slot/port-1</i>	The location of the ATM controller to be connected.
<i>name of PVC_SVC</i>	Specifies the permanent or switched virtual circuit.
<i>vpi/vci</i>	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.
<i>slot/port-2</i>	The location of the T1 or E1 controller to be connected.
<i>TDM-group-number</i>	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	This command 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** 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 voice/WAN interface card (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. Use with the l2transport keyword to specify that it is a switched PVC, not terminated.

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-name voice-port voice-port-number {t1 | e1} controller-number
ds0-group-number
```

```
no connect connection-name voice-port voice-port-number {t1 | e1} controller-number
ds0-group-number
```

Syntax Description

<i>connection-name</i>	A name for this connection.
voice-port	Specifies that a voice port is used in the connection.
<i>voice-port-number</i>	The voice port slot number and port number.
t1	Specifies a T1 port.
e1	Specifies an E1 port.
<i>controller-number</i>	The location of the first T1 or E1 controller to be connected. Valid values for the slot and port are 0 and 1.
<i>ds0-group-number</i>	The number identifier of the DS0 group associated with the first T1 or E1 controller port. 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 connect 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 fxo-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 id {t1 | e1} slot/port-1 tdm-group-no-1 {t1 | e1} slot/port-2 tdm-group-no-2
```

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

Syntax Description

<i>id</i>	A name for this connection.
t1	Specifies a T1 port.
e1	Specifies an E1 port.
<i>slot/port-1</i>	The location of the first T1 or E1 controller to be connected. Range for <i>slot</i> and <i>port</i> is 0 and 1.
<i>tdm-group-no-1</i>	The number identifier of the time-division multiplexing (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.
<i>slot/port-2</i>	The location of the second T1 or E1 controller port to be connected. Range for <i>slot</i> is from 0 to 5, depending on the platform. Range for <i>port</i> is from 0 to 3, depending on the platform and the presence of a network module.
<i>tdm-group-no-2</i>	The number identifier of the time-division multiplexing (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 card.

The VWIC card 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 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 CallManager configuration mode. To reset to the default value, use the **no** form of this command.

connect interval *seconds*

no connect interval

Syntax Description	<i>seconds</i>	Timer value, in seconds. Range is 1 to 3600. Default is 60.
---------------------------	----------------	---

Command Default	60 seconds
------------------------	------------

Command Modes	SCCP Cisco CallManager configuration
----------------------	--------------------------------------

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-cm) # connect interval 1200
```

Related Commands	Command	Description
	associate ccm	Associates a Cisco 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.
	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 CallManger group and enters SCCP Cisco 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	<i>number</i>	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-cm) # 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] } digits | { trunk digits
[answer-mode] }
```

```
no connection { plar | tie-line | plar opx [cut-through-wait | immediate] } digits | { trunk digits
[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. Note This command 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. Note This option cannot be configured on an FXO port that is configured as a Centralized Automatic Message Accounting (CAMA) port.
<i>digits</i>	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

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 was implemented on the Cisco MC3810.
	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 implemented.
	12.4(11)XW	The immediate keyword was implemented.

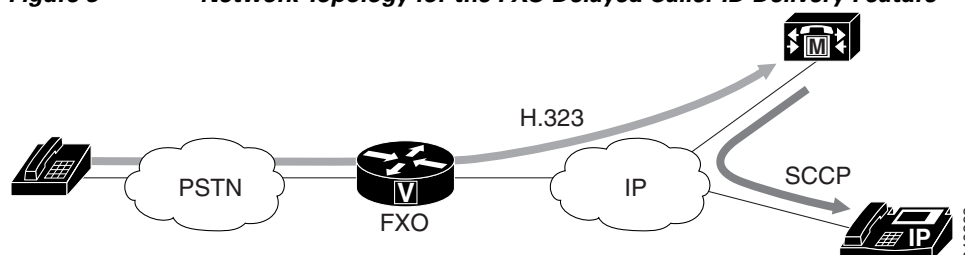
Usage Guidelines

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.

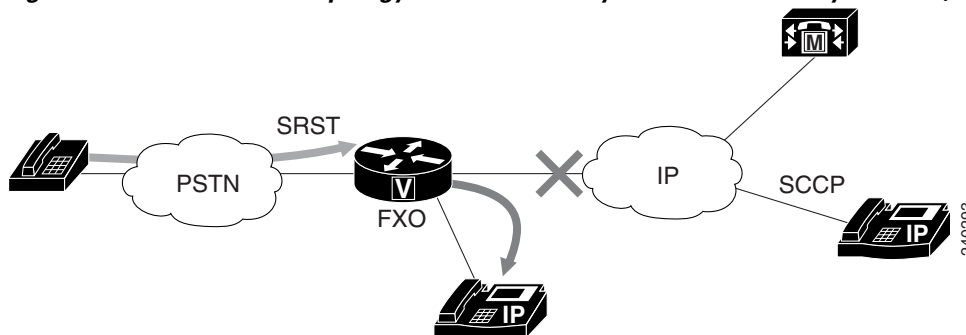
Figure 3 and Figure 4 show the network topology and call flow for the FXO Delayed Caller ID feature. The caller is in the PSTN, and the call arrives via an FXO port at the gateway. In Figure 3, the gateway is connected via H.323 to Cisco CallManager. Cisco CallManager extends the call to the called party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

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



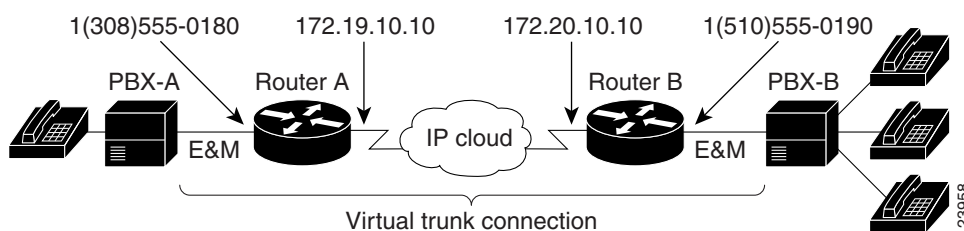
In [Figure 4](#), the gateway is on the same router, and Survivable Remote Site Telephony (SRST) is active. SRST extends the call to the called party, which is a Skinny Client Control Protocol (SCCP)-based IP phone (Cisco 7941).

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



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

- You can use the following voice port combinations:
 - Ear and mouth (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 additional 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 the digits 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	<i>seconds</i>	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	<i>slot-number</i>	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
	11.3 NA	This command was introduced on the Cisco AS5300.

Usage Guidelines	<p>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.</p> <p>Once the VCWare file has been copied, use the unbundle vfc command to uncompress and install VCWare.</p>
-------------------------	--

Examples	<p>The following example copies a new version of VCWare from the Cisco AS5300 universal access server motherboard to VFC Flash memory:</p> <pre>Router# copy flash vfc 0</pre>
-----------------	--

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	<i>slot-number</i>	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
	11.3 NA	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	<i>cor-list-name</i>	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 list_100, dial peer 200 has list_200 as the outgoing COR list name. If list_100 does not include all the members of list_200 (that is, if list_100 is not a superset of list_200), 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 900_call

dial-peer list list2
  member 800_call
  member other_call

dial-peer voice 526 pots
  answer-address 408526....
  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	<i>cor-list-name</i>	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 900_call

dial-peer list list2
member 800_call
member other_call

dial-peer voice 526 pots
answer-address 408526....
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
```

cptune

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

cptune {*locale*}

no cptune {*locale*}

Syntax Description

locale Voice-interface-related default tone, ring, and cadence setting for a specified country (for ISDN PRI and E1 R2 signaling). Keywords are shown in [Table 14](#) and [Table 15](#). The **default keyword is us in Cisco IOS Release 12.0(4)T and later**.

Command Default

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

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	The full keyword names for the countries were first supported on the Cisco MC3810.
12.0(4)T	ISO 3166 two-letter country codes were added on the Cisco MC3810.
12.1(5)XM	The following keywords were added: eg, gh, jo, ke, lb, np, ng, pk, pa, 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	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	The c1 and c2 keywords were integrated into Cisco IOS Release 12.3(4)T.

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 **cptune** 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 u-law E1 R2 signaling, use the **us** value for the **cptone** command.

Table 14 lists valid entries for the *locale* argument.

Table 14 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	Mexico	mx
Brazil	br	Nepal	np
Custom 1 ¹	c1	Netherlands	nl
Custom 2 ¹	c2	New Zealand	nz
Canada	ca	Nigeria	ng
China	cn	Norway	no
Colombia	co	Pakistan	pk
Czech Republic	cz	Panama	pa
Denmark	dk	Peru	pe
Egypt	eg	Philippines	ph
Finland	fi	Poland	pl
France	fr	Portugal	pt
Germany	de	Russian Federation	ru
Ghana	gh	Saudi Arabia	sa
Great Britain	gb	Singapore	sg
Greece	gr	Slovakia	sk
Hong Kong	hk	Slovenia	si
Hungary	hu	South Africa	za
Iceland	is	Spain	es
India	in	Sweden	se
Indonesia	id	Switzerland	ch
Ireland	ie	Taiwan	tw
Israel	il	Thailand	th
Italy	it	Turkey	tr
Japan	jp	United States	us
Jordan	jo	Venezuela	ve

Table 14 Valid Command Entries for locale Argument (continued)

Country	cptone locale Command Entry	Country	cptone locale Command Entry
Kenya	ke	Zimbabwe	zw
Korea Republic	kr		

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

Table 15 describes the *locale* keywords for ISDN PRI signaling.

Table 15 *cptone locale Argument Values for ISDN PRI Signaling*

Argument Value	Description
australia	Analog voice-interface-related default tone, ring, and cadence setting for Australia.
brazil	Analog voice-interface-related default tone, ring, and cadence setting for Brazil.
china	Analog voice-interface-related default tone, ring, and cadence setting for China.
finland	Analog voice-interface-related default tone, ring, and cadence setting for Finland.
france	Analog voice-interface-related default tone, ring, and cadence setting for France.
germany	Analog voice-interface-related default tone, ring, and cadence setting for Germany.
japan	Analog voice-interface-related default tone, ring, and cadence setting for Japan.
northamerica	Analog voice-interface-related default tone, ring, and cadence setting for North America.
sweden	Analog voice-interface-related default tone, ring, and cadence setting for Sweden.
unitedkingdom	Analog voice-interface-related default tone, ring, and cadence setting for the United Kingdom.

Examples

The following example configures United States as the call-progress tone locale beginning in global configuration mode:

```
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.

credential load

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

credential load *tag*

Syntax Description

<i>tag</i>	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.

