Cisco IOS Voice Command Reference - A through C

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aal2-profile custom

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To specify custom numbers and user-to-user information (UUI) code points for ATM adaptation layer 2 (AAL2) profiles and codecs, use the **aal2-profile custom**command in global configuration mode. To disable the configuration, use the **no** form of this command.

aal2-profile custom *number number number* {**clear-channel** | **g711alaw** | **g711ulaw** | **g726r32** | **g729br8** | **g720r8** | **llcc**} *packet-length minimum-UUI-codepoint maximum-UUI-codepoint*
no aal2-profile custom *number*

Syntax Description

number	AAL profile number. For more information, use the question mark (?) online help function.
clear-channel | **g711alaw** | **g711ulaw** | **g726r32** | **g729br8** | **g720r8** | **llcc**	Specifies the types of codec as follows: • Clear Channel • G.711 a-law • G.711-mu-law • G.726r32 • G.729 ANNEX-B 8000 bits per second • G.729 8000 bps • Lossless Compression
packet-length	Packet length in octets. The range is from 5 to 64.
minimum-UUI-codepoint	Minimim UUI code point. The range is from 0 to 15.
maximum-UUI-codepoint	Maximum UUI code point. The range is from 0 to 15.

Command Default

One of the predefined International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) profiles can be used.

Command Modes

Global configuration (config)

Command History

Release	Modification
15.0(1)M	This command was introduced in a release earlier than Cisco IOS Release 15.0(1)M.

Usage Guidelines

AAL2 custom profiles are used to define additional profiles that are not present in the ITU-T specifications.

After defining a custom profile, apply that profile under a Voice over ATM (VoATM) dial peer for it to take affect using the **codec aal2-profile** command. The **codec aal2-profile** command can be used only if the session protocol is "aal2-trunk".

Examples

The following example shows how to specify custom numbers and UUI code points for AAL2 profiles and codecs:

```
Router# configure terminal
Router(config)# aal2-profile custom  2 1 1 g711ulaw 6 3 3
```

aaa nas port voip

To send out the standard NAS-port attribute (RADIUS IETF Attribute 5) on voice interfaces, use the **aaa nas port voip** command in global configuration mode. To disable the command, use the **no** form of the command.

aaa nas port voip
no aaa nas port voip

Syntax Description

This command has no arguments or keywords.

Command Default

Disabled

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(11)T	This command was introduced on the Cisco AS5300.

Usage Guidelines

This command brings back the original behavior of the Authentication, Authorization, and Accounting (AAA). NAS-Port on VoIP interfaces. By default this feature is disabled.

Examples

The following example shows how to return to the original behavior of the AAA NAS-Port:

```
aaa nas port voip
```

Related Commands

Command	Description
aaa nas port extended	Replaces the NAS-port attribute with RADIUS IETF attribute 26 and displays extended field information.

aaa username

To determine the information with which to populate the username attribute for Authentication, Authorization, and Accounting (AAA). billing records, use the **aaa username**command in SIP user agent configuration mode. To achieve default capabilities, use the **no** form of this command.

aaa username {**calling-number** | **proxy-auth**}
no aaa username

Syntax Description

calling-number	Uses the FROM: header in the SIP INVITE (default value). This keyword is used in most implementations.
proxy-auth	Parses the Proxy-Authorization header. Decodes the Microsoft Passport user ID (PUID) and password, and then populates the PUID into the username attribute and a "." into the password attribute. The username attribute is used for billing, and the "." is used for the password, because the user has already been authenticated before this point.

Command Default

calling-number

Command Modes

SIP user agent configuration (config-sip-ua)

Command History

Release	Modification
12.2(2)XB	This command was introduced on the Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco AS5300, Cisco AS5350, and the Cisco AS5400.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T. This command does not support the Cisco AS5300, Cisco AS5350, and Cisco AS5400 in this release.
12.2(11)T	This command was integrated Cisco IOS Release 12.2(11)T and was implemented on the Cisco AS5850. This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5800 in this release.

Usage Guidelines

Parsing the Proxy-Authorization header, decoding the PUID and password, and populating the username attribute with the PUID must be enabled through this command. If this command is not issued, the Proxy-Authorization header is ignored.

The keyword **proxy-auth** is a nonstandard implementation, and Session Initiation Protocol (SIP) gateways do not normally receive or process the Proxy-Authorization header.

Examples

The following example enables the processing of the SIP username from the Proxy-Authorization header:

```
Router(config)# sip-ua
Router(config-sip-ua)# aaa username proxy-auth
```

Related Commands

Command	Description
show call active voice	Displays sactive call information for voice calls or fax transmissions in progress.
show call history voice	Displays the voice call history table.

access-list (voice source-group)

To assign an access list to a voice source group, use the **access-list** command in voice source-group configuration mode. To delete the access list, use the **no** form of this command.

access-list *access-list-number*
no access-list *access-list-number*

Syntax Description

access -list-number	Number of an access list. The range is from 1 to 99.

Command Default

No default behavior or values

Command Modes

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

Command History

Release	Modification
12.2(11)T	This command was introduced in voice source-group configuration mode.

Usage Guidelines

An access list defines a range of IP addresses for incoming calls that require additional scrutiny. Two related commands are used for voice source groups:

- Use the **access-list** *access-list-number* {**deny** | **permit**} *source*[*source-wildcard*] [**log**] command in global configuration mode to define the contents of the access list.
- Use the **access-list** *access-list-number*command in voice source-group configuration mode to assign the defined access list to the voice source group.

The terminating gateway uses the source IP group to identify the source of the incoming VoIP call before selecting an inbound dial peer. If the source is found in the access list, then the call is accepted or rejected, depending on how the access list is defined.

The terminating gateway uses the access list to implement call blocking. If the call is rejected, the terminating gateway returns a disconnect cause to the source. Use the **disconnect-cause** command to specify a disconnect cause to use for rejected calls.

Use the **show access-lists** privileged EXEC command to display the contents of all access lists.

Use the **show ipaccess-lis**t privileged EXEC command to display the contents of one access list.

Examples

The following example assigns access list 1 to voice source-group alpha. Access list 1 was defined previously using another command. An incoming source IP group call is checked against the conditions defined for access list 1 and is processed based on the permit or deny conditions of the access list.

```
Router(config)# voice source-group alpha
Router(cfg-source-grp)# access-list 1
```

Related Commands

Command	Description
carrier-id (dial-peer)	Specifies the carrier as the source of incoming VoIP calls (for carrier ID routing).
disconnect-cause	Specifies a cause for blocked calls.
h323zone-id (voice source group)	Associates a zone for an incoming H.323 call.
show access-lists	Displays the contents of all access lists.
show ip access-list	Displays the contents of one access list.
translation-profile (source group)	Associates a translation profile with incoming source IP group calls.
trunk-group-label (voice source group)	Specifies the trunk group as the source of incoming VoIP calls (for trunk group label routing).
voice source-group	Initiates the source IP group profile definition.

access-policy

To require that a neighbor be explicitly configured in order for requests to be accepted, use the **access-policy**command in Annex G configuration mode. To reset the configuration to accept all requests, use the **no** form of this command.

access-policy [**neighbors-only**]
no access-policy

Syntax Description

neighbors-only	(Optional) Requires that a neighbor be configured.

Command Default

Border elements accept any and all requests if service relationships are not configured.

Command Modes

Annex G configuration (config-annexg)

Command History

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

Usage Guidelines

Border elements accept any and all requests if service relationships are not configured. The **access-policy** command eliminates arbitrary requests from unknown border elements, and is a required prerequisite for configuring service relationships.

Examples

The following example shows how to enable the service relationship between border elements:

```
Router(config-annexg)# access-policy neighbors-only
```

Related Commands

Command	Description
call-router	Enables the Annex G border element configuration commands.
domain-name	Sets the domain name reported in service relationships.

access-secure

To specify that the secure (encrypted) mode is to be used for accessing the session border controller (SBC), use the **access-secure** command in phone proxy configuration mode. To remove the secure mode, use the **no** form of the command.

access-secure
no access-secure

This command has no arguments or keywords.

Command Default

The non-secure mode is used for communication with the SBC.

Command Modes

Phone proxy configuration mode (config-phone-proxy)

Command History

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

Usage Guidelines

Example

The following example shows how to specify that the secure (encrypted) mode is to be used for accessing the SBC:

```
Device(config)# voice-phone-proxy first-pp
Device(config-phone-proxy)# access-secure
```

accounting method

To set an accounting method at login for calls that come into a dial peer, use the **accounting method** command in voice class AAA configuration mode. To disable the accounting method set at login, use the **no** form of this command.

accounting method *MethListName* [**out-bound**]
no accounting method *MethListName* [**out-bound**]

Syntax Description

MethListName	Defines an accounting method list name.
out-bound	(Optional) Defines the outbound leg.

Command Default

When this command is not used to specify an accounting method, the system uses the **aaa accounting connection h323** command as the default . If the method list name is not specified, the outbound call leg uses the same method list name as the inbound call leg

Command Modes

Voice class AAA configuration

Command History

Release	Modification
12.2(11)T	This command was introduced on the Cisco 3660, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.

Usage Guidelines

This command sets the accounting method for dial peers in voice class AAA configuration mode. To initially define a method list, refer to the *Cisco IOS Security Configuration Guide,* Release 12.2.

If the outbound option is specified, the outbound call leg on the dial peer uses the method list name specified in the command. If the method list name is not specified, by default, the outbound call leg uses the same method list name as the inbound call leg.

Examples

The following example sets the dp-out method for the outbound leg:

```
voice class aaa 1
 accounting method dp-out out-bound
```

Related Commands

Command	Description
aaa accounting connection h323	Defines the accounting method list H.323 with RADIUS, using **stop-only** or **start-stop** accounting options.
voice class aaa	Enables dial-peer-based VoIP AAA configurations.

accounting suppress

To disable accounting that is automatically generated by a service provider module for a specific dial peer, use the **accounting suppress** command invoice class AAA configuration mode. To allow accounting to be automatically generated, use the **no** form of this command.

accounting suppress [{**in-bound** | **out-bound**}]
no accounting suppress [{**in-bound** | **out-bound**}]

Syntax Description

in-bound	(Optional) Defines the call leg for incoming calls.
out-bound	(Optional) Defines the call leg for outbound calls.

Command Default

Accounting is automatically generated by the service provider module.

Command Modes

Voice class AAA configuration

Command History

Release	Modification
12.2(11)T	This command was introduced on the Cisco 3660, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.

Usage Guidelines

If a call leg option is not specified by the command, accounting is disabled for both inbound and outbound calls. For accounting to be automatically generated in the service provider module, you must first configure**gw-accounting aaa**command in global configuration mode before configuring dial-peer-based accounting in voice class AAA configuration mode.

Examples

In the example below, accounting is suppressed for the incoming call leg.

```
voice class aaa 1
 accounting suppress in-bound
```

Related Commands

Command	Description
gw-accounting aaa	Enables VoIP gateway accounting.
suppress	Turns off accounting for a call leg on a POTS or VoIP dial peer.
voice class aaa	Enables dial-peer-based VoIP AAA configurations.

accounting template

To allow each dial peer to choose and send a customized accounting template to the RADIUS server, use the **accounting template** command in voice class AAA configuration mode. To disable the dial peer from choosing and sending a customized accounting template, use the **no** form of this command.

accounting template *acctTempName* [**out-bound**]
no accounting template *acctTempName* [**out-bound**]

Syntax Description

acctTempName	Defines an accounting template name.
out-bound	(Optional) Defines the outbound leg.

Command Default

The dial peer does not choose and send a customized accounting template to the RADIUS server.

Command Modes

Voice class AAA configuration

Command History

Release	Modification
12.2(11)T	This command was introduced on the Cisco 3660, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.

Usage Guidelines

By default, non-RFC-mandatory vendor-specific attributes (VSAs) are not included in accounting records if you do not configure the accounting template. The accounting template enables you to manage accounting records at a per-VSA level. When an accounting template is used for customizing the accounting record, the VSA name release source has to be included in the template file so that it is included in the accounting record and sent to the RADIUS server.

This command overrides the **acct-template** command in gateway accounting AAA configuration mode when a customized accounting template is used.

If you use a Tool Command Language (Tcl) script, the Tcl verb**aaa accounting start** [**-tacctTempName**] takes precedence over the**accounting template** command in voice class AAA configuration mode.

Examples

The following example sets the template temp-dp for the outbound leg

```
voice class aaa 1
 accounting template temp-dp out-bound
```

Related Commands

Command	Description
acct-template	Sends a selected group of voice accounting VSAs.
voice class aaa	Enables dial-peer-based VoIP AAA configurations.

acc-qos

To define the acceptable quality of service (QoS) for any inbound and outbound call on a VoIP dial peer, use the **acc-qos** command in dial-peer configuration mode. To restore the default QoS setting, use the **no** form of this command.

acc-qos {**best-effort** | **controlled-load** | **guaranteed-delay**} [{**audio** | **video**}]
no acc-qos

Syntax Description

best-effort	Indicates that Resource Reservation Protocol (RSVP) makes no bandwidth reservation. This is the default.
controlled-load	Indicates that RSVP guarantees a single level of preferential service, presumed to correlate to a delay boundary. The controlled load service uses admission (or capacity) control to ensure that preferential service is received even when the bandwidth is overloaded.
guaranteed-delay	Indicates that RSVP reserves bandwidth and guarantees a minimum bit rate and preferential queueing if the bandwidth reserved is not exceeded.
audio	(Optional) Configures acceptable QoS for audio traffic.
video	(Optional) Configures acceptable QoS for video traffic.

Command Default

RSVP makes no bandwidth reservations.

Command Modes

Dial-peer configuration (config-dial-peer)

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series routers.
12.1(5)T	The description of the command was modified.
12.3(4)T	The **audio** and **video** keywords were added.
Cisco IOS XE Release 3.3S	This command was integrated into Cisco IOS XE Release 3.3S.

Usage Guidelines

This command is applicable only to VoIP dial peers.

When VoIP dial peers are used, the Cisco IOS software uses RSVP to reserve a certain amount of bandwidth so that the selected QoS can be provided by the network. Call setup is aborted if the RSVP resource reservation does not satisfy the acceptable QoS for both peers.

To select the most appropriate value for this command, you need to be familiar with the amount of traffic this connection supports and what kind of impact you are willing to have on it. The Cisco IOS software generates a trap message when the bandwidth required to provide the selected quality of service is not available.

If **audio** or **video** is not configured, the bearer capability information element (IE) is not checked against max values during SETUP.

You must use the **iprsvpbandwidth** command to enable RSVP on an IP interface before you can specify RSVP QoS.

In order to use this command, you have to have the "req-qos" statement present.

Examples

The following example selects **guaranteed-delay**as the acceptable QoS for inbound and outbound audio calls on VoIP dial peer 10:

```
dial-peer voice 10 voip
 acc-qos guaranteed-delay
```

The following example selects **controlled-load** as the acceptable QoS for audio and video:

```
dial-peer voice 100 voip
 acc-qos controlled-load audio
 acc-qos controlled-load video
```

Related Commands

Command	Description
req-qos	Requests a particular QoS using RSVP to be used in reaching a specified dial peer in VoIP.
ip rsvp bandwidth	Enables Resource Reservation Protocol (RSVP) for IP on an interface.

acct-template

To select a group of voice attributes to collect in accounting records, use the **acct-template** command in gateway accounting AAA or gateway accounting file configuration mode. To disable collection of a group of voice attributes, use the **no** form of this command.

acct-template {*template-name* | **callhistory-detail**}
no **acct-template** {*template-name* | **callhistory-detail**}

Syntax Description

template-name	Name of the custom accounting template.
callhistory-detail	Collects all voice vendor-specific attributes (VSAs) for accounting.

Command Default

No voice attributes are collected.

Command Modes

Gateway accounting AAA configuration (config-gw-accounting-aaa)
Gateway accounting file configuration (config-gw-accounting-file)

Command History

Release	Modification
12.2(11)T	This command was introduced.
12.4(15)XY	This command was added to gateway accounting file configuration mode.
12.4(20)T	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to collect only the voice attributes that are defined in an accounting template. The accounting template is a text file that you create by selecting specific attributes that are applicable to your billing needs. Use the **call accounting-template voice** command to define your accounting template before using the**acct - template** command.

The **show call accounting-template voice** command displays all the voice attributes that can be filtered by accounting templates.

Use the **callhistory-detail** keyword to send all voice VSAs to the accounting server. For a description of supported voice VSAs, see the "VSAs Supported by Cisco Voice Products" section in the *RADIUS VSA Voice Implementation Guide* .

When you send only those VSAs defined in your accounting template, the default call-history records that are created by the service provider are automatically suppressed.

Examples

The example below uses the **acct-template** command to specify temp-global, a custom template.

```
gw-accounting aaa
 acct-template temp-global
```

Related Commands

Command	Description
call accounting-template voice	Defines a customized accounting template.
gw-accounting	Enables the method of collecting accounting data.
show call accounting-template voice	Displays attributes defined in accounting templates.

activation-key

To define an activation key that can be dialed by phone users to activate Call Back on Busy on an analog phone, use the **activation-key** command in STC application feature callback configuration mode. To return the code to its default, use the **no** form of this command.

activation-key *string*
no activation-key

Syntax Description

string	Character string that can be dialed on a telephone keypad (0-9, *, #). Length of string is one to five characters. Default: #1.

Command Default

Callback activation key is #1.

Command Modes

STC application feature callback configuration (config-stcapp-callback)

Command History

Release	Modification
12.4(20)YA	This command was introduced.
12.4(22)T	This command was integrated into Cisco IOS Release 12.4(22)T.

Usage Guidelines

This command changes the value of the callback activation key for Call Back on Busy from the default (#1) to the specified value.

To display information about the Call Back configuration, use the **show stcapp feature codes** command.

Examples

The following example shows how to change the value of the callback activation key sequence from the default (#1) to a new value (*22).

```
Router(config)# stcapp feature callback
Router(config-stcapp-callback)# activation-key *22
Router(config-stcapp-callback)#
```

The following partial output from the **show stcapp feature codes** command displays values for the call back feature:

```
Router# show stcapp feature codes

.
.
.
  stcapp feature callback
    key *1
    timeout 30
```

Related Commands

Command	Description
ringing-timeout	Defines the timeout period for Callback on Busy.
show stcapp feature codes	Displays all feature codes for FACs, FSDs, and call back.

address-family (tgrep)

To set the global address family to be used on all dial peers, use the **address-family**command in TGREP configuration mode. To change back to the default address family, use the **no** form of this command.

address family {**e164** | **decimal** | **penta-decimal**}
no address family {**e164** | **decimal** | **penta-decimal**}

Syntax Description

e164	E.164 address family.
decimal	Digital address family.
penta-decimal	Pentadecimal address family.

Command Default

E.164 address family

Command Modes

TGREP configuration (config-tgrep)

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

The E. 164 address family is used if the telephony network is a public telephony network. Decimal and pentadecimal options can be used to advertise private dial plans. For example, if a company wants to use TRIP in within its enterprise telephony network using five-digit extensions, then the gateway would advertise the beginning digits of the private numbers as a decimal address family. These calls cannot be sent out of the company's private telephony network because they are not E.164-compliant.

The pentadecimal family allows numbers 0 through 9 and alphabetic characters A through E and can be used in countries where letters are also carried in the called number.

Examples

The following example shows that the address family for itad 1234 is set for E.164 addresses:

```
Router(config)# tgrep local-itad 1234
Router(config-tgrep)# address family e164
```

Related Commands

Command	Description
tgrep local-itad	Enters TGREP configuration mode and defines an ITAD.

address-hiding

To hide signaling and media peer addresses from endpoints other than the gateway, use the **address-hiding** command in voice service voip configuration mode. To allow the peer address known to all endpoints, use the **no** form of this command.

address-hiding
no address-hiding

Syntax Description

There are no keywords or arguments.

Command Default

Signaling and media addresses are visible to all endpoints.

Command Modes

Voice service voip configuration (config-voi-serv)

Command History

Release	Modification
12.4(9)T	This command was introduced.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

All SIP methods or messages must terminate at IP-to-IP gateway and re-originate with IP-to-IP gateway address, address-hiding makes the peer address known only to the IP-to-IP gateway. Hiding address in flow-through mode is required for SIP-to-SIP in an IP-to-IP gateway network.

This command modifies specific supplementary service headers and changes them appropriately from the in-leg to the out-leg.

Those headers include the following:

- Refer-To
- Referred-by
- 3xx response contact header
- History-Info
- Diversion

In these headers, an inside IP would be silently passed from the in-leg to the out-leg by an IP-to-IP gateway resulting in inside IP being sent to the ITSP/Public Internet. When configured with address-hiding the IP-to-IP gateway specifically looks for and changes those headers appropriately to mask the inside IP with its own.

Note Distinctive ringing headers include ringing information and server address where the ringtone can be obtained. These headers are forwarded as is to the peer side even if address hiding is enabled.

Examples

The following example shows address-hiding being configured for all VoIP calls:

```
Router(config)# voice service voip
Router(config-voi-serv)# address-hiding
```

Related Commands

Command	Description
voice service	Enters voice service configuration mode.

advertise (annex g)

To control the types of descriptors that the border element (BE) advertises to its neighbors, use the **advertise** command in Annex G configuration mode. To reset this command to the default value, use the **no** form of this command.

advertise [{**static** | **dynamic** | **all**}]
no advertise

Syntax Description

static	(Optional) Only the descriptors provisioned on this BE is advertised. This is the default.
dynamic	(Optional) Only dynamically learned descriptors is advertised.
all	(Optional) Both static and dynamic descriptors are advertised.

Command Default

Static

Command Modes

Annex G configuration (config-annexg)

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T. Support for the Cisco AS5300 universal access server, Cisco AS5350, Cisco AS5400 is not included in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850 universal gateway.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Examples

The following example configures a BE that advertises both static and dynamic descriptors to its neighbors:

```
Router(config)# call-router h323-annexg be20
Router(config-annexg)# advertise all
```

Related Commands

Command	Description
call-router	Enables the Annex G border element configuration commands.
show call history	Displays the routes stored in cache for the BE.
show call-router status	Displays the Annex G BE status.

advertise (tgrep)

To turn on reporting for a specified address family, use the **advertise** command in TGREP configuration mode. To turn off reporting for a specified address family, use the **no** form of this command.

advertise {**e164** | **decimal** | **penta-decimal**} [**csr**] [**ac**] [**tc**] [{**trunk-group** | **carrier**}]
advertise {**trunk-group** | **carrier**} [**csr**] [**ac**] [**tc**]
no advertise {**e164** | **decimal** | **penta-decimal** | **trunk-group** | **carrier**}

Syntax Description

e164	E.164 address family.
decimal	Decimal address family
penta-decimal	Penta-decimal address family
trunk-group	(Optional) Trunk group address family
carrier	(Optional) Carrier code address family
csr	(Optional) Call success rate
ac	(Optional) Available circuits
tc	(Optional) Total circuits

Command Default

No attributes for address families are advertised.

Command Modes

TGREP configuration (config-tgrep)

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

If you specify **e164**, **decimal** or **penta-decimal** for the address family, you can stipulate whether the related **carrier** or **trunk-group** parameters are advertised. If you stipulate **carrier** or **trunk-group** for the address family, you can stipulate that the related address family prefix is advertised. If you stipulate **carrier** or **trunk-group** for the address family, you cannot stipulate **carrier** or **trunk-group** attributes for advertising.

When the **no** version of this command is used, it turns off the advertisement of that particular address family altogether.

Examples

The following example shows that the E.164 address family with call success rate, available circuits, total circuits, and trunk group attributes is being advertised for ITAD 1234:

```
Router(config)# tgrep local-itad 1234
Router(config-tgrep)# advertise e164 csr ac tc trunk-group
```

Related Commands

Command	Description
tgrep local-itad	Enters TGREP configuration mode and defines an ITAD.

alarm-trigger

To configure a T1 or E1 controller to send an alarm to the public switched telephone network (PSTN) or switch if specified T1 or E1 DS0 groups are out of service, use the **alarm-trigger** command in controller configuration mode. To configure a T1 or E1 controller not to send an alarm, use the **no** form of this command.

alarm-trigger blue *ds0-group-list*
no alarm-trigger

Syntax Description

blue	Specifies the alarm type to be sent is "blue," also known as an Alarm Indication Signal (AIS).
ds0-group-list	Specifies the DS0 group or groups to be monitored for permanent trunk connection status or busyout status.

Command Default

No alarm is sent.

Command Modes

Controller configuration (config-controller)

Command History

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

Usage Guidelines

Any monitored time slot can be used for either permanent trunk connections or switched connections. Permanent virtual circuits (PVCs) and switched virtual circuits (SVCs) can be combined on a T1 or E1 controller and monitored for alarm conditioning.

An alarm is sent only if all of the time slots configured for alarm conditioning on a T1 or E1 controller are out of service. If one monitored time slot remains in service or returns to service, no alarm is sent.

Examples

The following example configures T1 0 to send a blue (AIS) alarm if DS0 groups 0 and 1 are out of service:

```
controller t1 0
 alarm-trigger blue 0,1
 exit
```

Related Commands

Command	Description
busyout monitor	Configures a voice port to monitor an interface for events that would trigger a voice-port busyout.
connection trunk	Creates a permanent trunk connection (private line or tie-line) between a voice port and a PBX.
voice class permanent	Creates a voice class for a Cisco or FRF-11 permanent trunk.

alias static

To create a static entry in the local alias table, use the **alias static** command in gatekeeper configuration mode. To remove a static entry, use the **no** form of this command.

alias static *ip-signaling-addr* [*port*] **gkid** *gatekeeper-name* [**ras** *ip-ras-addr port*] [{**terminal** | **mcu** | **gateway** {**h320** | **h323-proxy** | **voip**}}] [**e164** *e164-address*] [**h323id** *h323-id*]
no alias static *ip-signaling-addr* [*port*] **gkid** *gatekeeper-name* [**ras** *ip-ras-addr port*] [{**terminal** | **mcu** | **gateway** {**h320** | **h323-proxy** | **voip**}}] [**e164** *e164-address*] [**h323id** *h323-id*]

Syntax Description

ip-signaling-addr	IP address of the H.323 node, used as the address to signal when establishing a call.
port	(Optional) Port number other than the endpoint Call Signaling well-known port number (1720).
gkid *gatekeeper-name*	Name of the local gatekeeper of whose zone this node is a member.
ras *ip-ras-addr*	(Optional) Node remote access server (RAS) signaling address. If omitted, the *ip-signaling-addr* parameter is used in conjunction with the RAS well-known port.
port	(Optional) Port number other than the RAS well-known port number (1719).
terminal	(Optional) Indicates that the alias refers to a terminal.
mcu	(Optional) Indicates that the alias refers to a multiple control unit (MCU).
gateway	(Optional) Indicates that the alias refers to a gateway.
h320	(Optional) Indicates that the alias refers to an H.320 node.
h323-proxy	(Optional) Indicates that the alias refers to an H.323 proxy.
voip	(Optional) Indicates that the alias refers to VoIP.
e164 *e164-address*	(Optional) Specifies the node E.164 address. This keyword and argument can be used more than once to specify as many E.164 addresses as needed. Note that there is a maximum number of 128 characters that can be entered for this address. To avoid exceeding this limit, you can enter multiple **alias static** commands with the same call signaling address and different aliases.
h323id *h323-id*	(Optional) Specifies the node H.323 alias. This keyword and argument can be used more than once to specify as many H.323 identification (ID) aliases as needed. Note that there is a maximum number of 256 characters that can be entered for this address. To avoid exceeding this limit, you can enter multiple **alias static** commands with the same call signaling address and different aliases.

Command Default

No static aliases exist.

Command Modes

Gatekeeper configuration (config-gk)

Command History

Release	Modification
11.3(2)NA	This command was introduced on the Cisco 2500 series and Cisco 3600 series.
12.0(3)T	This command was integrated into Cisco IOS Release 12.0(3)T.

Usage Guidelines

The local alias table can be used to load static entries by performing as many of the commands as necessary. Aliases for the same IP address can be added in different commands, if required.

Typically, static aliases are needed to access endpoints that do not belong to a zone (that is, they are not registered with any gatekeeper) or whose gatekeeper is inaccessible.

Examples

The following example creates a static terminal alias in the local zone:

```
zone local gk.zone1.com zone1.com
alias static 192.168.8.5 gkid gk.zone1.com terminal e164 14085551212 h323id terminal1
```

allow-connections

To allow connections between specific types of endpoints in a VoIP network, use the **allow-connections** command in voice service configuration mode. To refuse specific types of connections, use the **no** form of this command.

allow-connections *from-type* **to** *to-type*
no allow-connections *from-type* **to** *to-type*

Syntax Description

from-type	Originating endpoint type. The following choices are valid: • **h323** --H.323. • **sip** --Session Interface Protocol (SIP).
to	Indicates that the argument that follows is the connection target.
to-type	Terminating endpoint type. The following choices are valid: • **h323** --H.323. • **sip** --Session Interface Protocol (SIP).

Command Default

H.323-to-H.323 connections are enabled by default and cannot be changed, and POTS-to-any and any-to-POTS connections are disabled.

H.323-to-H.323 connections are disabled by default and can be changed, and POTS-to-any and any-to-POTS connections are enabled.

H.323-to-SIP and SIP-to-H.323 connections are disabled by default, and POTS-to-any and any-to-POTS connections are enabled.

SIP-to-SIP connections are disabled by default, and POTS-to-any and any-to-POTS connections are enabled.

Command Modes

Voice-service configuration (config-voi-serv)

Command History

Cisco IOS Release	Modification
12.2(13)T3	This command was introduced.
12.3(7)T	The default was changed.
12.3(11)T	The **sip** endpoint option was introduced for use with Cisco CallManager Express.
12.4(4)T	This command was modified. The **sip** endpoint option was implemented for use in IP-to-IP gateway networks.
12.2(33)SB	This command was integrated into Cisco IOS Release 12.2(33)SB.
12.4(22)T	Support for IPv6 was added.
Cisco IOS XE Release 2.5	This command was integrated into Cisco IOS XE Release 2.5.

Cisco IOS Release	Modification
Cisco IOS XE Amsterdam 17.2.1r	Introduced support for YANG models.

Usage Guidelines

Cisco IOS Release 12.3(4)T, Cisco IOS Release 12.3, and Earlier Releases

This command is used to allow connections between specific types of endpoints in a Cisco multiservice IP-to-IP gateway. The command is enabled by default and cannot be changed. Connections to or from POTS endpoints are not allowed. Only H.323-to-H.323 connections are allowed.

Cisco IOS Release 12.3(7)T and Later Releases

This command is used with Cisco Unified Communications Manager Express 3.1 or later systems and with the Cisco Multiservice IP-to-IP Gateway feature. In Cisco Unified Communications Manager Express, the **allow-connections**command enables the VoIP-to-VoIP connections used for hairpin call routing or routing to an H.450 tandem gateway.

Examples

The following example specifies that connections between H.323 and SIP endpoints are allowed:

```
Router(config-voi-serv)# allow-connections h323 to sip
```

The following example specifies that connections between H.323 endpoints are allowed:

```
Router(config-voi-serv)# allow-connections h323 to h323
```

The following example specifies that connections between SIP endpoints are allowed:

```
Router(config-voi-serv)# allow-connections sip to sip
```

Related Commands

Command	Description
voice service	Enters voice service configuration mode.

allow subscribe

To allow internal watchers to monitor external presentities, use the **allow subscribe** command in presence configuration mode. To disable external watching, use the **no** form of this command.

allow subscribe
no allow subscribe

Syntax Description

This command has no arguments or keywords.

Command Default

Only internal presentities can be watched when presence is enabled.

Command Modes

Presence configuration (config-presence)

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 allows internal watchers to receive Busy Lamp Field (BLF) status notification for external directory numbers on a remote router connected through a SIP trunk. An external directory number must be enabled as a presentity with the **allow watch** command.

The router sends SUBSCRIBE requests through the SIP trunk to an external presence server on behalf of the internal watcher and returns presence status to the watcher. To permit the external directory numbers to be watched, you must enable the **watcher all** command on the remote router.

Examples

The following example shows how to enable internal watchers to monitor external presentities:

```
Router(config)# presence
Router(config-presence)# allow subscribe
```

Related Commands

Command	Description
allow watch	Allows a line on a phone registered to Cisco Unified CME to be watched in a presence service.
blf-speed-dial	Enables BLF monitoring for a speed-dial number on a phone registered to Cisco Unified CME.
presence	Enables presence service on the router and enters presence configuration mode.
presence call-list	Enables BLF monitoring for call lists and directories on phones registered to Cisco Unified CME.
presence enable	Allows incoming presence requests from SIP trunks.

Command	Description
server	Specifies the IP address of a presence server for sending presence requests from internal watchers to external presence entities.
show presence global	Displays configuration information about the presence service.
show presence subscription	Displays information about active presence subscriptions.
watcher all	Allows an external watcher to monitor an internal presentity.

alt-dial

To configure an alternate dial-out string for dial peers, use the **alt-dial** command in dial-peer configuration mode. To delete the alternate dial-out string, use the **no** form of this command.

alt-dial *string*
no alt-dial *string*

Syntax Description

string	The alternate dial-out string.

Command Default

No alternate dial-out string is configured

Command Modes

Dial-peer configuration (config-dial-peer)

Command History

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

Usage Guidelines

This command applies to plain old telephone service (POTS), Voice over Frame Relay (VoFR), and Voice ATM (VoATM) dial peers.

The **alt-dial** command is used for the on-net-to-off-net alternative dialing function. The string replaces the destination-pattern string for dialing out.

Examples

The following example configures an alternate dial-out string of 95550188:

```
alt-dial 95550188
```

anat

To enable Alternative Network Address Types (ANAT) on a Session Initiation Protocol (SIP) trunk, use the **anat** command in voice service SIP configuration mode, or or voice class tenant, or dial peer configuration mode. To disable ANAT on SIP trunks, use the **no** form of this command.

anat system
no anat system

Syntax Description

This command has no arguments or keywords.

Command Default

ANAT is enabled on SIP trunks.

Command Modes

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

Dial peer configuration (config-dial-peer)

Voice class tenant configuration (config-class)

Command History

Release	Modification
12.4(22)T	This command was introduced.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: **system**. This command is now available under voice class tenants.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Both the Cisco IOS SIP gateway and the Cisco Unified Border Element are required to support Session Description Protocol (SDP) ANAT semantics for SIP IPv6 sessions. SDP ANAT semantics are intended to address scenarios that involve different network address families (for example, different IP versions). Media lines grouped using ANAT semantics provide alternative network addresses of different families for a single logical media stream. The entity creating a session description with an ANAT group must be ready to receive or send media over any of the grouped "m" lines.

By default, ANAT is enabled on SIP trunks. However, if the SIP gateway is configured in IPv4-only or IPv6-only mode, the gateway will not use ANAT semantics in its SDP offer.

Examples

The following example enables ANAT on a SIP trunk:

```
Router(conf-serv-sip)# anat
```

The following example shows ANAT being configured per tenant:

```
Router(config-class)# anat system
```

ani mapping

To preprogram the Numbering Plan Area (NPA), or area code, into a single Multi Frequency (MF) digit, use the **ani mapping** command in voice-port configuration mode. To disable Automatic Number Identification (ANI) mapping, use the **no** form of this command.

ani mapping *npd-value npa-number*
no ani mapping

Syntax Description

npd-value	Value of the Numbering Plan Digit (NPD). Range is 0 to 3. There is no default.
npa-number	Number (area code) of the NPA. Range is 100 to 999. There is no default value.

Command Default

No default behavior or values

Command Modes

Voice-port configuration (config-voiceport)

Command History

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

Usage Guidelines

The **ani mapping** command table translates the NPA into a single MF digit. The number of NPDs programmed is determined by local policy as well as by the number of NPAs that the public service answering point (PSAP) serves. Repeat this command until all NPDs are configured or until the NPD maximum range is reached.

Examples

The following example shows the voice port preprogramming the NPA into a single MF digit:

```
voice-port 1/1/0
 timing digit 100
 timing inter-digit 100
 ani mapping 1 408
 signal cama KP-NPD-NXX-XXXX-ST
!
voice-port 1/1/1
 timing digit 100
 timing inter-digit 100
 ani mapping 1 408
 signal cama KP-NPD-NXX-XXXX-ST
```

Related Commands

Command	Description
signal	Specifies the type of signaling for a CAMA port.
voice-port	Enters voice-port configuration mode.

answer-address

To specify the full E.164 telephone number to be used to identify the dial peer of an incoming call, use the **answer-address** command in dial-peer configuration mode. To disable the configured telephone number, use the **no** form of this command.

answer-address[{**+**}]*string*[{**T**}]
no answer-address

Syntax Description

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

Command Default

The default value is enabled with a null string

Command Modes

Dial peer configuration Router (config-dial-peer)

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 3600 series routers.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Use the **answer-address** command to identify the origin (or dial peer) of incoming calls from the IP network. Cisco IOS software identifies the dial peers of a call in one of two ways: by identifying either the interface through which the call is received or the telephone number configured with the **answer-address** command. In the absence of a configured telephone number, the peer associated with the interface is associated with the incoming call.

For calls that come in from a plain old telephone service (POTS) interface, the **answer-address** command is not used to select an incoming dial peer. The incoming POTS dial peer is selected on the basis of the port configured for that dial peer.

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

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

Examples

The following example shows the E.164 telephone number 555-0104 as the dial peer of an incoming call being configured:

```
dial-peer voice 10 pots
 answer-address +5550104
```

Related Commands

Command	Description
destination-pattern	Specifies either the prefix or the full E.164 telephone number to be used for a dial peer.
port (dial peer)	Associates a dial peer with a specific port.
prefix	Specifies the prefix of the dialed digits for a dial peer.

application (dial-peer)

To enable a specific application on a dial peer, use the **application** command in dial-peer configuration mode. To remove the application from the dial peer, use the **no** form of this command.

application *application-name* [**out-bound**]
no application *application-name* [**out-bound**]

Syntax Description

application-name	Name of the predefined application that you wish to enable on the dial peer. See the "Usage Guidelines" section for valid application names.
out-bound	(Optional) Outbound calls are handed off to the named application. This keyword is used for store-and-forward fax applications and VoiceXML applications.

Command Default

No default behavior or values

Command Modes

Dial peer voice configuration (config-dial-peer)

Command History

Release	Modification
11.3(6)NA2	This command was introduced on the Cisco 2500 series, Cisco 3600 series, and Cisco AS5300.
12.0(5)T	The SGCPAPP application was supported initially on the Cisco AS5300.
12.0(7)XK	Support for the SGCPAPP application was implemented on the Cisco MC3810 and the Cisco 3600 series (except for the Cisco 3620).
12.1(2)T	The SGCPAPP application was integrated into Cisco IOS Release 12.1(2)T.
12.1(3)T	The MGCPAPP application was implemented on the Cisco AS5300.
12.1(3)XI	The **out-bound** keyword was added for store-and-forward fax on the Cisco AS5300.
12.1(5)T	The **out-bound** keyword was integrated into Cisco IOS Release 12.1(5)T, and the command was implemented on the Cisco AS5800.
12.2(2)T	This command was implemented on the Cisco 7200 series.
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series, Cisco 3600 series, and Cisco VG200.
12.2(4)T	This command was implemented on the Cisco 1750.
12.2(4)XM	This command was implemented on the Cisco 1751.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the following platforms: The Cisco 3725 and Cisco 3745. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.

Release	Modification
12.2(11)T	This command was integrated into Cisco CallManager Version 3.2 and implemented on the Cisco 1760 and Cisco IAD2420 series routers. This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 in this release.
12.2(13)T	The *application-name* argument was removed from the **no** form of this command.
12.2(15)T	Malicious Caller Identification (MCID) was added as a valid *application-name* argument.
12.2(15)ZJ	The session application referred to by the **default** value of the *application-name* argument was updated to include support for Open Settlement Protocol (OSP), call transfer, and call forwarding. The version of the session application referred to by **default** in Cisco IOS Release 12.2(13)T and earlier releases was renamed default.c.old.
12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(14)T	This command is obsolete in Cisco IOS Release 12.3(14)T. For Cisco IOS Release 12.3(14)T and later releases, use the **application** command in global configuration mode to configure applications on a dial peer.

Usage Guidelines

Use this command when configuring interactive voice response (IVR) or any of the IVR-related features to associate a predefined session application with an incoming POTS dial peer and an outgoing Multimedia Mail over IP (MMoIP) dial peer. Calls that use the incoming POTS dial peer and the outgoing MMoIP dial peer are handed off to the specified predefined session application.

Note In Cisco IOS Release 12.2(15)ZJ and later releases, the application name default refers to the application that supports OSP, call transfer, and call forwarding. The default session application in Cisco IOS Release 12.2(13)T and earlier releases has been renamed to default.old.c and can still be configured for specific dial peers through the **application** command or globally configured for all inbound dial peers through the **call application global** command.

For Media Gateway Control Protocol (MGCP) and Simple Gateway Control Protocol (SGCP) networks, enter the application name in uppercase characters. For example, for MGCP networks, you would enter MGCPAPP for the *application-name* argument. The application can be applied only to POTS dial peers. Note that SGCP dial peers do not use dial-peer hunting.

Note In Cisco IOS Release 12.2, you cannot mix SGCP and non-SGCP endpoints in the same T1 controller, nor can you mix SGCP and non-SGCP endpoints in the same DS0 group.

Note MGCP scripting is not supported on the Cisco 1750 router or on Cisco 7200 series routers.

For H.323 networks, the application is defined by a Tool Command Language/interactive voice response (Tcl/IVR) filename and location. Incoming calls that use POTS dial peers and outgoing calls that use MMoIP dial peers are handed off to this application**.**

For Session Initiation Protocol (SIP) networks, use this command to associate a predefined session application. The default Tcl application (from the Cisco IOS image) for SIP is session and can be applied to both VoIP and POTS dial peers.

Examples

The following example defines an application and applies it to an outbound MMoIP dial peer for the fax on-ramp operation:

```
call application voice fax_on_vfc_onramp http://santa/username/clid_4digits_npw_3.tcl
dial-peer voice 3 mmoip
 application fax_on_vfc_onramp out-bound
 destination-pattern 57108..
 session target mailto:$d$@mail-server.cisco.com
```

The following example applies the MGCP application to a dial peer:

```
dial-peer voice 1 pots
 application MGCPAPP
```

The following example applies a predefined application to an incoming POTS dial peer:

```
dial-peer voice 100 pots
 application c4
```

The following example applies a predefined application to an outbound MMoIP dial peer for the on-ramp operation:

```
dial-peer voice 3 mmoip
 application fax_on_vfc_onramp_ap out-bound
 destination-pattern 57108..
 session target mailto:$d$@mail-server.cisco.com
```

The following example applies the predefined SIP application to a dial peer:

```
dial-peer voice 10 pots
 application session
```

For Cisco IOS Release 12.2(15)T, MCID was added as a valid *application-name* argument. The following is a sample configuration using the MCID application name:

```
call application voice mcid http://santa/username/app_mcid_dtmf.2.0.0.28.tcl
dial-peer voice 3 pots
 application mcid
 incoming called-number 222....
 direct-inward-dial
 port 1:D
```

Related Commands

Command	Description
application	Enables a specific application on a dial peer.
call application voice	Defines the name to be used for an application and indicates the location of the appropriate IVR script to be used with this application.
mgcp	Starts the MGCP daemon.
sgcp	Starts and allocates resources for the SGCP daemon.

Command	Description
sgcp call-agent	Defines the IP address of the default SGCP call agent.

application (global)

To enter application configuration mode to configure applications, use the **application** command in global configuration mode.

application

Syntax Description

This command has no keywords or arguments.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Voice class tenant configuration (config-class)

Command History

Release	Modification
12.3(14)T	This command was introduced to replace the **application** command in dial-peer configuration mode.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.
Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models (under voice class tenat configuration) and **package auth**.

Usage Guidelines

Note **custom application** under service is not supported in YANG configuration.

application service APPNAME <app-url> paramspace <options>

Use this command to enter application configuration mode. You can use related commands in application configuration mode to configure standalone applications (services) and linkable functions (packages).

Examples

The following example shows how to enter application configuration mode and configure debit card service:

Enter application configuration mode to configure applications and services:

```
Router(config)# application
```

Load the debit card script:

```
Router(config-app)# service debitcard

tftp://server-1/tftpboot/scripts/app_debitcard.2.0.2.8.tcl
```

Configure language parameters for the debit card service:

```
Router(config-app-param)# paramspace english language en

paramspace english index 1
  paramspace english prefix en
  paramspace english location tftp://server-1/tftpboot/scripts/au/en/
```

Related Commands

Command	Description
call application voice	Defines the name of a voice application and specify the location of the Tcl or VoiceXML document to load for this application.

aqm-register-fnf

To export the audio and video call quality statistics to flow record using Flexible NetFlow collector, use the **aqm-register-fnf** command in global configuration mode. To disable the export of audio and video call quality statistics, use the **no** form of this command.

aqm-register-fnf
no aqm-register-fnf

Syntax Description

This command has no arguments or keywords.

Command Default

The **aqm-register-fnf** command is enabled.

Command Modes

Global configuration (config)

Command History

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

Usage Guidelines

Use the **aqm-register-fnf** command when you want to export metrics related to media (voice) quality; for example, conversational mean opinion score (MOS), packet loss rate, conceal ratio, and so on. The **aqm-register-fnf** command must be configured before you use the **media monitoring** command to configure voice quality metrics.

Note Configuring the **no aqm-register-fnf** command does not disable the command in the device's running and startup configurations.

Examples

The following example shows how to enable exporting of audio quality statistics to the flow record:

```
Device> enable
Device# configure terminal
Device(config)# aqm-register-fnf
```

arq reject-resource-low

To configure the gatekeeper to send an Admission Reject (ARJ) message to the requesting gateway if destination resources are low, use the **arq reject-resource-low** command in gatekeeper configuration mode. To disable the gatekeeper from checking resources, use the **no** form of this command.

arq reject-resource-low
no arq reject-resource-low

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

Gatekeeper configuration (config-gk)

Command History

Release	Modification
12.3(1)	This command was introduced.

Examples

The following example shows that the gatekeeper is configured to send an ARJ message to the requesting gateway if destination resources are low:

```
gatekeeper
 arq reject-resource-low
```

Related Commands

Command	Description
lrq reject-resource-low	Configures a gatekeeper to notify a sending gatekeeper on receipt of an LRQ message that no terminating endpoints are available.

arq reject-unknown-prefix

To enable the gatekeeper to reject admission requests (ARQs) for zone prefixes that are not configured, use the**arqreject-unknown-prefix** command in gatekeeper configuration mode. To reenable the gatekeeper to accept and process all incoming ARQs, use the **no** form of this command.

arq reject-unknown-prefix
no arq reject-unknown-prefix

Syntax Description

This command has no arguments or keywords

Command Default

The gatekeeper accepts and processes all incoming ARQs.

Command Modes

Gatekeeper configuration (config-gk)

Command History

Release	Modification
11.3(6)Q,	This command was introduced.
11.3(7)NA	This command was introduced.
12.0(3)T	This command was integrated into Cisco IOS Release 12.0(3)T.

Usage Guidelines

Use the **arqreject-unknown-prefix** command to configure the gatekeeper to reject any incoming ARQs for a destination E.164 address that does not match any of the configured zone prefixes.

When an endpoint or gateway initiates an H.323 call, it sends an ARQ to its gatekeeper. The gatekeeper uses the configured list of zone prefixes to determine where to direct the call. If the called address does not match any of the known zone prefixes, the gatekeeper attempts to *hairpin* the call out through a local gateway. If you do not want your gateway to do this, then use the **arqreject-unknown-prefix** command. (The term *hairpin*is used in telephony. It means to send a call back in the direction from which it came. For example, if a call cannot be routed over IP to a gateway that is closer to the target phone, the call is typically sent back out through the local zone, back the way it came.)

This command is typically used to either restrict local gateway calls to a known set of prefixes or deliberately fail such calls so that an alternate choice on a gateway's rotary dial peer is selected.

Examples

Consider a gatekeeper configured as follows:

```
zone local gk408 cisco.com
zone remote gk415 cisco.com 172.21.139.91
zone prefix gk408 1408.......
zone prefix gk415 1415.......
```

In this example configuration, the gatekeeper manages a zone containing gateways to the 408 area code, and it knows about a peer gatekeeper that has gateways to the 415 area code. Using the **zoneprefix** command, the gatekeeper is then configured with the appropriate prefixes so that calls to those area codes hop off in the optimal zone.

If the **arqrequest-unknown-prefix** command is not configured, the gatekeeper handles calls in the following way:

- A call to the 408 area code is routed out through a local gateway.
- A call to the 415 area code is routed to the gk415 zone, where it hops off on a local gateway.
- A call to the 212 area code is routed to a local gateway in the gk408 zone.

If the **arqreject-unknown-prefix** command is configured, the gatekeeper handles calls in the following way:

- A call to the 408 area code is routed out through a local gateway.
- A call to the 415 area code is routed to the gk415 zone, where it hops off on a local gateway.
- A call to the 212 area code is rejected because the destination address does not match any configured prefix.

Related Commands

Command	Description
zone prefix	Adds a prefix to the gatekeeper zone list.

as

To define an application server for backhaul, use the **as** command in IUA configuration mode. To disable the backhaul ability from an application server, use the **no** form of this command.

as *as-name* *localip1* [*localip2*] [**local-sctp-port**] [**fail-over-timer**] [**sctp-startup-rtx**] [**sctp-streams**] [**sctp-t1init**]
no as *name*

Syntax Description

as-name	Defines the protocol name (only ISDN is supported).
localip1	Defines the local IP address(es) for all the ASPs in a particular AS.
localip2	(Optional) Defines the local IP address(es) for all the ASPs in a particular application server .
local-sctp-port	(Optional) Defines a specific local Simple Control Transmission Protocol (SCTP) port rather than an ISDN Q.921 User Adaptation Layer (IUA) well-known port.
fail-over-timer	(Optional) Configures the failover timer for a particular application server .
sctp-startup-rtx	(Optional) Configures the SCTP maximum startup retransmission timer.
sctp-streams	(Optional) Configures the number of SCTP streams for a particular application server .
sctp-t1init	(Optional) Configures the SCTP T1 initiation timer.

Command Default

No application server is defined.

Command Modes

IUA configuration (config-iua)

Command History

Release	Modification
12.2(4)T	This command was introduced.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and support was added for the Cisco AS5300 platform.
12.2(13)T1	This command was implemented on the Cisco AS5850.
12.2(15)T	This command was integrated into Cisco IOS Release xx.x(x)X and implemented on the Cisco 2420, Cisco 2600 series, Cisco 3600 series, and Cisco 3700 series; Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 network access server (NAS) platforms.

Usage Guidelines

A maximum of two local IP addresses can be specified. (Note that SCTP has built-in support for multihomed machines.)

Note All of the ASPs in an application server must be removed before an application server can be unconfigured.

The default value of the SCTP streams is determined by the hardware that you have installed. The value of the failover timer is found in the **showiuaasall**command output.

The number of streams to assign to a given association is implementation dependent. During the initialization of the IUA association, you need to specify the total number of streams that can be used. Each D channel is associated with a specific stream within the association. With multiple trunk group support, every interface can potentially be a separate D channel.

At startup, the IUA code checks for all the possible T1, E1, or T3 interfaces and sets the total number of inbound and outbound streams supported accordingly. In most cases, there is only a need for one association between the gateway (GW) and the Media Gateway Controller (MGC). For the rare case that you are configuring multiple AS associations to various MGCs, the overhead from the unused streams would have minimal impact. The NFAS D channels are configured for one or more interfaces, where each interface is assigned a unique stream ID.

The total number of streams for the association needs to include an additional stream for the SCTP management messages. So during startup, the IUA code adds one to the total number of interfaces (streams) found.

You have the option to manually configure the number of streams per association. In the backhaul scenario, if the number of D channel links is limited to one, allowing the number of streams to be configurable avoids the unnecessary allocation of streams in an association that is never used. For multiple associations between a GW and multiple MGCs, the configuration utility is useful in providing only the necessary number of streams per association. The overhead from the streams allocated but not used in the association is negligible.

If the number of streams is manually configured through the CLI, the IUA code cannot distinguish between a startup event, which automatically sets the streams to the number of interfaces, or if the value is set manually during runtime. If you are configuring the number of SCTP streams manually, you must add one plus the number of interfaces using the **sctp-streams** keyword. Otherwise, IUA needs to always add one for the management stream, and the total number of streams increments by one after every reload.

When you set the SCTP stream with the CLI, you cannot change the inbound and outbound stream support once the association is established with SCTP. The value takes effect when you first remove the IUA AS configuration and then configure it back as the same application server or a new one. The other option is to reload the router.

Examples

An application server and the application server process (ASP) should be configured first to allow a National ISDN-2 with Cisco extensions (NI2+) to be bound to this transport layer protocol. The application server is a logical representation of the SCTP local endpoint. The local endpoint can have more than one IP address but must use the same port number.

The following is an example of an application server configuration on a gateway. The configuration shows that an application server named as5400-3 is configured to use two local IP addresses and a port number of 2577:

```
Router(config-iua)# as as5400-3 10.1.2.34 10.1.2.35 2577
```

The following output shows that the application server (as1) is defined for backhaul:

```
AS as1 10.21.0.2 9900
```

Related Commands

Command	Description
asp	Defines an ASP for backhaul.

asp

To define an application server process (ASP) for backhaul, use the **asp** command in IUA configuration mode. To disable the ASP, use the **no** form of this command.

asp *asp-name* **as** **as-name** *as-name*{*remote-p1* [{[*remoteip2*]}]}[{**remote-sctp-port**}] [{[**ip-precedence**]}][{**sctp-keepalives**}][{**sctp-max-associations**}][{**sctp-path-retransmissions**}][{[**sctp-t3-timeout**]}]

no **asp** *asp-name*

Syntax Description

asp-name	Names the current ASP.
as	The application server to which the ASP belongs.
as-name	Name of the application server to which the ASP belongs.
remoteip1	(Optional) Designates the remote IP address for this Simple Control Transmission Protocol (SCTP) association.
remoteip2	Designates the remote IP address for this SCTP association.
remote-sctp-port	Connects to a remote SCTP port rather than the IUA well-known port.
ip-precedence	(Optional) Sets IP Precedence bits for protocol data units (PDUs). • IP precedence is expressed in the type of service (ToS) field of the**showipsctpassociationparameters** output. The default type of service (ToS) value is 0. • Valid precedence values range from 0 to 7. You can also use the default IP precedence value for this address by choosing the default option.
sctp-keepalives	(Optional) Modifies the keepalive behavior of an IP address in a particular ASP. • Valid keepalive interval values range from 1000 to 60000. The default value is 500 ms (see the **showipsctpassociationparameters** output under **heartbeats**).
sctp-max-associations	(Optional) Sets the SCTP maximum association retransmissions for a particular ASP. Valid values range from 2 to 20. The default is 5.
sctp-path-retransmissions	(Optional) Sets the SCTP path retransmissions for a particular ASP. Valid values range from 2 to 10. The default is 3.
sctp-t3-timeout	(Optional) Sets the SCTP T3 retransmission timeout for a particular ASP. The default value is 900 ms.

Command Default

No ASP is defined.

Command Modes

IUA configuration (config-iua)

Command History

Release	Modification
12.2(4)T	This command was introduced.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and support was added for the Cisco AS5300.
12.2(11)T1	This command was implemented on the Cisco AS5850.
12.2(15)T	This command was integrated into Cisco IOS Release 12.2(15)T and implemented on the Cisco 2420, Cisco 2600 series, Cisco 3600 series, and Cisco 3700 series; and Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 network access server (NAS) platforms.

Usage Guidelines

This command establishes SCTP associations. There can be only a maximum of three ASPs configured per AS. IP precedence is expressed in the ToS field of **showipsctpassociationparameters** output. The default ToS value is 0.

Note All of the ASPs in an application server must be removed before an application serever can be unconfigured.

You can configure the precedence value in IUA in the range of 0 to 7 for a given IP address. Within IUA, the upper three bits representing the IP precedence in the ToS byte (used in the IP header) is set based on the user input before passing down the value to SCTP. In turn, SCTP passes the ToS byte value to IP. The default value is 0 for "normal" IP precedence handling.

The *asp-name* argument specifies the name of this ASP. The**ip-precedence** keyword sets the precedence and ToS field. The *remote-ip-address* argument specifies the IP address of the remote end-point (the address of MGC, for example). The *number* argument can be any IP precedence bits in the range 1 to 255.

The **no** form of the command results in precedence bits not being explicitly set by SCTP.

In the case of a hot-standby Cisco PGW2200 pair, from the gateway (GW) perspective there is usually one ASP active and another in the INACTIVE state. The ASP_UP message is used to bring the ASP state on the GW to the INACTIVE state, followed by the ASPTM message, ASP_ACTIVE to ready the IUA link for data exchange. (Eventually the QPTM Establish Request message actually initiates the start of the D channel for the given interface.) In the event that the GW detects a failure on the active ASP, it can send a NTFY message to the standby ASP to request that it become active.

Examples

An ASP can be viewed as a local representation of an SCTP association because it specifies a remote endpoint that is in communication with an AS local endpoint. An ASP is defined for a given AS. For example, the following configuration defines a remote signaling controller *asp-name* at two IP addresses for AS as1. The remote SCTP port number is 2577:

```
Router(config-iua)# as as1 10.4.8.69, 10.4.9.69 2477
Router(config-iua)# asp asp1 as as1 10.4.8.68 10.4.9.68 2577
```

Multiple ASPs can be defined for a single AS for the purpose of redundancy, but only one ASP can be active. The ASPs are inactive and only become active after fail-over.

In the Cisco Media Gateway Controller (MGC) solution, a signaling controller is always the client that initiates the association with a gateway. During the initiation phase, you can request outbound and inbound stream numbers, but the gateway only allows a number that is at least one digit higher than the number of interfaces (T1/E1) allowed for the platform.

The following example specifies the IP precedence level on the specified IP address. This example uses IP precedence level 7, which is the maximum level allowed:

```
Router(config-iua)# asp asp1 as ip-precedence 10.1.2.345 7
```

The following example specifies the IP address to enable and disable keepalives:

```
Router(config-iua)# asp asp1 as sctp-keepalive 10.1.2.34
```

The following example specifies the keepalive interval in milliseconds. In this example, the maximum value of 60000 ms is used:

```
Router(config-iua)# asp asp1 as sctp-keepalive 10.10.10.10 60000
```

The following example specifies the IP address for the SCTP maximum association and the maximum association value. In this example, a maximum value of 20 is used:

```
Router(config-iua)# asp asp1 as sctp-max-association 10.10.10.10 20
```

The following example specifies the IP address for the SCTP path retransmission and the maximum path retransmission value. In this example, a maximum value of 20 is used:

```
Router(config-iua)# asp asp1 as sctp-path-retransmissions 10.10.10.10 10
```

The following example specifies the IP address for SCTP T3 timeout and specifies the T3 timeout value in milliseconds. In this example, the maximum value of 60000 is used:

```
Router(config-iua)# asp asp1 as sctp-t3-timeout 10.10.10.10 60000
```

Related Commands

Command	Description
as	Defines an application server for backhaul.

asserted-id

To enable support for the asserted ID header in incoming Session Initiation Protocol (SIP) requests or response messages, and to send the asserted ID privacy information in outgoing SIP requests or response messages, use the **asserted-id** command in voice service VoIP-SIP configuration mode or voice class tenant configuration mode. To disable the support for the asserted ID header, use the **no** form of this command.

asserted-id {**pai** | **ppi**}**system**
no asserted-id system

Syntax Description

pai	(Optional) Enables the P-Asserted-Identity (PAI) privacy header in incoming and outgoing SIP requests or response messages.
ppi	(Optional) Enables the P-Preferred-Identity (PPI) privacy header in incoming SIP requests and outgoing SIP requests or response messages.
system	Specifies that the asserted-id use the global forced CLI setting. This keyword is available only for the tenant configuration mode.

Command Default

The privacy information is sent using the Remote-Party-ID (RPID) header or the FROM header.

Command Modes

Voice service VoIP-SIP configuration (conf-serv-sip)

Voice class tenant configuration (config-class)

Command History

Release	Modification
12.4(15)T	This command was introduced.
15.1(3)T	This command was modified. Support for incoming calls was added.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: **system**.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

If you choose the **pai** keyword or the **ppi** keyword, the gateway builds the PAI header or the PPI header, respectively, into the common SIP stack. The **pai** keyword or the **ppi** keyword has the priority over the Remote-Party-ID (RPID) header, and removes the RPID header from the outbound message, even if the router is configured to use the RPID header at the global level.

Examples

The following example shows how to enable support for the PAI privacy header:

```
Router> enable
Router# configure terminal
Router(config)# voice service voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# asserted-id pai
```

The following example shows asserted ID used in the voice class tenant configuration mode:

```
Router(config-class)# asserted-id system
```

Related Commands

Command	Description
calling-info pstn-to-sip	Specifies calling information treatment for PSTN-to-SIP calls.
privacy	Sets privacy in support of RFC 3323.
voice-class sip asserted-id	Enables support for the asserted ID header in incoming and outgoing SIP requests or response messages in dial-peer configuration mode.

associate application

To associate an application to the digital signal processor (DSP) farm profile, use the **associateapplication**command in DSP farm profile configuration mode. To remove the protocol, use the **no** form of this command.

associate application {**cube** | **sbc** | **sccp**} *profile-description-text*
no associate application sccp

Syntax Description

cube	Associates the Cisco Unified Border Element application to a defined profile in the DSP farm.
sbc	Associates the SBC application to a defined profile in the DSP farm.
sccp	Associates the skinny client control protocol application to a defined profile in the DSP farm.
profile-description-text	(Optional) User defined name for the associated applicaion.

Command Default

No application is associated with the DSP farm profile.

Command Modes

DSP farm profile configuration (config-dspfarm-profile)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.2(33)SB	This command was integrated into Cisco IOS Release 12.2(33)SB.
12.4(22)T	Support for IPv6 was added.
Cisco IOS XE Release 3.2S	This command was modified. The **cube**and **sbc**keywords and the *profile-description-text*argument were added.

Usage Guidelines

Use the associate application command to associate an application to a predefinded DSP farm profile.

Examples

The following example associates SCCP to the DSP farm profile:

```
Router(config-dspfarm-profile)#
associate application sccp
```

The following example associates Cisco Unified Border Element to the DSP farm profile:

```
Router(config-dspfarm-profile)#
associate application cube
```

Related Commands

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

associate ccm

To associate a Cisco Unified Communications Manager with a Cisco Unified Communications Manager group and establish its priority within the group, use the **associate ccm** command in the SCCP Cisco CallManager configuration mode. To disassociate a Cisco Unified Communications Manager from a Cisco Unified Communications Manager group, use the **no** form of this command.

associate ccm *identifier-number* **priority** *priority-number*
no associate ccm *identifier-number* **priority** *priority-number*

Syntax Description

identifier-number	Number that identifies the Cisco Unified Communications Manager. Range is 1 to 50. There is no default value.
priority *priority-number*	Priority of the Cisco Unified Communications Manager within the Cisco Unified Communications Manager group. Range is 1 to 4. There is no default value. The highest priority is 1.

Command Default

No default behavior or values

Command Modes

SCCP Cisco CallManager configuration (config-sccp-ccm)

Command History

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

Examples

The following example associates Cisco Unified Communications Manager 25 with Cisco Unified Communications Manager group 9 and sets the priority of the Cisco Unified Communications Manager within the group to 2:

```
Router(config)# sccp ccm group 9
Router(config-sccp-ccm)# associate ccm 25 priority 2
```

Related Commands

Command	Description
connect interval	Specifies the amount of time that a DSP farm profile waits before attempting to connect to a Cisco Unified Communications Manager when the current Cisco Unified Communications Manager fails to connect.
connect retries	Specifies the number of times that a DSP farm attempts to connect to a Cisco Unified Communications Manager when the current Cisco Unified Communications Manager connections fails.
sccp ccm group	Creates a Cisco CallManger group and enters SCCP Cisco CallManager configuration mode.

associate profile

To associate a digital signal processor (DSP) farm profile with a Cisco CallManager group, use the **associateprofile**command in SCCP Cisco CallManager configuration mode. To disassociate a DSP farm profile from a Cisco Unified CallManager, use the **no** form of this command.

associate profile *profile-identifier* **register** *device-name*
no associate profile *profile-identifier* **register** *device-name*

Syntax Description

profile-identifier	Number that identifies the DSP farm profile. Range is 1 to 65535. There is no default value.
register *device-name*	User-specified device name in Cisco Unified CallManager. A maximum number of 15 characters can be entered for the device name.

Command Default

This command is not enabled.

Command Modes

SCCP Cisco CallManager configuration (conig-sccp-ccm)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.4(22)T	Support for IPv6 was added.

Usage Guidelines

The device name must match the name configured in Cisco UnifiedCallManager; otherwise the profile is not registered to Cisco Unified CallManager.

Note Each profile can be associated to only one Cisco CallManager group.

Examples

The following example associates DSP farm profile abgz12345 to Cisco CallManager group 999:

```
Router(config)# sccp ccm group 999
Router(config-sccp-ccm)# associate profile 1 register abgz12345
```

Related Commands

Command	Description
bind interface	Binds an interface to a Cisco CallManager group.
dspfarm profile	Enters DSP farm profile configuration mode and defines a profile for DSP farm services.
sccp ccm group	Creates a Cisco CallManager group and enters SCCP Cisco CallManager configuration mode.

associate registered-number

To associate the preloaded route and outbound proxy details with the registered number, use the **associateregistered-number** command in voice service VoIP SIP configuration mode or voice class tenant configuration mode. To remove the association, use the **no** form of this command.

associate registered-number *number* **system**
no associate registered-number

Syntax Description

number	Registered number. The number must be between 4 and 32.
system	Use the global sip-ua associate configuration.

Command Default

The preloaded route and outbound proxy details are not associated with the registered number by default.

Command Modes

Voice service VoIP SIP configuration (conf-serv-sip)

Voice class tenant configuration (config-class)

Command History

Release	Modification
15.1(2)T	This command was introduced.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: **system**. This command is now available under voice class tenants.

Examples

The following example shows how to associate a registered number in the SIP configuration mode:

```
Router# enable
Router# configure terminal
Router(config)# voice service voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# associate registered-number 5
```

The following example shows how to associate a registered number in the voice class tenant configuration mode:

```
Router(config-class)# associate registered-number system
```

Related Commands

Command	Description
voice-class sip associate registered-number	Associates preloaded route and outbound proxy details with the registered number in the dial-peer configuration level.

asymmetric payload

To configure Session Initiation Protocol (SIP) asymmetric payload support, use the **asymmetricpayload** command in SIP configuration mode or voice class tenant configuration mode. To disable asymmetric payload support, use the **no** form of this command.

asymmetric payload {**dtmf** | **dynamic-codecs** | **full** | **system**}
no asymmetric payload

Syntax Description

dtmf	(Optional) Specifies that the asymmetric payload support is dual-tone multi-frequency (DTMF) only.
dynamic-codecs	(Optional) Specifies that the asymmetric payload support is for dynamic codec payloads only.
full	(Optional) Specifies that the asymmetric payload support is for both DTMF and dynamic codec payloads.
system	(Optional) Specifies that the asymmetric payload uses the global value.

Command Default

This command is disabled.

Command Modes

Voice service SIP configuration (conf-serv-sip)

Voice class tenant configuration (config-class)

Command History

Release	Modification
12.4(15)T	This command was introduced.
Cisco IOS XE Release 3.1S	This command was integrated into Cisco IOS Release IOS XE 3.1.
15.6(2) and IOS XE Denali 16.3.1	This command was modified to include the keyword: **system**. This command is now available under voice class tenants.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Enter SIP configuration mode from voice-service configuration mode, as shown in the example.

For the Cisco UBE the SIP asymmetric payload-type is supported for audio/video codecs, DTMF, and NSE. Hence, **dtmf** and **dynamic-codecs** keywords are internally mapped to the **full** keyword to provide asymmetric payload-type support for audio/video codecs , DTMF, and NSE.

Examples

The following example shows how to set up a full asymmetric payload globally on a SIP network for both DTMF and dynamic codecs:

```
Router(config)# voice service voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# asymmetric payload full
```

The following example shows how to set up a full asymmetric payload globally in the voice class tenant configuration mode:

```
Router(config-class)# asymmetric payload system
```

Related Commands

Command	Description
sip	Enters SIP configuration mode from voice-service VoIP configuration mode.
voice-class sip asymmetric payload	Configures SIP asymmetric payload support on a dial peer.

atm scramble-enable

To enable scrambling on E1 links, use the **atmscramble-enable** command in interface configuration mode. To disable scrambling, use the **no**form of this command.

atm scramble-enable
no atm scramble-enable

Syntax Description

This command has no arguments or keywords.

Command Default

By default, payload scrambling is set off

Command Modes

Interface configuration

Command History

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

Usage Guidelines

Enable scrambling on E1 links only. On T1 links, the default binary 8-zero substitution (B8ZS) line encoding normally ensures sufficient reliability. Scrambling improves data reliability on E1 links by randomizing the ATM cell payload frames to avoid continuous nonvariable bit patterns and to improve the efficiency of the ATM cell delineation algorithms.

The scrambling setting must match that of the far end.

Examples

The following example shows how to set the ATM0 E1 link to scramble payload:

```
interface atm0
 atm scramble-enable
```

atm video aesa

To set the unique ATM end-station address (AESA) for an ATM video interface that is using switched virtual circuit (SVC) mode, use the **atmvideoaesa** command in ATM interface configuration mode. To remove any configured address for the interface, use the **no** form of this command.

atm video aesa [{**default***esi-address*}]
no atm video aesa

Syntax Description

default	(Optional) Automatically creates a network service access point (NSAP) address for the interface, based on a prefix from the ATM switch (26 hexadecimal characters), the MAC address (12 hexadecimal characters) as the end station identifier (ESI), and a selector byte (two hexadecimal characters).
esi-address	(Optional) Defines the 12 hexadecimal characters used as the ESI. The ATM switch provides the prefix (26 hexadecimal characters), and the video selector byte provides the remaining two hexadecimal characters.

Command Default

default

Command Modes

ATM Interface 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.

Usage Guidelines

You cannot specify the ATM interface NSAP address in its entirety. The system creates either all of the address or part of it, depending on how you use this command.

Examples

The following example shows the ATM interface NSAP address set automatically:

```
interface atm0
 atm video aesa default
```

The following example shows the ATM interface NSAP address set to a specific ESI value:

```
interface atm0/1
 atm video aesa 444444444444
```

Related Commands

Command	Description
show atm video-voice address	Displays the NSAP address for the ATM interface.

attribute acct-session-id overloaded

To overload the acct-session-id attribute with call detail records, use the **attributeacct-session-idoverloaded** command in gateway accounting AAA configuration mode. To disable overloading the acct-session-id attribute with call detail records, use the **no** form of this command.

attribute acct-session-id overloaded
no attribute acct-session-id overloaded

Syntax Description

This command has no arguments or keywords.

Command Default

The acct-session-id attribute is not overloaded with call detail records.

Command Modes

Gateway accounting AAA configuration (config-gw-accounting-aaa)

Command History

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

Usage Guidelines

The **attributeacct-session-idoverloaded**command replaces the **gw-accountingh323**command.

The acct-session-id attribute is RADIUS attribute 44. For more information on this attribute, see the document *RADIUS Attribute 44 (Accounting Session ID) in Access Requests* .

Attributes that cannot be mapped to standard RADIUS attributes are packed into the acct-session-id attribute field as ASCII strings separated by the forward slash ("/") character.

The Accounting Session ID (acct-session-id) attribute contains the RADIUS account session ID, which is a unique identifier that links accounting records associated with the same login session for a user. This unique identifier makes it easy to match start and stop records in a log file.

Accounting Session ID numbers restart at 1 each time the router is power-cycled or the software is reloaded.

Examples

The following example shows the acct-session-id attribute being overloaded with call detail records:

```
gw-accounting aaa
 attribute acct-session-id overloaded
```

Related Commands

Command	Description
call accounting-template voice	Defines and loads the template file at the location defined by the URL.
gw-accounting aaa	Enables VoIP gateway accounting.

attribute h323-remote-id resolved

To resolve the h323-remote-id attribute, use the **attributeh323-remote-idresolved**command in gateway accounting AAA configuration mode. To keep the h323-remote-id attribute unresolved, use the **no** form of this command.

attribute h323-remote-id resolved
no attribute h323-remote-id resolved

Syntax Description

This command has no arguments or keywords.

Command Default

The h323-remote-id attribute is not resolved.

Command Modes

Gateway accounting aaa configuration (config-gw-accounting-aaa)

Command History

Release	Modification
12.2(11)T	This command was introduced on the Cisco 3660, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.

Usage Guidelines

In Cisco IOS Release 12.2(11)T, the **attributeh323-remote-idresolved** command replaces the **gw-accountingh323resolve**command, and the h323-remote-id attribute has been added as a Cisco vendor-specific attribute (VSA). This attribute is a string that indicates the Domain Name System (DNS) name or locally defined host name of the remote gateway.

You can obtain the value of the h323-remote-id attribute by doing a DNS lookup of the h323-remote-address attribute. The h323-remote-address attribute indicates the IP address of the remote gateway.

Examples

The following example sets the h323-remote-id attribute to resolved:

```
gw-accounting aaa
 attribute h323-remote-id resolved
```

Related Commands

Command	Description
gw-accounting aaa	Enables VoIP gateway accounting.

audio

To enable the incoming and outgoing IP-IP call gain/loss feature for audio volume control on the incoming dial peer and the outgoing dial peer, enter the **audio** command in dial-peer configuration mode. To disable this feature, use the**no** form of this command.

audio {**incoming** | **outgoing**} **level adjustment** *value*
no audio {**incoming** | **outgoing**} **level adjustment** *value*

Syntax Description

incoming	Enables the incoming IP-IP call volume control on either the incoming dial peer or the outgoing dial peer.
outgoing	Enables the outgoing IP-IP call volume control on either the incoming dial peer or the outgoing dial peer.
value	Range is -27 to 16.

Command Default

This command is disabled by default, and there is no volume control available.

Command Modes

Dial peer configuration (config-dial-peer)

Command History

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

Usage Guidelines

This feature enables the adjustment of the audio volume within a Cisco Unified Border Element (Cisco UBE) call. As with codec repacketization, dissimilar networks that have different built-in loss/gain characteristics may experience connectivity problems. By adding the ability to control the loss/gain within the Cisco UBE, you can more easily connect your networks.

The DSP requires one level for each stream, so the *value* for audio incoming level-adjustment and the *value* for audio outgoing level-adjustment will be added together. If the combined values are outside of the limit the DSP can perform, the value sent to the DSP will be either the minimum (-27) or maximum (+16) supported by the DSP.

Caution For gain/loss control, be aware that adding gain in a network with echo can generate feedback loud enough to cause hearing damage. Always exercise extreme caution when configuring gain into your network.

To configure IP-IP Call Gain/Loss Control on a voice gateway, you must configure the incoming and outgoing VoIP dial peers.

Examples

The following example shows how to configure audio incoming level to 5 and the audio outgoing level to -5:

```
Router(config-dial-peer)# audio incoming level-adjustment 5
Router(config-dial-peer)# audio outgoing level-adjustment -5
```

Related Commands

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

audio forced

To allow only audio and image (for T.38 Fax) media types, and drop all other media types (such as video and application), use the **audio forced** command in voice service voip sip configuration mode. To disable, use **no** form of this command.

audio forced
no audio forced

Command Default

Along with audio and image (for T38 fax) media types, all other media types (such as video and application) are also allowed.

Command Modes

voice service voip sip configuration (conf-serv-sip).
Voice class tenant configuration (config-class).

Command History

Release	Modification
Cisco IOS 15.6(2)T	This command was introduced.
Cisco IOS XE Denali 16.3.1	This command was integrated into Cisco IOS XE Denali 16.3.1.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Use **audio forced** command to globally allow only audio and image media types.

Example

```
Router> enable
 Router# configure terminal
 Router(config)# voice service voip
 Router(conf-voi-serv)# sip
 Router(conf-serv-sip)# audio forced
```

audio-prompt load

To initiate loading the selected audio file (.au), which contains the announcement prompt for the caller, from Flash memory into RAM, use the **audio-promptload**command in privileged EXEC mode. This command does not have a **no** form.

audio-prompt **load** *name*

Syntax Description

name	Location of the audio file that you want to have loaded from memory, flash memory, an FTP server, an HTTP server, or an HTTPS (HTTP over Secure Socket Layer (SSL)) server.

Command Default

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(6)NA2	This command was introduced. **Note** With Cisco IOS Release 11.3(6)NA2, the URL pointer refers to the directory where Flash memory is stored.
12.0(3)T	This command was integrated into Cisco IOS Release 12.0(3)T.
12.1(5)T	This command was implemented on the Cisco AS5800.
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on the Cisco 1750 and Cisco 1751. Support for other Cisco platforms is not included in this release.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T. This command is supported on the Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 in this release.
12.4(15)T	The *name* argument was modified to accept an HTTPS server URL.

Usage Guidelines

The first time the interactive voice response (IVR) application plays a prompt, it reads it from the URL (or the specified location for the .au file, such as Flash or FTP) into RAM. Then it plays the script from RAM. An example of the sequence of events follows:

- When the first caller is asked to enter the account and personal identification numbers (PINs), the enter_account.au and enter_pin.au files are loaded into RAM from Flash memory.
- When the next call comes in, these prompts are played from the RAM copy.

- If all callers enter valid account numbers and PINs, the auth_failed.au file is not loaded from Flash memory into RAM.

The router loads the audio file only when the script initially plays that prompt after the router restarts. If the audio file is changed, you must run this privileged EXEC command to reread the file. This generates an error message if the file is not accessible or if there is a format error.

Examples

The following example shows how to load the enter_pin.au audio file from Flash memory into RAM:

```
audio-prompt load flash:enter_pin.au
```

The following example shows how to load the hello.au audio file from an HTTPS server into RAM:

```
audio-prompt load https://http-server1/audio/hello.au
```

authenticate redirecting-number

To enable a Cisco IOS voice gateway to authenticate and pass Session Initiation Protocol (SIP) credentials based on the redirecting number when available instead of the calling number of a forwarded call, use the **authenticateredirecting-number** command in voice service SIP configuration mode or voice class tenant configuration mode. To return a Cisco IOS voice gateway to the default setting so that the gateway uses only the calling number for SIP credentials, use the **no** form of this command.

authenticate redirecting-number system
no authenticate redirecting-number

Syntax Description

system	Specifies that the authenticate redirecting-number use the global forced CLI setting. This keyword is available only for the tenant configuration mode.

Command Default

The Cisco IOS voice gateway uses only the calling number of a forwarded call for SIP credentials even when the redirecting number information is available for that call.

Command Modes

Voice service SIP configuration (conf-serv-sip)

Voice class tenant configuration (config-class)

Command History

Release	Modification
12.4(24)T	This command was introduced.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: **system**. This command is now available under voice class tenants.

Usage Guidelines

When an INVITE message sent out by the gateway is challenged, it must respond with the appropriate SIP credentials before the call is established. The default global behavior for the gateway is to authenticate and pass SIP credentials based on the calling number and all dial peers on a gateway default to the global setting. However, for forwarded calls, it is sometimes more appropriate to use the redirecting number and this can be specified at either the global or dial peer level (configuring behavior for a specific dial peer supersedes the global setting).

Use the **authenticateredirecting-number** command in voice service SIP configuration mode to globally enable a Cisco IOS voice gateway to authenticate and pass SIP credentials based on the redirecting number when available. Use the **no** form of this command to configure the gateway to authenticate and pass SIP credentials based only on the calling number of forwarded calls unless otherwise configured at the dial peer level:

- Use the **voice-classsipauthenticateredirecting-number** command in dial peer voice configuration mode to supersede global settings and force a specific dial peer on the gateway to authenticate and pass SIP credentials based on the redirecting number when available.
- Use the **no** form of the **voice-classsipauthenticateredirecting-number** command in dial peer voice configuration mode to supersede global settings and force a specific dial peer on the gateway to authenticate and pass SIP credentials based only on the calling number regardless of the global setting.

The redirecting number is present only in the headers of forwarded calls. When this command is disabled or the redirecting number is not available (nonforwarded calls), the gateway uses the calling number for SIP credentials.

Examples

The following example shows how to globally enable a Cisco IOS voice gateway to authenticate and pass the redirecting number of a forwarded call when a SIP INVITE message is challenged:

```
Router> enable
Router# configure terminal
Router(config)# voice service voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# authenticate redirecting-number
```

The following example shows how to authenticate a re-directed number in the voice class tenant configuration mode:

```
Router(config-class)# authenticate redirecting-number system
```

Related Commands

Command	Description
voice-class sip authenticate redirecting-number	Supersedes global settings and enables a dial peer on a Cisco IOS voice gateway to authenticate and pass SIP credentials based on the redirecting number of forwarded calls.

authentication (dial peer)

To enable SIP digest authentication on an individual dial peer, use the **authentication** command in dial peer voice configuration mode. To disable SIP digest authentication, use the **no** form of this command.

authentication username *username* **password** {**0** | **6** | **7**} *password* [**realm** *realm* [**challenge**]]
no authentication {**username** *username* **password** {**0** | **6** | **7**} *password* [**realm** *realm* [**challenge**]] | **all**}

Syntax Description

username	Specifies the username for the user who is providing authentication.
username	A string representing the username for the user who is providing authentication. A username must be at least four characters.
password	Specifies password settings for authentication.
0	Specifies encryption type as cleartext (no encryption).
6	Specifies secure reversible encryption for passwords using type **6** Advanced Encryption Scheme (AES). **Note** Requires AES primary key to be preconfigured.
7	Specifies encryption type as encrypted.
password	A string representing the password for authentication. If no encryption type is specified, the password will be cleartext format. The string must be between 4 and 128 characters.
realm	(Optional) Specifies the domain where the credentials are applicable.
realm	(Optional) A string representing the domain where the credentials are applicable.
all	(Optional) Specifies all the authentication entries for the user (dial-peer).

Command Default

SIP digest authentication is disabled.

Command Modes

Dial peer voice configuration (config-dial-peer)

Command History

Release	Modification
12.3(8)T	This command was introduced.
15.1(3)T	This command was modified. The **challenge** keyword was added.
15.2(3)T	This command was modified. The **all** keyword was added to the **no** form of the command.
IOS XE 16.11.1a	Secure reversible encryption for passwords using type **6** Advanced Encryption Scheme (AES) was introduced.

Usage Guidelines

The following configuration rules are applicable when enabling digest authentication:

- Only one username can be configured per dial peer. Any existing username configuration must be removed before configuring a different username.
- A maximum of five *password* or *realm* arguments can be configured for any one username.

The *username* and *password* arguments are used to authenticate a user. An authenticating server/proxy issuing a 407/401 challenge response includes a realm in the challenge response and the user provides credentials that are valid for that realm. Because it is assumed that a maximum of five proxy servers in the signaling path can try to authenticate a given request from a user-agent client (UAC) to a user-agent server (UAS), a user can configure up to five password and realm combinations for a configured username.

Note The user provides the password in plain text but it is encrypted and saved for 401 challenge response. If the password is not saved in encrypted form, a junk password is sent and the authentication fails.

- The realm specification is optional. If omitted, the password configured for that username applies to all realms that attempt to authenticate.
- Only one password can be configured at a time for all configured realms. If a new password is configured, it overwrites any previously configured password.

This means that only one global password (one without a specified realm) can be configured. If you configure a new password without configuring a corresponding realm, the new password overwrites the previous one.

- If a realm is configured for a previously configured username and password, that realm specification is added to that existing username and password configuration. However, once a realm is added to a username and password configuration, that username and password combination is valid only for that realm. A configured realm cannot be removed from a username and password configuration without first removing the entire configuration for that username and password--you can then reconfigure that username and password combination with or without a different realm.
- In an entry with both a password and realm, you can change either the password or realm.
- Use the **no authentication all** command to remove all the authentication entries for the user.

It is mandatory to specify the encryption type for the password. If a clear text password (type **0**) is configured, it is encrypted as type **6** before saving it to the running configuration.

If you specify the encryption type as **6** or **7**, the entered password is checked against a valid type **6** or **7** password format and saved as type **6** or **7** respectively.

Type-6 passwords are encrypted using AES cipher and a user-defined primary key. These passwords are comparatively more secure. The primary key is never displayed in the configuration. Without the knowledge of the primary key, type **6** passwords are unusable. If the primary key is modified, the password that is saved as type 6 is re-encrypted with the new primary key. If the primary key configuration is removed, the type **6** passwords cannot be decrypted, which may result in the authentication failure for calls and registrations.

Note When backing up a configuration or migrating the configuration to another device, the primary key is not dumped. Hence the primary key must be configured again manually.

To configure an encrypted preshared key, see Configuring an Encrypted Preshared Key.

Note The encryption type **7** is supported in IOS XE Release 16.11.1a, but will be deprecated in the later releases. Following warning message is displayed when encryption type **7** is configured.

```
Warning: Command has been added to the configuration using a type 7
password. However, type 7 passwords will soon be deprecated. Migrate to
a supported password type 6.
```

Examples

The following example shows how to enable the digest authentication:

```
Router> enable
Router# configure terminal
Router(config)# dial-peer voice 1 pots
Router(config-dial-peer)# authentication username MyUser password 6 MyPassword realm
MyRealm.example.com
```

The following example shows how to remove a previously configured digest authentication:

```
Router> enable
Router# configure terminal
Router(config)# dial-peer voice 1 pots
Router(config-dial-peer)# no authentication username MyUser 6 password MyPassword
```

Related Commands

Command	Description
authentication (SIP UA)	Enables SIP digest authentication globally.
credentials (SIP UA)	Configures a Cisco UBE to send a SIP registration message when in the UP state.
localhost	Configures global settings for substituting a DNS local hostname in place of the physical IP address in the From, Call-ID, and Remote-Party-ID headers of outgoing messages.
registrar	Enables Cisco IOS SIP gateways to register E.164 numbers on behalf of FXS, EFXS, and SCCP phones with an external SIP proxy or SIP registrar.
voice-class sip localhost	Configures settings for substituting a DNS local hostname in place of the physical IP address in the From, Call-ID, and Remote-Party-ID headers of outgoing messages on an individual dial peer, overriding the global setting.

authentication (SIP UA)

To enable SIP digest authentication, use the **authentication** command in SIP UA or voice class tenant configuration mode. To disable SIP digest authentication, use the **no** form of this command.

authentication username *username* **password** { **0** | **6** | **7** } *password* [**realm** *realm*]
no authentication { **username** *username* **password** { **0** | **6** | **7** } *password* [**realm** *realm*] | **all** }

Syntax Description

username *username*	A string representing the username for the user who is providing authentication (must be at least four characters).
password	Specifies password settings for authentication.
0	Specifies encryption type as cleartext (no encryption), which is the default.
6	Specifies secure reversible encryption for passwords using type **6** Advanced Encryption Scheme (AES). **Note** Requires AES primary key to be preconfigured.
7	Specifies encryption type as encrypted.
password	A string representing the password for authentication. If no encryption type is specified, the password is cleartext format. The string must be between 4 and 128 characters.
realm *realm*	(Optional) A string representing the domain where the credentials are applicable.
all	(Optional) Specifies all the authentication entries for the user (sip-ua).

Command Default

SIP digest authentication is disabled.

Command Modes

SIP UA configuration (config-sip-ua)

Voice class tenant configuration (config-class)

Command History

Release	Modification
12.3(8)T	This command was introduced.
15.2(3)T	This command was modified. The **all** keyword was added to the **no** form of the command.
15.6(2)T and IOS XE Denali 16.3.1	This command is now available under voice class tenants.
IOS XE 16.11.1a	Secure reversible encryption for passwords using type **6** Advanced Encryption Scheme (AES) was introduced.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

The following configuration rules are applicable when enabling digest access authentication:

- Only one username can be configured globally in SIP UA configuration mode. Any existing username configuration must be removed before configuring a different username.
- A maximum of five *password* or *realm* arguments are allowed for a given *username*argument.

The *username* and *password* arguments are used to authenticate a user. An authenticating server/proxy issuing a 407/401 challenge response includes a realm in the challenge response and you provide credentials that are valid for that realm. Because it is assumed that a maximum of five proxy servers in the signaling path can try to authenticate a given request from a user-agent client (UAC) to a user-agent server (UAS), a user can configure up to five password and realm combinations for a configured username.

- The realm specification is optional. If omitted, the password that is configured for that username applies to all realms that attempt to authenticate.
- Only one password can be configured at a time for all configured realms. If a new password is configured, it overwrites any previously configured password.

This means that only one global password (one without a specified realm) can be configured. If you configure a new password without configuring a corresponding realm, the new password overwrites the previous one.

- If a realm is configured for a previously configured username and password, that realm specification is added to that existing username and password configuration. However, once a realm is added to a username and password configuration, that username and password combination is valid only for that realm. A configured realm cannot be removed from a username and password configuration without first removing the entire configuration for that username and password--you can then reconfigure that username and password combination with or without a different realm.
- In an entry with both a password and realm, you can change either the password or realm.
- Use the **no authentication all** command to remove all the authentication entries for the user.

It is mandatory to specify the encryption type for the password. If a cleartext password (type **0**) is configured, it is encrypted as type **6** before saving it to the running configuration.

If you specify the encryption type as **6** or **7**, the entered password is checked against a valid type **6** or **7** password format and saved as type **6** or **7** respectively.

Type-6 passwords are encrypted using AES cipher and a user-defined primary key. These passwords are comparatively more secure. The primary key is never displayed in the configuration. Without the knowledge of the primary key, type **6** passwords are unusable. If the primary key is modified, the password that is saved as type 6 is re-encrypted with the new primary key. If the primary key configuration is removed, the type **6** passwords cannot be decrypted, which may result in the authentication failure for calls and registrations.

Note In YANG, you cannot configure the same username across two different realms.

Note When backing up a configuration or migrating the configuration to another device, the primary key is not dumped. Hence the primary key must be configured again manually.

To configure an encrypted preshared key, see Configuring an Encrypted Preshared Key.

Note The encryption type **7** is supported in IOS XE Release 16.11.1a, but will be deprecated in the later releases. Following warning message is displayed when encryption type **7** is configured.

```
Warning: Command has been added to the configuration using a type 7
password. However, type 7 passwords will soon be deprecated. Migrate to
a supported password type 6.
```

Examples

The following example shows how to enable digest access authentication:

```
Router> enable
Router# configure terminal
Router(config)# sip-ua
Router(config-sip-ua)# authentication username MyUser password 6 MyPassword realm example.com
```

The following example shows how to remove a previously configured digest access authentication:

```
Router> enable
Router# configure terminal
Router(config)# sip-ua
Router(config-sip-ua)# no authentication username MyUser password 6 MyPassword realm
example.com
```

Related Commands

Command	Description
authentication (dial peer)	Enables SIP digest authentication on an individual dial peer.
credentials (SIP UA)	Configures a Cisco UBE to send a SIP registration message when in the UP state.
localhost	Configures global settings for substituting a DNS local host name in place of the physical IP address in the From, Call-ID, and Remote-Party-ID headers of outgoing messages.
registrar	Enables Cisco IOS SIP gateways to register E.164 numbers on behalf of FXS, EFXS, and SCCP phones with an external SIP proxy or SIP registrar.
voice-class sip localhost	Configures settings for substituting a DNS local hostname in place of the physical IP address in the From, Call-ID, and Remote-Party-ID headers of outgoing messages on an individual dial peer, overriding the global setting.

authentication method

To set an authentication method at login for calls that come into a dial peer, use the **authenticationmethod** command in voice class AAA configuration mode. To disable the authentication method set at login, use the **no** form of this command.

authentication method *MethListName*
no authentication method *MethListName*

Syntax Description

MethListName	Authentication method list name.

Command Default

When this command is not used to specify a login authentication method, the system uses the **aaaauthenticationloginh323** command as the default.

Command Modes

Voice class AAA configuration

Command History

Release	Modification
12.2(11)T	This command was introduced on the Cisco 3660, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.

Usage Guidelines

This command is used to direct authentication requests to a RADIUS server based on dialed number information service (DNIS) or trunk grouping.

This command is used for directing dial-peer-based authentication requests. The method list must be defined during initial authentication setup.

Examples

In the example below, "dp" is the method list name used for authentication. The method list name is defined during initial authentication setup.

```
voice class aaa 1
 authentication method dp
```

Related Commands

Command	Description
aaa authentication login	Sets AAA authentication at login.
voice class aaa	Enables dial-peer-based VoIP AAA configurations.

authorization method

To set an authorization method at login for calls that are into a dial peer, use the **authorizationmethod** command in voice class AAA configuration mode. To disable the authorization method set at login, use the **no** form of this command.

authorization method *MethListName*
no authorization method *MethListName*

Syntax Description

MethListName	Defines an authorization method list name.

Command Default

When this command is not used to specifiy a login authorization method, the system uses the **aaaauthorizationexech323** command as the default.

Command Modes

Voice class AAA configuration

Command History

Release	Modification
12.2(11)T	This command was introduced on the Cisco 3660, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.

Usage Guidelines

This command is used to direct authentication requests to a RADIUS server based on dialed number information service (DNIS) or trunk grouping.

This command is used for directing dial-peer-based authentication requests. The method list must be defined during initial authentication setup.

Examples

The following example set an authorization method of "dp":

```
voice class aaa 1
 authorization method dp
```

Related Commands

Command	Description
aaa authorization exec	Runs authorization to determine if the user is allowed to run an EXEC shell.
voice class aaa	Enables dial-peer-based VoIP AAA configurations.

auto-config

To enable auto-configuration or to enter auto-config application configuration mode for the Skinny Client Control Protocol (SCCP) application, use the **auto-config**command in global configuration mode. To disable auto-configuration, use the **no** form of this command.

auto-config [**application sccp**]
no auto-config

Syntax Description

application sccp	(Optional) Enters auto-config application configuration mode for the SCCP application.

Command Default

Auto-configuration is disabled.

Command Modes

Global configuration (config)

Command History

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

Examples

The following example shows the**auto-config** command used to enter auto-configuration application configuration mode for the SCCP application and the**noshutdown** command used to enable the SCCP application for download:

```
Router(config)# auto-config application sccp
Router(auto-config-app)#
no shutdown
```

Related Commands

Command	Description
shutdown (auto-config application)	Disables an auto-configuration application for download.
show auto-config	Displays the current status of auto-configuration applications.

auto-cut-through

To enable call completion when a PBX does not provide an M-lead response, use the **auto-cut-through** command in voice-port configuration mode. To disable the auto-cut-through operation, use the **no** form of this command.

auto-cut-through
no auto-cut-through

Syntax Description

This command has no arguments or keywords.

Command Default

Auto-cut-through is enabled.

Command Modes

Voice-port configuration (config-voiceport)

Command History

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

Usage Guidelines

The **auto-cut-through** command applies to ear and mouth (E&M) voice ports only.

Examples

The following example shows enabling of call completion on a router when a PBX does not provide an M-lead response:

```
voice-port 1/0/0
 auto-cut-through
```

Related Commands

Command	Description
show voice port	Displays voice port configuration information.

accounting (gatekeeper)

To enable and define the gatekeeper-specific accounting method, use the **accounting** command in gatekeeper configuration mode. To disable gatekeeper-specific accounting, use the **no**form of this command.

accounting {**username** **h323id** | **vsa**}
no **accounting**

Syntax Description

username h323id	Enables H323ID in the user name field of accounting record.
vsa	Enables the vendor specific attribute accounting format.

Command Default

Accounting is disabled.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced.
12.0(3)T	This command was integrated into Cisco IOS Release 12.0(3)T.
12.1(5)XM	The **vsa** keyword was added.
12.2(2)T	The **vsa** keyword was integrated into Cisco IOS Release 12.2(2)T.
12.2(2)XB1	This command was implemented on the Cisco AS5850 universal gateway.
12.2(33)SRA	This command was integrated into Cisco IOS Release 12.2(33)SRA.
12.3(9)T	This **username h323id**keyword was added.
12.2SX	This command is supported in the Cisco IOS Release 12.2SX train. Support in a specific 12.2SX release of this train depends on your feature set, platform, and platform hardware.

Usage Guidelines

To collect basic start-stop connection accounting data, the gatekeeper must be configured to support gatekeeper-specific H.323 accounting functionality. The **accounting** command enables you to send accounting data to the RADIUS server via IETF RADIUS or VSA attriibutes.

Specify a RADIUS server before using the **accounting** command.

There are three different methods of accounting. The H.323 method sends the call detail record (CDR) to the RADIUS server, the syslog method uses the system logging facility to record the CDRs, and the VSA method collects VSAs.

Examples

The following example enables the gateway to report user activity to the RADIUS server in the form of connection accounting records:

```
aaa accounting connection start-stop group radius
```

```
gatekeeper
 accounting
```

The following example shows how to enable VSA accounting:

```
aaa accounting connection start-stop group radius
gatekeeper
 accounting exec vsa
```

The following example configures H.323 accounting using IETF RADIUS attributes:

```
Router(config-gk)# accounting
username
 h323id
```

The following example configures H.323 accounting using VSA RADIUS attributes:

Router(config-gk)# **accounting vsa**

Related Commands

Command	Description
aaa accounting	Enables AAA accounting of requested services for billing or security purposes.
gatekeeper	Enters gatekeeper configuration mode.

B

backhaul-session-manager

To enter backhaul session manager configuration mode, use the **backhaul-session-manager**command in global configuration mode.

backhaul-session-manager

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(1)T	This command was introduced.
12.2(2)T	This command was implemented on the Cisco 7200.
12.2(4)T	This command was implemented on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.2(2)XB	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850 platform.
12.2(8)T	This command was implemented on Cisco IAD2420. Support for the Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command is supported on the Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release.

Usage Guidelines

Use the **backhaul-session-manager** command to switch to backhaul session manager configuration mode from global configuration mode. Use the **exit** command to exit backhaul session manager configuration mode and return to global configuration mode.

Examples

The following example enters backhaul session manager configuration mode:

```
Router(config)# backhaul-session-manager
Router(config-bsm)#
```

Related Commands

Command	Description
clear backhaul-session-manager group	Resets the statistics or traffic counters for a specified session group.
clear rudpv1 statistics	Clears the RUDP statistics and failure counters.

Command	Description
group	Creates a session group and associates it with a specified session set.
group auto-reset	Configures the maximum auto-reset value.
group cumulative-ack	Configures maximum cumulative acknowledgments.
group out-of-sequence	Configures maximum out-of-sequence segments that are received before an EACK is sent.
group receive	Configures maximum receive segments.
group retransmit	Configures maximum retransmits.
group timer cumulative-ack	Configures cumulative acknowledgment timeout.
group timer keepalive	Configures keepalive (or null segment) timeout.
group timer retransmit	Configures retransmission timeout.
group timer transfer	Configures state transfer timeout.
isdn bind-l3	Configures the ISDN serial interface for backhaul.
session group	Associates a transport session with a specified session group.
set	Creates a fault-tolerant or non-fault-tolerant session set with the client or server option.
show backhaul-session-manager group	Displays status, statistics, or configuration of a specified or all session groups.
show backhaul-session-manager session	Displays status, statistics, or configuration of sessions.
show backhaul-session-manager set	Displays session groups associated with a specific or all session sets.
show rudpv1	Displays RUDP statistics.

bandwidth (dial peer)

To set the maximum bandwidth on a POTS dial peer for an H.320 call, use the **bandwidth** command in dial peer configuration mode. To remove the bandwidth setting, use the **no** form of this command.

bandwidth maximum *value* [**maximum** *value*]
no bandwidth

Syntax Description

maximum *value*	Sets the maximum bandwidth for an H.320 call on a POTS dial peer. The range is 64 to 1024, entered in increments of 64 kilobits per second (kbps). The default is 64.
minimum *value*	(Optional)Sets the minimum bandwidth. Acceptable values are 64 kbps or **minimum***value*=**maximum***value*.

Command Default

No maximum bandwidth is set.

Command Modes

Dial peer configuration (config-dial-peer)

Command History

Release	Modification
12.4(11)T	This command was introduced.

Usage Guidelines

Use this command to set the maximum and minimum bandwidth for an H.320 POTS dial-peer. Only the maximum bandwidth is required. The value must be entered in increments of 64 kbps. The minimum bandwidth setting is optional, and the value must be either 64 kbps or equal to the maximum value setting.

Examples

The following example shows configuration for POTS dial peer 200 with a maximum bandwidth of 1024 kbps:

```
dial-peer voice 200 pots
 bandwidth maximum 1024
```

The following example shows configuration for POTS dial peer 11 with a maximum bandwidth of 640 and a minimum of 64:

```
dial-peer voice 11 pots
 bandwidth maximum 640 minimum 64
```

Related Commands

Command	Description
bandwidth	Specifies the maximum aggregate bandwidth for H.323 traffic and verifies the available bandwidth of the destination gatekeeper.

bandwidth

To specify the maximum aggregate bandwidth for H.323 traffic and verify the available bandwidth of the destination gatekeeper, use the**bandwidth**command in gatekeeper configuration mode. To disable maximum aggregate bandwidth, use the **no** form of this command.

bandwidth {**interzone** | **total** | **session**} {**default** | **zone** *zone-name*} *bandwidth-size*
no bandwidth {**interzone** | **total** | **session**} {**default** | **zone** *zone-name*}

Syntax Description

interzone	Total amount of bandwidth for H.323 traffic from the zone to any other zone.
total	Total amount of bandwidth for H.323 traffic allowed in the zone.
session	Maximum bandwidth allowed for a session in the zone.
default	Default value for all zones.
zone	A particular zone.
zone-name	Name of the particular zone.
bandwidth-size	Maximum bandwidth, in kbps. For**interzone** and **total**, range : 1 to 10000000. For **session**, range:1 to 5000.

Command Default

Maximum aggregate bandwidth is unlimited by default.

Command Modes

Gatekeeper configuration (config-gk)

Command History

Release	Modification
11.3(2)NA	This command was introduced on the Cisco 2500, Cisco 3600 series and the Cisco AS5300.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T. The **bandwidth** command replaced the**zonebw**command.
12.1(5)XM	The **bandwidth** command was recognized without using the **zonegatekeeper** command.
12.2(2)T	The changes in Cisco IOS Release 12.1(5)XM were integrated into Cisco IOS Release 12.2(2)T.
12.2(2)XB1	This command was implemented on the Cisco AS5850.

Usage Guidelines

This command, in conjunction with the **bandwidthremote**command, replaces the **zonegatekeeper**command.

To specify maximum bandwidth for traffic between one zone and any other zone, use the **default** keyword with the **interzone**keyword.

To specify maximum bandwidth for traffic within one zone or for traffic between that zone and another zone (interzone or intrazone), use the**default** keyword with the **total** keyword.

To specify maximum bandwidth for a single session within a specific zone, use the**zone** keyword with the**session** keyword.

To specify maximum bandwidth for a single session within any zone, use the **default** keyword with the **session** keyword.

Examples

The following example configures the default maximum bandwidth for traffic between one zone and another zone to 5000 kbps:

```
gatekeeper
 bandwidth interzone default 5000
```

The following example configures the default maximum bandwidth for all zones to 5000 kbps:

```
gatekeeper
 bandwidth total default 5000
```

The following example configures the default maximum bandwidth for a single session within any zone to 2000 kbps:

```
gatekeeper
 bandwidth session default 2000
```

The following example configures the default maximum bandwidth for a single session with a specific zone to 1000 kbps:

```
gatekeeper
 bandwidth session zone example 1000
```

Related Commands

Command	Description
bandwidth check-destination	Enables the gatekeeper to verify available bandwidth resources at the destination endpoint.
bandwidth remote	Specifies the total bandwidth for H.323 traffic between this gatekeeper and any other gatekeeper.
h323 interface	Defines on which port the proxy listens.
h323 t120	Enables the T.120 capabilities on the router and specifies bypass or proxy mode.

bandwidth check-destination

To enable the gatekeeper to verify available bandwidth resources at the destination endpoint, use the**bandwidthcheck-destinationcommandin**gatekeeper configuration mode. To disable resource verification, use the **no** form of this command.

bandwidth check-destination
no bandwidth check-destination

Syntax Description

This command has no arguments or keywords.

Command Default

Resource verification is disabled by default.

Command Modes

Gatekeeper configuration (config-gk)

Command History

Release	Modification
12.3(1)	This command was introduced.

Examples

The following example activates bandwidth resource verification at the destination:

```
gatekeeper
 bandwidth check-destination
```

Related Commands

Command	Description
bandwidth	Specifies the maximum aggregate bandwidth for H.323 traffic from a zone to another zone, within a zone, or for a session in a zone.
bandwidth remote	Specifies the total bandwidth for H.323 traffic between this gatekeeper and any other gatekeeper.
h323 interface	Defines the port on which port the proxy listens.
h323 t120	Enables the T.120 capabilities on your router and specifies bypass or proxy mode.

bandwidth remote

To specify the total bandwidth for H.323 traffic between this gatekeeper and any other gatekeeper, use the **bandwidthremote** command in gatekeeper configuration mode. To disable total bandwidth specified, use the**no** form of this command.

bandwidth remote *bandwidth-size*
no bandwidth remote

Syntax Description

bandwidth-size	Maximum bandwidth, in kbps. Range: 1 to 10000000.

Command Default

Total bandwidth is unlimited by default.

Command Modes

Gatekeeper configuration (config-gk)

Command History

Release	Modification
12.1(3)XI	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco 7200 series.
12.2(2)T	This command was integrated into Cisco IOS Release 12.2(2)T.
12.2(2)XB1	This command was implemented on the Cisco AS5850.

Usage Guidelines

This command, with the **bandwidth**command, replaces the **zonegatekeeper** command.

Examples

The following example configures the remote maximum bandwidth to 100,000 kbps:

```
gatekeeper
 bandwidth remote 100000
```

Related Commands

Command	Description
bandwidth	Specifies the maximum aggregate bandwidth for H.323 traffic from a zone to another zone, within a zone, or for a session in a zone.
bandwidth check-destination	Enables the gatekeeper to verify available bandwidth resources at the destination endpoint.
h323 interface	Defines which port the proxy listens on.
h323 t120	Enables the T.120 capabilities on your router and specifies bypass or proxy mode.

battery-reversal

To specify battery polarity reversal on a Foreign Exchange Office (FXO) or Foreign Exchange Station (FXS) port, use the **battery-reversal** command in voice-port configuration mode. To disable battery reversal, use the **no** form of this command.

battery-reversal [**answer**]
no battery-reversal [**answer**]

Syntax Description

answer	(Optional) Configures an FXO port to support answer supervision by detection of battery reversal.

Command Default

Battery reversal is enabled

Command Modes

Voice-port configuration (config-voiceport)

Command History

Release	Modification
12.0(7)XK	This command was introduced on the Cisco 2600 series and Cisco 3600 series and on the Cisco MC3810.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.2(2)T	The **answer** keyword was added.

Usage Guidelines

The **battery-reversal** command applies to FXO and FXS voice ports. On Cisco 2600 and 3600 series routers, only analog voice ports in VIC-2FXO-M1 and VIC-2FXO-M2 voice interface cards are able to detect battery reversal; analog voice ports in VIC-2FXO and VIC-2FXO-EU voice interface cards do not detect battery reversal. On digital voice ports, battery reversal is supported only on E1 Mercury Exchange Limited Channel Associated Signaling (MEL CAS); it is not supported in T1 channel associated signaling (CAS) or E1 CAS.

FXS ports normally reverse battery upon call connection. If an FXS port is connected to an FXO port that does not support battery reversal detection, you can use the **nobattery-reversal** command on the FXS port to prevent unexpected behavior.

FXO ports in loopstart mode normally disconnect calls when they detect a second battery reversal (back to normal). You can use the **nobattery-reversal** command on FXO ports to disable this action.

The **battery-reversal** command restores voice ports to their default battery-reversal operation.

If an FXO voice port is connected to the PSTN and supports battery reversal, use the **battery-reversal**command with the **answer** keyword to configure answer supervision. This configures the FXO voice port to detect when a call is answered in order to provide correct billing information.

If the voice port, PSTN, or PBX does not support battery reversal, do not use the **battery-reversal**command because it prevents outgoing calls from being connected. Use the **supervisoryanswerdualtone** command instead.

If an FXO port or its peer FXS port does not support battery reversal, avoid configuring **battery-reversal** or **battery-reversalanswer** on the FXO port. On FXO ports that do not support battery reversal, the **battery-reversal** command can cause unpredictable behavior, and the **battery-reversalanswer** command prevents calls from being answered. To ensure that battery reversal answer is disabled on FXO ports that do not support battery reversal, use the **nobattery-reversal** command.

Examples

The following example disables battery reversal on voice port 1/0/0 on a router:

```
voice-port 1/0/0
 no battery-reversal
```

The following example enables battery reversal to provide answer supervision on voice port 1/0/0 on a router:

```
voice-port 1/0/0
 battery-reversal answer
```

Related Commands

Command	Description
show voice port	Displays voice port configuration information.
supervisory answer dualtone	Enables answer supervision on an FXO voice port on which battery reversal is not supported.

battery-reversal detection-delay

To configure delay detection interval of battery-reversal signal on analog FXO voice port. Use the battery-reversal detection-delay command in voice-port configuration mode. To reset to default, use the no form of this command or battery-reversal detection-delay 0.

This command is only applicable to analog FXO voice port.

battery-reversal detection-delay [**time**]
no battery-reversal detection-delay

Syntax Description

time	0-800 - detection delay time in milliseconds (default to 0)

Command Default

no battery-reversal detection-delay

or

battery-reversal detection-delay 0

Command Modes

Voice-port configuration

bearer-capability clear-channel

To specify the information transfer capability of the bearer capability information element (IE) in the outgoing ISDN SETUP message for Session Initiation Protocol (SIP) early-media calls that negotiate the clear-channel codec, use the **bearer-capability clear-channel** command in SIP configuration mode. To reset the information transfer capability of the bearer capability IE to **speech** (default), use the **no** form of this command.

bearer-capability clear-channel {**audio** | **rdi** | **speech** | **tones** | **udi** [{**bidirectional**}] | **video**}
no bearer-capability clear-channel

Syntax Description

audio	Specifies 3.1 kHz audio.
rdi	Specifies restricted digital information (RDI).
speech	Specifies speech as the information transfer capability. This is the default.
tones	Specifies UDI with tones and announcements.
udi	Specifies unrestricted digital information (UDI).
bidirectional	(Optional) Enables clear-channel codec to UDI bearer capability mapping and UDI bearer capability to clear-channel codec mapping.
video	Specifies video as the information transfer capability.

Command Default

The default information transfer capability setting for the bearer-capability IE is **speech**.

Command Modes

SIP configuration (conf-serv-sip)

Command History

Release	Modification
12.4(15)T	This command was introduced.
15.2(2)T	This command was modified. The **bidirectional** keyword was added.

Usage Guidelines

When a Cisco voice gateway receives a SIP early-media call and negotiates the clear-channel codec, the default for the information transfer capability octet (octet 3) of the bearer capability IE in the outgoing ISDN SETUP message is set to **speech**. Use the **bearer-capability clear-channel** command to change the information transfer capability of the bearer capability IE to a different value.

Note Changing the information transfer capability of the bearer capability IE affects only SIP early-media calls. The information transfer capability value is always **speech** for SIP delayed-media calls, even when the clear-channel codec is negotiated.

You can display the current information transfer capability setting for the bearer capability IE using the **show running-config** command. To show only voice service configuration information, limit the display output to the section on voice service (see the “Examples” section).

Note When the information transfer capability is set to the default value (**speech**), the output of the **show running-config** command does not include the bearer-capability information line.

When you configure the **bearer-capability clear-channel udi bidirectional** command, the ISDN UDI bearer capability is mapped only to the clear-channel codec. Non-UDI bearer capability, like speech, is mapped only to the configured voice codecs. However, the configuration does not indicate the encapsulation type to be used for the clear-channel codec. You can configure the **encap clear-channel standard** or the **voice-class sip encap clear-channel standard** command to use the clear-channel codec mode for negotiation.

Examples

The following examples show how to configure the information transfer capability of the bearer capability IE to UDI to allow for 64 kb/s data transfer over ISDN and how to display the current setting.

Use the following commands to change the information transfer capability setting in the bearer capability IE to UDI:

```
voice service voip
 sip
  bearer-capability clear-channel udi
```

Use the following command to display the current information transfer capability setting:

```
Router# show running-config | section voice service
voice service voip
 h323
 sip
  bearer-capability clear-channel udi
```

Related Commands

Command	Description
encap clear-channel standard	Globally enables RFC 4040-based clear-channel codec negotiation for SIP calls on a Cisco IOS voice gateway or Cisco UBE.
voice-class sip encap clear-channel standard	Enables RFC 4040-based clear-channel codec negotiation for SIP calls on an individual dial peer, overriding the global setting on a Cisco IOS voice gateway or Cisco UBE.

billing b-channel

To enable the H.323 gateway to access B-channel information for all H.323 calls, use the **billingb-channel** command in H.323 voice service configuration mode. To return to the default setting, use the **no** form of this command.

billing b-channel
no billing b-channel

Syntax Description

This command has no arguments or keywords.

Command Default

B-channel information is disabled.

Command Modes

H.323 voice service configuration

Command History

Release	Modification
12.3(7)T	This command was introduced.

Usage Guidelines

This command enables the H.323 application to receive B-channel information of incoming ISDN calls. The B-channel information appears in H.323 ARQ / LRQ messages and can be used during call transfer or to route a call.

Examples

The following example adds B-channel information to the H.323 gateway:

```
Router(config)# voice service voip
Router(conf-voi-serv)# h323
Router(conf-serv-h323)# billing b-channel
```

Related Commands

Command	Description
h323	Enables H.323 voice service configuration commands.
voice service	Enters voice-service configuration mode and specifies the voice encapsulation type.

bind

To bind the source address for signaling and media packets to the IPv4 or IPv6 address of a specific interface, use the **bind** command in SIP configuration mode. To disable binding, use the **no** form of this command.

bind {**control** | **media** | **all**} **source-interface** *interface-id* [{**ipv4-address** *ipv4-address* | **ipv6-address** *ipv6-address*}]
no bind

Syntax Description

control	Binds Session Initiation Protocol (SIP) signaling packets.
media	Binds only media packets.
all	Binds SIP signaling and media packets. The source address (the address that shows where the SIP request came from) of the signaling and media packets is set to the IPv4 or IPv6 address of the specified interface.
source-interface	Specifies an interface as the source address of SIP packets.
interface-id	Specifies one of the following interfaces: • **Async** : ATM interface • **BVI** : Bridge-Group Virtual Interface • **CTunnel** : CTunnel interface • **Dialer** : Dialer interface • **Ethernet** : IEEE 802.3 • **FastEthernet** : Fast Ethernet • **Lex** : Lex interface • **Loopback** : Loopback interface • **Multilink** : Multilink-group interface • **Null** : Null interface • **Serial** : Serial interface (Frame Relay) • **Tunnel** : Tunnel interface • **Vif** : PGM Multicast Host interface • **Virtual-Template** : Virtual template interface • **Virtual-TokenRing** : Virtual token ring
ipv4-address *ipv4-address*	(Optional) Configures the IPv4 address. Several IPv4 addresses can be configured under one interface.
ipv6-address *ipv6-address*	(Optional) Configures the IPv6 address under an IPv4 interface. Several IPv6 addresses can be configured under one IPv4 interface.

Command Default

Binding is disabled.

Command Modes

SIP configuration (conf-serv-sip)

Voice class tenant

Command History

Release	Modification
12.2(2)XB	This command was introduced on the Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco AS5300, Cisco AS5350, and Cisco AS5400.
12.2(2)XB2	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T. This command does not support the Cisco AS5300, Cisco AS5350, Cisco AS5850, and Cisco AS5400 in this release.
12.3(4)T	The **media** keyword was added.
12.4(22)T	Support for IPv6 was added.
Cisco IOS XE Release 2.5	This command was integrated into Cisco IOS XE Release 2.5
Cisco IOS XE Amsterdam 17.2.1r	Introduced support for YANG models.

Usage Guidelines

Async, Ethernet, FastEthernet, Loopback, and Serial (including Frame Relay) are interfaces within the SIP application.

If the **bind** command is not enabled, the IPv4 layer still provides the best local address.

Examples

The following example sets up binding on a SIP network:

```
Router(config)# voice serv voip
Router(config-voi-serv)# sip
Router(config-serv-sip)# bind control source-interface FastEthernet 0
```

Related Commands

Command	Description
sip	Enters SIP configuration mode from voice service VoIP configuration mode.

bind interface

To bind an interface to a Cisco CallManager group, use the **bindinterface**command in SCCP Cisco CallManager configuration mode. To unbind the selected interface, use the **no** form of this command.

bind interface {**dynamic** | *interface-type interface-number*}
no bind interface {**dynamic** | *interface-type interface-number*}

Syntax Description

dynamic	The transcoder interface is chosen based on the remote IP address.
interface-type	Type of selected interface.
interface-number	Number of the selected interface.

Command Default

Interfaces are not associated with any Cisco CallManager group.

Command Modes

SCCP Cisco CallManager configuration (config-sccp-ccm)

Command History

Release	Modification
12.3(8)T	This command was introduced.
15.1(3)T1	This command was modified. The **dynamic** keyword was added.
Cisco IOS XE Amsterdam 17.2.1r	Introduced support for YANG models.

Usage Guidelines

Normally a firewall only opens certain addresses or port combination to the outside world and those addresses can change dynamically. The VoIP technology requires the use of more than one address or port combination to pass information. The **bindinterface** command allows administrators to dictate the use of one network to transport the signaling and another network to transport the media by assigning an interface to a Cisco CallManager group for a specific interface for the signaling or media application.

The selected interface is used for all calls that belong to the profiles that are associated to this Cisco CallManager group. If the **dynamic** keyword is configured, the networking device chooses the transcoder interface based on the remote address. If the interface is not configured, the Skinny Call Control Protocol (SCCP) selects the best interface IP address in the gateway. Interfaces are selected according to user requirements. If there is only one group interface, configuration is not needed.

Note Only one interface can be selected. A given interface can be bound to more than one Cisco CallManager group.

Examples

The following example shows how to bind the interface to a specific Cisco CallManager group:

```
Router(config-sccp-ccm)#bindinterfacefastethernet2:1
```

Related Commands

Command	Description
associate profile	Associates a DSP farm profile with a Cisco CallManager group.
sccp ccm group	Creates a Cisco CallManger group and enters SCCP Cisco CallManager configuration mode.

block

To configure global settings to drop (not pass) specific incoming Session Initiation Protocol (SIP) provisional response messages on a Cisco IOS voice gateway or Cisco Unified Border Element (CUBE), use the **block** command in voice service SIP configuration mode or voice class tenant configuration mode. To disable a global configuration to drop incoming SIP provisional response messages, use the **no** form of this command.

block {**180** | **181** | **183**} [**sdp** {**absent** | **present**}**[system]**]
no block {**180** | **181** | **183**}

Syntax Description

180	Specifies that incoming SIP 180 Ringing messages should be dropped (not passed to the other leg).
181	Specifies that incoming SIP 181 Call is Being Forwarded messages should be dropped (not passed to the other leg).
183	Specifies that incoming SIP 183 Session in Progress messages should be dropped (not passed to the other leg).
sdp	(Optional) Specifies that either the presence or absence of Session Description Protocol (SDP) information in the received response determines when the dropping of specified incoming SIP messages takes place.
absent	Configures the SDP option so that specified incoming SIP messages are dropped only if SDP is absent from the received provisional response.
present	Configures the SDP option so that specified incoming SIP messages are dropped only if SDP is present in the received provisional response.
system	Specifies that the block use the global forced CLI setting. This keyword is available only for the tenant configuration mode.

Command Default

Incoming SIP 180, 181, and 183 provisional responses are forwarded.

Command Modes

Voice service SIP configuration (conf-serv-sip)

Voice class tenant configuration (config-class)

Command History

Release	Modification
12.4(22)YB	This command was introduced. Only SIP 180 and SIP 183 messages are supported on Cisco UBEs.
15.0(1)M	This command was integrated into Cisco IOS Release 15.0(1)M.
15.0(1)XA	This command was modified. Support was added for SIP 181 messages on the Cisco IOS SIP gateway, SIP-SIP Cisco UBEs, and the SIP trunk of Cisco Unified Communications Manager Express (Cisco Unified CME).
15.1(1)T	This command was integrated into Cisco IOS Release 15.1(1)T.

Release	Modification
Cisco IOS XE Release 3.1S	This command was integrated into Cisco IOS XE Release 3.1S
Cisco IOS 15.4(1)T	The **block 183 sdp absent** command was modified to provide support for PRACK and 18x with SDP.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: **system**. This command is now available under voice class tenants.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Use the **block** command in voice service SIP configuration mode to globally configure Cisco IOS voice gateways and Cisco UBEs to drop specified SIP provisional response messages. Additionally, you can use the **sdp** keyword to further control when the specified SIP message is dropped based on either the absence or presence of SDP information.

To configure settings for an individual dial peer, use the **voice-classsipblock** command in dial peer voice configuration mode. To disable global configurations for dropping specified incoming SIP messages on a Cisco IOS voice gateway or Cisco UBE, use the **noblock** command in voice service SIP configuration mode.

Note This command is supported only on outbound dial peers--it is nonoperational if configured on inbound dial peers. You should configure this command on the outbound SIP leg that sends out the initial INVITE message. Additionally, this feature applies only to SIP-to-SIP calls and will have no effect on H.323-to-SIP calls.

Note When the **block 183 sdp absent** command is enabled, the Require: rel1xx header is not disabled, thus supporting for PRACK and 18x with SDP.

Examples

The following example shows how to globally configure dropping of incoming SIP provisional response messages:

```
Router> enable
Router# configure
 terminal
Router(config)# voice
 service
 voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# block 181
```

The following example shows how to globally configure dropping of incoming SIP with SDP provisional response messages:

```
Router> enable
Router# configure
 terminal
Router(config)# voice
 service
 voip
```

```
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# block 183 sdp present
```

The following example shows how to globally configure dropping of incoming SIP without SDP provisional response messages:

```
Router> enable
Router# configure
 terminal
Router(config)# voice
 service
 voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# block 180 sdp absent
```

The following example shows how to globally configure passing all specified incoming SIP provisional response messages (except for those on individual dial peers that are configured to override the global configuration):

```
Router> enable
Router# configure
 terminal
Router(config)# voice
 service
 voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# no block 181
```

The following example shows how to block responses in CUBE in the voice class tenant configuration mode:

```
Router(config-class)# block 181 system
```

Related Commands

Command	Description
map resp-code	Configures global settings on a CUBE for mapping specific incoming SIP provisional response messages to a different SIP response message.
voice-class sip block	Configures an individual dial peer on a Cisco IOS voice gateway or CUBE to drop specified SIP provisional response messages.
voice-class sip map resp-code	Configures a specific dial peer on a CUBE to map specific incoming SIP provisional response messages to a different SIP response message.

block-caller

To configure call blocking on caller ID, use the **block-caller** command in dial peer voice configuration mode. To disable call blocking on caller ID, use the **no** form of this command.

block-caller *number*
no block-caller *number*

Syntax Description

number	Specifies the telephone number to block. You can use a period (.) as a digit wildcard. For example, the command **block-caller5.51234**blocks all numbers beginning with the digit 5, followed by any digit, and then sequentially followed by the digits 5, 1, 2, 3, and 4.

Command Default

Call blocking is disabled; the router does not block any calls for any listed directory numbers (LDNs) based on caller ID numbers

Command Modes

Dial peer configuration (config-dial-peer)

Command History

Release	Modification
12.1(2)XF	This command was introduced on the Cisco 800 series routers.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.

This command is available on Cisco 800 series routers that have plain old telephone service (POTS) ports. For each dial peer, you can enter up to ten caller ID numbers to block. The routers do not accept additional caller ID numbers if ten numbers are already present. In that case, a number must be removed before another caller ID number can be added for blocking.

If you do not specify the **block-caller** command for a local directory, all voice calls to that local directory are accepted. If you specify the **block-caller** command for a local directory, the router verifies that the incoming calling-party number does not match any caller ID numbers in that local directory before processing or accepting the voice call. Each specified caller ID number and incoming calling-party number is compared from right to left, up to the number of digits in the specified caller ID number or incoming calling-party number, whichever has fewer digits.

This command is effective only if you subscribe to caller ID service. If you enable call blocking on caller ID without subscribing to the caller ID service, the routers do not perform the verification process on calling-party numbers and do not block any calls.

Examples

The following example configures a router to block calls from a caller whose caller ID number is 408-555-0134.

```
dial-peer voice 1 pots
 block-caller 4085550134
```

Related Commands

Command	Description
caller-id	Identifies incoming calls with caller ID.

Command	Description
debug pots csm csm	Activates events from which an application can determine and display the status and progress of calls to and from POTS ports.
isdn i-number	Configures several terminal devices to use one subscriber line.
pots call-waiting	Enables local call waiting on a router.
registered-caller ring	Configures the Nariwake service registered caller ring cadence.

bootup e-lead off

To prevent an analog ear and mouth (E&M) voice port from keying the attached radio on router boot up, use the **bootupe-leadoff** command in voice-port configuration mode. To allow the analog E&M voice port to key the attached radio on boot up, use the **no** form of this command.

bootup e-lead off
no bootup e-lead off

Syntax Description

This command has no arguments or keywords.

Command Default

The analog E&M voice port keys the attached radio on radio boot up.

Command Modes

Voice-port configuration (config-voiceport)

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This feature was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

This command configures the E-lead behavior on boot up for both voice ports on the voice interface card (VIC).

Examples

The following example configures the analog E&M voice port to not key the attached radio on router boot up:

```
voice-port 1/0/0
 bootup e-lead off
```

busyout forced

To force a voice port into the busyout state, use the **busyoutforced**command in voice-port configuration mode. To remove the voice port from the busyout state, use the **no** form of this command.

busyout forced
no busyout forced

Syntax Description

This command has no arguments or keywords.

Command Default

The voice-port is not in the busyout state.

Command Modes

Voice-port configuration (config-voiceport)

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was implemented on the Cisco 2600s series and Cisco 3600 series. On the Cisco MC3810, the **voice-portbusyout**command was eliminated in favor of this command.
12.1(2)T	The command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

If a voice port is in the forced busyout state, only the **nobusyoutforced** command can restore the voice port to service.

To avoid conflicting command-line interface (CLI) commands, do not use the **busyoutforced** command and the **ds0busyout**command on the same controller.

Examples

The following example forces analog voice port 3/1/1 on a Cisco 3600 router into the busyout state:

```
voice-port 3/1/1
 busyout forced
```

The following example forces digital voice port 0/0:12 on a Cisco 3600 router into the busyout state:

```
voice-port 0/0:12
 busyout forced
```

Related Commands

Command	Description
busyout-monitor interface	Configures a voice port to monitor a serial interface for events that would trigger a voice-port busyout.
busyout seize	Changes the busyout seize procedure for a voice port.
show voice busyout	Displays information about the voice busyout state.

busyout monitor

To place a voice port into the busyout monitor state, enter the **busyoutmonitor** command in voice-portconfiguration mode. To remove the busyout monitor state from the voice port, use the **no** form of this command.

busyout monitor {**serial** *interface-number* | **ethernet** *interface-number* | **keepalive**} [**in-service**]
no busyout monitor {**serial** *interface-number* | **ethernet** *interface-number* | **keepalive**}

Syntax Description

serial	Specifies monitoring of a serial interface. More than one interface can be entered for a voice port.
ethernet	Specifies monitoring of an Ethernet interface. More than one interface can be entered for a voice port.
interface-number	The interface to be monitored for the voice port busyout function.
keepalive	In case of keepalive failures, the selected voice port or ports are busied out.
in-service	(Optional) Configures the voice port to be busied out when any monitored interface comes into service (its state changes to up). If the keyword is not entered, the voice port is busied out when all monitored interfaces go out of service (that is, the state changes to down).

Command Default

The voice port does not monitor any interfaces.

Command Modes

Voice-port configuration (config-voiceport)

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco MC3810.
12.0(5)XE	This command was implemented on the Cisco 7200 series.
12.0(5)XK	This command was implemented on the Cisco 2600 series and Cisco 3600 series.
12.0(7)T	This command was implemented on the Cisco 2600 series and Cisco 3600 series and integrated into Cisco IOS Release 12.0(7)T.
12.0(7)XK	The ability to monitor an Ethernet port was introduced and the **in-service** keyword was added. The **serial** keyword was first supported on the Cisco 2600 series and Cisco 3600 series.
12.1(1)T	The implementation of this command on the Cisco 7200 series was integrated into Cisco IOS Release 12.1(1)T.
12.1(2)T	The **serial** and **ethernet** keywords were added, the **in-service** keyword was integrated into Cisco IOS Release 12.1(2)T, and the *interface-number*argument was added to the **serial** and **ethernet** keywords.
12.1(3)T	The **interface** keyword was removed.

Release	Modification
12.4(6)T	The **keepalive** keyword was added.

Usage Guidelines

When you place a voice port in the busyout monitor state, the voice port monitors the specified interface and enters the busyout state when the interface is down. This down state forces the rerouting of calls.

The **busyoutmonitor**command monitors only the up or down status of an interface--not end-to-end TCP/IP connectivity.

When an interface is operational, a busied-out voice port returns to its normal state.

This feature can monitor LAN, WAN, and virtual subinterfaces.

A voice port can monitor multiple interfaces at the same time. To configure a voice port to monitor multiple interfaces, reenter the **busyoutmonitor**command for each additional interface to be monitored.

If you specify more than one monitored interface for a voice port, all the monitored interfaces must be down to trigger busyout on the voice port.

You can combine in-service and out-of-service monitoring on a voice port. The following rule describes the action if monitored interfaces change state. A voice port is busied out if either of the following occurs:

- Any interface monitored for coming into service comes up.
- All interfaces monitored for going out of service go down.

Examples

The following example shows configuration of analog voice port 1/2 to busy out if serial port 0 or 1 comes into service:

```
voice-port 1/2
 busyout monitor serial 0 in-service
 busyout monitor serial 1 in-service
```

The following example shows configuration of digital voice port 1/2/2 on a Cisco 3600 series router to busy out if serial port 0 goes out of service:

```
voice-port 1/2/2
 busyout monitor serial 0
```

The following example shows configuration of the voice port to monitor two serial interfaces and an Ethernet interface. When all these interfaces are down, the voice port is busied out. When at least one interface is operating, the voice port is put back into a normal state.

```
voice-port 3/0:0
 busyout monitor ethernet 0/0
 busyout monitor serial 1/0
 busyout monitor serial 2/0
```

The following example shows configuration of the voice port to be busied out in case of a keepalive failure:

```
voice-port 10
 busyout monitor keepalive
```

Related Commands

Command	Description
busyout forced	Forces a voice port into the busyout state.
busyout monitor probe	Configures a voice port to enter busyout state if an SAA probe signal returned from a remote interface crosses a delay or loss threshold.
busyout seize	Changes the busyout seize procedure for a voice port.
show voice busyout	Displays information about the voice busyout state.
voice-port busyout	Places all voice ports associated with a serial or ATM interface into a busyout state.

busyout monitor action

To place a voice port into graceful or shutdown busyout state when triggered by the busyout monitor, use the **busyoutmonitoraction**command in voice-port configuration mode. To remove the voice port from the busyout state, use the **no** form of this command.

busyout monitor action {**graceful** | **shutdown** | **alarm blue**}
no busyout monitor action {**graceful** | **shutdown** | **alarm blue**}

Syntax Description

graceful	Graceful busyout state.
shutdown	D-channel shutdown busyout state.
alarm blue	Shutdown state with a blue alarm, also known as an alarm-indication signal (AIS).

Command Default

Default voice busyout behavior without this command is a forced busyout.

Default voice busyout behavior for PRI depends on whether or not the ISDN switch type supports service messages:

- If the switch type supports service messages, default voice busyout behavior is to transmit B-channel out-of-service (OOS) messages and to keep the D channel active. D-Channel service-messages are supported on the following ISDN switch-types: NI, 4ESS (User Side only), 5ESS (User Side only), DMS100.
- If the switch type does not support service messages, default voice busyout behavior is to bring down the D channel.
- For switch-types not specified above, the D-channel is taken down when the **busyoutmonitoractiongraceful** is configured.

Command Modes

Voice-port configuration (config-voiceport)

Command History

Release	Modification
12.2(13)T	The **busyoutmonitoractiongraceful**command was introduced on the following platforms: Cisco 2600 series, Cisco 2600XM, Cisco 2691, Cisco 3640, Cisco 3660, Cisco 3725, and Cisco VG200.
12.3(6)	The **busyoutmonitoractionshutdown** command was introduced on the following platforms: Cisco 1700 series, Cisco IAD2420 series, Cisco 2600 series, Cisco 2600XM series, Cisco 2691, Cisco 3600 series, Cisco 3700 series, Cisco 4224, Cisco 7200 series, Cisco 7301, Cisco 7400 series, Cisco MC3810, Cisco WS-X4604-GWY, and Cisco VG200.
12.3(7)T	The **busyoutmonitoractionshutdown** command was integrated into Cisco IOS Release 12.3(7)T and support was added for the Cisco IAD2430 series.
12.4(6)T	The **busyoutmonitoractiongraceful** and **busyoutmonitoractionshutdown**commands were introduced to replace the **busyoutactiongraceful**and**busyoutactionshutdown**commands.

Release	Modification
12.4(9)T	The **busyoutmonitoraction**command was introduced to combine the **busyoutmonitoractiongraceful** and **busyoutmonitoractionshutdown** commands. The **shutdownalarmblue** keywords were added.

Usage Guidelines

Use this command to control busyout behavior that is triggered by the**busyoutmonitor**command.

This command with the **graceful**keyword busies out the voice port immediately or, if there is an active call on this voice port, waits until the call is over.

This command with the **shutdown** keyword has the following attributes:

- Before Cisco IOS Release 12.2(8)T, when voice busyout is triggered on a PRI voice port, the D channel is deactivated until the busyout trigger is cleared. Some ISDN switch types, however, support in-service and OOS Q.931 messages that permit B channels to be taken out of service while still keeping the D channel active. Starting in Cisco IOS Release 12.3(8)T for these ISDN switch types, OOS messages are sent and the D channel is kept active when a voice busyout is triggered.
- This keyword is available only for PRI voice ports.
- For switch-types not specified above, the D-channel is be taken down when the **busyoutmonitoractiongraceful**command is configured.

Examples

The following example shows analog voice-port busyout state set to graceful:

```
voice-port 2/0:15
 busyout monitor action graceful
```

The following example shows E1 PRI voice-port busyout state set to shutdown:

```
voice-port 1/1:15 (E1 PRI)
 busyout monitor gatekeeper
 busyout monitor action shutdown
```

The following example shows T1 PRI voice-port busyout state set to shutdown:

```
voice-port 0/1:23 (T1 PRI)
 busyout monitor gatekeeper
 busyout monitor action shutdown
```

Related Commands

Command	Description
busyout forced	Forces a voice port into busyout state.
busyout monitor	Configures a voice port to monitor an interface for events that would trigger voice-port busyout.
busyout monitor backhaul	Configures a voice port to enter busyout-monitor state with backhaul-L3 connectivity monitoring during a WAN failure.
busyout monitor gatekeeper	Configures a voice port to enter busyout state if connectivity to the gatekeeper is lost.

Command	Description
busyout monitor probe	Configures a voice port to enter busyout state if an SAA probe signal returned from a remote, IP-addressable interface crosses a specified delay or loss threshold.
busyout seize	Changes the busyout seize procedure for a voice port.
show voice busyout	Displays information about voice-busyout state.
voice-port	Enters voice-port configuration mode and identifies the voice port to be configured.

busyout monitor backhaul

To configure a voice port to enter busyout-monitor state with backhaul-L3 connectivity monitoring during a wide-area-network (WAN) failure, use the **busyoutmonitorbackhaul**command in voice-port configuration mode. To disable busyout-monitor state, use the **no** form of this command.

busyout monitor backhaul
no busyout monitor backhaul

Syntax Description

This command has no arguments or keywords.

Command Default

If this command is not used, the voice port is not configured to enter busyout state during a WAN failure.

Command Modes

Voice-port configuration (config-voiceport)

Command History

Release	Modification
12.4(9)T	This command was introduced.

Usage Guidelines

Use this command to implement backhaul-L3 connectivity monitoring.

Examples

The following example configures a voice port to enter busyout-monitor state with backhaul-L3 connectivity monitoring during a WAN failure:

```
Router(config-voiceport)# busyout monitor backhaul
```

Related Commands

Command	Description
busyout monitor action	Places a voice port into busyout state.
busyout monitor	Configures a voice port to enter busyout-monitor state.

busyout monitor gatekeeper

To configure a voice port to enter the busyout state if connectivity to the gatekeeper is lost, use the **busyoutmonitorgatekeeper** command in voice-port configuration mode. To configure the monitor to trigger a busyout when any voice port assigned to a specific voice class loses connectivity to the gatekeeper, use the **busyoutmonitorgatekeeper** command in voice-class configuration mode. To disable the busyout monitoring state for the gatekeeper, use the **no** form of this command.

busyout monitor gatekeeper
no busyout monitor gatekeeper

Syntax Description

This command has no arguments or keywords.

Command Default

If this command is not used, the voice port or voice class is not configured to enter a busyout state if connectivity to the gatekeeper is lost.

Command Modes

Voice-class configuration (config-voice-class)
Voice-port configuration (config-voiceport)

Command History

Release	Modification
12.2(13)T	This command was introduced on the following platforms: Cisco 2600 series, Cisco 2600XM, Cisco 2691, Cisco 3640, Cisco 3660, Cisco 3725 and Cisco VG200.
12.4(6)T	This command was extended to include functionality in voice-class configuration mode.

Usage Guidelines

Use this command to monitor the connection between the gateway and gatekeeper. In voice-port configuration mode, if a voice port loses connectivity to the gatekeeper, the voice port enters a busyout state. In voice configuration mode, if any voice port assigned to a specific voice class loses connectivity to the gatekeeper, a busyout is triggered.

Examples

The following example shows the busyout monitor state set to busy out the port according to the state of the gatekeeper:

```
voice-port 1/1/1
 busyout monitor gatekeeper
```

The following example enters voice-class (busyout) configuration mode and creates a voice class named 33. The monitor is set to busyout when any voice port in voice class 33 loses connectivity to the gatekeeper:

```
voice-class busyout 33
 busyout monitor gatekeeper
```

Related Commands

Command	Description
busyout monitor action graceful	Places a voice port into the graceful busyout state when triggered by the busyout monitor.

Command	Description
busyout monitor action shutdown	Shuts down the voice port immediately, but if there is an active call it waits until the call is over.
busyout forced	Forces a voice port into the busyout state.
busyout monitor	Configures a voice port to monitor an interface for events that would trigger a voice-port busyout.
busyout monitor probe	Configures a voice port to enter the busyout state if an SAA probe signal returned from a remote, IP-addressable interface crosses a specified delay or loss threshold.
busyout seize	Changes the busyout seize procedure for a voice port.
show voice busyout	Displays information about the voice busyout state.
voice-port	Enters voice-port configuration mode and identifies the voice port to be configured.

busyout monitor probe

To configure a voice port to enter the busyout state if a Service Assurance Agent (SAA) probe signal is returned from a remote IP-addressable interface after the expiration of a specified delay or loss threshold, use the **busyoutmonitorprobe** command invoice-port configuration mode or voice class busyout mode. To configure a voice port not to monitor SAA probe signals, use the **no** form of this command.

busyout monitor probe [**icmp-ping**] *ip-address* [{**codec** *codec-type* | **size** *bytes*}] [{**icpif** *number* | **loss** *percent* **delay** *milliseconds*}] [**grace-period** *seconds*] **size**
no busyout monitor probe *ip-address*

Syntax Description

icmp-ping	(Optional) Configures voice-port parameters to use ICMP pings to monitor IP destinations.
ip -address	The IP address of a target interface for the SAA probe signal.
codec	(Optional) Configures the profile of the SAA probe signal to mimic the packet size and interval of a specific codec type.
codec -type	(Optional) The codec type for the SAA probe signal. Available options are as follows: • **g711a** --G.711 a-law • **g711u** --G.711 mu-law (the default) • **g729** --G.729 • **g729a** --G.729 Annex A • **g729b** --G.729 Annex B
size *bytes*	(Optional) Size (in bytes) of the ping packet. Default is 32.
icpif	(Optional) Configures the busyout monitor probe to use an Impairment/Calculated Planning Impairment Factor (ICPIF) loss/delay busyout threshold, in accordance with ITU-T G.113. The ICPIF numbers represent predefined combinations of loss and delay.
number	(Optional) The ICPIF threshold for initiating a busyout condition. Range is from 0 to 30. Low numbers are equivalent to low loss and delay thresholds.
loss	(Optional) Configures the percentage-of-packets-lost threshold for initiating a busyout condition.
percent	(Optional) The loss value (expressed as a percentage) for initiating a busyout condition. Range is from 1 to 100.
delay	(Optional) Configures the average packet delay threshold for initiating a busyout condition.
milliseconds	(Optional) The delay threshold, in milliseconds, for initiating a busyout condition. Range is from 1 to 2,147,483,647.
grace-period	(Optional) Configures a time limit that the system waits before initiating a busyout condition after the loss of SAA probe connectivity.

seconds	(Optional) Number of seconds for the duration of the grace period. Range is from 30 to 300.

Command Default

If the **busyoutmonitorprobe**command is not entered, the voice port does not monitor SAA probe signals.

If the **busyoutmonitorprobe**command is entered with no optional keywords or arguments, the default codec type is G.711 a-law, the default loss and delay thresholds are the threshold values that are configured with the **callfallbackthresholddelay-loss**command, and the loss of SAA connectivity causes an immediate forced busyout condition.

Command Modes

Voice-port configuration and voice class busyout

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco 2600 and Cisco 3600 series and on the Cisco MC3810.
12.3(15)	This command was integrated into Cisco IOS Release 12.3(15) and the **grace-period** keyword and *seconds* argument were added.
12.4(1)	This command was integrated into Cisco IOS Release 12.4(1).
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

A voice port can monitor multiple interfaces at the same time. To configure a voice port to monitor multiple interfaces, enter the **busyoutmonitorprobe**command for each additional interface to be monitored.

Caution The **busyoutmonitorprobe**command is effective only if the call fallback function is enabled on the source router, and the SAA responder is enabled on the target router. To enable the call fallback function, you must enter the **callfallbackactive** command for the **busyoutmonitorprobe** command to work.

The SAA probe is transmitted periodically with a period determined by the call fallback function.

Low thresholds of ICPIF, loss, and delay result in early busyout when the link deteriorates, thereby raising the voice minimum quality level. High thresholds prevent busyout until loss and delay are long, allowing transmission of lower-quality voice.

Caution If thresholds are set too low, the link can alternate between in-service and out-of-service states, causing repeated interruptions of traffic.

Before the introduction of the **grace-period** keyword to the **busyoutmonitorprobe** command, the loss of SAA probe connectivity was sufficient to immediately enforce busyout, causing service and connectivity problems in some networks because busyout conditions could occur frequently and abruptly. To improve busyout monitoring via SAA probes, the **grace-period** setting allows for an additional timer that must expire before a busyout condition is enforced. That is, the SAA probes and the period of grace must both expire before a busyout condition is invoked. If the SAA IP connectivity is restored within the period of grace, the busyout condition does not occur.

Note: To disable the **grace-period** option, you must first enter the**nobusyoutmonitorprobe** command and then re-enter the **busyoutmonitorprobe**command without the **grace-period** option.

The **grace-period** keyword is not available in Cisco IOS Release 12.3T.

Examples

The following example shows how to configure analog voice port 1/1/0 to use an SAA probe with a G.711a-law profile to probe the link to two remote interfaces that have IP addresses and to busy out the voice port if SAA probe connectivity is lost for at least 5 seconds. Both links have a loss exceeding 25 percent or a packet delay of more than 1.5 seconds.

```
voice-port 1/1/0
 busyout monitor probe 209.165.202.128 codec g711a loss 25 delay 1500 grace-period 45
 busyout monitor probe 209.165.202.129 codec g711a loss 25 delay 1500 grace-period 45
```

Related Commands

Command	Description
busyout monitor	Places a voice port into the busyout monitor state.
call fallback active	Enables the ICMP-ping or SAA (formerly RTR) probe mechanism for use with the dial-peer **monitorprobe** or voice-port **busyoutmonitorprobe** commands.
call fallback threshold delay-loss	Forces a voice port into the busyout state.
show voice busyout	Displays information about the voice busyout state.
voice class busyout	Creates a voice class for local voice busyout functions.

busyout seize

To change the busyout action for a Foreign Exchange Office (FXO) or Foreign Exchange Station (FXS) voice port, use the **busyoutseize** command in voice-port configuration mode. To restore the default busyout action, use the **no** form of this command.

busyout seize {**ignore** | **repeat**}
no busyout seize

Syntax Description

ignore	Specifies the type of ignore procedure, depending on the type of voice port signaling. See the table below for more information.
repeat	Specifies the type of repeat procedure, depending on the type of voice port signaling. See the table below for more information.

Command Default

See the table below for the default actions for different voice ports and signaling types

Command Modes

Voice-port configuration (config-voiceport)

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was implemented on the Cisco 2600 and Cisco 3600 series.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

The **busyoutseize** command is valid for both analog and digital voice ports. On digital voice ports, the busyout actions are valid whether the busyout results from a voice-port busyout event or from the **ds0-busyout** command.

The voice port returns to an idle state when the event that triggered the busyout disappears.

The table below describes the busyout actions for the **busyoutseize** settings on each voice port type.

The busyout action for E and M voice ports is to seize the far end by setting lead busy.

Table 1: Busyout Seize Actions for Voice Ports

Voice Port Signaling Type	Procedure Setting (busyout-option command)	Busyout Actions
FXS loop start	Default	Removes the power from the loop. For analog voice ports, this is equivalent to removing the ground from the tip lead. For digital voice ports, the port generates the bit pattern equivalent to removing the ground from the tip lead, or it busies out if the bit pattern exists.

Voice Port Signaling Type	Procedure Setting (busyout-option command)	Busyout Actions
FXS loop start	Ignore	Ignores the ground on the ring lead.
FXS ground start	Default	Grounds the tip lead and stays at this state.
FXS ground start	Ignore	1. Leaves the tip lead open. 2. Ignores the ground on the ring lead.
FXS ground start	Repeat	1. Grounds the tip lead. 2. Waits for the far end to close the loop. 3. The far end closes the loop. 4. If the far end then opens the loop, FXS removes the ground from the tip lead. 5. FXS waits for several seconds before returning to Step 1.
FXO loop start	Default	Closes the loop and stays at this state.
FXO loop start	Ignore	1. Leaves the loop open. 2. Ignores the ringing current on the ring level.
FXO loop start	Repeat	1. Closes the loop. 2. After the detected far end starts the power denial procedure, FXO opens the loop. 3. After the detected far end has completed the power denial procedure, FXO waits for several seconds before returning to Step 1.
FXO ground start	Default	Grounds the tip lead.
FXO ground start	Ignore	1. Leaves the loop open. 2. Ignores the running current on the ring lead, or the ground current on the tip lead.
FXO ground start	Repeat	1. Grounds the ring lead. 2. Removes the ground from the ring lead and closes the loop after the detected far end grounds the tip lead. 3. When the detected far end removes the ground from tip lead, FXO opens the loop. 4. FXO waits for several seconds before returning to Step 1.

Examples

The following example shows configuration of analog voice port 1/1 to perform the ignore actions when busied out:

```
voice-port 1/1
 busyout seize ignore
```

The following example shows configuration of digital voice port 0:2 to perform the repeat actions when busied out:

```
voice-port 0:2
 busyout seize repeat
```

Related Commands

Command	Description
busyout forced	Forces a voice port into the busyout state.
busyout-monitor interface	Configures a voice port to monitor an interface for events that would trigger a voice port busyout.
ds0 busyout	Forces a DS0 time slot on a controller into the busyout state.
show voice busyout	Displays information about the voice busyout state.
voice-port busyout	Places all voice ports associated with a serial or ATM interface into a busyout state.

cable detect through call application stats

cable-detect

To enable cable polling on an analog Foreign Exchange Office (FXO) voice port, use the **cable-detect** command in voice port configuration mode. To disable cable polling, use the **no** form of this command.

cable-detect
no cable-detect

Syntax Description

This command has no arguments or keywords.

Command Default

Cable polling on an analog FXO voice port is disabled.

Command Modes

Voice port configuration (config-voiceport)

Command History

Release	Modification
15.1(1)T	This command was introduced.
15.2(4)M	This command was modified. Cable polling was extended to analog Foreign Exchange Office End Ground Start (FXOGS), Foreign Exchange Office End Loop Start (FXOLS), Foreign Exchange Station End Ground Start (FXSGS), and Foreign Exchange Station End Loop Start (FXSLS) voice ports.

Usage Guidelines

The FXOLS voice port is in the busyout state if the Digital Signal Processor (DSP) detects that no cable is connected between the analog FXSLS and FXOLS ports. If you have configured the **no cable-detect** command and the analog FXOLS voice port is in busyout state because no cable is connected, the Cisco software stops polling the cable connection. The analog FXOLS voice port remains in the busyout state until you use the **shutdown** and **no shutdown** commands to switch the analog FXOLS voice port to the idle state. The **cable-detect** command supports loop start and Central Automatic Message Accounting (CAMA) signaling.

Unlike FXOLS, for analog FXOGS, FXSLS, and FXSGS voice ports, the voice port state does not change when the cable status changes from connected to disconnected or disconnected to connected; only a syslog message is printed to indicate the new cable status. The **cable-detect** command will not show up under the voice port if the analog voice interface does not support cable polling.

For analog FXOGS, the **cable-detect** command can be configured on all FXO voice interface cards.

This command can be configured on the following analog FXOLS voice interface cards (VICs):

- VIC2-2FXS
- VIC2-4FXS
- EM-HDA-6FXO
- EM-HDA-3FXS-4FXO
- EM-HDA-4FXO

For analog FXSLS and FXSGS, this command can be configured on the following FXS voice interface cards:

- VIC3-2FXS/DID

- VIC3-4FXS/DID
- VIC3-2FXS-E/DID
- EM3-HDA-8FXS/DID
- SM-D-72FXS
- SM-D-48FXS-E
- Onboard analog FXS on Cisco 8xx platforms
- Onboard analog FXS on Cisco VG20x and VG2435 platforms

Examples

The following example shows how to enable cable polling on an FXOLS voice port:

```
Device> enable
Device# configure terminal
Device(config)# voice-port 1/2/3
Device(config-voiceport)# cable-detect
```

Related Commands

Command	Description
shutdown	Changes the state of the voice ports for a specific voice interface card to offline.

cable-detect-poll-timer

To configure the cable polling timer value for background polling processes on an analog voice port, use the **cable-detect-poll-timer** command in voice service configuration mode. To disable the polling timer, use the **no** form of this command.

cable-detect-poll-timer *timer-value*
no cable-detect-poll-timer

Syntax Description

timer-value	Cable polling timer value in minutes. The range is from 0 to 1440.

Command Default

The cable polling on analog voice ports is disabled.

Command Modes

Voice service configuration (conf-voi-serv)

Command History

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

Usage Guidelines

Use the **cable-detect-poll-timer** command to configure the cable polling timer value on analog Foreign Exchange Office End Ground Start (FXOGS), Foreign Exchange Office End Loop Start (FXOLS), Foreign Exchange Station End Ground Start (FXSGS), and Foreign Exchange Station End Loop Start (FXSLS) voice ports.

Examples

The following example shows how to enable cable polling on an FXOLS voice port:

```
Device> enable
Device# configure terminal
Device(config)# voice service pots
Device(conf-voi-serv)# cable-detect-poll-timer 100
```

Related Commands

Command	Description
cable-detect	Enables cable polling on analog FXOGS, FXOLS, FXSGS, and FXSLS voice ports.

cac_off

To disable connection admission control (CAC), use the **cac_off**command in interface-ATM-VC configuration mode. To enable CAC, use the **no** form of this command.

cac_off
no cac_off

Syntax Description

This command has no keywords or arguments.

Command Default

Call admission control is enabled.

Command Modes

Interface-ATM-VC configuration

Command History

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

Usage Guidelines

Connection admission control (CAC) is a set of actions taken by each ATM switch during connection setup to determine whether the requested quality of service (QoS) will violate the QoS guarantees for established connections. CAC reserves bandwidth for voice calls, however, the bandwidth required when the lossless compression codec (LLCC) is used is dynamic and usually less than what is generally reserved by CAC. Disabling CAC can help in better utilization of bandwidth when LLCC is used.

Examples

The following example disables call admission control on a PVC:

```
interface ATM0/IMA1.1 point-to-point
 pvc test1 15/135
  cac_off
```

cache (neighbor BE)

To configure the local border element (BE) to cache the descriptors received from its neighbors, use the **cache** command in neighbor BE configuration mode. To disable caching, use the **no** form of this command.

cache
no cache

Syntax Description

This command has no arguments or keywords.

Command Default

Caching is not enabled

Command Modes

Neighbor BE configuration (config-annexg-neigh)

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T. Support for the Cisco AS5300 universal access server, Cisco AS5350, Cisco AS5400 is not included in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

Use this command to configure the local BE to cache the descriptors received from its neighbor. If caching is enabled, the neighbors are queried at the specified interval for their descriptors.

Examples

The following example shows the border element enabled to cache the descriptors from its neighbors.

```
Router(config-annexg-neigh)# id neighbor-id
Router(config-annexg-neigh)# cache
```

Related Commands

Command	Description
id	Configures the local ID of the neighboring BE.
port	Configures the neighbor’s port number that is used for exchanging Annex G messages.
query-interval	Configures the interval at which the local BE queries the neighboring BE.

cache reload time (global application configuration mode)

To configure the router to reload scripts from cache on a regular interval, use the **cachereloadtime** command in global application configuration mode. To set the value to the default, use the **no** form of this command.

cache reload time *bg-minutes*
no cache reload time

Syntax Description

bg -minutes	Number of minutes after which the background process is awakened. This background process checks the time elapsed since the script was last used and whether the script is current: • If the script has not been used in the last "unload time," it unloads the script and quits. The unload time is not configurable. • If the script has been used, the background process loads the script from the URL. It compares the scripts, and if they do not match, it begins using the new script for new calls.

Command Default

30 minutes

Command Modes

Global application configuration

Command History

Release	Modification
12.3(14)T	The **callapplicationcachereloadtime** command was moved to global application configuration mode and changed to **cachereloadtime**.

Examples

The following example displays the **cachereloadtime** command configured to specify 15 minutes before a background process is awakened:

Enter application configuration mode to configure applications and services:

```
application
```

Enter global application configuration mode:

```
global
```

Configure the cache reload time:

```
cache reload time 15
```

Related Commands

Command	Description
call application cache reload time	Configures the router to reload the MGCP scripts from cache on a regular interval.
show call application voice	Displays all Tcl or MGCP scripts that are loaded.

cadence

To define the tone-on and tone-off durations for a call-progress tone, use the **cadence** command in call-progress dualtone configuration mode. To restore the default cadence, use the **no** form of this command.

{**cadence** *cycle-1-on-time* *cycle-1-off-time* [*cycle-2-on-time* *cycle-2-off-time*] [*cycle-3-on-time* *cycle-3-off-time*] [*cycle-4-on-time* *cycle-4-off-time*] | **continuous**}
no **cadence**

Syntax Description

cycle-1-on-time	Tone-on duration for the first cycle of the cadence pattern, in milliseconds (ms). Range is from 0 to 1000. The default is 0.
cycle-1-off-time	Tone-off duration for the first cycle of the cadence pattern, in milliseconds. Range is from 0 to 1000. The default is 0.
cycle-2-on-time	(Optional) Tone-on duration for the second cycle of the cadence pattern, in milliseconds. Range is from 0 to 1000. The default is 0.
cycle-2-off-time	(Optional) Tone-off duration for the second cycle of the cadence pattern, in milliseconds. Range is from 0 to 1000. The default is 0.
cycle-3-on-time	(Optional) Tone-on duration for the third cycle of the cadence pattern, in milliseconds. Range is from 0 to 1000. The default is 0.
cycle-3-off-time	(Optional) Tone-off duration for the third cycle of the cadence pattern, in milliseconds. Range is from 0 to 1000. The default is 0.
cycle-4-on-time	(Optional) Tone-on duration for the fourth cycle of the cadence pattern, in milliseconds. Range is from 0 to 1000. The default is 0.
cycle-4-off-time	(Optional) Tone-off duration for the fourth cycle of the cadence pattern, in milliseconds. Range is from 0 to 1000. The default is 0.
continuous	Continuous call-progress tone is detected.

Command Default

Continuous

Command Modes

Call-progress
dualtone configuration

Command History

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

Usage Guidelines

This command specifies the cadence for a class of custom call-progress tones.

You must define each cadence that you want a voice port to detect. Reenter the command for each additional cadence to be detected.

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

Examples

The following example defines a cadence for a busy tone in the custom-cptone voice class with the name "country-x." This example defines 500 ms tone on and 500 ms tone off.

```
voice class custom-cptone country-x
 dualtone busy
 cadence 500 500
```

The following example configures detection of the default frequency and cadence values for the busy tone in the custom-cptone voice class with the name "country-x". The default frequency is a 300 Hz tone, and the default cadence is continuous.

```
voice class custom-cptone country-x
 dualtone busy
 no cadence
 no frequency
```

Related Commands

Command	Description
supervisory custom-cptone	Associates a class of custom call-progress tones with a voice port.
voice class custom-cptone	Creates a voice class for defining custom call-progress tones.
voice class dualtone-detect-params	Modifies the boundaries and limits for custom call-progress tones defined by the **voiceclasscustom-cptone**command.

cadence-list

To specify a tone cadence pattern to be detected, use the **cadence-list**command in voice-class configuration mode. To delete a cadence pattern, use the **no** form of this command.

cadence-list *cadence-id cycle-1-on-time cycle-1-off-time* [*cycle-2-on-time cycle-2-off-time*] [*cycle-3-on-time cycle-3-off-time*] [*cycle-4-on-time cycle-4-off-time*]
no cadence-list *cadence-id*

Syntax Description

cadence-id	A tag to identify this cadence list. The range is from 1 to 10.
cycle-1-on-time	The tone duration for the first cycle of the cadence pattern. Range is from 0 to 1000 (0 milliseconds to 100 seconds). The default is 0.
cycle-1-off-time	The silence duration for the first cycle of the cadence pattern. Range is from 0 to 1000 (0 milliseconds to 100 seconds). The default is 0.
cycle-2-on-time	(Optional) The tone duration for the second cycle of the cadence pattern. Range is from 0 to 1000 (0 milliseconds to 100 seconds). The default is 0.
cycle-2-off-time	(Optional) The silence duration for the second cycle of the cadence pattern. Range is from 0 to 1000 (0 milliseconds to 100 seconds). The default is 0.
cycle-3-on-time	(Optional) The tone duration for the third cycle of the cadence pattern. Range is from 0 to 1000 (0 milliseconds to 100 seconds). The default is 0.
cycle-3-off-time	(Optional) The silence duration for the third cycle of the cadence pattern. Range is from 0 to 1000 (0 milliseconds to 100 seconds). The default is 0.
cycle-4-on-time	(Optional) The tone duration for the fourth cycle of the cadence pattern. Range is from 0 to 1000 (0 milliseconds to 100 seconds). The default is 0.
cycle-4-off-time	(Optional) The silence duration for the fourth cycle of the cadence pattern. Range is from 0 to 1000 (0 milliseconds to 100 seconds). The default is 0.

Command Default

No cadence pattern is configured.

Command Modes

Voice-class configuration (config-voice-class)

Command History

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

Usage Guidelines

A cadence list enables the router to match a complex tone pattern from a PBX or public switched telephone network (PSTN). A tone is detected if it matches any configured cadence list. You can create up to ten cadence lists, enabling the router to detect up to ten different tone patterns. If the tone to be detected consists of only one on-off cycle, you can configure this in either of two ways:

- Create a cadence list using only the *cycle-1-on-time*and *cycle-1-off-timevariables.*

- Use the **cadence-max-off-time** and **cadence-min-on-time** commands.

You must also configure the times of the **cadence-max-off-time** and **cadence-min-on-time**commands to be compatible with the on and off times specified by the **cadence-list** command. The time of the **cadence-max-off-time** must be equal to or greater than the longest off-time in the cadence list; the **cadence-min-on-time** must be equal to or less than the shortest on-time in the cadence list.

Examples

The following example shows configuration of cadence list 1 with three on/off cycles and cadence list 2 with two on/off cycles for voice class 100:

```
voice class dualtone 100
 cadence-list 1 100 100 300 300 100 200
 cadence-list 2 100 200 100 400
```

Related Commands

Command	Description
cadence-max-off-time	Specifies the maximum off duration for detection of a tone.
cadence-min-on-time	Specifies the minimum on duration for detection of a tone.
voice class dualtone	Creates a voice class for FXO tone detection parameters.

cadence-max-off-time

To specify the maximum time that a tone can be off and still detected as part of a cadence, use the **cadence-max-off-time** command in voice-class configuration mode. To restore the default, use the **no** form of this command.

cadence-max-off-time *time*
no cadence-max-off-time

Syntax Description

time	The maximum off time of a tone that can be detected, in 10-millisecond increments. Range is from 0 to 5000 (0 milliseconds to 50 seconds). The default is 0.

Command Default

0 (no off time)

Command Modes

Voice-class configuration (config-voice-class)

Command History

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

Usage Guidelines

Specify a time value greater than the off time of the tone to be detected, and use a time value greater than 0 to enable detection of a cadenced tone. With the default (0), the router detects only a continuous tone.

Examples

The following example shows configuration of a maximum off duration of 20 seconds for voice class 100:

```
voice class dualtone 100
 cadence-max-off-time 2000
```

Related Commands

Command	Description
cadence-min-on-time	Specifies the minimum on duration for detection of a tone.
cadence-variation	Specifies the cadence variation time allowed for detection of a tone.
voice class dualtone	Creates a voice class for FXO tone detection parameters.

cadence-min-on-time

To specify the minimum time that a tone can be on and still detected as part of a cadence, use the **cadence-min-on-time** command in voice-class configuration mode. To restore the default, use the **no** form of this command.

cadence-min-on-time *time*
no cadence-min-on-time

Syntax Description

time	The minimum *on* time of a tone that can be detected, in 10-millisecond increments. Range is from 0 to 100 (0 milliseconds to 1 seconds). The default is 0.

Command Default

0 (no minimum on time)

Command Modes

Voice-class configuration (config-voice-class)

Command History

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

Usage Guidelines

Specify a time value shorter than the on time of the tone to be detected. With the default (0), a tone of any length is detected.

Examples

The following example shows configuration of a minimum on duration of 30 milliseconds (three 10-ms time intervals) for voice class 100:

```
voice class dualtone 100
 cadence-min-on-time 3
```

Related Commands

Command	Description
cadence-max-off-time	Specifies the maximum off duration for detection of a tone.
cadence-variation	Specifies the cadence variation time allowed for detection of a tone.
voice class dualtone	Creates a voice class for FXO tone detection parameters.

cadence-variation

To specify the cadence variation time allowed for detection of a tone, use the **cadence-variation** command in voice-class configuration mode. To restore the default cadence variation time, use the no form of this command.

cadence-variation *time*
no cadence-variation

Syntax Description

time	The maximum time by which the tone onset can vary from the specified onset time and still be detected, in 10-millisecond increments. Range is from 0 to 200 (0 milliseconds to 2 seconds). The default is 10.

Command Default

10 milliseconds

Command Modes

Voice-class configuration (config-voice-class)

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and the Cisco MC3810.
12.1(5)XM	This command was implemented on the Cisco 2600 series, Cisco 3600 series, and the Cisco MC3810.
12.2(2)T	This command was integrated into Cisco IOS Release 12.2(2)T and implemented on the Cisco 1750 router.

Usage Guidelines

Specify a time value greater than the cadence variation of the tone to be detected. With the default of 0, only those tones that match the configured cadence are detected.

This command creates a detection limit for one parameter within a voice class. You can apply the detection limit to any voice port.

Examples

The following example specifies a cadence variation time of 30 milliseconds for voice class 100:

```
voice class dualtone 100
 cadence-variation 3
```

The following example specifies 80 ms (eight 10-ms time intervals) as the maximum allowable cadence variation in voice class 70:

```
voice class dualtone-detect-params 70
 cadence-variation 8
```

Related Commands

Command	Description
cadence-max-off-time	Specifies the maximum off duration for detection of a tone.

Command	Description
cadence-min-on-time	Specifies the minimum on duration for detection of a tone.
supervisory answer dualtone	Enables answer supervision on a voice port.
supervisory dualtone-detect-params	Assigns the boundary and detection tolerance parameters defined by the**voiceclassdualtone-detect-params**command to a voice port.

call accounting-template

To select an accounting template at a specific location, use the **callaccounting-template**command in global configuration or application configuration mode. To deselect a specific accounting template, use the **no** form of this command.

call accounting-template *acctTempName url*
no call accounting-template *acctTempName url*

Syntax Description

acctTempName	Template name.
url	Location of the template.

Command Default

No default behavior or values

Command Modes

Global configuration (config)
Application configuration

Command History

Release	Modification
12.2(11)T	This command was introduced on the following platforms: Cisco 3660, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.
12.3(14)T	This command was added to the application configuration mode to replace the **callapplicationvoiceaccounting-template**command.

Usage Guidelines

For call detail records, the template name must have a .cdr extension. To select call records based on your accounting needs and to specify the location of an accounting template that defines the applicable vendor-specific attributes (VSAs) for generating those selected call records, use the **callaccounting-template**command in global configuration mode.

The*acctTempName* argument refers to a specific accounting template file that you want to send to the RADIUS server. This template file defines only specific VSAs selected by you to control your call records based on your accounting needs.

Examples

The example below shows the accounting template cdr1 selected from a specific TFTP address.

```
call accounting-template temp-ivr tftp://kyer/sample/cdr/cdr1.cdr
```

Related Commands

Command	Description
call application voice accounting-template	Configures T.37 fax accounting with VoIP AAA nonblocking API.
show call accounting-template voice	Selects an accounting template at a specific location.

call accounting-template voice

To select an accounting template at a specific location, use the **callaccounting-templatevoice** command in global configuration mode. To remove a specific accounting template, use the **no** form of this command.

call accounting-template voice *acctTempName url*
no call accounting-template voice *acctTempName url*

Syntax Description

acctTempName	Template name.
url	Location of the template.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(11)T	This command was introduced on the following platforms: Cisco 3660, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.
12.3(14)T	The **callaccounting-templatevoice**command is replaced by the **callaccounting-template**command in application configuration mode. See the **callaccounting-template**command for more information.

Usage Guidelines

The template name must have a .cdr extension.

To select call records based on your accounting needs and to specify the location of an accounting template that defines the applicable vendor-specific attributes (VSAs) for generating those selected call records, use the **callaccounting-templatevoice** command in global configuration mode.

The*acctTempName* argument refers to a specific accounting template file that you want to send to the RADIUS server. This template file defines only specific VSAs selected by you to control your call records based on your accounting needs.

Examples

The example below shows the accounting template cdr1 selected from a specific TFTP address.

```
call accounting-template voice temp-ivr tftp://kyer/sample/cdr/cdr1.cdr
```

Related Commands

Command	Description
call accounting-template voice reload	Reloads the accounting template.
show call accounting-template voice	Selects an accounting template at a specific location.

call accounting-template voice reload

To reload the accounting template, use the **callaccounting-templatevoicereload** command in privileged EXEC mode.

call accounting-template voice reload *acctTempName*

Syntax Description

reload	Reloads the accounting template from the address (for example, a tftp address) where the template is stored.
acctTempName	Name of the accounting template.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.2(11)T	This command was introduced on the following platforms: Cisco 3660, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.

Usage Guidelines

Use the **callaccounting-templatevoicereload** command to reload the template from the URL defined in the **callaccounting-templatevoice** command. After bootup, if the template file fails to load from the TFTP server, the system tries to automatically reload the file at 5-minute intervals.

Examples

The example below shows how to reload accounting template cdr2:

```
call accounting-template voice reload cdr2
```

Related Commands

Command	Description
call accounting-template voice	Selects an accounting template at a specific location
gw-accounting aaa	Defines and loads the template file at the location defined by the URL.
show call accounting-template voice	Displays the VSAs that are contained in the accounting template.

call-agent

To define the call agent for a Media Gateway Control Protocol (MGCP) profile, use the **call-agent** command in MGCP profile configuration mode. To return to the default values, use the **no** form of this command.

call-agent {*dns-nameip-address*} [*port*] [**service-type** *type*] [**version** *protocol-version*]
no call-agent

Syntax Description

dns-name	Fully qualified domain name (including host portion) for the call agent. For example, "ca123.example.net".
ip-address	IP address of the call agent.
port	(Optional) User Datagram Protocol (UDP) port number over which the gateway sends messages to the call agent. Range is from 1025 to 65535. • The default call-agent UDP port is 2727 for MGCP 1.0, Network-based Call Signaling (NCS) 1.0, and Trunking Gateway Control Protocol (TGCP) 1.0. • The default call-agent UDP port is 2427 for MGCP 0.1 and Simple Gateway Control Protocol (SGCP).
service-type *type*	(Optional) Protocol service type valid values for the *type* argument are **mgcp**, **ncs**, **sgcp**, and **tgcp**. The default service type is **mgcp**.
version *protocol-version*	(Optional) Version number of the protocol. Valid values follow: • Service-type MGCP--**0.1**, **1.0** • Service-type NCS--**1.0** • Service-type SGCP--**1.1**, **1.5** • Service-type TGCP--**1.0** The default service type and version are **mgcp** and **0.1**.

Command Default

The default call-agent UDP port is 2727 for MGCP 1.0, Network-based Call Signaling (NCS) 1.0, and Trunking Gateway Control Protocol (TGCP) 1.0. The default call-agent UDP port is 2427 for MGCP 0.1 and Simple Gateway Control Protocol (SGCP). The default service type and version are MGCP 0.1.

Command Modes

MGCP profile configuration (config-mgcp-profile)

Command History

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

Usage Guidelines

This command is used when values for a MGCP profile are configured.

Call-agent configuration for an MGCP profile (with this command) and global call-agent configuration (with the **mgcpcall-agent** command) are mutually exclusive; the first to be configured on an endpoint blocks configuration of the other on the same endpoint.

Identifying call agents by Domain Name System (DNS) name rather than by IP address in the **call-agent** command provides call-agent redundancy, because a DNS name can have more than one IP address associated with it. If a call agent is identified by a DNS name and a message from the gateway fails to reach the call agent, the **max1lookup** and **max2lookup** commands enable a search from the DNS lookup table for a backup call agent at a different IP address.

The *port* argument configures the call agent port number (the UDP port over which the gateway sends messages to the call agent). The reverse, or the gateway port number (the UDP port over which the gateway receives messages from the call agent), is configured by specifying a port number in the **mgcp** command.

The service type **mgcp** supports the Restart In Progress (RSIP) error messages sent by the gateway if the **mgcpsgcprestartnotify** command is enabled. The service type **sgcp** ignores the RSIP messages.

Examples

The following example defines a call agent for the MGCP profile named "tgcp_trunk":

```
Router(config)# mgcp profile tgcp_trunk
Router(config-mgcp-profile)# call-agent 10.13.93.3 2500 service-type tgcp version 1.0
```

Related Commands

Command	Description
max1 lookup	Enables DNS lookup of the MGCP call agent address when the suspicion threshold value is reached.
max2 lookup	Enables DNS lookup of the MGCP call agent address when the disconnect threshold value is reached.
mgcp	Starts and allocates resources for the MGCP daemon.
mgcp call-agent	Configures the address of the call agent (media gateway controller).
mgcp profile	Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile.

call application alternate

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationalternate**command is replaced by the **service**command in global application configuration mode. See the **service**command for more information.

To specify an alternate application to use if the application that is configured in the dial peer fails, use the **callapplicationalternate** command in global configuration mode. To return to the default behavior, use the **no** form of this command.

call application alternate [*application-name*]
no call application alternate

Syntax Description

application-name	(Optional) Name of the specific voice application to use if the application in the dial peer fails. If a specific application name is not entered, the gateway uses the DEFAULT application.

Command Default

The call is rejected if the application in the dial peer fails.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(11)T	This command was introduced.
12.3(14)T	This command was replaced by the **service**command in global application configuration mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

If this command is not configured, calls are rejected when the dial peer that matches the call does not specify a valid voice application.

In releases before Cisco IOS Release 12.2(11)T, the default application (DEFAULT) was automatically triggered if no application was configured in the dial peer or if the configured application failed. The default application is no longer automatically executed unless the **callapplicationalternate** command is configured.

The application named DEFAULT is a simple application that outputs dial tone, collects digits, and places a call to the dialed number. This application is included in Cisco IOS software; you do not have to download it or configure it by using the **callapplicationvoice** command.

The **callapplicationalternate** command specifies that if the application that is configured in the dial peer fails, the default voice application is executed. If the name of a specific application is entered, that application is triggered if the application configured in the dial peer fails. If the alternate application also fails, the call is rejected.

If an application name is entered, that application must first be configured on the gateway by using the **callapplicationvoice** command.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application alternate
 Warning: This command has been deprecated. Please use the following:
  service
```

The following example configures the DEFAULT application as the alternate:

```
call application alternate
```

The following example configures the application session as the alternate:

```
call application alternate session
```

Related Commands

Command	Description
application	Enables a voice application on a dial peer.
call application voice	Defines the name of a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
service	Loads and configures a specific, standalone application on a dial peer.
show call application voice	Displays information about voice applications.

call application cache reload time

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationcachereloadtime** command is replaced by the **cachereloadtime** command in application configuration global mode. See the **cachereloadtime** command for more information.

To configure the router to reload the Media Gateway Control Protocol (MGCP) scripts from cache on a regular interval, use the **callapplicationcachereloadtime** command in global configuration mode. To set the value to the default, use the **no** form of this command.

call application cache reload time *bg-minutes*
no call application cache reload time

Syntax Description

bg-minutes	Specifies the number of minutes after which the background process is awakened. This background process checks the time elapsed since the script was last used and whether the script is current: • If the script has not been used in the last "unload time," it unloads the script and quits. The unload time is not configurable. • If the script has been used, the background process loads the script from the URL. It compares the scripts, and if they do not match, it begins using the new script for new calls.

Command Default

30 minutes

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco AS5300.
12.3(14)T	This command was replaced by the **cachereloadtime** command in application configuration global mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application cache reload 20
 Warning: This command has been deprecated. Please use the following:
  cache reload time
```

The following example displays the **callapplicationcachereloadtime** command configured to specify 30 minutes before a background process is awakened:

```
call application cache reload time 30
```

Related Commands

Command	Description
cache reload time	Configures the router to reload scripts from cache on a regular interval.
call application voice load	Allows reload of an application that was loaded via the MGCP scripting package.
show call application voice	Displays all Tcl or MGCP scripts that are loaded.

call application dump event-log

To flush the event log buffer for application instances to an external file, use the **callapplicationdumpevent-log**command in privileged EXEC mode.

call application dump event-log

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

This command immediately writes the event log buffer to the external file whose location is defined with the **callapplicationevent-logdumpftp** command in global configuration mode.

Note The**callapplicationdumpevent-log** command and the **callapplicationevent-logdumpftp** command are two different commands.

Examples

The following example flushes the application event log buffer:

```
Router# call application dump event-log
```

Related Commands

Command	Description
call application event-log	Enables event logging for voice application instances.
call application event-log dump ftp	Enables the gateway to write the contents of the application event log buffer to an external file.
call application event-log max-buffer-size	Sets the maximum size of the event log buffer for each application instance.
show call application session-level	Displays event logs and statistics for voice application instances.

call application event-log

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationevent-log** command is replaced by the **event-log**command in application configuration monitor mode. See the **event-log** command for more information.

To enable event logging for all voice application instances, use the **callapplicationevent-log**command in global configuration mode. To reset to the default, use the **no** form of this command.

call application event-log
no call application event-log

Syntax Description

This command has no arguments or keywords.

Command Default

Event logging for voice applications is disabled.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.3(14)T	This command was replaced by the **event-log**command in application configuration monitor mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command enables event logging globally for all voice application instances. To enable or disable event logging for a specific application, use the **callapplicationvoiceevent-log** command.

Note To prevent event logging from adversely impacting system resources for production traffic, the gateway uses a throttling mechanism. When free processor memory drops below 20%, the gateway automatically disables all event logging. It resumes event logging when free memory rises above 30%. While throttling is occurring, the gateway does not capture any new event logs even if event logging is enabled. You should monitor free memory and enable event logging only when necessary for isolating faults.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application event-log
        Warning: This command has been deprecated. Please use the following:
             event-log
```

The following example enables event logging for all application instances:

```
call application event-log
```

Related Commands

Command	Description
call application event-log error-only	Restricts event logging to error events only for application instances.
call application event-log max-buffer-size	Sets the maximum size of the event log buffer for each application instance.
call application interface event-log	Enables event logging for external interfaces used by voice applications.
call application stats	Enables statistics collection for voice applications.
call application voice event-log	Enables event logging for a specific voice application.
call leg event-log	Enables event logging for voice, fax, and modem call legs.
event-log	Enables event logging for applications.
monitor call application event-log	Displays the event log for an active application instance in real-time.
show call application session-level	Displays event logs and statistics for voice application instances.

call application event-log dump ftp

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationevent-logdumpftp**command is replaced by the **event-logdumpftp**command in application configuration monitor mode. See the **event-logdumpftp**command for more information.

To enable the gateway to write the contents of the application event log buffer to an external file, use the **callapplicationevent-logdumpftp**command in global configuration mode. To reset to the default, use the **no** form of this command.

call application event-log dump ftp *server*[{*:port*}]*/file* **username** *username* **password** [{[*encryption-type*]}]*password*
no call application event-log dump ftp

Syntax Description

server	Name or IP address of FTP server where file is located.
: *port*	(Optional) Specific port number on server.
/ *file*	Name and path of file.
username	Username required to access file.
encryption-type	(Optional) The Cisco proprietary algorithm used to encrypt the password. Values are 0 or 7. To disable encryption enter 0; to enable encryption enter 7. If you specify 7, you must enter an encrypted password (a password already encrypted by a Cisco router).
password	Password required to access file.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.3(14)T	This command was replaced by the **event-logdumpftp**command in application configuration monitor mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command enables the gateway to automatically write the event log buffer to the named file either after an active application instance terminates or when the event log buffer becomes full. The default buffer size is 4 KB. To modify the size of the buffer, use the **callapplicationevent-logmax-buffer-size** command. To manually flush the event log buffer, use the **callapplicationdumpevent-log** command in privileged EXEC mode.

Note The**callapplicationdumpevent-log** command and the **callapplicationevent-logdumpftp** command are two different commands.

Enabling the gateway to write event logs to FTP could adversely impact gateway memory resources in some scenarios, for example, when:

- The gateway is consuming high processor resources and FTP does not have enough processor resources to flush the logged buffers to the FTP server.
- The designated FTP server is not powerful enough to perform FTP transfers quickly
- Bandwidth on the link between the gateway and the FTP server is not large enough
- The gateway is receiving a high volume of short-duration calls or calls that are failing

You should enable FTP dumping only when necessary and not enable it in situations where it might adversely impact system performance.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application event-log dump ftp
 Warning: This command has been deprecated. Please use the following:
  event-log dump ftp
```

The following example enables the gateway to write application event logs to an external file named app_elogs.log on a server named ftp-server:

```
call application event-log dump ftp ftp-server/:elogs/app-elogs.log username myname password
 0 mypass
```

The following example specifies that application event logs are written to an external file named app_elogs.log on a server with the IP address of 10.10.10.101:

```
call application event-log dump ftp 10.10.10.101/:elogs/app-elogs.log username myname
password 0 mypass
```

Related Commands

Command	Description
call application dump event-log	Flushes the event log buffer for application instances to an external file.
call application event-log	Enables event logging for voice application instances.
call application event-log max-buffer-size	Sets the maximum size of the event log buffer for each application instance.
event-log dump ftp	Enables the gateway to write the contents of the application event log buffer to an external file.

Command	Description
show call application session-level	Displays event logs and statistics for voice application instances.

call application event-log error-only

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationevent-logerror-only**command is replaced by the **event-logerror-only**command in application configuration monitor mode. See the **event-logerror-only**command for more information.

To restrict event logging to error events only for application instances, use the **callapplicationevent-logerror-only** command in global configuration mode. To reset to the default, use the **no** form of this command.

call application event-log error-only
no call application event-log error-only

Syntax Description

This command has no arguments or keywords.

Command Default

All application events are logged.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.3(14)T	This command was replaced by the **event-logerror-only**command in application configuration monitor mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command limits new event logging to error events only; it does not enable logging. You must use this command with either the **callapplicationevent-log** command, which enables event logging for all voice applications, or with the **callapplicationvoiceevent-log** command, which enables event logging for a specific application. Any events logged before this command is issued are not affected.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application event-log error-only
 Warning: This command has been deprecated. Please use the following:
  event-log error-only
```

The following example enables event logging for error events only:

```
call application event-log
call application event-log error-only
```

Related Commands

Command	Description
call application event-log	Enables event logging for voice application instances.
call application history session event-log save-exception-only	Saves in history only the event logs for application instances that have at least one error.
call application voice event-log	Enables event logging for a specific voice application.
event-log error-only	Restricts event logging to error events only for application instances.
show call application app-level	Displays application-level statistics for voice applications.
show call application session-level	Displays event logs and statistics for voice application instances.

call application event-log max-buffer-size

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationevent-logmax-buffer-size**command is replaced by the **event-logmax-buffer-size**command in application configuration monitor mode. See the **event-logmax-buffer-size**command for more information.

To set the maximum size of the event log buffer for each application instance, use the **callapplicationevent-logmax-buffer-size**command in global configuration mode. To reset to the default, use the **no** form of this command.

call application event-log max-buffer-size *kilobytes*
no call application event-log max-buffer-size

Syntax Description

kilobytes	Maximum buffer size, in kilobytes. Range is 1 to 50. Default is 4.

Command Default

4 *kilobytes*

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.3(14)T	This command was replaced by the **event-logmax-buffer-size**command in application configuration monitor mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

If the event log buffer reaches the limit set by this command, the gateway allocates a second buffer of equal size. The contents of both buffers is displayed when you use the **showcallapplicationsession-level** command. When the first event log buffer becomes full, the gateway automatically appends its contents to an external FTP location if the **callapplicationevent-logdumpftp** command is used.

A maximum of two buffers are allocated for an event log. If both buffers are filled, the first buffer is deleted and another buffer is allocated for new events (buffer wraps around). If the **callapplicationevent-logdumpftp** command is configured and the second buffer becomes full before the first buffer is dumped, event messages are dropped and are not recorded in the buffer.

Do not set the maximum buffer size to more than you need for a typical application session. After an active session terminates, the amount of memory used by the buffer is allocated to the history table and is maintained for the length of time set by the **callapplicationhistorysessionretain-timer** command. Also consider that most fatal errors are captured at the end of an event log.

To conserve memory resources, write the event log buffer to FTP by using the **callapplicationevent-logdumpftp** command.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application event-log max-buffer-size
 Warning: This command has been deprecated. Please use the following:
  event-log max-buffer-size
```

The following example sets the application event log buffer to 8 kilobytes:

```
call application event-log
call application event-log max-buffer-size 8
```

Related Commands

Command	Description
call application dump event-log	Flushes the event log buffer for application instances to an external file.
call application event-log	Enables event logging for voice application instances.
call application event-log dump ftp	Enables the gateway to write the contents of the application event log buffer to an external file.
event-log max-buffer-size	Sets the maximum size of the event log buffer for each application instance.
show call application session-level	Displays event logs and statistics for voice application instances.

call application global

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationglobal**command is replaced by the **global**command in application configuration mode. See the **global**command for more information.

To configure an application to use for incoming calls whose incoming dial peer does not have an explicit application configured, use the **callapplicationglobal** command in global configuration mode. To remove the application, use the**no** form of this command.

call application global *application-name*
no call application global *application-name*

Syntax Description

application-name	Character string that defines the name of the application.

Command Default

The default application is **default** for all dial peers.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(15)ZJ	This command was introduced.
12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(14)T	This command was replaced by the **global**command in application configuration mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

The application defined in the dial peer always takes precedence over the global application configured with the **callapplicationglobal** command. The application configured with this command executes only when a dial peer has no application configured.

The application you configure with this command can be an application other than the default session application, but it must be included with the Cisco IOS software or be loaded onto the gateway with the **callapplicationvoice** command before using this command. If the application does not exist in Cisco IOS software or has not been loaded onto the gateway, this command will have no effect.

Note In Cisco IOS Release 12.3(4)T and later releases, the application-name default refers to the application that supports Open Settlement Protocol (OSP), call transfer, and call forwarding. The default session application in Cisco IOS Release 12.2(13)T and earlier releases has been renamed default.old.c and can still be configured for specific dial peers through the **application** command or globally configured for all inbound dial peers through the **callapplicationglobal** command.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application global
 Warning: This command has been deprecated. Please use the following:
  global
```

In the following example, the clid_authen_collect application is configured as the global application for all inbound dial peers that do not have a specific application configured:

```
call application global clid_authen_collect
```

Related Commands

Command	Description
application	Enables a specific IVR application on a dial peer.
call application voice	Defines the name to be used for an application and indicates the location of the appropriate IVR script to be used with this application.
global	Enters application configuration mode.

call application history session event-log save-exception-only

Note Effective with Cisco IOS Release 12.3(14)T, the**callapplicationhistorysessionevent-logsave-exception-only** command is replaced by the **historysessionevent-logsave-exception-only**command in application configuration monitor mode. See the **historysessionevent-logsave-exception-only**command for more information.

To save in history only the event logs for application sessions that have at least one error, use the **callapplicationhistorysessionevent-logsave-exception-only** command in global configuration mode. To reset to the default, use the **no** form of this command.

call application history session event-log save-exception-only
no call application history session event-log save-exception-only

Syntax Description

This command has no arguments or keywords.

Command Default

All event logs for sessions are saved to history.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.3(14)T	This command was replaced by the **historysessionevent-logsave-exception-only**command in application configuration monitor mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

Application event logs move from active to history after an instance terminates. If you use this command, the voice gateway saves event logs only for instances that had one or more errors. Event logs for normal instances that do not contain any errors are not saved to history.

Note This command does not affect records saved to an FTP server by using the **callapplicationdumpevent-log** command.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application history session event-log save-exception-only
 Warning: This command has been deprecated. Please use the following:
  history session event-log save-exception-only
```

The following example saves an event log in history only if the instance had an error:

```
call application history session event-log save-exception-only
```

Related Commands

Command	Description
call application event-log	Enables event logging for voice application instances.
call application event-log error-only	Restricts event logging to error events only for application instances.
call application event-log max-buffer-size	Sets the maximum size of the event log buffer for each application instance.
call application history session max-records	Sets the maximum number of application instance records saved in history.
call application history session retain-timer	Sets the maximum number of minutes for which application instance records are saved in history.
history session event-log save-exception-only	Saves in history only the event logs for application sessions that have at least one error.

call application history session max-records

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationhistorysessionmax-records**command is replaced by the **historysessionmax-records**command in application configuration monitor mode. See the **historysessionmax-records** command for more information.

To set the maximum number of application instance records saved in history, use the **callapplicationhistorysessionmax-records**command in global configuration mode. To reset to the default, use the **no** form of this command.

call application history session max-records *number*
no call application history session max-records

Syntax Description

number	Maximum number of records to save in history. Range is 0 to 2000. Default is 360.

Command Default

360

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.3(14)T	This command was replaced by the **historysessionmax-records**command in application configuration monitor mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command affects the number of records that display when you use the **showcallapplicationhistorysession-level** command.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application history session max-records
 Warning: This command has been deprecated. Please use the following:
  history session max-records
```

The following example sets the maximum record limit to 500:

```
call application history session max-records 500
```

Related Commands

Command	Description
call application event-log	Enables event logging for voice application instances.
call application history session event-log save-exception-only	Saves in history only the event logs for application instances that have at least one error.
call application history session retain-timer	Sets the maximum number of minutes that application instance records are saved in history.
history session max-records	Sets the maximum number of application instance records saved in history.
show call application session-level	Displays event logs and statistics for voice application instances.

call application history session retain-timer

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationhistorysessionretain-timer**command is replaced by the **historysessionretain-timer** command in application configuration monitor mode. See the **historysessionretain-timer**command for more information.

To set the maximum number of minutes for which application instance records are saved in history, use the **callapplicationhistorysessionretain-timer**command in global configuration mode. To reset to the default, use the **no** form of this command.

call application history session retain-timer *minutes*
no **call application history session retain-timer**

Syntax Description

minutes	Maximum time, in minutes, for which history records are saved. Range is 0 to 4294,967,295. Default is 15.

Command Default

15

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.3(14)T	This command was replaced by the **historysessionretain-timer** command in application configuration monitor mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command affects the number of records that display when you use the **showcallapplicationhistorysession-level** command.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application history session retain-timer
 Warning: This command has been deprecated. Please use the following:
  history session retain-timer
```

The following example sets the maximum time to save history records to 1 hour:

```
call application history session retain-timer 60
```

Related Commands

Command	Description
call application event-log	Enables event logging for voice application instances.
call application history session event-log save-exception-only	Saves in history only the event logs for application instances that have at least one error.
call application history session max-records	Sets the maximum number of application instance records saved in history.
history session retain-timer	Sets the maximum number of minutes for which application instance records are saved in history.
show call application session-level	Displays event logs and statistics for voice application instances.

call application interface dump event-log

To flush the event log buffer for application interfaces to an external file, use the **callapplicationinterfacedumpevent-log**command in privileged EXEC mode.

call application interface dump event-log

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

This command immediately writes the event log buffer to the external file whose location is defined with the **callapplicationinterfaceevent-logdumpftp** command in global configuration mode.

Note The **callapplicationinterfacedumpevent-log** command and the **callapplicationinterfaceevent-logdumpftp** command are two different commands.

Examples

The following example writes the event log buffer to the external file named int_elogs:

```
Router(config)# call application interface event-log dump ftp ftp-server/int_elogs.log
username myname password 0 mypass
Router(config)# exit
Router# call application interface dump event-log
```

Related Commands

Command	Description
call application interface event-log	Enables event logging for external interfaces used by voice applications.
call application interface event-log dump ftp	Enables the voice gateway to write the contents of the interface event log buffer to an external file.
call application interface event-log max-buffer-size	Sets the maximum size of the event log buffer for each application interface.
show call application interface	Displays event logs and statistics for application interfaces.

call application interface event-log

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationinterfaceevent-log** command is replaced by the **interfaceevent-log** command in application configuration monitor mode. See the **interfaceevent-log** command for more information.

To enable event logging for interfaces that provide services to voice applications, use the **callapplicationinterfaceevent-log**command in global configuration mode. To reset to the default, use the **no** form of this command.

call application interface event-log [{**aaa** | **asr** | **flash** | **http** | **ram** | **rtsp** | **smtp** | **tftp** | **tts**} [**server** *server*] [**disable**]]
no call application interface event-log [{**aaa** | **asr** | **flash** | **http** | **ram** | **rtsp** | **smtp** | **tftp** | **tts**} [**server** *server*] [**disable**]]

Syntax Description

aaa	Authentication, authorization, and accounting (AAA) interface type.
asr	Automatic speech recognition (ASR) interface type.
flash	Flash memory of the Cisco gateway.
http	Hypertext Transfer Protocol (HTTP) interface type.
ram	Memory of the Cisco gateway.
rtsp	Real Time Streaming Protocol (RTSP) interface type.
smtp	Simple Mail Transfer Protocol (SMTP) interface type.
tftp	Trivial File Transfer Protocol (TFTP) interface type.
tts	Text-to-speech (TTS) interface type.
server *server*	(Optional) Server name or IP address.
disable	(Optional) Disables event logging for the specified interface type or server.

Command Default

Event logging for application interfaces is disabled.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.3(14)T	This command was replaced by the **interfaceevent-log** command in application configuration monitor mode.

Release	Modification
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command enables event logging globally for all interface types and servers unless you select a specific interface type or server. Specifying an interface type takes precedence over the global command for a specific interface type. Specifying an individual server takes precedence over the interface type.

Note To prevent event logging from adversely impacting system resources for production traffic, the gateway uses a throttling mechanism. When free processor memory drops below 20%, the gateway automatically disables all event logging. It resumes event logging when free memory rises above 30%. While throttling is occurring, the gateway does not capture any new event logs even if event logging is enabled. You should monitor free memory and enable event logging only when necessary for isolating faults.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application interface event-log
 Warning: This command has been deprecated. Please use the following:
  interface event-log
```

The following example enables event logging for all interfaces:

```
call application interface event-log
```

The following example enables event logging for HTTP interfaces only:

```
call application interface event-log http
```

The following example enables event logging for all interfaces except HTTP:

```
call application interface event-log
call application interface event-log http disable
```

The following example enables event logging for all HTTP servers except the server with the IP address of 10.10.1.1:

```
call application interface event-log http
call application interface event-log http server http://10.10.1.1 disable
```

Related Commands

Command	Description
call application interface event-log dump ftp	Enables the gateway to write the contents of the interface event log buffer to an external file.
call application interface event-log error-only	Restricts event logging to error events only for application interfaces.

Command	Description
call application interface event-log max-buffer-size	Sets the maximum size of the event log buffer for each application interface.
call application interface max-server-records	Sets the maximum number of application interface records that are saved.
call application interface stats	Enables statistics collection for application interfaces.
interface event-log	Enables event logging for interfaces providing services to voice applications.
show call application interface	Displays event logs and statistics for application interfaces.

call application interface event-log dump ftp

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationinterfaceevent-logdumpftp** command is replaced by the **interfaceevent-logdumpftp** command in application configuration monitor mode. See the **interfaceevent-logdumpftp** command for more information.

To enable the gateway to write the contents of the interface event log buffer to an external file, use the **callapplicationinterfaceevent-logdumpftp**command in global configuration mode. To reset to the default, use the **no** form of this command.

call application interface event-log dump ftp *server*[{*:port*}]*/file* **username** *username* **password** [{[*encryption-type*]}]*password*
no call application interface event-log dump ftp

Syntax Description

server	Name or IP address of FTP server where the file is located.
: *port*	(Optional) Specific port number on server.
/ *file*	Name and path of file.
username *username*	Username required to access file.
encryption-type	(Optional) Cisco proprietary algorithm used to encrypt the password. Values are 0 or 7. To disable encryption enter 0; to enable encryption enter 7. If you specify 7, you must enter an encrypted password (a password already encrypted by a Cisco router).
password *password*	Password required to access file.

Command Default

Interface event log buffer is not written to an external file.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.3(14)T	This command was replaced by the **interfaceevent-logdumpftp** command in application configuration monitor mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command enables the gateway to automatically write the interface event log buffer to the named file when the buffer becomes full. The default buffer size is 4 KB. To modify the size of the buffer, use the

callapplicationinterfaceevent-logmax-buffer-size command. To manually flush the event log buffer, use the **callapplicationinterfacedumpevent-log** command in privileged EXEC mode.

Note The **callapplicationinterfacedumpevent-log** command and the **callapplicationinterfaceevent-logdumpftp** command are two different commands.

- Enabling the gateway to write event logs to FTP can adversely impact gateway-memory resources in scenarios such as the following:
  - The gateway is consuming high processor resources and FTP does not have enough processor resources to flush the logged buffers to the FTP server.
  - The designated FTP server is not powerful enough to perform FTP transfers quickly
  - Bandwidth on the link between the gateway and the FTP server is not large enough
  - The gateway is receiving a high volume of short-duration calls or calls that are failing

You should enable FTP dumping only when necessary and not enable it in situations where it might adversely impact system performance.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application interface event-log dump ftp
 Warning: This command has been deprecated. Please use the following:
  interface event-log dump ftp
```

The following example specifies that interface event log are written to an external file named int_elogs.log on a server named ftp-server:

```
call application interface event-log dump ftp ftp-server/elogs/int_elogs.log username myname
 password 0 mypass
```

The following example specifies that application event logs are written to an external file named int_elogs.log on a server with the IP address of 10.10.10.101:

```
call application interface event-log dump ftp 10.10.10.101/elogs/int_elogs.log username
myname password 0 mypass
```

Related Commands

Command	Description
call application interface dump event-log	Flushes the event log buffer for application interfaces to an external file.
call application interface event-log	Enables event logging for external interfaces used by voice applications.
call application interface event-log max-buffer-size	Sets the maximum size of the event log buffer for each application interface.
call application interface max-server-records	Sets the maximum number of application interface records that are saved.

Command	Description
interface event-log dump ftp	Enables the gateway to write the contents of the interface event log buffer to an external file.
show call application interface	Displays event logs and statistics for application interfaces.

call application interface event-log error-only

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationinterfaceevent-logerror-only** command is replaced by the**interfaceevent-logerroronlycommandin**application configuration monitor mode. See the **interfaceevent-logerroronly** command for more information.

To restrict event logging to error events only for application interfaces, use the **callapplicationinterfaceevent-logerror-only** command in global configuration mode. To reset to the default, use the **no** form of this command.

call application interface event-log error-only
no call application interface event-log error-only

Syntax Description

This command has no arguments or keywords.

Command Default

All events are logged.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.3(14)T	This command was replaced by the**interfaceevent-logerroronly** command in application configuration monitor mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command limits the severity level of the events that are logged; it does not enable logging. You must use this command with the **callapplicationinterfaceevent-log** command, which enables event logging for all application interfaces.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application interface event-log error-only
        Warning: This command has been deprecated. Please use the following:
             interface event-log error only
```

The following example enables event logging for error events only:

```
call application interface event-log error-only
```

Related Commands

Command	Description
call application interface event-log	Enables event logging for external interfaces used by voice applications.
call application interface event-log max-buffer-size	Sets the maximum size of the event log buffer for each application interface.
call application interface max-server-records	Sets the maximum number of application interface records that are saved.
interface event-log error-only	Restricts event logging to error events only for application interfaces.
show call application interface	Displays event logs and statistics for application interfaces.

call application interface event-log max-buffer-size

Note Effective with Cisco IOS Release 12.3(14)T, the**callapplicationinterfaceevent-logmax-buffer-size**command is replaced by the **interfaceevent-logmax-buffer-size** command in application configuration monitor mode. See the **interfaceevent-logmax-buffer-size** command for more information.

To set the maximum size of the event log buffer for each application interface, use the **callapplicationinterfaceevent-logmax-buffer-size**command in global configuration mode. To reset to the default, use the **no** form of this command.

call application interface event-log max-buffer-size *kilobytes*
no call application interface event-log max-buffer-size

Syntax Description

kilobytes	Maximum buffer size, in kilobytes. Range is 1 to 10. Default is 4.

Command Default

4 kilobytes

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.3(14)T	This command was replaced by the **interfaceevent-logmax-buffer-size** command in application configuration monitor mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

If the event log buffer reaches the limit set by this command, the gateway allocates a second buffer of equal size. The contents of both buffers is displayed when you use the **showcallapplicationinterface** command. When the first event log buffer becomes full, the gateway automatically appends its contents to an external FTP location if the **callapplicationinterfaceevent-logdumpftp** command is used.

A maximum of two buffers are allocated for an event log. If both buffers are filled, the first buffer is deleted and another buffer is allocated for new events (buffer wraps around). If the **callapplicationinterfaceevent-logdumpftp** command is configured and the second buffer becomes full before the first buffer is dumped, event messages are dropped and are not recorded in the buffer.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application interface event-log max-buffer-size
 Warning: This command has been deprecated. Please use the following:
   interface event-log max-buffer-size
```

The following example sets the maximum buffer size to 8 kilobytes:

```
call application interface event-log max-buffer-size 8
```

Related Commands

Command	Description
call application interface dump event-log	Flushes the event log buffer for application interfaces to an external file.
call application interface event-log dump ftp	Enables the gateway to write the contents of the interface event log buffer to an external file.
call application interface max-server-records	Sets the maximum number of application interface records that are saved.
interface event-log max-buffer-size	Sets the maximum size of the event log buffer for each application interface.
show call application interface	Displays event logs and statistics for application interfaces.

call application interface max-server-records

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationinterfacemax-server-records** command is replaced by the**interfacemax-server-records** command in application configuration monitor mode. See the **interfacemax-server-records**command for more information.

To set the maximum number of application interface records that are saved, use the **callapplicationinterfacemax-server-records**command in global configuration mode. To reset to the default, use the **no** form of this command.

call application interface max-server-records *number*
no call application interface max-server-records

Syntax Description

number	Maximum number of records to save. Range is 1 to 100. Default is 10.

Command Default

10

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.3(14)T	This command was replaced by the**interfacemax-server-records** command in application configuration monitor mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

Only the specified number of records from the most recently accessed servers are kept.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application interface max-server-records
        Warning: This command has been deprecated. Please use the following:
             interface max-server-records
```

The following example sets the maximum saved records to 50:

```
call application interface max-server-records 50
```

Related Commands

Command	Description
call application interface event-log	Enables event logging for external interfaces used by voice applications.
call application interface event-log max-buffer-size	Sets the maximum size of the event log buffer for each application interface.
interface max-server-records	Sets the maximum number of application interface records that are saved.
show call application interface	Displays event logs and statistics for application interfaces.

call application interface stats

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationinterfacestats** command is replaced by the **interfacestats** command in application configuration monitor mode. See the **interfacestats** command for more information.

To enable statistics collection for application interfaces, use the **callapplicationinterfacestats** command in global configuration mode. To reset to the default, use the **no** form of this command.

call application interface stats
no call application interface stats

Syntax Description

This command has no arguments or keywords.

Command Default

Statistics collection is disabled.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.3(14)T	This command was replaced by the **interfacestats** command in application configuration monitor mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

To display the interface statistics enabled by this command, use the **showcallapplicationinterface** command. To reset the interface counters to zero, use the **clearcallapplicationinterface** command.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application interface stats

 Warning: This command has been deprecated. Please use the following:
  interface stats
```

The following example enables statistics collection for application interfaces:

```
call application interface stats
```

Related Commands

Command	Description
call application interface event-log	Enables event logging for external interfaces used by voice applications.
clear call application interface	Clears application interface statistics or event logs.
interface stats	Enables statistics collection for application interfaces.
show call application interface	Displays event logs and statistics for application interfaces.
stats	Enables statistics collection for voice applications.

call application session start (global)

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationsessionstart**(global) command is replaced by the the **sessionstart**command in application configuration mode. See the **sessionstart**command for more information.

To start a new instance (session) of a Tcl IVR 2.0 application, use the **callapplicationsessionstart**command in global configuration mode. To stop the session and remove the configuration, use the **no** form of this command.

call application session start *instance-name application-name*
no call application session start *instance-name*

Syntax Description

instance-name	Alphanumeric label that uniquely identifies this application instance.
application-name	Name of the Tcl application. This is the name of the application that was assigned with the **callapplicationvoice** command.

Command Default

This command has no default behavior or values.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(4)T	This command was introduced.
12.3(14)T	The **callapplicationsessionstart** (global configuration) command was replaced by the **sessionstart**command in application configuration mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command starts a new session, or instance, of a Tcl IVR 2.0 application. It cannot start a session for a VoiceXML application because Cisco IOS software cannot start a VoiceXML application without an active call leg.

You can start an application instance only after the Tcl application is loaded onto the gateway with the **callapplicationvoice** command.

If this command is used, the session restarts if the gateway reboots.

The **nocallapplicationsessionstart** command stops the Tcl session and removes the configuration from the gateway. You can stop an application session without removing the configuration by using the**callapplicationsessionstop** command.

VoiceXML sessions cannot be stopped with the **nocallapplicationsessionstart** command because VoiceXML sessions cannot be started with Cisco IOS commands.

If the application session stops running, it does not restart unless the gateway reboots. A Tcl script might intentionally stop running by executing a "call close" command for example, or it might fail because of a script error.

You can start multiple instances of the same application by using different instance names.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application session start
 Warning: This command has been deprecated. Please use the following:
  session start
```

The following example starts a session named my_instance for the application named demo:

```
call application session start my_instance demo
```

The following example starts another session for the application named demo:

```
call application session start my_instance2 demo
```

Related Commands

Command	Description
call application session start (privileged EXEC)	Starts a new instance (session) of a Tcl application from privileged EXEC mode.
call application session stop	Stops a voice application session that is running.
debug voip ivr	Displays debug messages for VoIP IVR interactions.
session start	Starts a new instance (session) of a Tcl IVR 2.0 application.
show call application services registry	Displays a one-line summary of all registered services.
show call application sessions	Displays summary or detailed information about voice application sessions.

call application session start (privileged EXEC)

To start a new instance (session) of a Tcl IVR 2.0 application, use the **callapplicationsessionstart** command in privileged EXEC mode.

call application session start *instance-name* [*application-name*]

Syntax Description

instance-name	Alphanumeric label that uniquely identifies this application instance.
application-name	(Optional) Name of the Tcl application. This is the name of the application that was assigned with the **callapplicationvoice** command. **Note** This argument is optional if the application instance was previously started and stopped.

Command Default

None

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

This command starts a new session, or instance, of a Tcl IVR 2.0 application. It cannot start a session for a VoiceXML application because Cisco IOS software cannot start a VoiceXML application without an active call leg.

You can start an application instance only after the Tcl application is loaded onto the gateway with the **callapplicationvoice** command.

Using this command does not restart the session if the gateway reboots. To automatically restart the session if the gateway reboots, use the **callapplicationsessionstart** command in global configuration mode.

To stop an application session once it starts running, use the **callapplicationsessionstop** command.

If the application session stops running, it does not restart unless the gateway reboots and the **callapplicationsessionstart** command is used in global configuration mode. A Tcl script might intentionally stop running by executing a "call close" command for example, or it might fail due to a script error.

You can start multiple instances of the same application by using different instance names.

Examples

The following example restarts an application session called my_instance:

```
call application session start my_instance
```

Related Commands

Command	Description
call application session start (global configuration)	Starts a new instance (session) of a Tcl application in global configuration mode.

Command	Description
call application session stop	Stops a voice application session that is running.
show call application services registry	Displays a one-line summary of all registered services.
show call application sessions	Displays summary or detailed information about voice application sessions.

call application session stop

To stop a voice application session that is running, use the **callapplicationsessionstop** command in privileged EXEC mode.

call application session stop {**callid** *call-id* | **handle** *handle* | **id** *session-id* | **name** *instance-name*}

Syntax Description

callid *call-id*	Call-leg ID that can be displayed in the output from the **debugvoipivrscript** command if the Tcl script uses puts commands.
handle *handle*	Handle of a session from the Tcl mod_handle infotag.
id *session-id*	Session ID that can be displayed with the **showcallapplicationsessions** command.
name *instance-name*	Instance name that was configured with the **callapplicationsessionstart**command.

Command Default

None

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

This command stops a Tcl IVR 2.0 or VoiceXML application session that is identified by one of four different methods: call ID, handle, session ID, or instance name. To see a list of currently running applications, use the **showcallapplicationsessions** command.

A Tcl session that is stopped with this command receives a session terminate event. The session is expected to close all call legs and stop. If a session does not close itself after a 10-second timer, it is forcibly stopped and all call legs that it controls disconnect.

Using this command to stop a VoiceXML session immediately stops the document interpretation and disconnects the call leg. No VoiceXML events are thrown.

If you stop a Tcl session that is configured to start with the **callapplicationsessionstart** command in global configuration mode, you must remove the session by using the **nocallapplicationsessionstart** command before you can restart it.

To see a list of stopped sessions, use the **showcallapplicationsessions** command. Only stopped sessions that are configured to start with the **callapplicationsessionstart** command in global configuration mode are displayed. If a session was started with the **callapplicationsessionstart** command in privileged EXEC mode, it is not tracked by the system and it is not shown as stopped in the output of the **showcallapplicationsessions** command.

Examples

The following example stops an application session called my_instance:

```
call application session stop name my_instance
```

Related Commands

Command	Description
call application session start (global configuration)	Starts a new instance (session) of a Tcl application from global configuration mode.
show call application services registry	Displays a one-line summary of all registered services.
show call application sessions	Displays summary or detailed information about voice application sessions.

call application stats

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationstats**command is replaced by the **stats** command in application configuration monitor mode. See the **stats** command for more information.

To enable statistics collection for voice applications, use the**callapplicationstats** command in global configuration mode. To reset to the default, use the **no** form of this command.

call application stats
no call application stats

Syntax Description

This command has no arguments or keywords.

Command Default

Statistics collection is disabled.

Command Modes

Global configuration (#)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.3(14)T	This command was replaced by the **stats** command in application configuration monitor mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

To display the application statistics, use the **showcallapplicationsession-level**, **showcallapplicationapp-level**, or **showcallapplicationgateway-level**command. To reset the application counters in history to zero, use the **clearcallapplicationstats** command.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application stats
        Warning: This command has been deprecated. Please use the following:
             stats
```

The following example enables statistics collection for voice applications:

```
call application stats
```

Related Commands

Command	Description
call application event-log	Enables event logging for voice application instances.
call application interface stats	Enables statistics collection for application interfaces.

Command	Description
clear call application stats	Clears application-level statistics in history and subtracts the statistics from the gateway-level statistics.
show call application app-level	Displays application-level statistics for voice applications.
show call application gateway-level	Displays gateway-level statistics for voice application instances.
show call application session-level	Displays event logs and statistics for voice application instances.
stats	Enables statistics collection for voice applications.

call application voice through call denial

call application voice

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoice**command is replaced by the commands shown in the table below. See these commands for more information.

To define the name of a voice application and specify the location of the Tool Command Language (Tcl) or VoiceXML document to load for this application, use the **callapplicationvoice** command in global configuration mode. To remove the defined application and all configured parameters associated with it, use the**no** form of this command.

call application voice *application-name* {*locationav-pair*}
no call application voice *application-name*

Syntax Description

application-name	Character string that defines the name of the voice application.
location	Location of the Tcl script or VoiceXML document in URL format. Valid storage locations are TFTP, FTP, HTTP, and flash memory.
av-pair	Text string that defines attribute-value (AV) pairs specified by the Tcl script and understood by the RADIUS server. Multiple AV pairs can be enclosed in quotes; up to 512 entries are supported.

Command Default

None

Command Modes

Global configuration (config)

Command History

Release	Modification
12.0(4)XH	This command was introduced.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0(7)T. The *location*argument was added.
12.1(3)T	The *av-pair* argument was added for AV pairs.
12.1(5)T	This command was implemented on the Cisco AS5800.
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB	This command was modified to support VoiceXML applications and HTTP server locations on the Cisco AS5300, Cisco AS5350, and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T and implemented on the Cisco 1750.

Release	Modification
12.2(4)XM	This command was implemented on the Cisco 1751. Support for other Cisco platforms is not included in this release.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 2600 series, Cisco 3600 series, Cisco 3725, Cisco 3745, and Cisco 7200 series.
12.2(11)T	This command was implemented for VoiceXML applications. This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 in this release.
12.2(15)T	MCID AV-pairs were added for the *av-pair* argument; they are mcid-dtmf, mcid-release-timer, and mcid-retry-limit.
12.3(8)T	Support was added to allow up to 512 multiple AV pairs (enclosed in quotes) to be used in a single command.
12.3(14)T	This command was replaced. The **callapplicationvoice** command was replaced by the commands shown in the table below.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

The **callapplicationvoice** command was replaced by the commands shown in the table below.

Table 2: call application voice Command Replacements

Command	Command Mode	Purpose
application	Global configuration	Enters application configuration mode to configure voice applications and services.
service	Application configuration	Enters service configuration mode to configure a standalone application, such as a debit card script.
package	Application configuration	Use to load and configure a package. A package is a linkable set of C or Tcl functions that provide functionality invoked by applications or other packages.
param	Application parameter configuration	Use to configure parameters for services or packages.

Use this command when configuring interactive voice response (IVR) or one of the IVR-related features (such as Debit Card) to define the name of an application and to identify the location of the Tcl script or VoiceXML document associated with the application.

A voice application must be configured by using this command before the application can be configured with the **application** command in a dial peer.

Tcl scripts and VoiceXML documents can be stored in any of the following locations: on TFTP, FTP, or HTTP servers, in the flash memory of the gateway, or on the removable disks of the Cisco 3600 series. The audio files that they use can be stored in any of these locations, and on Real-Time Streaming Protocol (RTSP) servers.

HTTP is the recommended protocol for loading applications and audio prompts because of its efficient design for loading information over the web. For example, it has methods for determining how long a file can be cached and whether a cached file is still valid.

Include the file type extension in the filename (.vxml or .tcl) when specifying the document used by the application. Tcl files require the extension .tcl, and VoiceXML documents require .vxml.

Note The **nocallapplicationvoice** command causes all related call application commands--for instance,**callapplicationvoicelanguage** and **callapplicationvoiceset-location**--to be deleted. The **nocallapplicationvoiceapplication-name**command removes the entire application and all parameters, if configured.

Examples

Effective with Cisco IOS Release 12.3(14)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice
        Warning: This command has been deprecated. Please use the following:
             application
               service
                 package
     param
```

The following example defines the fax-relay application and the TFTP server location of the associated Tcl script:

```
call application voice fax-relay tftp://keyer/faxrelay.tcl
```

The following example defines the application "prepaid" and the TFTP server location of the associated Tcl script:

```
call application voice prepaid tftp://keyer/debitcard.tcl
```

The following is an example of AV pair configuration:

```
set avsend(h323-ivr-out,)) "payphone:true"
set avsend(323-ivr-out,1) "creditTime:3400"
```

The AV pair (after the array is defined, as in the prior example) must be sent to the server using the authentication, authorization, and accounting (AAA) authenticate or AAA authorize verbs as follows:

```
aaa authenticate $account $password $avsend
```

The script would use this AV pair whenever it is needed to convey information to the RADIUS server that cannot be represented by the standard vendor-specific attributes (VSAs).

The following example shows how to define the VoiceXML application "vapptest1" and the flash memory location of the associated VoiceXML document "demo0.vxml":

```
call application voice vapptest1 flash:demo0.vxml
```

The following example specifies the MCID application name, the TFTP server location of the associated Tcl script, and the AV-pairs associated with the MCID application:

```
call application voice mcid tftp://keyer/app_mcid.2.0.0.40.tcl
call application voice mcid mcid-dtmf #99
call application voice mcid-retry-limit 3
call application voice mcid mcid-release-timer 90
```

Related Commands

Command	Description
application (dial peer)	Defines the call application in the dial peer.
application (global configuation)	Enters application configuration mode to configure applications.
call application voice language	Defines the language of the audio file for the designated application and passes that information to the application.
call application voice load	Reloads the designated Tcl script or VoiceXML document.
call application voice pin-len	Defines the number of characters in the PIN for the application and passes that information to the application.
call application voice redirect-number	Defines the telephone number to which a call is redirected for the designated application.
call application voice retry-count	Defines the number of times a caller is permitted to reenter the PIN for a designated application and passes that information to the application.
call application voice security trusted	Sets the security level of a VoiceXML application to trusted so that ANI is not blocked.
call application voice set-location	Defines the location, language, and category of the audio files for the designated application and passes that information to the application.
call application voice uid-len	Defines the number of characters in the UID for the designated application and passes that information to the application.
call application voice warning-time	Defines, in seconds, how long in advance a user is warned before the allowed calling time expires for the designated application.
package	Enters application parameter configuration mode to load and configure a package.
param	Loads and configures parameters in a package or a service (application) on the gateway.
service	Loads and configures a specific, standalone application on a dial peer.
show call application voice	Displays information about voice applications.

call application voice access-method

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoiceaccess-method** command was replaced by the **paramaccess-method** command in application parameter configuration mode. See the **paramaccess-method**command for more information.

To specify the access method for two-stage dialing for the designated application, use the **callapplicationvoiceaccess-method**command in global configuration mode. To restore default values for this command, use the **no** form of this command.

call application voice *application-name* **access-method** {**prompt-user** | **redialer**}
no call application voice *application-name* **access-method**

Syntax Description

application-name	Name of the application.
prompt-user	Specifies that no DID is set in the incoming POTS dial peer and that a Tcl script in the incoming POTS dial peer is used for two-stage dialing.
redialer	Specifies that no DID is set in the incoming POTS dial peer and that the redialer device are used for two-stage dialing.

Command Default

Prompt-user (when DID is not set in the dial peer)

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)XI	This command was introduced on the Cisco AS5300.
12.1(5)T	This command was integrated into the Cisco IOS Release 12.1(5)T.
12.2(4)T	This command was introduced on the Cisco 1750.
12.3(14)T	This command was replaced by the **paramaccess-method** command in application parameter configuration mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

Use the **callapplicationvoiceaccess-method**command to specify the access method for two-stage dialing when DID is disabled in the POTS dial peer.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice access-method
        Warning: This command has been deprecated. Please use the following:
             param access-method
```

The following example specifies prompt-user as the access method for two-stage dialing for the app_libretto_onramp9 IVR application:

```
call application voice app_libretto_onramp9 access-method prompt-user
```

Related Commands

Command	Description
call application voice	Loads a specified application onto the router from the TFTP server and gives it an application name by which it is known on the router.
call application voice language	Defines the language of the audio file for the designated application and passes that information to the application.
call application voice load	Reloads the designated Tcl script.
call application voice pin len	Defines the number of characters in the PIN for the application and passes that information to the application.
call application voice redirect number	Defines the telephone number to which a call is redirected--for example, the operator telephone number of the service provider--for the designated application.
call application voice retry count	Defines the number of times a caller is permitted to reenter the PIN for a designated application and passes that information to the application.
call application voice set location	Defines the location, language, and category of the audio files for the designated application and passes that information to the application.
call application voice uid len	Defines the number of characters in the UID for the designated application and passes that information to the application.
call application voice warning time	Defines, in seconds, how long in advance a user is warned before the allowed calling time expires for the designated application.
param access-method	Specifies the access method for two-stage dialing for the designated application.

call application voice account-id-method

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoiceaccount-id-method** command is replaced by the **paramaccount-id-method**command in application parameter configuration mode. See the **paramaccount-id-method** command for more information.

To configure the fax detection IVR application to use a particular method to assign the account identifier, use the **callapplicationvoiceaccountidmethod**command in global configuration mode. To remove configuration of this account identifier, use the **no** form of this command.

call application voice *application-name* **account-id-method** {**none** | **ani** | **dnis** | **gateway**}
no call application voice *application-name* **account-id-method**

Syntax Description

application-name	Name of the defined fax detection IVR application.
none	Account identifier is blank. This is the default.
ani	Account identifier is the calling party telephone number (automatic number identification, or ANI).
dnis	Account identifier is the dialed party telephone number (dialed number identification service, or DNIS).
gateway	Account identifier is a router-specific name derived from the hostname and domain name, displayed in the following format: router-name.domain-name.

Command Default

No default behavior or values.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(5)XM	This command was introduced for the Cisco AS5300.
12.2(2)XB	This command was implemented on the Cisco AS5400 and Cisco AS5350.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745.
12.2(11)T	This command was implemented on the Cisco AS5300, the Cisco AS5350, and Cisco AS5400.
12.3(14)T	This command was replaced by the **paramaccount-id-method**command in application parameter configuration mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

When an on-ramp application converts a fax into an e-mail, the e-mail contains a field called x-account-id, which can be used for accounting or authentication. The x-account-id field can contain information supplied as a result of this command, such as the calling party's telephone number (**ani**), the called party's telephone number (**dnis**), or the name of the gateway (**gateway**).

This command is not supported by Cisco IOS help; that is, if you type**thecallapplicationvoicefax_detectaccount-id-methodcommandandaquestionmark(?)**,the Cisco IOS help does not supply a list of entries that are valid in place of the question mark.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application fax_detect account-id-method gateway
        Warning: This command has been deprecated. Please use the following:
             param account-id-method
```

The following example sets the fax detection IVR application account identifier to the router-specific name derived from the hostname and domain name:

```
call application voice fax_detect account-id-method gateway
```

Related Commands

Command	Description
call application voice	Loads a specified IVR application onto the router from the TFTP server and gives it an application name by which it is known on the router.
call application voice fax dtmf	Configures the fax detection IVR application to recognize a specified digit that indicates a fax call in default-voice and default-fax modes.
call application voice mode	Configures the fax detection IVR application to operate in one of its four modes.
call application voice prompt	Configures the fax detection IVR application to use the specified audio file as a user prompt in listen-first mode, default-voice mode, or default-fax mode.
call application voice voice dtmf	Configures the fax detection IVR application to recognize a specified digit that indicate a voice call in default-voice and default-fax modes.
param account-id-method	Configures an application to use a particular method to assign the account identifier.

call application voice authentication enable

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoiceauthenticationenable** command is replaced by the **paramauthenticationenable**command in application configuration mode. See the **paramauthenticationenable**command for more information.

To enable authentication, authorization, and accounting (AAA) services for a Tool Command Language (TCL) application, use the **callapplicationvoiceauthenticationenable**command in global configuration mode. To disable authentication for a TCL application, use the **no** form of this command.

call application voice *application-name* **authentication enable**
no call application voice *application-name* **authentication enable**

Syntax Description

application-name	Name of the application.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)XI	This command was introduced on the Cisco AS5300.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.2(4)T	This command was implemented on the Cisco 1750.
12.2(8)T	This command was implemented on the Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745.
12.3(14)T	This command was replaced by the **paramauthenticationenable**command in application configuration mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command enables AAA authentication services for a TCL application if a AAA authentication method list has been defined using both the **aaaauthentication**command and the**callapplicationvoiceauthen-list** command.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice authentication enable
        Warning: This command has been deprecated. Please use the following:
   param authentication enable
```

The following example enables a AAA authentication method list (called "sample") to be used with outbound store-and-forward fax.

```
call application voice app_eaample_onramp9 authen-list sample
call application voice app_example_onramp9 authentication enable
```

Related Commands

Command	Description
aaa authentication	Enables AAA accounting of requested services when you use RADIUS or TACACS+.
call application voice authen-list	Specifies the name of an authentication method list for a TCL application.
call application voice authen-method	Specifies the authentication method for a TCL application.

call application voice accounting-list

Note Effective with Cisco IOS Release 12.3(14)T, the**callapplicationvoiceaccounting-list**commandis replaced by the **paramaccounting-list** in application configuration mode. See the **paramaccounting-list** command for more information.

To define the name of the accounting method list to be used for authentication, authorization, and accounting (AAA) with store-and-forward fax on a voice feature card (VFC), use the **callapplicationvoiceaccountinglist** command in global configuration mode. To undefine the accounting method list, use the **no** form of this command.

call application voice *application-name* **accounting-list** *method-list-name*
no call application voice *application-name* **accounting-list** *method-list-name*

Syntax Description

application-name	Name of the application.
method-list-name	Character string used to name a list of accounting methods to be used with store-and-forward fax.

Command Default

No AAA accounting method list is defined

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)XI	This command was introduced on the Cisco AS5300.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.2(4)T	This command was implemented on the Cisco 1750.
12.2(8)T	This command was implemented on the following platforms: Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745.
12.3(14)T	This commandwas replaced by the **paramaccounting-list**in application configuration mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command defines the name of the AAA accounting method list to be used with store-and-forward fax. The method list itself, which defines the type of accounting services provided for store-and-forward fax, is defined using the **aaaaccounting** command. Unlike standard AAA (in which each defined method list can be applied to specific interfaces and lines), the AAA accounting method lists that are used in store-and-forward fax are applied globally.

After the accounting method lists have been defined, they are enabled by using the **mmoipaaareceive-accountingenable**command.

This command applies to both on-ramp and off-ramp store-and-forward fax functions on VFCs. The command is not used on modem cards.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice accounting-list
        Warning: This command has been deprecated. Please use the following:
   param accounting-list
```

The following example defines a AAA accounting method list "example" to be used with store-and-forward fax:

```
aaa new-model
call application voice app_libretto_onramp9 accounting-list example
```

Related Commands

Command	Description
aaa accounting	Enables AAA accounting of requested services when you use RADIUS or TACACS+.
call application voice accounting enable	Enables AAA accounting for a TCL application.
mmoip aaa receive-accounting enable	Enables on-ramp AAA accounting services.

call application voice accounting-template

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoiceaccounting-template** command is obsolete. Use the **callaccounting-template**command in application configuration mode to configure a voice accounting template.

To configure T.37 fax accounting with VoIP authentication, authorization, and accounting (AAA) nonblocking Application Programming Interface (API), use the **callapplicationvoiceaccountingtemplate**command in global configuration mode. To remove the defined application and all configured parameters associated with it, use the**no** form of this command.

call application voice *application-name* **accounting-template** *template-name*
no call application voice *application-name* **accounting-template** *template-name*

Syntax Description

application-name	Defines the name of the T.37 voice application. • Use the **callapplicationvoice**command to define the name of a voice application and specify the location of the Tool Command Language (Tcl) or VoiceXML document to load for this application.
template -name	Defines the name of the template. • Use the **callaccounting-templatevoice**command to define the template name.

Command Default

Disabled

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(1)	This command was introduced.
12.3(14)T	This command is obsolete. Use the **callaccounting-template**command in application configuration mode to configure a voice accounting template.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command enables T.37 fax to be consistent with VoIP AAA accounting services, which uses the Cisco IOS software nonblocking APIs. This command creates accounting templates for faxes by associating the template name with the T.37 onramp or offramp application.

You can define an accounting template to specify information that is included in an accounting packet.

Note This command applies only to T.37 fax.

Use the **showcallactivefax** and the **showcallhistoryfax** commands to check the configuration.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice accounting-template
        Warning: This command has been deprecated. Please use the following:
             call accounting-template
```

The following is an example configuration using the T.37 accounting template:

```
Router(config)# call application voice t37_onramp accounting-template sample-name
Router(config)# call application voice t37_offramp accounting-template sample-name
```

Related Commands

Command	Description
application	Defines the call application in the dial peer.
call accounting-template	Selects an accounting template at a specific location.
call accounting-template voice	Selects an accounting template at a specific location.
call application voice	Defines the name of a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
call application voice language	Defines the language of the audio file for the designated application and passes that information to the application.
call application voice load	Reloads the designated Tcl script or VoiceXML document.
call application voice pin-len	Defines the number of characters in the PIN for the application and passes that information to the application.
call application voice redirect-number	Defines the telephone number to which a call is redirected for the designated application.
call application voice retry-count	Defines the number of times a caller is permitted to reenter the PIN for a designated application and passes that information to the application.
call application voice security trusted	Sets the security level of a VoiceXML application to trusted so that ANI is not blocked.
call application voice set-location	Defines the location, language, and category of the audio files for the designated application and passes that information to the application.
call application voice uid-len	Defines the number of characters in the UID for the designated application and passes that information to the application.

Command	Description
call application voice warning-time	Defines, in seconds, how long in advance a user is warned before the allowed calling time expires for the designated application.
show call active fax	Displays call information for fax transmissions in progress.
show call application voice	Displays information about voice applications.
show call history fax	Displays the call history table for fax transmissions.

call application voice authen-list

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoiceauthen-list** command was replaced by the **paramauthen-list** command in application configuration mode. See the **paramauthen-list**command for more information.

To specify the name of an authentication method list for a Tool Command Language (Tcl) application, use the **callapplicationvoiceauthenlist**command in global configuration mode. To disable the authentication method list for a Tcl application, use the **no** form of this command.

call application voice *application-name* **authen-list** *method-list-name*
no call application voice *application-name* **authen-list** *method-list-name*

Syntax Description

application-name	Name of the application.
method-list-name	Character string used to name a list of authentication methods to be used with T.38 fax relay and T.37 store-and-forward fax.

Command Default

No default behavior or values.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)XI	This command was introduced on the Cisco AS5300.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.2(4)T	This command was implemented on the Cisco 1750.
12.2(8)T	This command was implemented on the Cisco 1751, Cisco 2600 series and Cisco 3600 series, Cisco 3725, and Cisco 3745.
12.3(14)T	This command was replaced by the **paramauthen-list** command in application configuration mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command defines the name of the authentication, authorization, and accounting (AAA) method list to be used with fax applications on voice feature cards. The method list itself, which defines the type of authentication services provided for store-and-forward fax, is defined using the **aaaauthentication** command. Unlike standard AAA (in which each defined method list can be applied to specific interfaces and lines), AAA method lists that are used with fax applications are applied globally.

After the authentication method lists have been defined, they are enabled by using the **callapplicationvoiceauthenticationenable**command.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice authen-list
        Warning: This command has been deprecated. Please use the following:
   param authen-list
```

The following example defines a AAA authentication method list (called "fax") to be used with T.38 fax relay and T.37 store-and-forward fax:

```
call application voice app_libretto_onramp9 authen-list fax
```

Related Commands

Command	Description
aaa authentication	Enable AAA accounting of requested services for billing or security purposes.
call application voice authen-method	Specifies the authentication method for a Tcl application.
call application voice authentication enable	Enables AAA authentication services for a Tcl application.

call application voice authen-method

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoiceauthen-method**command is replaced by the **paramauthen-method**command in application configuration mode. See the **paramauthen-method** command for more information.

To specify an authentication, authorization, and accounting (AAA) authentication method for a Tool Command Language (Tcl) application, use the **callapplicationvoiceauthen-method**command in global configuration mode. To disable the authentication method for a Tcl application, use the **no** form of this command.

call application voice *application-name* **authen-method** {**prompt-user** | **ani** | **dnis** | **gateway** | **redialer-id** | **redialer-dnis**}
no call application voice *application-name* **authen-method** {**prompt-user** | **ani** | **dnis** | **gateway** | **redialer-id** | **redialer-dnis**}

Syntax Description

application-name	Name of the application.
prompt user	User is prompted for the Tcl application account identifier.
ani	Calling party telephone number (automatic number identification or ANI) is used as the Tcl application account identifier.
dnis	Called party telephone number (dialed number identification service or DNIS) is used as the Tcl application account identifier.
gateway	Router-specific name derived from the host name and domain name is used as the Tcl application account identifier, displayed in the following format: *router-name.domain-name*.
redialer id	Account string returned by the external redialer device is used as the Tcl application account identifier. In this case, the redialer ID is either the redialer serial number or the redialer account number.
redialer dnis	Called party telephone number (dialed number identification service or DNIS) is used as the Tcl application account identifier captured by the redialer if a redialer device is present.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

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

Release	Modification
12.2(4)T	This command was implemented on the Cisco 1750.
12.2(8)T	This command was implemented on the Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745.
12.3(14)T	This command was replaced by the **paramauthen-method**command in application configuration mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

Normally, when AAA is used for simple user authentication, AAA uses the username information defined in the user profile for authentication. With T.37 store-and-forward fax and T.38 real-time fax, you can specify that the ANI, DNIS, gateway ID, redialer ID, or redialer DNIS be used to identify the user for authentication or that the user be prompted for the Tcl application.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice authen-method
        Warning: This command has been deprecated. Please use the following:
   param authen-method
```

The following example configures the router-specific name derived from the hostname and domain name as the Tcl application account identifier for the app_sample_onramp9 Tcl application:

```
call application voice app_sample_onramp9 authen-method gateway
```

Related Commands

Command	Description
call application voice authentication enable	Enables AAA authentication services for a Tcl application.
call application voice authen list	Specifies the name of an authentication method list for a Tcl application.

call application voice accounting enable

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoiceaccountingenable**command is replaced by the **paramaccountingenable**command in application configuration mode. See the **paramaccountingenable**command for more information.

To enable authentication, authorization, and accounting (AAA) accounting for a Tool Command Language (Tcl) application, use the **callapplicationvoiceaccountingenable**command in global configuration mode. To disable accounting for a Tcl application, use the **no** form of this command.

call application voice *application-name* **accounting enable**
no call application voice *application-name* **accounting enable**

Syntax Description

application-name	Name of the application.

Command Default

Disabled

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)XI	This command was introduced on the Cisco AS5300.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.2(4)T	This command was implemented on the Cisco 1750.
12.2(8)T	This command was implemented on the Cisco 1751, Cisco 2600 series and Cisco 3600 series, Cisco 3725, and Cisco 3745.
12.3(14)T	This command was replaced by the **paramaccountingenable**command in application configuration mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command enables AAA accounting services if a AAA accounting method list has been defined using both the **aaaaccounting** command and the**mmoipaaamethodfaxaccounting** command.

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

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice accounting enable
```

```
     Warning: This command has been deprecated. Please use the following:
   param accounting enable
```

The following example enables AAA accounting to be used with outbound store-and-forward fax:

```
call application voice app_libretto_onramp9 accounting enable
```

Related Commands

Command	Description
aaa accounting	Enables AAA accounting of requested services when you use RADIUS or TACACS+.
mmoip aaa method fax accounting	Defines the name of the method list to be used for AAA accounting with store-and-forward fax.

call application voice default disc-prog-ind-at-connect

Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoicedefaultdisc-prog-ind-at-connect**command is replaced. Use one of the following commands:

- **param convert-discpi-after-connect** (application parameter configuration mode)
- **paramspace session_xwork convert-discpi-after-connect** (service configuration mode)

To convert a DISCONNECT message with Progress Indicator set to PROG_INBAND (PI=8) to a regular DISCONNECT message when the call is in the active state, use the **callapplicationvoicedefaultdisc-prog-ind-at-connect** command in global configuration mode. To revert to a DISCONNECT message with Progress Indicator set to PROG_INBAND (PI=8) when the call is in the active state, use the**no** form of this command.

call application voice default disc-prog-ind-at-connect [{**1** | **0**}]
no call application voice default disc-prog-ind-at-connect [{**1** | **0**}]

Syntax Description

1	(Optional) Convert a DISCONNECT message with Progress Indicator set to PROG_INBAND (PI=8) to a regular DISCONNECT message when the call is in the active state.
0	(Optional) Revert to a DISCONNECT message with Progress Indicator set to PROG_INBAND (PI=8) when the call is in the active state.

Command Default

The DISCONNECT message has Progress Indicator set to PROG_INBAND (PI=8) when the call is in the active state.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(15)ZJ	This command was introduced.
12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(14)T	The**callapplicationvoicedefaultdisc-prog-ind-at-connect**command was replaced. Use one of the following commands: • **param convert-discpi-after-connect** (application parameter configuration mode) • **paramspace session_xwork convert-discpi-after-connect** (service configuration mode)
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command has no effect if the call is not in the active state.

This command is available for the default voice application. It may not be available when using some Tcl IVR applications.

The Cisco IOS command-line interface command completion and help features do not work with this command.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# voice default disc-prog-ind-at-connect

        Warning: This command has been deprecated. Please use the following:
   param convert-discpi-after-connect
    paramspace session_xwork convert-discpi-after-connec
```

In the following example, a DISCONNECT message with Progress Indicator set to PROG_INBAND (PI=8) is converted to a regular DISCONNECT message when the call is in the active state:

```
call application voice default disc-prog-ind-at-connect 1
```

Related Commands

Command	Description
param convert-discpi-after-connect	Enables or disables conversion of a DISCONNECT message with Progress Indicator set to PROG_INBAND (PI=8) to a regular DISCONNECT message when the call is in the active state.
paramspace session_xwork convert-discpi-after-connect	Enables or disables conversion of a DISCONNECT message with Progress Indicator set to PROG_INBAND (PI=8) to a regular DISCONNECT message when the call is in the active state.

call application voice dsn-script

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoicedsn-script**command is replaced by the **paramdsn-script**command in application parameter configuration mode.

To specify the VoiceXML application to which the off-ramp mail application hands off calls for off-ramp delivery status notification (DSN) and message disposition notification (MDN) e-mail messages, use the **callapplicationvoicedsnscript**command in global configuration mode. To remove the application, use the **no** form of this command.

call application voice *mail-application-name* **dsn-script** *application-name*
no call application voice *mail-application-name* **dsn-script** *application-name*

Syntax Description

mail-application-name	Name of the off-ramp mail application that launches the app_voicemail_offramp.tcl scr ipt when the gateway receives an e-mail trigger.
application-name	Name of the VoiceXML application to which the off-ramp mail application hands off the call when the destination answers.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(11)T	This command was introduced on the Cisco 3640, Cisco 3660, Cisco AS5300, Cisco AS5350, and Cisco AS5400.
12.3(14)T	This command was replaced by the **paramdsn-script**command in application parameter configuration mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

When the off-ramp gateway receives a DSN or MDN e-mail message, it handles it in the same way as a voice e-mail trigger message. The dial peer is selected on the basis of dialed number identification service (DNIS), and the mail application hands off the call to the VoiceXML application that is configured with this command.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice dsn-script
        Warning: This command has been deprecated. Please use the following:
   param dsn-script
```

The following example shows how to define the DSN application and how to apply it to a dial peer:

```
call application voice offramp-mapp tftp://sample/tftp-users/tcl/app_voicemail_offramp.tcl
call application voice dsn-mapp-test tftp://sample/tftp-users/vxml/dsn-mapp-test.vxml
call application voice offramp-mapp dsn-script dsn-mapp-test
!
dial-peer voice 1000 mmoip
 application offramp-mapp
 incoming called-number 555....
 information-type voice
```

Related Commands

Command	Description
application	Defines a specific voice application in the dial peer.
call application voice	Defines the name of a voice application and specifies the location of the document (Tcl or VoiceXML) to load for the application.
param dsn-script	Specifies the VoiceXML application to which the off-ramp mail application hands off calls for off-ramp DSN and MDN e-mail messages.
show call application voice	Displays information about the configured voice applications.

call application voice event-log

Effective with Cisco IOS Release 12.3(14)T, the**callapplicationvoiceevent-log** is obsolete. To enable event logging for a specific voice application, use one of the following commands:

- **param event-log** (application parameter configuration mode)
- **paramspace appcommon event-log** (service configuration mod

To enable event logging for a specific voice application, use the **callapplicationvoiceevent-log** command in global configuration mode. To reset to the default, use the **no** form of this command.

call application voice *application-name* **event-log** [**disable**]
no call application voice *application-name* **event-log**

Syntax Description

application-name	Name of the voice application.
disable	(Optional) Disables event logging for the named application.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.3(14)T	The**callapplicationvoiceevent-log** is obsolete. To enable event logging for a specific voice application, use one of the following commands: • **param event-log** (application parameter configuration mode) • **paramspace appcommon event-log** (service configuration mode)
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command is application-specific; it takes precedence over the global configuration command, **callapplicationevent-log**, which enables event logging for all voice applications.

Before you can use this command, you must configure the named application on the gateway by using the **callapplicationvoice** command.

Note To prevent event logging from adversely impacting system resources for production traffic, the gateway uses a throttling mechanism. When free processor memory drops below 20 percent, the gateway automatically disables all event logging. It resumes event logging when free memory rises above 30 percent. While throttling is occurring, the gateway does not capture any new event logs even if event logging is enabled. You should monitor free memory and enable event logging only when necessary for isolating faults.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call call application voice event-log
  Warning: This command has been deprecated. Please use the following:
   param event-log
    paramspace appcommon event-log
```

The following example enables event logging for all instances of the application named sample_app:

```
call application voice sample_app event-log
```

The following example enables event logging for all applications except the application sample_app:

```
call application event-log
call application voice sample_app event-log disable
```

Related Commands

Command	Description
call application event-log	Enables event logging for voice application instances.
call application event-log max-buffer-size	Sets the maximum size of the event log buffer for each application instance.
call application voice	Defines the name of a voice application and specifies the location of the script to load for the application.
monitor call application event-log	Displays the event log for an active application instance in real-time.
param event-log	Enables or disables logging for linkable Tcl functions (packages).
paramspace appcommon event-log	Enable or disables logging for a service (application).
show call application session-level	Displays event logs and statistics for voice application instances.

call application voice fax-dtmf

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoicefax-dtmf**command is replaced by the **paramfax-dtmf**command in application parameter configuration mode. See the**paramfax-dtmf**command for more information.

To direct the fax detection interactive voice response (IVR) application to recognize a specified digit to indicate a fax call in default-voice and default-fax modes, use the **callapplicationvoicefax-dtmf** command in global configuration mode. To remove configuration of this digit, use the **no** form of this command.

call application voice*application-name***fax-dtmf**{**0** | **1** | **2** | **3** | **4** | **5** | **6** | **7** | **8** | **9** | * | #}
no call application voice*application-name***fax-dtmf**{**0** | **1** | **2** | **3** | **4** | **5** | **6** | **7** | **8** | **9** | * | #}

Syntax Description

application-name	The name of the fax detection IVR application that you defined when you loaded the application on the router.
0 | **1**| **2**| **3**| **4**| **5**| **6**| **7**| **8**| **9**| *****| #	The telephone keypad digit processedby the calling party to indicate a fax call, in response to the audio prompt that plays during the default-voice or default-fax mode of the fax detection IVR application.

Command Default

2

Command Modes

Global configuration (config)

Command History

Release	**Modification**
12.1(5)XM	This command was introduced for the Cisco AS5300.
12.2(2)T	This command was integrated into Cisco IOS Release 12.2(2)T.
12.2(2)XB	This command was implemented on the Cisco AS5400 and Cisco AS5350.
12.2(8)T	This command was implemented on the Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745.
12.2(11)T	This command was implemented on the Cisco AS5300, Cisco AS5350, and Cisco AS5400.
12.3(14)T	This command was replaced by the **paramfax-dtmf**command in application parameter configuration mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command is useful only when the fax detection IVR application is being configured in default-voice mode or default-fax mode as defined by the **callapplicationvoicemode** command.

Only one digit can be specified in this command, and that digit must be different from the digit specified in the **callapplicationvoicevoice-dtmf**command. You are not notified immediately if you make the error of configuring them both to the same digit. To find this error, you must start the debugging with the **debugvoipivrscript** command and then observe some failing calls.

This command is not supported by Cisco IOS help; that is, if you type**callapplicationvoicefax_detectfax-dtmfandaquestionmark(?)**,Cisco IOS help does not supply a list of entries that are valid in place of the question mark.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice fax-dtmf
        Warning: This command has been deprecated. Please use the following:
   param fax-dtmf
```

The following example selects DTMF digit 1 to indicate a fax call:

```
call application voice fax_detect script_url
call application voice fax_detect fax-dtmf 1
dial-peer voice 302 pots
 application fax_detect
```

Related Commands

Command	Description
call application voice	Loads an IVR application onto a router and gives it an application name.
call application voice account-id-method	Configures the fax detection IVR application to use a particular method to assign the account identifier.
call application voice mode	Configures the fax detection IVR application to operate in one of its four modes.
call application voice prompt	Configures the fax detection IVR application to use the specified audio file as a user prompt.
call application voice voice-dtmf	Configures the fax detection IVR application to recognize the specified digit to indicate a voice call.
debug voip ivr script	Displays debug information from the fax detection IVR script.
param fax-dtmf	Directs the fax detection IVR application to recognize a specified digit to indicate a fax call in default-voice and default-fax modes.

call application voice global-password

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoiceglobal-password** command is replaced by the **paramglobal-password** command in application parameter configuration mode. See the **paramglobal-password** command for more information.

To define a password to be used with CiscoSecure for Windows NT when using store-and-forward fax on a voice feature card, use the **callapplicationvoiceglobalpassword** command in global configuration mode. To restore the default value, use the **no** form of this command.

call application voice *application-name* **global-password** *password*
no call application voice *application-name* **global-password** *password*

Syntax Description

application-name	The name of the application.
password	Character string used to define the CiscoSecure for Windows NT password to be used with store-and-forward fax. The maximum length is 64 alphanumeric characters.

Command Default

No password is defined

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)XI	This command was introduced on the Cisco AS5300.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.3(14)T	This command is replaced by the **paramglobal-password** command in application parameter configuration mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

CiscoSecure for Windows NT might require a separate password to complete authentication, no matter what security protocol you use. This command defines the password to be used with CiscoSecure for Windows NT. All records on the Windows NT server use this defined password.

This command applies to on-ramp store-and-forward fax functions on Cisco AS5300 universal access server voice feature cards. It is not used on modem cards.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice global-password
```

```
     Warning: This command has been deprecated. Please use the following:
param global-password
```

The following example shows a password (abercrombie) being used by AAA for the app_sample_onramp9 Tcl application:

```
call application voice app_sample_onramp9 global-password abercrombie
```

Related Commands

Command	Description
param global-password	Defines a password to be used with CiscoSecure for Windows NT when using store-and-forward fax on a voice feature card.

call application voice language

Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoicelanguage** is replaced by the following commands:

- **param language** (application parameter configuration mode)
- **paramspace language** (service configuration mode)

See these commands for more information.

To specify the language for dynamic prompts used by an interactive voice response (IVR) application (Tool Command Language (Tcl) or VoiceXML), use the **callapplicationvoicelanguage** command in global configuration mode. To remove this language specification from the application, use the **no** form of this command.

call application voice *application-name* **language** *digit language*
no call application voice *application-name* **language** *digit language*

Syntax Description

application-name	Name of the application to which the language parameters are being passed.
digit	Number that identifies the language used by the audio files. Any number can represent any language. Enter 1 to indicate the primary language and 2 to indicate the secondary language. Range is from 0 to 9.
language	Two-character code that identifies the language of the associated audio files. Valid entries are as follows: • **en** --English • **sp** --Spanish • **ch** --Mandarin • **aa** --all

Command Default

If this command is not configured, the default language is English.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.0(7)T	This command was introduced.
12.1(5)T	This command was implemented on the Cisco AS5800.
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB	This command was modified to support VoiceXML applications on the Cisco AS5300, Cisco AS5350, and Cisco AS5400.

Release	Modification
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)T	This command was implemented on the Cisco 1750.
12.2(4)XM	This command was implemented on the Cisco 1751.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800 and Cisco AS5850 is not included in this release.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T for VoiceXML applications. This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco 5800, and Cisco AS5850 in this release.
12.3(14)T	The **callapplicationvoicelanguage** was replaced by the following commands: **param language** (application parameter configuration mode) **paramspace language** (service configuration mode)
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command identifies the number that users enter for a language; for example, "Enter 1 for English. Enter 2 for French."

This number is used only with the Tcl IVR Debit Card feature. Although it is not used by VoiceXML, you still must enter a number from 0 to 9.

Instead of using this command, you can configure the language and location of the prerecorded audio files within a Tcl script or VoiceXML document. For more information, see the Tcl IVR API Version 2.0 Programmer’s Guide or Cisco VoiceXML Programmer’s Guide, respectively.

To identify the location of the language audio files that are used for the dynamic prompts, use the **callapplicationvoiceset-location** command.

Tcl scripts and VoiceXML documents can be stored in any of the following locations: On the TFTP, FTP, or HTTP servers, in the flash memory of the gateway, or on the removable disks of the Cisco 3600 series. The audio files that they use can be stored in any of these locations, and on RTSP servers.

With the Pre-Paid Debitcard Multi-Language feature, you can create Tcl scripts and a two-character code for any language. See the Cisco Pre-Paid Debitcard Multi-Language Programmer's Reference.

With the multilanguage support for Cisco IOS IVR, you can create a Tcl language module for any language and any set of TTS notations for use with Tcl and VoiceXML applications. See the Enhanced Multi-Language Support for Cisco IOS Interactive Voice Response document.

The table below lists Tcl script names and the corresponding commands that are required for each Tcl script.

Table 3: Tcl Scripts and Commands

Tcl Script Name	Description	Commands to Configure
app_libretto_onramp9.tcl	Authenticates the account and personal identification number (PIN) using the following: prompt-user, using automatic number identification (ANI), dialed number identification service (DNIS), gateway ID, redialer ID, and redialer DNIS.	None
app_libretto_offramp5.tcl	Authenticates the account and PIN using the following: envelope-from, envelope-to, gateway ID, and x-account ID.	None
clid_4digits_npw_3_cli.tcl	This script authenticates the account number and PIN, respectively, using ANI and NULL. The number of digits allowed for the account number and password, respectively, are configurable through the command-line interface (CLI). If the authentication fails, the script allows the caller to retry. The retry number is also configured through the CLI.	**call application voice uid-length** Range is 1 to 20. The default is 10. **call application voice pin-length** Range is 0 to 10. The default is 4. **call application voice retry-count** Range is 1 to 5. The default is 3.
clid_authen_col_npw_cli.tcl	This script authenticates the account number and PIN, respectively, using ANI and NULL. If the authentication fails, it allows the caller to retry. The retry number is configured through CLI. The account number and PIN are collected separately.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_authen_collect_cli.tcl	This script authenticates the account number and PIN using ANI and DNIS. If the authentication fails, the script allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_col_npw_3_cli.tcl	This script authenticates using ANI and NULL for account numbers and PINs, respectively. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_col_npw_npw_cli.tcl	This script authenticates using ANI and NULL for account and PIN, respectively. If authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected together.	**call application voice retry-count** Range is 1 to 5. The default is 3.
fax_rollover_on_busy.tcl	Used for on-ramp T.38 fax rollover to T.37 fax when the destination fax line is busy.	**voice hunt user-busy**

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)#
call application voice language
        Warning: This command has been deprecated. Please use the following:
   param language
    paramspace language
```

The following example shows how to define the application "prepaid" and then selects English and Spanish as the languages of the audio files that are associated with the application:

```
call application voice prepaid tftp://keyer/debitcard.tcl
call application voice prepaid language 1 en
call application voice prepaid language 2 sp
```

Related Commands

Command	Description
call application voice	Specifies the name to be used for an application and indicates the location of the appropriate IVR script to be used with this application.
call application voice load	Reloads the designated Tcl script.
call application voice pin-len	Defines the number of characters in the PIN for the application and passes that information to the application.
call application voice redirect-number	Specifies the telephone number to which a call is redirected for the designated application.
call application voice retry-count	Defines the number of times a caller is permitted to reenter the PIN for a designated application and passes that information to the application.
call application voice set-location	Defines the location, language, and category of the audio files for the designated application and passes that information to the application.
call application voice uid-len	Defines the number of characters in the UID for the designated application and passes that information to the application.
call application voice warning-time	Defines, in seconds, how long in advance a user is warned before the allowed calling time expires for the designated application.
param language	Configures the language parameter in a service or package on the gateway.
paramspace language	Defines the category and location of audio files that are used for dynamic prompts by an IVR application (Tcl or VoiceXML).
show call application voice	Displays information about voice applications.

call application voice load

To reload the selected voice application script after it has been modified, use the**callapplicationvoiceload** command in privileged EXEC mode. This command does not have a **no** form.

call application voice load *application-name*

Syntax Description

application-name	Name of the Tcl or VoiceXML application to reload.

Command Default

No default behavior or values

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.0(7)T	This command was introduced on the Cisco 2600 series and Cisco 3600 series (except for the Cisco 3660), and on the Cisco AS5300.
12.1(3)T	Support for dynamic script loading of Media Gateway Control Protocol (MGCP) was added.
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB	This command was modified to support VoiceXML applications.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on the Cisco 1751.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T and implemented on the Cisco 1750.
12.2(8)T	This command and implemented on the Cisco 7200 series.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T for VoiceXML applications. This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and the Cisco AS5850 in this release.

Usage Guidelines

Use this command to reload an application Tcl script or VoiceXML document onto the gateway after it has been modified.

The location of the Tcl script or VoiceXML document for the specified application must have already been configured using the **callapplicationvoice** command.

Do not include the file type extension in the filename (.vxml or .tcl) when specifying the document used by the application.

Tcl scripts and VoiceXML documents can be stored in any of the following locations: on TFTP, FTP, or HTTP servers, in the flash memory of the gateway, or on the removable disks of the Cisco 3600 series. The audio files that they use can be stored on any of these locations, and on RTSP servers.

Before Cisco IOS Release 12.1(3)T, the software checked the signature in a Tcl script to ensure that it was supported by Cisco. A signature on Tcl scripts is no longer required. A signature has never been required for VoiceXML documents.

A Tcl script or VoiceXML document cannot be reloaded if it has active calls. Use the **showcallapplicationvoice** command to verify that no active calls are using this application.

Tip If the **callapplicationvoiceload** command fails to load the Tcl script or VoiceXML document that is associated with the application, enable the **debugvoipivr** command and retry. This debugging command can provide information on why loading fails.

Note MGCP scripting is not supported on the Cisco 1750 router or on Cisco 7200 series routers.

Examples

The following example shows the loading of a Tcl script called "clid_4digits_npw_3.tcl":

```
call application voice load clid_4digits_npw_3.tcl
```

The following example shows how to reload the VoiceXML application called " vapptest":

```
call application voice load vapptest
```

Related Commands

Command	Description
call application cache reload time	Configures the interval for reloading MGCP scripts.
call application voice	Creates and calls the application that interacts with the IVR feature.
debug http client	Displays information about the load an application that was loaded with HTTP.
show call application voice	Displays a list of the voice applications that are configured.

call application voice mail-script

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoicemail-script** command is replaced by the **parammail-script** command in application parameter configuration mode. See the **parammail-script** command for more information.

To specify the VoiceXML application to which the off-ramp mail application hands off a call when the destination telephone answers, use the **callapplicationvoicemail-script**command in global configuration mode. To remove the application, use the **no** form of this command.

call application voice *mail-application-name* **mail-script** *application-name*
no call application voice *mail-application-name* **mail-script** *application-name*

Syntax Description

mail-application-name	Name of the off-ramp mail application that launches the app_voicemail_offramp.tcl script when the gateway receives an e-mail trigger.
application-name	Name of the VoiceXML application to which the off-ramp mail application hands off the call when the destination answers.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(11)T	This command was introduced on the Cisco 3640, Cisco 3660, Cisco AS5300, Cisco AS5350, and Cisco AS5400.
12.3(14)T	This command was replaced by the **parammail-script** command in application parameter configuration mode.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

To load the mail application onto the gateway, use the**callapplicationvoice** command.

The off-ramp mail application must be configured in the Multimedia Mail over Internet Protocol (MMoIP) dial peer that matches the telephone number contained in the header of the incoming e-mail message.

The off-ramp mail application must use the Tool Command Language (Tcl) script named "app_voicemail_offramp.tcl" that is provided by Cisco. This Tcl script can be downloaded from the Cisco website by following this path: Cisco > Technical Support Help - TAC > Select & Download Software > Software Center > Access Software > TclWare.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice mail-script
        Warning: This command has been deprecated. Please use the following:
   param mail-script
```

The following example shows that the off-ramp mail application named "offramp-mapp" hands calls to the application named "mapp-test" if the telephone number in the e-mail header is seven digits beginning with 555 :

```
call application voice offramp-mapp tftp://sample/tftp-users/tcl/app_voicemail_offramp.tcl
call application voice mapp-test tftp://sample/tftp-users/vxml/user-test.vxml
call application voice offramp-mapp mail-script mapp-test
!
dial-peer voice 1001 mmoip
 application offramp-mapp
 incoming called-number 555....
 information-type voice
```

Related Commands

Command	Description
application	Defines a specific voice application in the dial peer.
call application voice	Defines the name of a voice application and specifies the location of the document (Tcl or VoiceXML) to load for the application.
param mail-script	Specifies the VoiceXML application to which the off-ramp mail application hands off a call when the destination telephone answers.
show call application voice	Displays information about the configured voice applications.

call application voice mode

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoicemode** command is replaced by the **parammode**command in application parameter configuration mode. See the **parammode** command for more information.

To direct the fax detection interactive voice response (IVR) application to operate in one of its four connection modes, use the **callapplicationvoicemode** command in global configuration mode. To return to the default connection mode, use the **no** form of this command.

call application voice *application-name* **mode** {**connect-first** | **listen-first** | **default-voice** | **default-fax**}
no call application voice *application-name* **mode** {**connect-first** | **listen-first** | **default-voice** | **default-fax**}

Syntax Description

application-name	Fax detection IVR application that was defined when the application was loaded on the router.
connect first	Incoming calls are connected to the Real-Time Streaming Protocol (RTSP) server. This is the default.
listen-first	The gateway listens to the call first and then connects to the RTSP server. Any Dual tone multifrequency (DTMF) tones take the call to the voice server, but subsequent DTMF is forwarded as configured.
default voice	Incoming calls are connected as voice calls to the RTSP server.
default fax	Incoming calls are connected to the fax relay or store-and-forward fax application that is configured on the gateway.

Command Default

connect first

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(5)XM	This command was introduced on the Cisco AS5300.
12.2(2)XB	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745.
12.2(11)T	This command is supported on the Cisco AS5300, Cisco AS5350, and Cisco AS5400 in this release.
12.3(14)T	This command was replaced by the **parammode**command in application parameter configuration mode.

Release	Modification
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

The call application voice mode commands control the way that the gateway handles fax detection IVR applications calls.

When the **connect-first** keyword is selected and CNG (calling) tones from the originating fax machine are detected, the voice application is disconnected and the call is passed to the configured fax application. If the **listen-first**keyword is selected, the gateway listens for CNG and, if it is detected, passes the call to the fax relay or store-and-forward fax application, whichever is configured on the gateway. When the**default-voice**and **default-fax** keywords are selected, the gateway defaults to voice after listening for CNG or passes the call to the fax relay or store-and-forward fax application, whichever was configured on the gateway. If the gateway hears the Dual tone multifrequency (DTMF) tones that are specified in the **callapplicationvoicevoice-dtmf**or **callapplicationvoicefax-dtmf**commands , the call is forwarded as appropriate.

Note that in all four connection modes, the router continues to listen for CNG throughout the call, even if the call has been connected to the voice server; if CNG is detected, the call is connected to fax relay or store-and-forward fax, whichever has been configured.

This command is not supported by Cisco IOS help. If you type the **callapplicationvoicefax_detectmode**command and a question mark (?), Cisco IOS help does not supply a list of valid entries in place of the question mark.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice mode
        Warning: This command has been deprecated. Please use the following:
   param mode
```

The following example shows a selection of default-voice modefor the fax detection application:

```
call application voice fax_detect script_url
call application voice fax_detect mode default-voice
dial-peer voice 302 pots
 application fax_detect
```

Related Commands

Command	Description
call application voice	Loads a specified IVR application onto the router from the TFTP server and gives it an application name by which it is known on the router.
call application voice account-id-method	Configures the fax detection IVR application to use a particular method to assign the account identifier.
call application voice fax-dtmf	Configures the fax detection IVR application to recognize a specified digit to indicate a fax call .

Command	Description
call application voice prompt	Configures the fax detection IVR application to use the specified audio file as a user prompt in listen-first mode, default-voice mode, or default-fax mode.
call application voice voice-dtmf	Configures the fax detection IVR application to recognize a specified digit to indicate a voice call.
param mode	Configures the call transfer mode for a package.

call application voice pin-len

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoicepin-len** command is replaced with the **parampin-len** command in application parameter configuration mode. See the **parampin-len** command for more information.

To define the number of characters in the personal identification number (PIN) for the designated application, use the **callapplicationvoicepinlen**command in global configuration mode. To disable the PIN for the designated application, use the no form of this command.

call application voice *application-name* **pin-len** *number*
no call application voice *application-name* **pin-len** *number*

Syntax Description

application-name	Application name to which the PIN length parameter is being passed.
number	Number of allowable characters in PINs associated with the specified application. Range is from 0 to 10. The default is 4.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

Release	Modification
12.0(7)T	This command was introduced.
12.1(5)T	This command was implemented on the Cisco AS5800.
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T and implemented on the Cisco 1750.
12.2(4)XM	This command was implemented on the Cisco 1751.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series.
12.2(11)T	This command is supported on the Cisco AS5350, Cisco AS5400 Cisco AS5800, and the Cisco AS5850 in this release.
12.3(14)T	The **callapplicationvoicepin-len**command was replaced with the **parampin-len** command in application parameter configuration mode.

Release	Modification
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

Use this command when configuring interactive voice response (IVR)--depending on the Tool Command Language (Tcl) script being used--or one of the IVR-related features (such as Debit Card) to define the number of allowable characters in a PIN for the specified application and to pass that information to the specified application.

The table below lists Tcl script names and the corresponding commands that are required for each Tcl script.

Table 4: Tcl Scripts and Commands

Tcl Script Name	Description	Commands to Configure
app_libretto_onramp9.tcl	Authenticates the account and personal identification number (PIN) using the following: prompt-user, using automatic number identification (ANI), dialed number identification service (DNIS), gateway ID, redialer ID, and redialer DNIS.	None
app_libretto_offramp5.tcl	Authenticates the account and PIN using the following: envelope-from, envelope-to, gateway ID, and x-account ID.	None
clid_4digits_npw_3_cli.tcl	This script authenticates the account number and PIN, respectively, using ANI and NULL. The number of digits allowed for the account number and password, respectively, are configurable through the command-line interface (CLI). If the authentication fails, the script allows the caller to retry. The retry number is also configured through the CLI.	**call application voice uid-length** Range is 1 to 20. The default is 10. **call application voice pin-length** Range is 0 to 10. The default is 4 **call application voice retry-count** Range is 1 to 5. The default is 3.
clid_authen_col_npw_cli.tcl	This script authenticates the account number and PIN, respectively, using ANI and NULL. If the authentication fails, it allows the caller to retry. The retry number is configured through CLI. The account number and PIN are collected separately.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_authen_collect_cli.tcl	This script authenticates the account number and PIN using ANI and DNIS. If the authentication fails, the script allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_col_npw_3_cli.tcl	This script authenticates using ANI and NULL for account numbers and PINs, respectively. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_col_npw_npw_cli.tcl	This script authenticates using ANI and NULL for account and PIN, respectively. If authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected together.	**call application voice retry-count** Range is 1 to 5. The default is 3.

Tcl Script Name	Description	Commands to Configure
fax_rollover_on_busy.tcl	Used for on-ramp T.38 fax rollover to T.37 fax when the destination fax line is busy.	**voice hunt user-busy**

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice pin-len
        Warning: This command has been deprecated. Please use the following:
   param pin-len
```

The following example shows how to define a PIN length of 4 characters for the application named "prepaid":

```
call application voice prepaid pin-len 4
```

Related Commands

Command	Description
call application voice	Specifies the name to be used for an application and indicates the location of the appropriate IVR script to be used with the application.
call application voice language	Specifies the language of the audio file for the designated application and passes that information to the application.
call application voice load	Reloads the designated Tcl script.
call application voice redirect-number	Specifies the telephone number to which a call is redirected for the designated application.
call application voice retry-count	Defines the number of times a caller is permitted to reenter the PIN for a designated application and passes that information to the application.
call application voice set-location	Defines the location, language, and category of the audio files for the designated application and passes that information to the application.
call application voice uid-len	Defines the number of characters in the UID for the designated application and passes that information to the application.
call application voice warning-time	Defines, in seconds, how long in advance a user is warned before the allowed calling time expires for the designated application.
param pin-len	Defines the number of characters in the PIN for an application.

call application voice prompt

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoiceprompt** command is replaced by the **parampromptcommand**. See the **paramprompt**command for more information.

To direct the fax detection interactive voice response (IVR) application to use the specified audio file as a user prompt, use the **callapplicationvoiceprompt** command in global configuration mode. To disable use of this audio file, use the **no** form of this command.

call application voice *application-name* **prompt** *prompt-url*
no call application voice *application-name* **prompt** *prompt-url*

Syntax Description

application-name	Name of the fax detection IVR application that you defined when you loaded the application on the router.
prompt-url	URL or Cisco IOS file system location on the TFTP server for the audio file containing the prompt for the application.

Command Default

The prompt space is empty and no prompt is played.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(5)XM	This command was introduced for the Cisco AS5300.
12.2(2)XB	This command was implemented on the Cisco AS5400 and Cisco AS5350.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745.
12.2(11)T	This command was implemented on the Cisco AS5300, Cisco AS5350, and Cisco AS5400.
12.3(14)T	This command was replaced by the **paramprompt**command.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command is useful only in the listen-first, default-voice, and default-fax modes of the fax detection application.

Audio files should be a minimum of 9 seconds long so that callers do not hear silence during the initial CNG detection period. Any .au file can be used; formats are described in the Cisco IOS Voice, Video, and Fax Configuration Guide, Release 12.4.

This command is not supported by Cisco IOS help. If you type the**callapplicationvoicefax_detectprompt**command with a question (?), the Cisco IOS help does not supply a list of entries that are valid in place of the question mark.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice prompt
        Warning: This command has been deprecated. Please use the following:
   param prompt
```

The following example associates the audio file" promptfile.au" with the application file "fax_detect", and the application with the inbound POTS dial peer:

```
call application voice fax_detect script_url
call application voice fax_detect mode default-voice
call application voice fax_detect prompt promptfile.au
dial-peer voice 302 pots
 application fax_detect
```

Related Commands

Command	Description
call application voice	Loads a specified IVR application onto the router from the TFTP server and gives it an application name by which it is known on the router.
call application voice account-id-method	Configures the fax detection IVR application to use a particular method to assign the account identifier.
call application voice fax-dtmf	Configures the fax detection IVR application to recognize a specified digit to indicate a fax call.
call application voice mode	Configures the fax detection IVR application to operate in one of its four modes.
call application voice voice-dtmf	Configures the fax detection IVR application to recognize a specified digit to indicate a voice call.
param prompt	Directs the fax detection IVR application to use the specified audio file as a user prompt.

call application voice redirect-number

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoiceredirect-number** command is replaced with the **paramredirect-number** command in application parameter configuration mode. See the the **paramredirect-number** command for more information.

To define the telephone number to which a call is redirected--for example, the operator telephone number of the service provider--for the designated application, use the **callapplicationvoiceredirectnumber** command in global configuration mode. To cancel the redirect telephone number, use the **no** form of this command.

call application voice *application-name* **redirect-number** *number*
no call application voice *application-name* **redirect-number** *number*

Syntax Description

application name	Name of the application to which the redirect telephone number parameter is being passed.
number	Designated operator telephone number of the service provider (or any other number designated by the customer). This is the number where calls are terminated when, for example, allowed debit time has run out or the debit amount is exceeded.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

Release	Modification
12.0(7)T	This command was introduced on the Cisco 2600 series, the Cisco 3600 series, and the Cisco AS5300.
12.1(5)T	This command was implemented on the Cisco AS5800.
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on the Cisco 1751.
12.2(4)T	This command was implemented on the Cisco 1750.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T. This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 in this release.

Release	Modification
12.3(14)T	This command was replaced by the **paramredirect-number**.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

Use this command when configuring interactive voice response (IVR)--depending on the Tool Command Language (Tcl) script being used--or one of the IVR-related features (such as Debit Card) to define the telephone number to which a call is redirected.

The table below lists Tcl script names and the corresponding commands that are required for each Tcl script.

Table 5: Tcl Scripts and Commands

Tcl Script Name	Description	Commands to Configure
app_libretto_onramp9.tcl	Authenticates the account and personal identification number (PIN) using the following: prompt-user, using automatic number identification (ANI), dialed number identification service (DNIS), gateway ID, redialer ID, and redialer DNIS.	None
app_libretto_offramp5.tcl	Authenticates the account and PIN using the following: envelope-from, envelope-to, gateway ID, and x-account ID.	None
clid_4digits_npw_3_cli.tcl	This script authenticates the account number and PIN, respectively, using ANI and NULL. The number of digits allowed for the account number and password, respectively, are configurable through the command-line interface (CLI). If the authentication fails, the script allows the caller to retry. The retry number is also configured through the CLI.	**call application voice uid-length** Range is 1 to 20. The default is 10. **call application voice pin-length** Range is 0 to 10. The default is 4. **call application voice retry-count** Range is 1 to 5. The default is 3.
clid_authen_col_npw_cli.tcl	This script authenticates the account number and PIN, respectively, using ANI and NULL. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_authen_collect_cli.tcl	This script authenticates the account number and PIN using ANI and DNIS. If the authentication fails, the script allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_col_npw_3_cli.tcl	This script authenticates using ANI and NULL for account numbers and PINs, respectively. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_col_npw_npw_cli.tcl	This script authenticates using ANI and NULL for account and PIN, respectively. If authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected together.	**call application voice retry-count** Range is 1 to 5. The default is 3.

Tcl Script Name	Description	Commands to Configure
fax_rollover_on_busy.tcl	Used for on-ramp T.38 fax rollover to T.37 fax when the destination fax line is busy.	**voice hunt user-busy**

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice redirect-number
        Warning: This command has been deprecated. Please use the following:
   param redirect-number
```

The following example shows how to define a redirect number for the application named "prepaid":

```
call application voice prepaid redirect-number 5550111
```

Related Commands

Command	Description
call application voice	Specifies the name to be used for an application and indicates the location of the appropriate IVR script to be used with this application.
call application voice language	Specifies the language of the audio file for the designated application and passes that information to the application.
call application voice load	Reloads the designated Tcl script.
call application voice pin-len	Defines the number of characters in the PIN for the application and passes that information to the application.
call application voice retry-count	Defines the number of times a caller is permitted to reenter the PIN for a designated application and passes that information to the application.
call application voice set-location	Defines the location, language, and category of the audio files for the designated application and passes that information to the application.
call application voice uid-len	Defines the number of characters in the UID for the designated application and passes that information to the application.
call application voice warning-time	Defines, in seconds, how long in advance a user is warned before the allowed calling time expires for the designated application.
param redirect-number	Defines the telephone number to which a call is redirected--for example, the operator telephone number of the service provider--for an application.

call application voice retry-count

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoiceretry-count** command is replaced by the **paramretry-count** command in application parameter configuration mode. See the the **paramretry-count** command for more information.

To define the number of times that a caller is permitted to reenter the personal identification number (PIN) for the designated application, use the **callapplicationvoiceretrycount** command in global configuration mode. To cancel the retry count, use the **no** form of this command.

call application voice *application-name* **retry-count** *number*
no call application voice *application-name* **retry-count** *number*

Syntax Description

application name	Name of the application to which the number of possible retries is being passed.
number	Number of times the caller is permitted to reenter PIN digits. Range is 1 to 5. The default is 3.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

Release	Modification
12.0(7)T	This command was introduced.
12.1(5)T	This command was implemented on the Cisco AS5800.
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on the Cisco 1751.
12.2(4)T	This command was introduced on the Cisco 1750.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series. Support for the Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T. This command is supported on the Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 in this release.
12.3(14)T	This command was replaced by the **paramretry-count** command.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

Use this command when configuring interactive voice response (IVR)--depending on the Tool Command Language (Tcl) script being used--or one of the IVR-related features (such as Debit Card) to define how many times a user can reenter a PIN.

The table below lists Tcl script names and the corresponding commands that are required for each Tcl script.

Table 6: Tcl Scripts and Commands

Tcl Script Name	Description	Commands to Configure
app_libretto_onramp9.tcl	Authenticates the account and personal identification number (PIN) using the following: prompt-user, using automatic number identification (ANI), dialed number identification service (DNIS), gateway ID, redialer ID, and redialer DNIS.	None
app_libretto_offramp5.tcl	Authenticates the account and PIN using the following: envelope-from, envelope-to, gateway ID, and x-account ID.	None
clid_4digits_npw_3_cli.tcl	This script authenticates the account number and PIN, respectively, using ANI and NULL. The number of digits allowed for the account number and password, respectively, are configurable through the command-line interface (CLI). If the authentication fails, the script allows the caller to retry. The retry number is also configured through the CLI.	**call application voice uid-length** Range is 1 to 20. The default is 10. **call application voice pin-length** Range is 0 to 10. The default is 4. **call application voice retry-count** Range is 1 to 5. The default is 3.
clid_authen_col_npw_cli.tcl	This script authenticates the account number and PIN, respectively, using ANI and NULL. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_authen_collect_cli.tcl	This script authenticates the account number and PIN using ANI and DNIS. If the authentication fails, the script allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_col_npw_3_cli.tcl	This script authenticates using ANI and NULL for account numbers and PINs, respectively. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_col_npw_npw_cli.tcl	This script authenticates using ANI and NULL for account and PIN, respectively. If authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected together.	**call application voice retry-count** Range is 1 to 5. The default is 3.
fax_rollover_on_busy.tcl	Used for on-ramp T.38 fax rollover to T.37 fax when the destination fax line is busy.	**voice hunt user-busy**

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application retry-count
        Warning: This command has been deprecated. Please use the following:
   param retry-count
```

The following example shows how to define that for the application named "prepaid" that a user can reenter a PIN three times before being disconnected:

```
call application voice prepaid retry-count 3
```

Related Commands

Command	Description
call application voice	Specifies the name to be used for an application and indicates the location of the appropriate IVR script to be used with this application.
call application voice language	Specifies the language of the audio file for the designated application and passes that information to the application.
call application voice load	Reloads the designated Tcl script.
call application voice pin-len	Defines the number of characters in the PIN for the application and passes that information to the application.
call application voice redirect-number	Specifies the telephone number to which a call is redirected for the designated application.
call application voice set-location	Defines the location, language, and category of the audio files for the designated application and passes that information to the application.
call application voice uid-len	Defines the number of characters in the UID for the designated application and passes that information to the application.
call application voice warning-time	Defines, in seconds, how long in advance a user is warned before the allowed calling time expires for the designated application.
param retry-count	Defines the number of times that a caller is permitted to reenter the PIN for a package.

call application voice security trusted

Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoicesecuritytrusted**command is replaced by the the following commands:

- **param security trusted** (application parameter configuration mode)
- **paramspace appcommon security trusted** (service configuration mode)

See these commands for more information.

To set the security level of a VoiceXML application to "trusted" so that automatic number identification (ANI) is not blocked, use the **callapplicationvoicesecuritytrusted**command in global configuration mode. To restore the default condition, use the **no** form of this command.

call application voice *application-name* **security trusted**
no call application voice *application-name* **security trusted**

Syntax Description

application name	Name of the application being configured as trusted.

Command Default

The security level of the application is not set to trusted, and ANI is blocked.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(2)XB	This command was introduced on the Cisco AS5300, Cisco AS5350, and Cisco AS5400.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 3640 and Cisco 3660.
12.3(14)T	The **callapplicationvoicesecuritytrusted** command was replaced by the following commands: • **param security trusted** (application parameter configuration mode) • **paramspace appcommon security trusted** (service configuration mode)
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command is applicable only for VoiceXML applications.

Note Tool Command Language (Tcl) applications provide the security parameter to the application but do not use it.

If an application is configured as a trusted application, it is trusted not to provide the calling number to the destination party, so ANI is always provided if available.

Normally, the voice gateway does not provide the calling number (ANI) to a VoiceXML application if the caller ID is blocked. Caller ID is blocked if a call that comes into the voice gateway has the presentation indication field set to "presentation restricted". The session.telephone.ani variable is set to "blocked". When the **callapplicationvoicesecuritytrusted**command is configured, the gateway does not block caller ID; it provides the calling number to the VoiceXML application.

If the keyword of this command is set to anything other than **trusted**, the value is accepted and the application is treated as not trusted. For example, in the following configuration, the application "sample" is treated as not trusted, and caller ID is blocked:

```
call application voice sample security not_trusted
```

To enable Generic Transparency Descriptor (GTD) parameters in call signaling messages to map to VoiceXML and Tcl session variables, configure the **callapplicationvoicesecuritytrusted** command. If this command is not configured, the VoiceXML variables that correspond to GTD parameters are marked as not available. For a detailed description of the VoiceXML and Tcl session variables, see the Cisco VoiceXML Programmer’s Guide and the Tcl IVR API Version 2.0 Programmer’s Guide, respectively.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice security trusted
        Warning: This command has been deprecated. Please use the following:
   param security trusted (application parameter configuration mode)
    paramspace appcommon security trusted
```

The following example configures the application "sample" as a trusted application. Caller ID is available to this VoiceXML application if it is supported by the service provider.

```
call application voice sample flash:sample.vxml
call application voice sample security trusted
```

The following example configures the application "example" as not trusted. Caller ID can be blocked.

```
call application voice coldcall tftp://joeserver/sellcars.vxml
no call application voice example security trusted
```

Related Commands

Command	Description
call application voice	Defines the name of a voice application and specifies the location of the document (Tcl or VoiceXML) to load for the application.
call application voice language	Defines the language of the audio files used for dynamic prompts by the designated application.
call application voice load	Reloads a Tcl or VoiceXML document.
call application voice pin-len	Defines the number of characters in the PIN for the Tcl application.
call application voice redirect-number	Defines the telephone number to which a call is redirected for the designated application.
call application voice retry-count	Defines the number of times that a caller is permitted to reenter the PIN for a designated application.

Command	Description
call application voice uid-len	Defines the number of characters in the UID for the designated application.
call application voice warning-time	Defines the number of seconds for which a warning prompt is played before a user's account time runs out.
param security	Configures security for linkable Tcl functions (packages).
paramspace appcommon security	Configures security for a service (application).
show call application voice	Displays the following information associated with a voice application: the audio files, the prompts, the caller interaction, and the abort key operation.

call application voice set-location

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoiceset-location** command is replaced by the **paramspacelanguage**command. See the **paramspacelanguage**command for more information.

To define the category and location of audio files that are used for dynamic prompts by the specified IVR application (Tcl or VoiceXML), use the **callapplicationvoicesetlocation** command in global configuration mode. To remove these definitions, use the **no** form of this command.

call application voice *application-name* **set-location** *language category location*
no call application voice *application-name* **set-location** *language category location*

Syntax Description

application name	Name of the application to which the **setlocation** parameters are being passed.
language	Two-character code that identifies the language associated with the audio files. Valid entries are as follows: • **en** --English • **sp** --Spanish • **ch** --Mandarin • **aa** --All This is the same language code that was entered when configuring the**callapplicationvoicelanguage** command .
category	Category group of the audio files (from 0 to 4). For example, audio files representing the days and months can be category 1, audio files representing units of currency can be category 2, and audio files representing units of time--seconds, minutes, and hours--can be category 3. Range is from 0 to 4; 0 means all categories.
location	URL of the audio files. Valid URLs refer to TFTP, FTP, HTTP, or RTSP servers, flash memory, or the removable disks on the Cisco 3600 series.

Command Default

No location or category is set.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.0(7)T	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and the Cisco AS5300.
12.1(5)T	This command was implemented on the Cisco AS5800.
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.

Release	Modification
12.2(2)XB	This command was modified to support VoiceXML applications on the Cisco AS5300, Cisco AS5350, and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on the Cisco 1751.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T and implemented on the Cisco 1750.
12.2(8)T	This command was implemented on the Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T for VoiceXML applications. This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 in this release.
12.3(14)T	This command was replaced by the **paramspacelanguage**command.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

Instead of using this command, you can configure the language and location of prerecorded audio files within a Tcl script or VoiceXML document. For more information, see the Tcl IVR API Version 2.0 Programmer’s Guide or Cisco VoiceXML Programmer’s Guide, respectively.

To identify the language of the audio files, use the**callapplicationvoicelanguage** command.

Tcl scripts and VoiceXML documents can be stored in any of the following locations: On TFTP, FTP, or HTTP servers, in the flash memory on the gateway, or on the removable disks of the Cisco 3600 series. The audio files that they use can be stored in any of these locations, and on RTSP servers.

You can configure multiple set-location lines for a single application.

With the Pre-Paid Debitcard Multi-Language feature, you can create Tcl scripts and a two-character code for any language. See the Cisco Pre-Paid Debitcard Multi-Language Programmer's Reference.

With the multilanguage support for Cisco IOS IVR, you can create a Tcl language module for any language and any set of Text-to-Speech (TTS) notations for use with Tcl and VoiceXML applications. See the Enhanced Multi-Language Support for Cisco IOS Interactive Voice Response document.

The table below lists Tcl script names and the corresponding commands that are required for each Tcl script.

Table 7: Tcl Scripts and Commands

Tcl Script Name	Description	Commands to Configure
app_libretto_onramp9.tcl	Authenticates the account and personal identification number (PIN) using the following: prompt-user, using automatic number identification (ANI), dialed number identification service (DNIS), gateway ID, redialer ID, and redialer DNIS.	None
app_libretto_offramp5.tcl	Authenticates the account and PIN using the following: envelope-from, envelope-to, gateway ID, and x-account ID.	None

Tcl Script Name	Description	Commands to Configure
clid_4digits_npw_3_cli.tcl	Authenticates the account number and PIN, respectively, using ANI and NULL. The number of digits allowed for the account number and password, respectively, are configurable through the command-line interface (CLI). If the authentication fails, the script allows the caller to retry. The retry number is also configured through the CLI.	**call application voice uid-length** Range is 1 to 20. The default is 10. **call application voice pin-length** Range is 0 to 10. The default is 4. **call application voice retry-count** Range is 1 to 5. The default is 3.
clid_authen_col_npw_cli.tcl	Authenticates the account number and PIN, respectively, using ANI and NULL. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_authen_collect_cli.tcl	Authenticates the account number and PIN using ANI and DNIS. If the authentication fails, the script allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_col_npw_3_cli.tcl	Authenticates using ANI and NULL for account numbers and PINs, respectively. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_col_npw_npw_cli.tcl	Authenticates using ANI and NULL for account and PIN, respectively. If authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected together.	**call application voice retry-count** Range is 1 to 5. The default is 3.
fax_rollover_on_busy.tcl	Used for on-ramp T.38 fax rollover to T.37 fax when the destination fax line is busy.	**voice hunt user-busy**

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice set-location
        Warning: This command has been deprecated. Please use the following:
   paramspace language
```

The following example shows how to configure the **callapplicationvoiceset-location** command for the application named "prepaid." In this example, the language specified is English, the category into which the audio files are grouped is category 0 (meaning all), and the location is the keyer directory on the TFTP server.

```
call application voice prepaid set-location en 0 tftp://keyer/
```

The following example shows how to configure the **callapplicationvoiceset-location** command for a fictitious VoiceXML application named "sample." In this example, as in the preceding example, the language defined is English, the category into which the audio files are grouped is category 0 (meaning "all") and the location is the example directory on an HTTP server.

```
call application voice sample set-location en 0 http://example/
```

The following example shows how to configure the **callapplicationvoiceset-location** command for multiple set locations:

```
call application voice sample set-location en 0 http://example/en_msg/
call application voice sample set-location sp 0 http://example/sp_msg/
call application voice sample set-location ch 0 http://example/ch_msg/
```

Related Commands

Command	Description
call application voice	Specifies the application name and indicates the location of the IVR script to be used with this application.
call application voice language	Specifies the audio file language for the designated application.
call application voice load	Reloads the designated Tcl script.
call application voice pin-len	Specifies the number of characters in the PIN.
call application voice redirect-number	Specifies the telephone number to which a call is redirected.
call application voice retry-count	Defines the number of times a caller is permitted to reenter the PIN.
call application voice uid-len	Defines the number of characters in the UID for the designated application.
call application voice warning-time	Defines, in seconds, how long in advance a user is warned before the allowed calling time expires for the designated application.
paramspace language	Defines the category and location of audio files that are used for dynamic prompts by an IVR application (Tcl or VoiceXML).
show call application voice	Displays information about voice applications.

call application voice transfer mode

Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoicetransfermode** command is replaced by the following commands:

- **param mode** (application parameter configuration mode)
- **paramspace callsetup mode** (service configuration mode)

See these commands for more information.

To specify the call-transfer method for Tool Command Language (Tcl) or VoiceXML applications, use the **callapplicationvoicetransfermode** command in global configuration mode. To reset to the default, use the **no** form of this command.

call application voice *application-name* **transfer mode** {**redirect** | **redirect-at-alert** | **redirect-at-connect** | **redirect-rotary** | **rotary**}
no call application voice *application-name* **transfer mode**

Syntax Description

application-name	Name of the voice application for which the transfer method is set.
redirect	Gateway redirects the call leg to the redirected destination number.
redirect-at-alert	Gateway places a new call to the redirected destination number and initiates a call transfer when the outgoing call leg is in the alert state. If the call transfer is successful, the two call legs are disconnected on the gateway. If the transfer fails, the gateway bridges the two call legs. Provides support for Two B-Channel Transfer (TBCT).
redirect-at-connect	Gateway places a new call to the redirected destination number and initiates a call transfer when the outgoing call leg is in the connect state. If the call transfer is successful, the two call legs are disconnected on the gateway. If the transfer fails, the gateway bridges the two call legs. Provides support for TBCT.
redirect-rotary	Gateway redirects the call leg to the redirected destination number. If redirection fails, the gateway places a rotary call to the redirected destination number and hairpins the two call legs. For TBCT, this mode is the same as for the **redirect-at-connect**keyword.
rotary	Gateway places a rotary call for the outgoing call leg and hairpins the two call legs. Call redirection is not invoked. This is the default.

Command Default

Rotary method; call redirection is not invoked.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(1)	This command was introduced.

Release	Modification
12.3(14)T	This command was replaced by the following commands: • **param mode** (application parameter configuration mode) • **paramspace callsetup mode** (service configuration mode)
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command determines whether a voice application can invoke TBCT or RTPvt. Before you can use this command, you must configure the named application on the gateway by using the **callapplicationvoice** command.

Redirect-rotary is the preferred transfer method because it ensures that a call-redirect method is always selected if the call leg is capable of it.

Tcl scripts can read the value of this command by using the info tag get cfg_avpair transfer-mode statement. For detailed information, see the Tcl IVR API Version 2.0 Programmer's Guide .

For VoiceXML applications, the value of this command becomes the default behavior if the com.cisco.transfer.mode property is not specified in the VoiceXML document. For detailed information, see the Cisco VoiceXML Programmer's Guide. The VoiceXML document property takes precedence over the gateway configuration.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice transfer mode
 Warning: This command has been deprecated. Please use the following:
  param mode
   paramspace callsetup mode
```

The following example sets the transfer method to redirect for the application callme:

```
Router(config)# call application voice callme transfer mode redirect
```

Related Commands

Command	Description
application	Enables a voice application on a dial peer.
call application voice	Defines the name of a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
call application voice transfer reroute-mode	Specifies the call-forwarding behavior of a Tcl application.
debug voip ivr callsetup redirect	Displays debugging information about H.450 calls that are redirected during setup.
debug voip ivr redirect	Displays debugging information about redirected H.450 calls.

Command	Description
isdn supp-service tbct	Enables ISDN TBCT on PRI trunks.
param mode	Configures the call transfer mode for a package.
paramspace callsetup mode	Configures the call transfer mode for an application.
show call active voice redirect	Displays information about active calls that are being redirected using RTPvt or TBCT.
show call application voice	Displays information about voice applications.
show call history voice redirect	Displays history information about calls that were redirected using RTPvt or TBCT.

call application voice transfer reroute-mode

Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoicetransferreroute-mode** command is replaced by the following commands:

- **param reroutemode** (application parameter configuration mode)
- **paramspace callsetup reroutemode** (service configuration mode)

See these commands for more information.

To specify the call-forwarding behavior of a Tool Command Language (Tcl) application, use the **callapplicationvoicetransferreroute-mode** command in global configuration mode. To reset to the default, use the **no** form of this command.

call application voice *application-name* **transfer reroute-mode** {**none** | **redirect** | **redirect-rotary** | **rotary**}
no call application voice *application-name* **transfer reroute-mode**

Syntax Description

application-name	Name of the voice application for which the transfer reroute method is set.
none	Call forwarding is not performed by the voice application.
redirect	Two call legs are directly connected. Provides support for RTPvt.
redirect-rotary	Two call legs are directly connected (redirect). If that fails, the two call legs are hairpinned on the gateway (rotary).
rotary	Gateway places a rotary call for the outgoing call leg and hairpins the two calls together. RTPvt is not invoked. This is the default.

Command Default

Rotary method; RTPvt is not invoked.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(1)	This command was introduced.
12.3(14)T	This command was replaced by the following commands: • **param reroutemode** (application parameter configuration mode) • **paramspace callsetup reroutemode** (service configuration mode)
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

Before you can use this command, you must configure the named application on the gateway by using the **callapplicationvoice** command. This command is not supported for VoiceXML applications or for TBCT.

Redirect-rotary is the preferred transfer method because it ensures that a call-redirect method is always selected, provided that the call leg is capable of it.

Tcl scripts can read the value of this command by using the info tag get cfg_avpair reroute-mode statement. For detailed information, see the Tcl IVR API Version 2.0 Programmer's Guide .

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice transfer reroute-mode
 Warning: This command has been deprecated. Please use the following:
  param reroutemode (application parameter configuration mode)
   paramspace callsetup reroutemode
```

The following example sets the call forwarding method to redirect for the application callme:

```
Router(config)# call application voice callme transfer reroute-mode redirect
```

Related Commands

Command	Description
application	Enables a voice application on a dial peer.
call application voice	Defines the name of a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
call application voice transfer mode	Specifies the call-transfer behavior of a Tcl or VoiceXML application.
isdn supp-service tbct	Enables ISDN TBCT on PRI trunks.
param reroutemode	Configures the call transfer reroutemode (call forwarding) for a package.
paramspace callsetup reroutemode	Configures the call reroute mode (call forwarding) for an application.
show call active voice redirect	Displays information about active calls that are being redirected using RTPvt or TBCT.
show call application voice	Displays information about voice applications.
show call history voice redirect	Displays history information about calls that were redirected using RTPvt or TBCT.

call application voice uid-length

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoiceuid-length** command is replaced by the **paramuid-len** command. See the **paramuid-len** command for more information.

To define the number of characters in the user identification (UID) number for the designated application and to pass that information to the specified application, use the **callapplicationvoiceuid-length**command in global configuration mode. To restore the default setting for this command, use the **no** form of this command.

call application voice *application-name* **uid-length** *number*
no call application voice *application-name* **uid-length** *number*

Syntax Description

application-name	Name of the application to which the UID length parameter is passed.
number	Number of allowable characters in UIDs that are associated with the specified application. Range is from 1 to 20. The default is 10.

Command Default

number

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)T	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and the Cisco AS5300.
12.1(5)T	This command was implemented on the Cisco AS5800.
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on the Cisco 1751. This release does not support any other Cisco platforms.
12.2(4)T	Support was added for the Cisco 1750.
12.2(8)T	This command was implemented on the Cisco 7200 series.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T. This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 in this release.
12.3(14)T	This command was replaced by the **paramuid-len** command.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

Use this command when configuring interactive voice response (IVR)--depending on the Tool Command Language (Tcl) script being used--or one of the IVR-related features (such as Debit Card) to define the number of allowable characters in a UID for the specified application and to pass that information to the specified application.

The table below lists Tcl script names and the corresponding commands that are required for each Tcl script.

Table 8: Tcl Scripts and Commands

Tcl Script Name	Description	Commands to Configure
app_libretto_onramp9.tcl	Authenticates the account and personal identification number (PIN) using the following: prompt-user, using automatic number identification (ANI), dialed number identification service (DNIS), gateway ID, redialer ID, and redialer DNIS.	None
app_libretto_offramp5.tcl	Authenticates the account and PIN using the following: envelope-from, envelope-to, gateway ID, and x-account ID.	None
clid_4digits_npw_3_cli.tcl	Authenticates the account number and PIN, respectively, using ANI and NULL. The number of digits allowed for the account number and password, respectively, are configurable through the command-line interface (CLI). If the authentication fails, the script allows the caller to retry. The retry number is also configured through the CLI.	**call application voice uid-length** Range is 1 to 20. The default is 10. **call application voice pin-length** Range is 0 to 10. The default is 4. **call application voice retry-count** Range is 1 to 5. The default is 3.
clid_authen_col_npw_cli.tcl	Authenticates the account number and PIN, respectively, using ANI and NULL. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_authen_collect_cli.tcl	Authenticates the account number and PIN using ANI and DNIS. If the authentication fails, the script allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_col_npw_3_cli.tcl	Authenticates using ANI and NULL for account numbers and PINs, respectively. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_col_npw_npw_cli.tcl	Authenticates using ANI and NULL for account and PIN, respectively. If authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected together.	**call application voice retry-count** Range is 1 to 5. The default is 3.
fax_rollover_on_busy.tcl	Used for on-ramp T.38 fax rollover to T.37 fax when the destination fax line is busy.	**voice hunt user-busy**

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice uid-length
 Warning: This command has been deprecated. Please use the following:
  param uid-len
```

The following example shows how to configure four allowable characters in the UID for the application named "sample":

```
Router(config)# all application voice sample uid-length 4
```

Related Commands

Command	Description
call application voice	Specifies the name to be used for an application and indicates the location of the appropriate IVR script to be used with this application.
call application voice language	Specifies the language of the audio file for the designated application and passes that information to the application.
call application voice load	Reloads the designated Tcl script.
call application voice pin-len	Defines the number of characters in the PIN for the application and passes that information to the application.
call application voice redirect-number	Specifies the telephone number to which a call is redirected for the designated application.
call application voice retry-count	Defines the number of times a caller is permitted to reenter the PIN for a designated application and passes that information to the application.
call application voice set-location	Defines the location, language, and category of the audio files for the designated application and passes that information to the application.
call application voice warning-time	Defines, in seconds, how long in advance a user is warned before the allowed calling time expires for the designated application.
param uid-length	Defines the number of characters in the UID for a package.

call application voice voice-dtmf

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoicevoice-dtmf** command is replaced by the **paramvoice-dtmf**command. See the **paramvoice-dtmf** command for more information.

To direct the fax detection interactive voice response (IVR) application to recognize a specified digit to indicate a voice call, use the **callapplicationvoicevoice-dtmf** command in global configuration mode. To remove configuration of this digit, use the **no** form of this command.

call application voice *application-name* **voice-dtmf** *keypad-character*
no call application voice *application-name* **voice-dtmf** *keypad-character*

Syntax Description

application-name	The name of the fax detection application that you defined when you loaded the application on the router.
keypad-character	Single character that can be dialed on a telephone keypad pressed by the calling party to indicate a voice call, in response to the audio prompt configured in default-voice and default-fax mode of the fax detection IVR application. Default is 1.

Command Default

1

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(5)XM	This command was introduced for the Cisco AS5300.
12.2(2)XB	This command was implemented on the Cisco AS5400 and Cisco AS5350.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745. Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 is not included in this release.
12.2(11)T	This command was supported on the Cisco AS5300, Cisco AS5350, and Cisco AS5400 in this release.
12.3(14)T	This command was replaced by the **paramvoice-dtmf**command.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

This command is useful only when the fax detection IVR application is being configured in default-voice mode or default-fax mode, as defined by the **callapplicationvoicemode** command. Only one digit can be specified in this command, and that digit must be different from the digit specified in the **callapplicationvoicefax-dtmfcommand**. You are not notified immediately if you make the error of configuring them both to the same digit. To find this error, you must start debugging with the **debugvoipivrscript** command and then observe some failing calls.

This command is not supported by Cisco IOS help. If you type the **callapplicationvoicefax_detectvoice-dtmf** command and a question mark (**?**), the Cisco IOS help does not supply a list of entries that are valid in place of the question mark.

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice voice-dtmf
 Warning: This command has been deprecated. Please use the following:
  param voice-dtmf
```

The following example selects digit 2 dual tone multifrequency (DTMF) to indicate a voice call:

```
call application voice fax_detect script_url
call application voice fax_detect voice-dtmf 2
dial-peer voice 302 pots
 application fax_detect
```

Related Commands

Command	Description
call application voice	Loads a specified IVR application onto the router from the TFTP server and gives it an application name by which it is known on the router.
call application voice account-id-method	Configures the fax detection IVR application to use a particular method to assign the account identifier.
call application voice fax-dtmf	Configures the fax detection IVR application to recognize a specified digit to indicate a fax call.
call application voice mode	Configures the fax detection IVR application to operate in one of its four modes.
call application voice prompt	Configures the fax detection IVR application to use the specified audio file as a user prompt.
param voice-dtmf	Directs the fax detection IVR application to recognize a specified digit to indicate a voice call.

call application voice warning-time

Note Effective with Cisco IOS Release 12.3(14)T, the **callapplicationvoicewarning-time** command is replaced by the **paramwarning-time**command. See the **paramwarning-time**command for more information.

To define the number of seconds of warning that a user receives before the allowed calling time expires use the **callapplicationvoicewarning-time** command in global configuration mode. To remove the configured warning period, use the **no** form of this command.

call application voice *application-name* **warning-time** *seconds*
no call application voice *application-name* **warning-time** *seconds*

Syntax Description

application-name	Name of the application to which the warning time parameter is being passed.
seconds	Length of the warning period, in seconds, before the allowed calling time expires. Range is from 10 to 600. This argument has no default value.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

Release	Modification
12.0(7)T	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco AS5300.
12.1(5)T	This command was implemented on the Cisco AS5800.
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on the Cisco 1751. Support for other Cisco platforms is not included in this release.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T and implemented on the Cisco 1750.
12.2(8)T	This command was implemented on the Cisco 7200 series.
12.2(11)T	This command was implemented on the Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 in this release.
12.3(14)T	This command was replaced by the **paramwarning-time**command.
12.4(24)T	This command was modified. The automatic conversion to the new CLI is replaced with an explicit error message.

Usage Guidelines

Use this command when configuring interactive voice response (IVR)--depending on the Tool Command Language (Tcl) script being used--or one of the IVR-related features (such as Debit Card) to define the number of seconds in the warning period before the allowed calling time expires for the specified application and to pass that information to the specified application.

The table below lists Tcl script names and the corresponding commands that are required for each Tcl script.

Table 9: Tcl Scripts and Commands

Tcl Script Name	Description	Commands to Configure
app_libretto_onramp9.tcl	Authenticates the account and personal identification number (PIN) using the following: prompt-user, using automatic number identification (ANI), dialed number identification service (DNIS), gateway ID, redialer ID, and redialer DNIS.	None
app_libretto_offramp5.tcl	Authenticates the account and PIN using the following: envelope-from, envelope-to, gateway ID, and x-account ID.	None
clid_4digits_npw_3_cli.tcl	This script authenticates the account number and PIN, respectively, using ANI and NULL. The number of digits allowed for the account number and password, respectively, are configurable through the command-line interface (CLI). If the authentication fails, the script allows the caller to retry. The retry number is also configured through the CLI.	**call application voice uid-length** Range is 1 to 20. The default is 10. **call application voice pin-length** Range is 0 to 10. The default is 4. **call application voice retry-count** Range is 1 to 5. The default is 3.
clid_authen_col_npw_cli.tcl	This script authenticates the account number and PIN, respectively, using ANI and NULL. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_authen_collect_cli.tcl	This script authenticates the account number and PIN using ANI and DNIS. If the authentication fails, the script allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_col_npw_3_cli.tcl	This script authenticates using ANI and NULL for account numbers and PINs, respectively. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI.	**call application voice retry-count** Range is 1 to 5. The default is 3.
clid_col_npw_npw_cli.tcl	This script authenticates using ANI and NULL for account and PIN, respectively. If authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected together.	**call application voice retry-count** Range is 1 to 5. The default is 3.
fax_rollover_on_busy.tcl	Used for on-ramp T.38 fax rollover to T.37 fax when the destination fax line is busy.	**voice hunt user-busy**

Examples

Effective with Cisco IOS Release 12.4(24)T, the following warning message is displayed to direct users to the replacement command options:

```
Router(config)# call application voice param warning-time

 Warning: This command has been deprecated. Please use the following:
  param warning-time
```

The following example shows how to configure a 30-second warning time for the application named "sample":

```
Router(config)# call application voice sample warning-time 30
```

Related Commands

Command	Description
call application voice language	Specifies the language of the audio file for the designated application and passes that information to the application.
call application voice load	Reloads the designated Tcl script.
call application voice location	Specifies the name to be used for an application and indicates the location of the appropriate IVR script to be used with this application.
call application voice pin-len	Defines the number of characters in the PIN for the application and passes that information to the application.
call application voice redirect-number	Specifies the telephone number to which a call is redirected for the designated application.
call application voice retry-count	Defines the number of times a caller is permitted to reenter the PIN for a designated application and passes that information to the application.
call application voice set-location	Defines the location, language, and category of the audio files for the designated application and passes that information to the application.
call application voice uid-length	Defines the number of characters in the UID for the designated application and passes that information to the application.
param warning-time	Defines the number of seconds of warning that a user receives before the allowed calling time expires.

call-block (dial peer)

To enable blocking of incoming calls, use the **call-block**command in dial peer configuration mode. To return to the default value, use the **no** form of this command.

call-block {**disconnect-cause incoming** {**call-reject** | **invalid-number** | **unassigned-number** | **user-busy**} | **translation-profile incoming** *name*}
no call-block {**disconnect-cause incoming** {**call-reject** | **invalid-number** | **unassigned-number** | **user-busy**} | **translation-profile incoming** *name*}

Syntax Description

disconnect-cause incoming	Associates a disconnect cause of incoming calls.
call-reject	Specifies call rejection as the cause for blocking a call during incoming call-number translation.
invalid-number	Specifies invalid number as the cause for blocking a call during incoming call-number translation.
unassigned-number	Specifies unassigned number as the cause for blocking a call during incoming call-number translation.
user-busy	Specifies busy as the cause for blocking a call during incoming call-number translation.
translation-profile incoming	Associates the translation profile for incoming calls.
name	Name of the translation profile.

Command Default

Disconnect cause: No Service (once the call-blocking translation profile is defined) Translation profile: No default behavior or values

Command Modes

Dial peer configuration (config-dial-peer)

Command History

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

Usage Guidelines

An incoming call can be blocked from the gateway if one of the call numbers (calling, called, or redirect) is matched with the reject translation rule of the incoming call-blocking translation profile.

The cause value is returned to the source of the call when a call is blocked during the incoming call-number translation.

This command is supported in POTS, VoIP, VoFR, and VoATM dial-peer configuration. For VoATM, only ATM Adaptation Layer 5 (AAL5) calls are supported.

The only option for call blocking is in the incoming direction. From the perspective of the voice gateway, the incoming direction can be either of the following:

- Incoming from a telephony device directly attached to a voice port on the gateway toward the gateway itself
- Incoming by way of an inbound Voice over X (VoX) call from a peer gateway

To configure incoming call blocking, define a translation rule with a **reject** keyword. For example:

```
voice translation-rule 1
 rule 1 reject /408252*/
```

Apply the rule to a translation profile for called, calling, or redirect-called numbers, such as:

```
voice translation profile call_block_profile
 translate calling 1
```

Include the translation profile within a dial peer definition. For example:

```
dial-peer voice 111 pots
 call-block translation-profile incoming call_block_profile
 call-block disconnect-cause incoming invalid_number
```

In this example, the gateway blocks any incoming time-division multiplexing (TDM) call that successfully matches inbound dial-peer 111 and has a calling number that starts with 408252. The gateway also returns the disconnect cause "invalid number" to the source of the call. (Other disconnect causes can be assigned: unassigned-number, user-busy, or call-rejected.)

Examples

The following example assigns the translation profile "example" to be used for incoming calls and returns the message "invalid number" as a cause for blocked calls:

```
Router(config)# dial-peer voice 5 pots
Router(config-dial-peer)# call-block translation-profile incoming example
Router(config-dial-peer)# call-block disconnect-cause incoming invalid-number
```

Following are two possible call-blocking scenarios:

Scenario 1: Block Inbound Calls from the PSTN/PBX/CO

We place the rejection profile on a POTS dial peer that is associated with the voice port on which we expect the inbound call. When the inbound call attempt is made, we see in the CCAPI debugs that POTS dial-peer 9 is matched for the telephony call leg. The call-block rule is checked and we send back user-busy to the switch.

```
voice translation-rule 1
 rule 1 reject /9193927582/   <<<<-------- filter out calls from this CallerID
voice translation-profile reject_ANI
 translate calling 1
dial-peer voice 9 pots
 destination-pattern 9T
 direct-inward-dial
 port 1/0:23
 call-block translation-profile incoming reject_ANI
 call-block disconnect-cause incoming user-busy
```

Scenario 2: Block Inbound VoX Calls from Using Local POTS Resources

We place the rejection profile on a VoIP/VoATM/VoFR dial peer that matches an inbound VoX call attempt. When the inbound call attempt is made, we see in the CCAPI debugs that VoIP dial-peer 7 is matched for the IP call leg. The call-block rule is checked and we send back user-busy to the switch.

```
voice translation-rule 1
 rule 1 reject /9193927582/   <<<<-------- filter out calls from this CallerID
voice translation-profile reject_ANI
 translate calling 1
dial-peer voice 7 voip
 destination-pattern 7T
 session target ipv4:A.B.C.D
 incoming called-number .   <<<<-------- force inbound IP call-leg match
 call-block translation-profile incoming reject_ANI
 call-block disconnect-cause incoming user-busy
```

Related Commands

Command	Description
dial-peer voice	Initiates the dial-peer voice configuration mode.
voice translation-profile	Defines a translation profile for voice calls.
voice translation-rule	Defines a translation rule for voice calls.

call-denial

The call - denial command is replaced by the **callthresholdglobal**command. See the **callthresholdglobal** command for more information.

call fallback through called-number (dial peer)

call fallback

To enable a call request to fall back to a specific dial peer in case of network congestion, use the **callfallback** command in dial peer configuration mode. To disable PSTN fallback for a specific dial peer, use the **no** form of this command.

call fallback
no call fallback

Syntax Description

This command has no arguments or keywords.

Command Default

This command is enabled by default if the **callfallbackactive** command is enabled in global configuration mode

Command Modes

Dial peer configuration (config-dial-peer)

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	The PSTN Fallback feature and enhancements were introduced on Cisco 7200 series and integrated into Cisco IOS Release 12.2(4)T.
12.2(4)T2	This command was implemented on the Cisco 7500 series.

Usage Guidelines

Disabling the **callfallback** command for a dial peer causes the call fallback subsystem not to fall back to the specified dial peer. Disabling the command is useful when internetworking fallback capable H.323 gateways with the Cisco CallManager or third-party equipment that does not run fallback. Connected calls are not affected by this feature.

Examples

The following example disables a PSTN fallback for a specific dial peer:

```
no call fallback
```

Related Commands

Command	Description
call fallback active	Enables a call request to fall back to alternate dial peers.
call fallback cache-size	Specifies the call fallback cache size for network traffic probe entries.
call fallback cache-timeout	Specifies the time after which the cache entries of network conditions are purged.
call fallback instantaneous-value-weight	Configures the call fallback subsystem to take an average from the last two cache entries for call requests.

Command	Description
call fallback jitter-probe num-packets	Specifies the number of packets in a jitter probe that are used to determine network conditions.
call fallback jitter-probe precedence	Specifies the priority of the jitter-probe transmission.
call fallback jitter-probe priority-queue	Assigns a priority queue for jitter-probe transmissions.
call fallback key-chain	Specifies use of MD5 authentication for sending and receiving SAA probes.
call fallback map address-list	Specifies that the call fallback router keep a cache table by IP addresses of distances for several destination peers that are sitting behind the router.
call fallback map subnet	Specifies that the call fallback router keep a cache table by subnet addresses of distances for several destination peers that are sitting behind the router.
call fallback probe-timeout	Sets the timeout for an SAA probe for call fallback purposes.
call fallback threshold delay loss	Specifies that the call fallback threshold use only packet delay and loss values.
call fallback threshold icpif	Specifies that call fallback use the ICPIF threshold.
dial-peer voice number	Enters dial peer configuration mode.
show call fallback config	Displays the call fallback configuration.

call fallback active

To enable the Internet Control Message Protocol (ICMP)-ping or Service Assurance Agent (SAA) (formerly Response Time Reporter [RTR]) probe mechanism for use with the dial-peer **monitorprobe** or voice-port **busyoutmonitorprobe** commands, use the **callfallbackactive** command in global configuration mode. To disable these probe mechanisms, use the **no** form of this command.

call fallback active [{**icmp-ping** | **rtr**}]
no call fallback active [{**icmp-ping** | **rtr**}]

Syntax Description

icmp-ping	Uses ICMP pings to monitor the IP destinations.
rtr	Uses SAA (formerly RTR) probes to monitor the IP destinations. SAA (RTR) probes are the default.

Command Default

This command is disabled by default. If the command is entered without an optional keyword, the default is RTR.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)T	This command was introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)T	The PSTN Fallback feature and enhancements were implemented on the Cisco 7200 series and integrated into Cisco IOS Release 12.2(4)T.
12.2(4)T2	This command was implemented for Cisco 7500 series.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

The **callfallbackactive** command creates and maintains a consolidated cache of probe results for use by the dial-peer **monitorprobe** or voice-port **busyoutmonitorprobe** commands.

Enabling the **callfallbackactive** command determines whether calls should be accepted or rejected on the basis of probing of network conditions. The **callfallbackactive** command checks each call request and rejects the call if the network congestion parameters are greater than the value of the configured threshold parameters of the destination. If this is the case, alternative dial peers are tried from the session application layer.

Use the**callfallbackthresholddelayloss** or **callfallbackthresholdicpif** command to set the threshold parameters.

Connected calls are not affected by this command.

Caution The **callfallbackactiveicmp-ping** command must be entered before the **callfallbackicmp-ping** command can be used. If you do not enter this command first, the **callfallbackicmpping** command will not work properly.

Note The Cisco SAA functionality in Cisco IOS software was formerly known as Response Time Reporter (RTR). The command-line interface still uses the keyword **rtr** for configuring RTR probes, which are now actually SAA probes.

Examples

The following example enables the **callfallbackactive**command and globally enables ICMP pinging to probe target destinations. The second command specifies values for the ping packets:

```
Router(config)# call fallback active icmp-ping
Router(config)# call fallback icmp-ping codec g729 interval 10 loss 10
```

Related Commands

Command	Description
call fallback cache-size	Specifies the call fallback cache size for network traffic probe entries.
call fallback cache-timeout	Specifies the time after which the cache entries of network conditions are purged.
call fallback instantaneous-value-weight	Specifies the call fallback subsystem to take an average from the last two cache entries for call requests.
call fallback jitter-probe num-packets	Specifies the number of packets in a jitter probe that are used to determine network conditions.
call fallback jitter-probe precedence	Specifies the priority of the jitter-probe transmission.
call fallback jitter-probe priority-queue	Assigns a priority queue for jitter-probe transmissions.
call fallback key-chain	Specifies use of MD5 authentication for sending and receiving SAA probes.
call fallback map address-list	Specifies that the call fallback router keep a cache table by IP addresses of distances for several destination peers that are sitting behind the router.
call fallback map subnet	Specifies that the call fallback router keep a cache table by subnet addresses of distances for several destination peers that are sitting behind the router.
call fallback probe-timeout	Sets the timeout for an SAA probe for call fallback purposes.
call fallback threshold delay loss	Specifies that the call fallback threshold use only packet delay and loss values.
call fallback threshold icpif	Specifies that call fallback use the ICPIF threshold.
dial-peer voice number	Enters dial peer configuration mode.

call fallback cache-size

To specify the call fallback cache size for network traffic probe entries, use the **callfallbackcachesize** command in global configuration mode. To restore the default value, use the **no** form of this command.

call fallback cache-size *number*
no call fallback cache-size

Syntax Description

number	Cache size, in number of entries. Range is from 1 to 256. The default is 128.

Command Default

128 entries

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)T	This command was introduced..
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)T	The PSTN Fallback feature and enhancements were introduced on the Cisco 7200 series and integrated into Cisco IOS Release 12.2(4)T.
12.2(4)T2	This command was implemented on the Cisco 7500 series.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

The cache size can be changed only when the **callfallbackactive** command is not enabled.

The overflow process deletes up to one-fourth of the cache entries to allow for additional calls beyond the specified cache size. The cache entries chosen for deletion are the oldest entries in the cache.

If the cache size is left unchanged, it can be changed only when fallback is off. Use the **no** form of the **callfallback** command to turn fallback off.

Examples

The following example specifies 120 cache entries:

```
Router(config)#
call fallback cache-size 120
```

Related Commands

Command	Description
call fallback	Enables a call request to fall back to a specific dial peer in case of network congestion
call fallback active	Enables a call request to fall back to alternate dial peers in case of network congestion.

Command	Description
call fallback cache-timeout	Specifies the time after which the cache entries of network conditions are purged.
show call fallback cache	Displays the current ICPIF estimates for all IP addresses in the cache.
show call fallback config	Displays the call fallback configuration.

call fallback cache-timeout

To specify the time after which the cache entries of network conditions are purged, use the **callfallbackcachetimeout** command in global configuration mode. To disable the **callfallbackcache-timeout** command, use the **no** form of this command.

call fallback cache-timeout *seconds*
no call fallback cache-timeout

Syntax Description

seconds	Cache timeout value, in seconds. Range is from 1 to 2147483. The default is 600.

Command Default

600 seconds

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)T	This command was introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)T	The PSTN Fallback feature and enhancements were implemented on the Cisco 7200 series and integrated into Cisco IOS Release 12.2(4)T.
12.2(4)T2	This command was implemented on the Cisco 7500 series.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

Enabling the **callfallbackcachetimeout** command sends a Service Assurance Agent (SAA) probe out to the network to determine the amount of congestion in terms of configured thresholds. The network condition is based upon delay and loss, or Calculated Planning Impairment Factor (ICPIF) thresholds. Use the**callfallbackthresholddelayloss** or **callfallbackthresholdicpif** command to set the threshold parameters.

The cache keeps entries for every network congestion - checking probe sent and received between timeouts. The cache updates after each probe returns the current condition of network traffic. To set the probe frequency, use the **callfallbackprobetimeout** command.

When a call comes into the router, the router matches a dial peer and obtains the destination information. The router calls the fallback subsystem to look up the specified destination in its network traffic cache. If the delay/loss or ICPIF threshold exists and is current, the router uses that value to decide whether to permit the call into the Voice over IP (VoIP) network. If the router determines that the network congestion is below the configured threshold (by looking at the value in the cache), the call is connected.

After each call request, the timer is reset. Purging of the cache occurs only when the cache has received no call requests during the timeout period (*seconds*). When the cache timeout expires, the entire cache is deleted, and a probe is sent to start a new cache entry. A call cannot be completed until this probe returns with network traffic information.

The network congestion probes continue in the background as long as the entry for the last call request remains in the cache.

Examples

The following example specifies an elapsed time of 1200 seconds before the cache times out:

```
Router(config)# call fallback cache-timeout 1200
```

Related Commands

Command	Description
call fallback active	Enables a call request to fall back to alternate dial peers in case of network congestion.
call fallback cache-size	Specifies the call fallback cache size.
call fallback probe-timeout	Specifies the time after which the cache entries of network conditions are purged.
call fallback threshold delay loss	Configures the call fallback threshold to use only packet delay and loss values.
call fallback threshold icpif	Specifies that call fallback use the ICPIF threshold.
show call fallback cache	Displays the current ICPIF estimates for all IP addresses in the cache.
show call fallback config	Displays the call fallback configuration.

call fallback expect-factor

To set a configurable value by which the call fallback expect factor feature will be activated, use the **callfallbackexpect-factor**command in global configuration mode. To disable the expect factor, use the **no** form of this command.

call fallback expect-factor *value*
no call fallback expect-factor

Syntax Description

value	Configures the expect-factor A. Range: 0 to 20. Default: 10.

Command Default

No value for the expect-factor is configured.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(3)	This command was introduced.
12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

The expect-factor is the level of expected voice quality that the user may have during a call. For example, you expect higher voice quality from a call on your home than on your cell phone. The expect-factor is a subjective value determined by the local administrators.

Call fallback is used by the software to generate a series of probes across an IP network to help make a Impairment/Calculated Impairment Planning Factor (ICPIF) calculation. The value calculated by the probes, ICPIF, is modified by the configured expect factor using the following formula:

ICPIF = Idd + Ie-A

Idd represents the impairment due to end-end delay, Ie, represents the impairment due to packet loss and the impact of the codec being used on the call, and A represents the expect-factor value. The expect-factor is the value to be subtracted from the calculated ICPIF value. This expect factor is known as the Advantage Factor (A) as specified in G.107 and takes into account the user’s expected level of voice quality based upon the type of call being made.

Examples

The following example shows the **callfallbackexpect-factor**command and the **callfallbackthresholdicpicf** command being configured. A calculated ICPIF value of 20 based on Idd and Ie from the probes set on a IP network would not activate the call fallback feature in this configuration. Even though the calculated ICPIF value of 20 exceeds the configured threshold of 10, subtraction of the expect-value of 15 would leave a value of 5, which is below the threshold value.

```
Router(config)# call fallback expect-factor 15
Router(config)# call fallback threshold icpif 10
```

Related Commands

Command	Description
call fallback active	Enables a call request to fall back to alternate dial peers.
call fallback cache-size	Specifies the call fallback cache size for network traffic probe entries.
call fallback cache-timeout	Specifies the time after which the cache entries of network conditions are purged.
call fallback instantaneous-value-weight	Configures the call fallback subsystem to take an average from the last two cache entries for call requests.
call fallback jitter-probe num-packets	Specifies the number of packets in a jitter probe that are used to determine network conditions.
call fallback jitter-probe precedence	Specifies the priority of the jitter-probe transmission.
call fallback jitter-probe priority-queue	Assigns a priority queue for jitter-probe transmissions.
call fallback key-chain	Specifies use of MD5 authentication for sending and receiving SAA probes.
call fallback map address-list	Specifies that the call fallback router keep a cache table by IP addresses of distances for several destination peers that are sitting behind the router.
call fallback map subnet	Specifies that the call fallback router keep a cache table by subnet addresses of distances for several destination peers that are sitting behind the router.
call fallback probe-timeout	Sets the timeout for an SAA probe for call fallback purposes.
call fallback threshold delay loss	Specifies that the call fallback threshold use only packet delay and loss values.
call fallback threshold icpif	Specifies that call fallback use the ICPIF threshold.
dial-peer voice number	Enters dial peer configuration mode.
show call fallback config	Displays the call fallback configuration.

call fallback icmp-ping

To specify Internet Control Message Protocol (ICMP) ping as the method for network traffic probe entries to IP destinations and configure parameters for the ping packets, use the **callfallbackicmp-ping** command in global configuration mode. To restore the default value, use the **no** form of this command.

call fallback icmp-ping[{**count** *packets* | **size***bytes*}]**interval***seconds*[{[**loss** [*percent*]]}]**timeout***milliseconds*
no call fallback icmp-ping[{**count** *packets* | **size***bytes*}]**interval***seconds*[{[**loss** [*percent*]]}]**timeout***milliseconds*

Syntax Description

count *packets*	(Optional) Number of ping packets that are sent to the destination address.
codec	(Optional) Configures the profile of the SAA probe signal to mimic the packet size and interval of a specific codec type.
codec -type	(Optional) The codec type for the SAA probe signal. Available options are as follows: • **g711a** --G.711 a-law • **g711u** --G.711 mu-law • **g729** --G.729 (the default) • **g729b** --G.729 Annex B
size *bytes*	(Optional) Size (in bytes) of the ping packet. Default is 32.
interval *seconds*	Time (in seconds) between ping packet sets. Default is 5. This number should be higher than the **timeout***milliseconds* value.
loss *percent*	(Optional) Configures the percentage-of-packets-lost threshold for initiating a busyout condition.
timeout *milliseconds*	(Optional) Timeout (in milliseconds) for echo packets. Default is 500.

Command Default

If this command is not configured, Response Time Reporter (RTR) is the probe method used.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.4(2)T	This command was introduced in a release earlier than Cisco IOS Release 12.4(2)T.

Usage Guidelines

The values configured by the global configuration version of the **callfallbackicmp-ping** command are appllied globally for measurements on probes and pings. If the **callfallbackicmp-ping** is configured in dial-peer configuration mode, these values override the global configuration for the specific dial peer.

One of these two commands must be in effect before the **monitorprobeicmp-ping** command can be used. If neither of the **callfallback** commands is in effect, the **monitorprobeicmp-ping** command will not work properly.

Examples

The following example shows how to configure an ICMP ping probe with a G.729 profile to probe the link with an interval value of 10 seconds and a packet-loss threshold of 10 percent:

```
call fallback active icmp-ping
call fallback icmp-ping codec g729 interval 10 loss 10
```

Related Commands

Command	Description
call fallback active	Forces a voice port into the busyout state.
call fallback icmp-ping (dial peer)	Specifies Internet Control Message Protocol (ICMP) ping as the method for network traffic probe entries to IP destinations.
monitor probe icmp-ping	Enables dial-peer status changes based on the results of probes.

call fallback icmp-ping (dial peer)

To specify Internet Control Message Protocol (ICMP) ping as the method for network traffic probe entries to IP destinations, use the **callfallbackicmp-ping** command in dial-peer configuration mode. To restore the default value, use the **no** form of this command.

call fallback [{**icmp-ping** | **rtr**}]
no call fallback [{**icmp-ping** | **rtr**}]

Syntax Description

icmp-ping	(Optional) Specifies ICMP ping as the method for monitoring the session target and updating the status of the dial peer.
rtr	(Optional) Specifies that the Response Time Reporter (RTR) probe is the method for monitoring the session target and updating the status of the dial peer.

Command Default

If this command is not entered, the globally configured method is used for measurements.

Command Modes

Dial peer configuration (config-dial-peer)

Command History

Release	Modification
12.2(11)T	This command was introduced in a release earlier than Cisco IOS Release 12.2(11)T.

Usage Guidelines

The principal use of this command is to specify ICMP ping as the probe method, even though the option for selecting RTR is also available.

If the **callfallbackicmp-ping** command is not entered, the **callfallbackactive** command in global configuration is used for measurements. If the **callfallbackicmp-ping** command is entered, these values override the global configuration.

One of these two commands must be in effect before the **monitorprobeicmp-ping** command can be used. If neither of the **callfallback** commands is in effect, the **monitorprobeicmp-ping** command will not work properly.

Note The Cisco Service Assurance Agent (SAA) functionality in Cisco IOS software was formerly known as Response Time Reporter (RTR). The command-line interface still uses the keyword **rtr** for configuring RTR probes, which are now actually the SAA probes.

Examples

The following example specifies that ICMP ping is used for monitoring the session target IP address and for updating the status of the dial peer:

```
Router(config)#
dial-peer voice 10 voip
Router(config-dial-peer)#
call fallback icmp-ping
```

Related Commands

Command	Description
call fallback	Enables a call request to fall back to a specific dial peer in case of network congestion
call fallback active	Enables a call request to fall back to alternate dial peers in case of network congestion.
monitor probe icmp-ping	Specifies that ICMP ping is the method used for probes.
show call fallback config	Displays the call fallback configuration.

call fallback instantaneous-value-weight

To configure the call fallback subsystem to take an average from the last two probes registered in the cache for call requests, use the **callfallbackinstantaneousvalueweight** command in global configuration mode. To return to the default before the average was calculated, use the **no** form of this command.

call fallback instantaneous-value-weight *percent*
no call fallback instantaneous-value-weight

Syntax Description

percent	Instantaneous value weight, in expressed as a percentage. Range is from 0 to 100. The default is 66.

Command Default

66 percent

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)T	This command was introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)T	The PSTN Fallback feature and enhancements were implemented on the Cisco 7200 series and integrated into Cisco IOS Release 12.2(4)T.
12.2(4)T2	This command was implemented on the Cisco 7500 series.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

Probes that return the network congestion information are logged into the cache to determine whether the next call request is granted. When the network is regularly busy, the cache entries reflect the heavy traffic conditions. However, one probe may return with low traffic conditions, which is in contrast to normal conditions. All call requests received between the time of this probe and the next use this entry to determine call acceptance. These calls are allowed through the network, but before the next probe is sent and received, the normal, heavy traffic conditions must have returned. The calls sent through congest the network and cause worsen traffic conditions.

Use the **callfallbackinstantaneousvalueweight**command to gradually recover from heavy traffic network conditions. While the system waits for a call, probes update the cache. When a new probe is received, the *percentage* is set and indicates how much the system is to rely upon the new probe and the previous cache entry. If the *percentage*is set to 50 percent, the system enters a cache entry based upon an average from the new probe and the most recent entry in the cache. Call requests use this blended entry to determine acceptance. This allows the call fallback subsystem to keep conservative measures of network congestion.

The configured *percentate*applies to the new probe first. If the **callfallbackinstantaneousvalueweight**command is configured with the default *percentage*of 66 percent, the new probe is given a higher value to calculate the average for the new cache entry.

Examples

The following example specifies a fallback value weight of 50 percent:

```
Router(config)# call fallback instantaneous-value-weight 50
```

Related Commands

Command	Description
call fallback active	Enables a call request to fall back to alternate dial peers in case of network congestion.
show call fallback config	Displays the call fallback configuration.

call fallback jitter-probe dscp

To specify the differentiated services code point (DSCP) of the jitter-probe transmission, use the **callfallbackjitter-probedscp**command in global configuration mode. To disable this feature and restore the default value of jitter-probe precedence, use the **no** form of this command.

call fallback jitter-probe dscp *dscp-number*
no call fallback jitter-probe dscp

Syntax Description

dscp-number	DSCP value. Range is from 0 to 63.

Command Default

None

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.3(9)	This command was implemented in Cisco IOS Release 12.3(9).

Usage Guidelines

Network devices that support differentiated services (DiffServ) use a DSCP in the IP header to select a per-hop behavior (PHB) for a packet. Cisco implements queuing techniques that can base their PHB on the IP precedence or DSCP value in the IP header of a packet. On the basic of DSCP or IP precedence, traffic can be put into a particular service class. Packets within a service class are treated alike.

The **callfallbackjitter-probedscp** command allows you to set a DSCP for jitter-probe packets. The specified DSCP is stored, displayed, and passed in probing packets to the Service Assurance Agent (SAA). This command enables the router to reserve some bandwidth so that during network congestion some of the jitter-probe packets do not get dropped. This command avoids the conflict that occurs with traditional precedence bits.

The **callfallbackjitter-probedscp** command is mutually exclusive with the **callfallbackjitter-probeprecedence** command. Only one of these command can be enabled on the router. When the **callfallbackjitter-probedscp** command is configured, the precedence value is replaced with the DSCP value. The **nocallfallbackjitter-probedscp** command restores the default value for precedence.

Examples

The following example specifies the jitter-probe DSCP as 10. DSCP configuration replaces the set jitter-probe precedence value with the DSCP value.

```
call fallback jitter-probe dscp 10
```

The following configuration disables the DSCP value and restores the default value for precedence, which is set to 2:

```
no call fallback jitter-probe dscp
```

Related Commands

Command	Description
call fallback active	Enables a call request to fall back to alternate dial peers in case of network congestion.
call fallback jitter-probe num-packets	Specifies the number of packets in a jitter probe that are used to determine network conditions.
call fallback jitter-probe precedence	Specifies the priority of the jitter-probe transmission.
call fallback jitter-probe priority-queue	Assigns a priority queue for jitter-probe transmissions.
show call fallback config	Displays the call fallback configuration.

call fallback jitter-probe num-packets

To specify the number of packets in a jitter probe used to determine network conditions, use the **callfallbackjitterprobenumpackets** command in global configuration mode. To restore the default number of packets, use the **no** form of this command.

call fallback jitter-probe num-packets *number-of-packets*
no call fallback jitter-probe num-packets

Syntax Description

number-of-packets	Number of packets. Range is from 2 to 50. The default is 15.

Command Default

15 packets

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)T	This command was introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)T	The PSTN Fallback feature and enhancements were implemented on Cisco 7200 series and integrated into Cisco IOS Release 12.2(4)T.
12.2(4)T2	This command was implemented on the Cisco 7500 series.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

A jitter probe, consisting of 2 to 50 packets, details the conditions of the network. More than one packet is used by the probe to calculate an average of delay/loss or Calculated Planning Impairment Factor (ICPIF). After the packets return to the probe, the probe delivers the traffic information to the cache where it is logged for call acceptance/denial. Use the**callfallbackthresholddelayloss** or **callfallbackthresholdicpif** command to set the threshold parameters. The newly specified number of packets take effect only for new probes.

To get a more realistic estimate on the network congestion, increase the number of packets. If more probing packets are sent, better estimates of network conditions are obtained, but the bandwidth for other network operations is negatively affected. Use fewer packets when you need to maximize bandwidth.

Examples

The following example specifies 20 packets in a jitter probe:

```
Router(config)# call fallback jitter-probe num-packets 20
```

Related Commands

Command	Description
call fallback threshold icpif	Specifies the ICPIF threshold.
call fallback threshold delay loss	Specifies the call fallback threshold delay and loss values.

call fallback jitter-probe precedence

To specify the priority of the jitter-probe transmission, use the **callfallbackjitter-probeprecedencecommandin**global configuration mode. To restore the default priority, use the **no** form of this command.

call fallback jitter-probe precedence *precedence-value*
no call fallback jitter-probe precedence

Syntax Description

precedence-value	Jitter-probe precedence. Range is from 0 to 6. The default is 2.

Command Default

Enabled Value set to 2

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)T	This command was introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)T	The PSTN Fallback feature and enhancements were implemented on the Cisco 7200 series and integrated into Cisco IOS Release 12.2(4)T.
12.2(4)T2	This command was implemented on the Cisco 7500 series.
12.2(8)T	Support for the Cisco AS5850 is not included in this release.
12.2(11)T	This command was implemented on the Cisco AS5850.

Usage Guidelines

Every IP packet has a precedence header. Precedence is used by various queueing mechanisms in routers to determine the priority of traffic passing through the system.

Use the **callfallbackjitter-probeprecedence**command if there are different queueing mechanisms in your network. Enabling the **callfallbackjitter-probeprecedence**command sets the precedence for jitter probes to pass through your network.

If you require your probes to be sent and returned quickly, set the *precedence* to a low number (0 or 1): the lower the precedence, the higher the priority given.

The **callfallbackjitter-probeprecedence** command is mutually exclusive with the **callfallbackjitter-probedscp** command. Only one of these commands can be enabled on the router. Usually the **callfallbackjitter-probeprecedence** command is enabled. When the **callfallbackjitter-probedscp** command is configured, the precedence value is replaced by the DSCP value. To disable DSCP and restore the default jitter probe precedence value, use the **nocallfallbackjitter-probedscp** command.

Examples

The following example specifies a jitter-probe precedence of 5, or low priority.

```
call fallback jitter-probe precedence 5
```

The following configuration restores the default value for precedence:

```
no call fallback jitter-probe precedence
```

Related Commands

Command	Description
call fallback active	Enables a call request to fall back to alternate dial peers in case of network congestion.
call fallback jitter-probe dscp	Specifies the dscp of the jitter-probe transmission.
call fallback jitter-probe num-packets	Specifies the number of packets in a jitter probe that are used to determine network conditions.
call fallback jitter-probe priority-queue	Assigns a priority queue for jitter-probe transmissions.
show call fallback config	Displays the call fallback configuration.

call fallback jitter-probe priority-queue

To assign a priority queue for jitter-probe transmissions, use the**callfallbackjitter-probepriority-queuecommandin**global configuration mode. To return to the default state, use the **no** form of this command.

call fallback jitter-probe priority-queue
no call fallback jitter-probe priority-queue

Syntax Description

This command has no arguments or keywords.

Command Default

Disabled

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)T	This command was introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)T	The PSTN Fallback feature and enhancements were implemented on the Cisco 7200 series and integrated into Cisco IOS Release 12.2(4)T.
12.2(4)T2	This command was implemented on the Cisco 7500 series.
12.2(8)T	Support for the Cisco AS5850 is not included in this release.
12.2(11)T	This command was implemented on the Cisco AS5850.

Usage Guidelines

This command is applicable only if the queueing method used is IP Real-Time Transport Protocol (RTP) priority. This command is unnecessary when low latency queueing (LLQ) is used because these packets follow the priority queue path (or not) based on the LLQ classification criteria.

This command works by choosing between sending the probe on an odd or even Service Assurance Agent (SAA) port number. The SAA probe packets go out on randomly selected ports chosen from within the top end of the audio User Datagram Protocol (UDP) defined port range (16384 to 32767). The port pair (RTP Control Protocol [RTCP] port) is selected, and by default, SAA probes for call fallback use the RTCP port (odd) to avoid going into the priority queue, if enabled. If call fallback is configured to use the priority queue, the RTP port (even) is selected.

Examples

The following example specifies that a probe be sent to an SAA port:

```
Router(config)# call fallback jitter-probe priority-queue
```

Note In order for this command to have any effect on the probes, the IP priority queueing must be set for UDP voice ports numbered from 16384 to 32767.

Related Commands

Command	Description
call fallback active	Enables a call request to fall back to alternate dial peers in case of network congestion.
call fallback jitter-probe num-packets	Specifies the number of packets in a jitter probe that are used to determine network conditions.
call fallback jitter-probe precedence	Specifies the jitter-probe precedence.
ip rtp priority	Provides a strict priority queueing scheme for delay-sensitive data.
show call fallback config	Displays the call fallback configuration.

call fallback key-chain

To specify the use of message digest algorithm 5 (MD5) authentication for sending and receiving Service Assurance Agents (SAA) probes, use the **callfallbackkeychain** command in global configuration mode. To disable MD5, use the **no** form of this command.

call fallback key-chain *name-of-chain*
no call fallback key-chain *name-of-chain*

Syntax Description

name-of-chain	Name of the chain. This name is alphanumeric and case-sensitive text. There is no default value.

Command Default

MD5 authentication is not used.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)T	This command was introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)T	The PSTN Fallback feature and enhancements were implemented on the Cisco 7200 series and integrated into Cisco IOS Release 12.2(4)T.
12.2(4)T2	This command was implemented on the Cisco 7500 series.
12.2(8)T	Support for the Cisco AS5850 is not included in this release.
12.2(11)T	This command was implemented on the Cisco AS5850.

Usage Guidelines

This command is used to enable the SAA probe authentication using MD5. If MD5 authentication is used, the keys on the sender and receiver routers must match.

Examples

The following example specifies "sample" as the fallback key chain:

```
Router(config)# call fallback key-chain sample
```

Related Commands

Command	Description
call fallback active	Enables a call request to fall back to alternate dial peers in case of network congestion.
key chain	Enables authentication for routing protocols by identifying a group of authentication keys.
key-string	Specifies the authentication string for a key.

Command	Description
show call fallback config	Displays the call fallback configuration.

call fallback map address-list

To specify that the call fallback router keep a cache table by IP addresses of distances for several destination peers, use the **callfallbackmapaddresslist** command in global configuration mode. To restore the default values, use the **no** form of this command.

call fallback map *map* **target** *ip-address* **address-list** *ip-address1* **. . .** *ip-address7*
no call fallback map *map* **target** *ip-address* **address-list** *ip-address1* **. . .** *ip-address7*

Syntax Description

map	Fallback map. Range is from 1 to 16. There is no default.
target *ip address*	Target IP address.
ip address1 ... ip-address7	Lists the IP addresses that are kept in the cache table. The maximum number of IP addresses is seven.

Command Default

No call fallback maps are defined.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)T	This command was introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)T	The PSTN Fallback feature and enhancements were implemented on the Cisco 7200 series and integrated into Cisco IOS Release 12.2(4)T.
12.2(4)T2	This command was implemented on the Cisco 7500 series.
12.2(8)T	Support for the Cisco AS5850 is not included in this release.
12.2(11)T	This command was implemented on the Cisco AS5850.

Usage Guidelines

Use this command when several destination peers are in one common node.

Call fallback map setup allows the decongestion of traffic caused by a high volume of call probes sent across a network to query a large number of dial peers. One router/common node can keep the distances in a cache table of the numerous IP addresses/destination peers in a network. When the fallback is queried for network congestion to a particular IP address (that is, the common node), the map addresses are searched to find the target IP address. If a match is determined, the probes are sent to the target address rather than to the particular IP address.

In the figure below, the three routers (1, 2, and 3) keep the cache tables of distances for the destination peers behind them. When a call probe comes from somewhere in the IP cloud, the cache routers check their distance tables for the IP address/destination peer where the call probe is destined. This distance checking limits congestion on the networks behind these routers by directing the probe to the particular IP address and not to the entire network.

Examples

The following example specifies call fallback map address-list configurations for 172.32.10.1 and 172.46.10.1:

```
Router(config)# call fallback map 1 target 172.32.10.1 address-list 172.32.10.2 172.32.10.3
 172.32.10.4 172.32.10.5 172.32.10.6 172.32.10.7 172.32.10.8
Router(config)# call fallback map 2 target 172.46.10.1 address-list 172.46.10.2 172.46.10.3
 172.46.10.4 172.46.10.5 172.46.10.6 172.46.10.7 172.46.10.8
```

Related Commands

Command	Description
call fallback active	Enables a call request to fall back to alternate dial peers in case of network congestion.
call fallback map subnet	Specifies that the call fallback router keep a cache table by subnet addresses of distances for several destination peers that are sitting behind the router.
show call fallback config	Displays the call fallback configuration.

call fallback map subnet

To specify that the call fallback router keep a cache table by subnet addresses of distances for several destination peers, use the **callfallbackmapsubnet** command in global configuration mode. To restore the default values, use the **no** form of this command.

call fallback map *map* **target** *ip-address* **subnet** *ip-network netmask*
no call fallback map *map* **target** *ip-address* **subnet** *ip-network netmask*

Syntax Description

map	Fallback map. Range is from 1 to 16. There is no default.
target *ip address*	Target IP address.
subnet *ip network*	Subnet IP address.
netmask	Network mask number.

Command Default

No call fallback maps are defined.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)T	This command was introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)T	The PSTN Fallback feature and enhancements were implemented on the Cisco 7200 series and integrated into Cisco IOS Release 12.2(4)T.
12.2(4)T2	This command was implemented on the Cisco 7500 series.
12.2(8)T	Support for the Cisco AS5850 is not included in this release.
12.2(11)T	This command is supported on the Cisco AS5850 in this release.

Usage Guidelines

Use this command when several destination peers are in one common node.

Call fallback map setup allows the decongestion of traffic caused by a high volume of call probes sent across a network to query a large number of dial peers. One router/common node can keep the distances in a cache table of the numerous IP addresses within a subnet (destination peers) in a network. When the fallback is queried for network congestion to a particular IP address (that is, the common node), the map addresses are searched to find the target IP address. If a match is determined, the probes are sent to the target address rather than to the particular IP address.

In the figure below, the three routers (1, 2, and 3) keep the cache tables of distances for the destination peers behind them. When a call probe comes from somewhere in the IP cloud, the cache routers check their distance tables for the subnet address/destination peer where the call probe is destined. This distance checking limits congestion on the networks behind these routers by directing the probe to the particular subnet address and not to the entire network.

Examples

The following examples specify the **callfallbackmapsubnet** configuration for two different IP addresses:

```
Router(config)#
call fallback map 1 target 209.165.201.225 subnet
209.165.201.224 255.255.255.224
Router(config)#
call fallback map 2 target 209.165.202.225 subnet
209.165.202.224 255.255.255.224
```

Related Commands

Command	Description
call fallback active	Enables a call request to fall back to alternate dial peers in case of network congestion.
call fallback map address-list	Specifies that the call fallback router keep a cache table by IP addresses of distances for several destination peers that are sitting behind the router.
show call fallback config	Displays the call fallback configuration.

call fallback monitor

To enable the monitoring of destinations without call fallback to alternate dial peers, use the **callfallbackmonitor**commandinglobal configuration mode. To disable monitoring without fallback, use the **no** form of this command.

call fallback monitor
no call fallback monitor

Syntax Description

This command has no arguments or keywords.

Command Default

Disabled

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)T	This command was introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)T	The PSTN Fallback feature and enhancements were introduced on the Cisco 7200 series and integrated into Cisco IOS Release 12.2(4)T.
12.2(4)T2	This command was implemented on the Cisco 7500 series.
12.2(8)T	Support for the Cisco AS5850 is not included in this release.
12.2(11)T	This command was implemented on the Cisco AS5850.

Usage Guidelines

The **callfallbackmonitor**command is used as a statistics collector of network conditions based upon probes (detailing network traffic) and connected calls. There is no H.323 call checking/rejecting as with the **callfallbackactive** command. All call requests are granted regardless of network traffic conditions.

Configure the**callfallbackthresholddelayloss** or **callfallbackthresholdicpif** command to set threshold parameters. The thresholds are ignored, but for statistics collecting, configuring one of the thresholds allows you to monitor cache entries for either delay/loss or Calculated Planning Impairment Factor (ICPIF) values.

Examples

The following example enables the **callfallbackmonitor**command:

```
Router(config)#
call fallback monitor
```

Related Commands

Command	Description
call fallback active	Enables a call request to fall back to alternate dial peers in case of network congestion.

Command	Description
call fallback threshold delay loss	Specifies that the call fallback threshold use only packet delay and loss values.
call fallback threshold icpif	Specifies that call fallback use the ICPIF threshold.
show call fallback config	Displays the call fallback configuration.

call fallback probe-timeout

To set the timeout for a Service Assurance Agent (SAA) probe for call fallback purposes, use the **callfallbackprobetimeoutcommand** in global configuration mode. To restore the default value, use the**no**form of this command.

call fallback probe-timeout *seconds*
no call fallback probe-timeout

Syntax Description

seconds	Interval, in seconds. Range is from 1 to 2147483. The default is 30.

Command Default

30 seconds

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)T	This command was introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)T	The PSTN Fallback feature and enhancements were implemented on the Cisco 7200 series and integrated into Cisco IOS Release 12.2(4)T.
12.2(4)T2	This command was implemented on the Cisco 7500 series.
12.2(8)T	Support for the Cisco AS5850 is not included in this release.
12.2(11)T	This command was implemented on the Cisco AS5850.

Usage Guidelines

SAA probes collect network traffic information based upon configured delay and loss or Calculated Planning Impairment Factor (ICPIF) values and report this information to the cache for call request determination. Use the**callfallbackthresholddelayloss** or **callfallbackthresholdicpif** command to set the threshold parameters.

When the probe timeout expires, a new probe is sent to collect network statistics. To reduce the bandwidth taken up by the probes, increase the probe - timeout interval (*seconds*). Probes do not have a great effect upon bandwidth unless several thousand destinations are involved. If this is the case in your network, use a longer timeout. If you need more network traffic information, and bandwidth is not an issue, use a lower timeout. The default interval, 30 seconds, is a low timeout.

When the **callfallbackcachetimeout** command is configured or expires, new probes are initiated for data collection.

Examples

The following example configures a 120-second interval:

```
Router(config)# call fallback probe-timeout 120
```

Related Commands

Command	Description
call fallback active	Enables a call request to fall back to alternate dial peers in case of network congestion.
call fallback cache-timeout	Specifies the time after which the cache entries of network conditions are purged.
call fallback threshold delay loss	Specifies that the call fallback threshold use only packet delay and loss values.
call fallback threshold icpif	Specifies that call fallback use the ICPIF threshold.
show call fallback config	Displays the call fallback configuration.

call fallback reject-cause-code

To enable a specific call fallback reject cause code in case of network congestion, use the **callfallbackrejectcausecode**command in global configuration mode. To reset the code to the default of 49, use the **no** form of this command.

call fallback reject-cause-code *number*
no call fallback reject-cause-code

Syntax Description

number	Specifies the cause code as defined in the International Telecommunication Union (ITU) standard Q.850 except the code for normal call clearing, which is code 16. The default is 49. See the table below for ITU cause-code numbers.

Command Default

49 (quality of service is unavailable)

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	The PSTN Fallback feature and enhancements were implemented on Cisco 7200 series and integrated into Cisco IOS Release 12.2(4)T.
12.2(4)T2	This command was implemented on the Cisco 7500 series.

Usage Guidelines

Enabling the **callfallbackrejectcausecode**commanddetermines the code to display when calls are rejected because of probing of network conditions.

Note Connected calls are not affected by this command.

Table 10: ITU cause codes and their associated display message and meanings.

Cause Code	Displayed Message	Meaning
1	Unallocated (unassigned) number	Indicates that the called party cannot be reached because, although the called party number is in a valid format, it is not currently allocated (assigned).
2	No route to specified transit network (national use)	Indicates that the equipment that is sending this code has received a request to route the call through a particular transit network that it does not recognize. The equipment that is sending this code does not recognize the transit network either because the transit network does not exist or because that particular transit network, although it does exist, does not serve the equipment that is sending this cause. This code is supported on a network-dependent basis.

Cause Code	Displayed Message	Meaning
3	No route to destination	Indicates that the called party cannot be reached because the network through which the call has been routed does not serve the destination desired. This code is supported on a network-dependent basis.
4	Send special information tone	Indicates that the called party cannot be reached for reasons that are of a long-term nature and that the special information tone should be returned to the calling party.
5	Misdialed trunk prefix (national use)	Indicates the erroneous inclusion of a trunk prefix in the called party number.
6	Channel unacceptable	Indicates that the channel most recently identified is not acceptable to the sending entity for use in this call.
7	Call awarded and being delivered in an established channel	Indicates that the user has been awarded the incoming call and that the incoming call is being connected to a channel that is already established to that user for similar calls (for example, packet-mode X.25 virtual calls).
8	Preemption	Indicates that the call is being preempted.
9	Preemption - circuit reserved for reuse	Indicates that the call is being preempted and that the circuit is reserved for reuse by the preempting exchange.
16	Normal call clearing	Indicates that the call is being cleared because one of the users involved in the call has requested that the call be cleared. Under normal situations, the source of this code is not the network.
17	User busy	Indicates that the called party is unable to accept another call. The user busy code may be generated by the called user or by the network. If the called user generates the user busy code, it is noted that the user equipment is compatible with the call.
18	No user responding	Indicates when a called party does not respond to a call establishment message with either an alerting or a connect indication within the prescribed period of time allocated.
19	No answer from user (user alerted)	Indicates when the called party has been alerted but does not respond with a connect indication within a prescribed period of time. **Note** This code is not necessarily generated by ITU standard Q.931 procedures but may be generated by internal network timers.
20	Subscriber absent	Indicates when a mobile station has logged off, when radio contact is not obtained with a mobile station, or when a personal telecommunication user is temporarily not addressable at any user-network interface.
21	Call rejected	Indicates that the equipment that is sending this code does not want to accept this call although it could have accepted the call because the equipment that is sending this code is neither busy nor incompatible. The network may also generate this code, indicating that the call was cleared because of a supplementary service constraint. The diagnostic field may contain additional information about the supplementary service and reason for rejection.

Cause Code	Displayed Message	Meaning
22	Number changed	Indicates when the called-party number indicated by the calling party is no longer assigned. The new called-party number may be included in the diagnostic field. If a network does not support this code, codeNo. 1, an unallocated (unassigned) number, shall be used.
26	Non-selected user clearing	Indicates that the user has not been sent the incoming call.
27	Destination out of order	Indicates that the destination indicated by the user cannot be reached because the interface to the destination is not functioning correctly. The term "not functioning correctly" indicates that a signaling message was unable to be delivered to the remote party; for example, a physical layer or data link layer failure at the remote party, or the equipment of the user is offline.
28	Invalid number format (address incomplete)	Indicates that the called party cannot be reached because the called party number is not in a valid format or is not complete.
29	Facility rejected	Indicates when a supplementary service requested by the user cannot be provided by the network.
30	Response to STATUS ENQUIRY	Indicates when the reason for generating the STATUS message was the prior receipt of a STATUS ENQUIRY message.
31	Normal, unspecified	Reports a normal event only when no other code in the normal class applies.
34	No circuit/channel available	Indicates that no appropriate circuit or channel is available to handle the call.
38	Network out of order	Indicates that the network is not functioning correctly and that the condition is likely to last a relatively long period of time; for example, immediately reattempting the call is not likely to be successful.
39	Permanent frame mode connection out-of-service	Indicates in a STATUS message that a permanently established frame mode connection is out-of-service (for example, due to equipment or section failure) (see the ITU standard, Annex A/Q.933).
40	Permanent frame mode connection operational	Indicates in a STATUS message to indicate that a permanently established frame mode connection is operational and capable of carrying user information (see the ITU standard, Annex A/Q.933).
41	Temporary failure	Indicates that the network is not functioning correctly and that the condition is not likely to last a long period of time; for example, the user may want to try another call attempt almost immediately.
42	Switching equipment congestion	Indicates that the switching equipment that is generating this code is experiencing a period of high traffic.
43	Access information discarded	Indicates that the network could not deliver access information to the remote user as requested, that is, user-to-user information, low layer compatibility, high layer compatibility, or subaddress, as indicated in the diagnostic. It is noted that the particular type of access information discarded is optionally included in the diagnostic.

Cause Code	Displayed Message	Meaning
44	Requested circuit/channel not available	Indicates when the circuit or channel indicated by the requesting entity cannot be provided by the other side of the interface.
46	Precedence call blocked	Indicates that there are no preemptable circuits or that the called user is busy with a call of an equal or higher preemptable level.
47	Resource unavailable, unspecified	Reports a resource-unavailable event only when no other cause in the resource-unavailable class applies.
49	Quality of service not available	Reports that the requested quality of service, as defined in ITU recommendation X.213, cannot be provided (for example, throughput or transit delay cannot be supported).
50	Requested facility not subscribed	Indicates that the user has requested a supplementary service that is implemented by the equipment that generated this cause but that the user is not authorized to use this service.
53	Outgoing calls barred within CUG	Indicates that, although the calling party is a member of the closed user group (CUG) for the outgoing CUG call, outgoing calls are not allowed for this member of the CUG.
55	Incoming calls barred within CUG	Indicates that, although the called party is a member of the CUG for the incoming CUG call, incoming calls are not allowed for this member of the CUG.
57	Bearer capability not authorized	Indicates that the user has requested a bearer capability that is implemented by the equipment that generated this cause but that the user is not authorized to use this capability.
58	Bearer capability not presently available	Indicates that the user has requested a bearer capability that is implemented by the equipment that generated this cause but that is not available at this time.
62	Inconsistency in designated outgoing access information and subscriber class	Indicates that there is an inconsistency in the designated outgoing access information and subscriber class.
63	Service or option not available, unspecified	Reports a service or option not available event only when no other cause in the service or option not available class applies.
65	Bearer capability not implemented	Indicates that the equipment that is sending this code does not support the bearer capability requested.
66	Channel type not implemented	Indicates that the equipment that is sending this code does not support the channel type requested.
69	Requested facility not implemented	Indicates that the equipment that is sending this code does not support the requested supplementary service.
70	Only restricted digital information bearer capability is available (national use)	Indicates that the calling party has requested an unrestricted bearer service but that the equipment that is sending this cause supports only the restricted version of the requested bearer capability.

Cause Code	Displayed Message	Meaning
79	Service or option not implemented, unspecified	Reports a service or option not implemented event only when no other code in the service or option not implemented class applies.
81	Invalid call reference value	Indicates that the equipment that is sending this code has received a message with a call reference that is not currently in use on the user-network interface.
82	Identified channel does not exist	Indicates that the equipment that is sending this code has received a request to use a channel not activated on the interface for a call. For example, if a user has subscribed to those channels on a PRI numbered from 1 to 12 and the user equipment or the network attempts to use channels 13 through 23, this cause is generated.
83	A suspended call exists, but this call identity does not	Indicates that a call resume has been attempted with a call identity that differs from that in use for any suspended calls.
84	Call identity in use	Indicates that the network has received a call suspended request that contains a call identity (including the null call identity) that is already in use for a suspended call within the domain of interfaces over which the call might be resumed.
85	No call suspended	Indicates that the network has received a call resume request that contains a call identity information element that does not indicate any suspended call within the domain of interfaces over which calls may be resumed.
86	Call having the requested call identity has been cleared	Indicates that the network has received a call resume request that contains a call identity information element that indicates a suspended call that has in the meantime been cleared while suspended (either by network timeout or by the remote user).
87	User not member of CUG	Indicates that the called user for the incoming CUG call is not a member of the specified CUG or that the calling user is an ordinary subscriber that is calling a CUG subscriber.
88	Incompatible destination	Indicates that the equipment that is sending this code has received a request to establish a call that has low layer compatibility, high layer compatibility, or other compatibility attributes (for example, data rate) that cannot be accommodated.
90	Non-existent CUG	Indicates that the specified CUG does not exist.
91	Invalid transit network selection (national use)	Indicates that a transit network identification was received that is of an incorrect format as defined in ITU standard Annex C/Q.931.
95	Invalid message, unspecified	Reports an invalid message event only when no other code in the invalid message class applies.
96	Mandatory information element is missing	Indicates that the equipment that is sending this code has received a message that is missing an information element that must be present in the message before that message can be processed.
97	Message type non-existent or not implemented	Indicates that the equipment that is sending this code has received a message with a message type that it does not recognize because this is a message not defined or defined but not implemented by the equipment that is sending this cause.

Cause Code	Displayed Message	Meaning
98	Message not compatible with call state or message type non-existent or not implemented	Indicates that the equipment that is sending this code has received a message that the procedures do not indicate as a permissible message to receive while in the call state, or that a STATUS message that indicates an incompatible call state was received.
99	Information element/parameter non-existent or not implemented	Indicates that the equipment that is sending this code has received a message that includes information elements or parameters not recognized because the information element identifiers or parameter names are not defined or are defined but not implemented by the equipment sending the code. This code indicates that the information elements or parameters were discarded. However, the information element is not required to be present in the message for the equipment that is sending the code to process the message.
100	Invalid information element contents	Indicates that the equipment that is sending this code has received an information element that it has implemented; however, one or more fields in the information element are coded in a way that has not been implemented by the equipment that is sending this code.
101	Message not compatible with call state	Indicates that a message has been received that is incompatible with the call state.
102	Recovery on timer expired	Indicates that a procedure has been initiated by the expiration of a timer in association with error-handling procedures.
103	Parameter non-existent or not implemented - passed on	Indicates that the equipment that is sending this code has received a message that includes parameters not recognized because the parameters are not defined or are defined but not implemented by the equipment that is sending the code. The code indicates that the parameters were ignored. In addition, if the equipment that is sending this code is an intermediate point, this code indicates that the parameters were passed on unchanged.
110	Message with unrecognized parameter discarded	Indicates that the equipment that is sending this code has discarded a received message that includes a parameter that is not recognized.
111	Protocol error, unspecified	Reports a protocol error event only when no other code in the protocol error class applies.
127	Interworking, unspecified	Indicates that there has been interworking with a network that does not provide codes for actions it takes. Thus, the precise code for a message that is being sent cannot be ascertained.

Examples

The following example enables the **callfallbackrejectcausecode**command and specifies cause code 34:

```
call fallback reject-cause-code 34
```

Related Commands

Command	Description
call fallback cache-size	Specifies the call fallback cache size for network traffic probe entries.
call fallback cache-timeout	Specifies the time after which the cache entries of network conditions are purged.
call fallback instantaneous-value-weight	Specifies that the call fallback subsystem take an average from the last two cache entries for call requests.
call fallback jitter-probe num-packets	Specifies the number of packets in a jitter probe that are used to determine network conditions.
call fallback jitter-probe precedence	Specifies the priority of the jitter - probe transmission.
call fallback jitter-probe priority-queue	Assigns a priority queue for jitter-probe transmissions.
call fallback key-chain	Specifies MD5 authentication for sending and receiving SAA probes.
call fallback map address-list	Specifies that the call fallback router keep a cache table by IP addresses of distances for several destination peers that are sitting behind the router.
call fallback map subnet	Specifies that the call fallback router keep a cache table by subnet addresses of distances for several destination peers that are sitting behind the router.
call fallback probe-timeout	Sets the timeout for an SAA probe for call fallback purposes.
call fallback threshold delay loss	Specifies that the call fallback threshold use only packet delay and loss values.
call fallback threshold icpif	Specifies that call fallback use the ICPIF threshold.
show call fallback config	Displays the call fallback configuration.

call fallback threshold delay loss

To specify that the call fallback threshold use only packet delay and loss values, use the **callfallbackthresholddelaylosscommandin**global configuration mode. To restore the default value, use the **no** form of this command.

call fallback threshold delay *milliseconds* **loss** *percent*
no call fallback threshold delay *milliseconds* **loss** *percent*

Syntax Description

milliseconds	The delay value, in milliseconds (ms). Range is from 1 to 2147483647. There is no default value.
percent	The loss value, expressed as a percentage. The valid range is from 0 to 100. There is no default value.

Command Default

None

Command Modes

Global configuration (config)

Command History

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

Usage Guidelines

During times of heavy voice traffic, two parties in a conversation may notice a significant delay in transmission or hear only part of a conversation because of voice-packet loss.

Use the **callfallbackthresholddelayloss**command to configure parameters for voice quality. Lower values of delay and loss allow higher quality of voice. Call requests match the network information in the cache with the configured thresholds of delay and loss.

The amount of delay set by the **callfallbackthresholddelayloss** command should not be more than half the amount of the time-to-wait value set by the **callfallbackwait-timeout** command; otherwise the threshold delay will not work correctly. Because the default value of the **callfallbackwait-timeout** command is set to 300 ms, the user can configure a delay of up to 150 ms for the **callfallbackthresholddelayloss** command. If the user wants to configure a higher threshold, the time-to-wait delay has to be increased from its default (300 ms) using the **callfallbackwait-timeout**command.

Note The delay configured by the **callfallbackthresholddelayloss** command corresponds to a one-way delay, whereas the time-to-wait period configured by the **callfallbackwait-timeout** command corresponds to a round-trip delay.

If you enable the **callfallbackactive**command, the call fallback subsystem uses the last cache entry compared with the configured delay/loss threshold to determine whether the call is connected or denied. If you enable the **callfallbackmonitor**command, all calls are connected, regardless of the configured threshold or voice quality. In this case, configuring the **callfallbackthresholddelayloss**command allows you to collect network statistics for further tracking.

Note

The **callfallbackthresholddelayloss**command differs from the call fallback threshold icpif command because the**callfallbackthresholddelayloss**commanduses only packet delay and loss parameters, and the call fallback threshold icpif command uses packet delay and loss parameters plus other International Telecommunication Union (ITU) G.113 factors to gather impairment information.

Setting this command does not affect bandwidth. Available bandwidth for call requests is determined by the call fallback subsystem using probes. The number of probes on the network affects bandwidth.

Examples

The following example configures a threshold delay of 20 ms and a threshold loss of 50 percent:

```
Router(config)#

call fallback threshold delay 20 loss 50
```

Related Commands

Command	Description
call fallback active	Enables a call request to fall back to alternate dial peers in case of network congestion.
call fallback monitor	Enable the monitoring of destinations without call fallback to alternate dial peers.
call fallback threshold icpif	Specifies the ICPIF threshold.
call fallback wait-timeout	Specifies the time to wait for a response to a probe.
show call fallback config	Displays the call fallback configuration.

call fallback threshold icpif

To specify that call fallback use the Calculated Planning Impairment Factor (ICPIF) threshold, use the **callfallbackthresholdicpif**command in global configuration mode. To restore the default value, use the **no** form of this command.

call fallback threshold icpif *threshold-value*
no call fallback threshold icpif

Syntax Description

threshold-value	Threshold value. Range is from 0 to 34. The default is 5.

Command Default

5

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)T	This command was introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)T	The PSTN Fallback feature and enhancements were introduced on the Cisco 7200 series routers and integrated into Cisco IOS Release 12.2(4)T.
12.2(4)T2	This command was implemented on the Cisco 7500 series.
12.2(8)T	Support for the Cisco AS5850 is not included in this release.
12.2(11)T	This command was implemented on the Cisco AS5850.

Usage Guidelines

During times of heavy voice traffic, the parties in a conversation may notice a significant delay in transmission or hear only part of a conversation because of voice-packet loss.

Use the **callfallbackthresholdicpif**command to configure parameters for voice quality. A low ICPIF value allows for higher quality of voice. Call requests match the network information in the cache with the configured ICPIF threshold. If you enable the **callfallbackactive**command, the call fallback subsystem uses the last cache entry compared with the configured ICPIF threshold to determine whether the call is connected or denied. If you enable **thecallfallbackmonitor**command, all calls are connected regardless of the configured threshold or voice quality. In this case, configuring the **callfallbackthresholdicpif**command allows you to collect network statistics for further tracking.

A lower ICPIF value tolerates less delay and loss of voice packets (according to ICPIF calculations). Use lower values for higher quality of voice. Configuring a value of 34 equates to 100 percent packet loss.

The ICPIF is calculated and used according to the International Telecommunication Union (ITU) G.113 specification.

Note The **callfallbackthresholddelayloss**command differs from the call fallback threshold icpif command because the **callfallbackthresholddelayloss**command uses only packet delay and loss parameters, while the call fallback threshold icpif command uses packet delay and loss parameters plus other ITU G.113 factors to gather impairment information.

Setting this command does not affect bandwidth. Available bandwidth for call requests is determined by the call fallback subsystem using probes. The number of probes on the network affects bandwidth.

Examples

The following example sets the **ICPIFthreshold**to 20:

```
Router(config)#
call fallback threshold icpif 20
```

Related Commands

Command	Description
call fallback active	Enables a call request to fall back to alternate dial peers in case of network congestion.
call fallback monitor	Enables the monitoring of destinations without call fallback to alternate dial peers.
call fallback threshold delay loss	Specifies the call fallback threshold delay and loss values.
show call fallback config	Displays the call fallback configuration.

call fallback wait-timeout

To modify the time to wait for a response to a probe, use the **callfallbackwait-timeout**command in global configuration mode. To return to the default value, use the **no** form of this command.

call fallback wait-timeout *milliseconds*
no call fallback wait-timeout *milliseconds*

Syntax Description

milliseconds	The time-to-wait value in milliseconds (ms). The range is 100 to 3000 milliseconds.

Command Default

300 milliseconds

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(15)T9	This command was introduced.

Usage Guidelines

This command is enabled by default. The time to wait for a response to a probe is set to 300 ms. This command allows the user to modify the amount of time to wait for a response to a probe. The *milliseconds* argument allows the user to configure a time-to-wait value from 100 ms and 3000 ms. A user that has a higher-latency network may want to increase the value of the default timer.

The time-to-wait period set by the **callfallbackwait-timeout** command should always be greater than or equal to twice the amount of the threshold delay time set by the **callfallbackthresholddelayloss** command; otherwise the probe will fail.

Note The delay configured by the **callfallbackthresholddelayloss** command corresponds to a one-way delay, whereas the time-to-wait period configured by **callfallbackwait-timeout** command corresponds to a round-trip delay. The threshold delay time should be set at half the value of the time-to-wait value.

Examples

The following example sets the amount of time to wait for a response to a probe to 200 ms:

```
call fallback wait-timeout 200
```

Related Commands

Command	Description
call fallback threshold delay loss	Specifies the call fallback threshold delay and loss values.

call filter match-list

To enter the call filter match list configuration mode and create a call filter match list for debugging voice calls, use the **callfiltermatch-list**command in global configuration mode. To remove the filter, use the **no** form of this command.

call filter match-list *number* {**voice** | **gatekeeper**}
no call filter match-list *number* {**voice** | **gatekeeper**}

Syntax Description

number	Numeric label that uniquely identifies the match list. The range is 1 to 16.
voice	Sets the conditions for filtering voice call debugging.
gatekeeper	Defines the conditions on the gatekeeper. The **gatekeeper** keyword is available only if the Cisco IOS image contains the gatekeeper debug filter functionality or a combination of gateway and gatekeeper debug filter functionality.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(4)T	This command was introduced.
15.0(1)M	This command was modified in a release earlier than Cisco IOS Release 15.0(1)M. The **gatekeeper** keyword was added.

Usage Guidelines

After the conditions are set with this command, use the **debugconditionmatch-list** command in privileged EXEC mode to get the filtered debug output and debug voice calls.

Examples

The following example shows that the call filter match list designated as list 1 filters the debug output for an incoming calling number matching 8288807, an incoming called number matching 6560729, and on incoming port 7/0:D:

```
call filter match-list 1 voice
 incoming calling-number 8288807
 incoming called-number 6560729
 incoming port 7/0:D
```

Related Commands

Command	Description
debug condition match-list	Runs a filtered debug on a voice call.
show call filter match-list	Displays call filter match lists.

call forward all

To define a a feature code for a Feature Access Code (FAC) to access Call Forward All (CFA) on an analog phone, use the **callforwardall**command in STC application feature access-code configuration mode. To return the code to its default, use the **no** form of this command.

call forward all *keypad-character*
no call forward all

Syntax Description

keypad-character	Character string that can be dialed on a telephone keypad (0-9, *, #). Default: 1. Before Cisco IOS Release 12.4(20)YA, this is a single character. In Cisco IOS Release 12.5(20)YA and later releases, the string can be any of the following: • A single character (0-9, *, #) • Two digits (00-99) • Two to four characters (0-9, *, #) and the leading or ending character must be an asterisk (*) or number sign (#) In Cisco IOS Release 15.0(1)M and later releases, the string can also be any of the following: • Three digits (000-999) • Four digits (0000-9999)

Command Default

The default value of the feature code for CFA is 1.

Command Modes

STC application feature access-code configuration (config-stcapp-fac)

Command History

Release	Modification
12.4(2)T	This command was introduced.
12.4(20)YA	This command was modified. The length of the *keypad-character* argument was changed to 1 to 4 characters.
12.4(22)T	This command was integrated into Cisco IOS Release 12.4(22)T.
15.0(1)M	This command was modified.

Usage Guidelines

This command changes the value of the feature code for Call Forward All from the default (1) to the specified value.

In Cisco IOS Release 12.4(20)YA and later releases, if the length of the *keypad-character* argument is at least two characters and the leading or ending character of the string is an asterisk (*) or a number sign (#), phone users are not required to dial a prefix to access this feature. Typically, phone users dial a special feature access code (FAC) consisting of a prefix plus a feature code, for example **2. If the feature code is 78#, the phone user dials only 78#, without the FAC prefix, to access the corresponding feature.

In Cisco IOS Release 15.0(1)M and later releases, if the length of the keypad-character argument is three or four digits, phone users are not required to dial a prefix or any special characters to access this feature. Typically, phone users dial a special feature access code (FAC) consisting of a prefix plus a feature code, for example **2. If the feature code is 788, the phone user dials only 788, without the FAC prefix, to access the corresponding feature.

In Cisco IOS Release 12.4(20)YA and later releases, if you attempt to configure this command with a value that is already configured for another FAC, for a speed-dial code, or for the Redial FSD, you receive a message. If you configure a duplicate code, the system implements the first matching feature in the order of precedence shown in the output of the **showstcappfeaturecodes** command.

In Cisco IOS Release 12.4(20)YA and later releases, if you attempt to configure this command with a value that precludes or is precluded by another FAC, by a speed-dial code, or by the Redial FSD, you receive a message. If you configure a feature code to a value that precludes or is precluded by another code, the system always executes the call feature with the shortest code and ignores the longer code. For example, #1 will always preclude #12 and #123. You must configure a new value for the precluded code in order to enable phone user access to that feature.

To display a list of all FACs, use the **showstcappfeaturecodes** command.

Examples

The following example shows how to change the value of the feature code for Call Forward All from the default (1). This configuration also changes the value of the prefix for all FACs from the default (**) to ##. With this configuration, a phone user must press ##3 on the keypad and then dial a target number, to forward all incoming calls to the target number.

```
Router(config)# stcapp feature access-code
Router(config-stcapp-fac)# prefix ##
Router(config-stcapp-fac)# call forward all 3
Router(config-stcapp-fac)# exit
```

The following example shows how to configure all-numeric three or four digit flexible feature access codes so that users are not required to dial a prefix or special characters:

```
VG224(config-stcapp-fac)# call forward all 111
do not use prefix. call forward all is 111
```

Related Commands

Command	Description
call-forward all	Configures call forwarding so that all incoming calls to a particular directory number are forwarded to another directory number.
call forward cancel	Defines a feature code for a feature access code (FAC) to cancel the call-forward-all condition.
call forward to voicemail	Configures call forwarding to voicemail so that all incoming calls are forwarded to voicemail.
prefix (stcapp-fac)	Defines the prefix for feature access codes (FACs).
show stcapp feature codes	Displays all feature access codes (FACs).

Command	Description
stcapp feature access-code	Enables feature access codes (FACs) and enters STC application feature access-code configuration mode for changing values of the prefix and features codes from the default.

call forward cancel

To define a a feature code for a Feature Access Code (FAC) to access Call Forward All Cancel, use the **callforwardcancel**command in STC application feature access-code configuration mode. To return the feature code to its default, use the **no** form of this command.

call forward cancel *keypad-character*
no call forward cancel

Syntax Description

keypad-character	Character string that can be dialed on a telephone keypad (0-9, *, #). Default: 2. Before Cisco IOS Release 12.4(20)YA, this is a single character. In Cisco IOS Release 12.4(20)YA and later releases, the string can be any of the following: • A single character (0-9, *, #) • Two digits (00-99) • Two to four characters (0-9, *, #) and the leading or ending character must be an asterisk (*) or number sign (#) In Cisco IOS Release 15.0(1)M and later releases, the string can also be any of the following: • Three digits (000-999) • Four digits (0000-9999)

Command Default

The default value of the feature code is 2.

Command Modes

STC application feature access-code configuration (config-stcapp-fac)

Command History

Release	Modification
12.4(2)T	This command was introduced.
12.4(20)YA	The length of the *keypad-character* argument was changed to 1 to 4 characters.
12.4(22)T	This command was integrated into Cisco IOS Release 12.4(22)T.
15.0(1)M	This command was modified.

Usage Guidelines

This command changes the value of the feature code for Call Forward All Cancel from the default (2) to the specified value.

In Cisco IOS Release 12.4(20)YA and later releases, if the length of the *keypad-character* argument is at least two characters and the leading or ending character of the string is an asterisk (*) or a number sign (#), phone users are not required to dial a prefix to access this feature. Typically, phone users dial a special feature access code (FAC) consisting of a prefix plus a feature code, for example **2. If the feature code is 78#, the phone user dials only 78#, without the FAC prefix, to access the corresponding feature.

In Cisco IOS Release 15.0(1)M and later releases, if the length of the keypad-character argument is three or four digits, phone users are not required to dial a prefix or any special characters to access this feature. Typically, phone users dial a special feature access code (FAC) consisting of a prefix plus a feature code, for example **2. If the feature code is 788, the phone user dials only 788, without the FAC prefix, to access the corresponding feature.

In Cisco IOS Release 12.4(20)YA and later releases, if you attempt to configure this command with a value that is already configured for another FAC, for a speed-dial code, or for the Redial FSD, you receive a message. If you configure a duplicate code, the system implements the first matching feature in the order of precedence shown in the output of the **showstcappfeaturecodes** command.

In Cisco IOS Release 12.4(20)YA and later releases, if you attempt to configure this command with a value that precludes or is precluded by another FAC, by a speed-dial code, or by the Redial FSD, you receive a message. If you configure a feature code to a value that precludes or is precluded by another code, the system always executes the call feature with the shortest code and ignores the longer code. For example, #1 will always preclude #12 and #123. You must configure a new value for the precluded code in order to enable phone user access to that feature.

To display a list of all FACs, use the **showstcappfeaturecodes** command.

Note To disable call-forward-all on a particular directory number associated with SCCP endpoints connected to Cisco Unified CME through an analog voice gateway, use the **nocall-forwardall**command in ephone-dn or ephone-dn-template configuration mode.

Examples

The following example shows how to change the value of the feature code for Call Forward Cancel from the default (2). This configuration also changes the value of the prefix for all FACs from the default (**) to ##. With this configuration, a phone user must press ##3 on the phone keypad to cancel all-call forwarding.

```
Router(config)# stcapp feature access-code
Router(config-stcapp-fac)# prefix ##
Router(config-stcapp-fac)# call forward cancel 3
Router(config-stcapp-fac)# exit
```

Related Commands

Command	Description
call forward all	Defines the feature code in the feature access code (FAC) for forwarding all calls.
call-forward all	Configures call forwarding so that all incoming calls to a particular directory number are forwarded to another directory number.
prefix (stcapp-fac)	Defines the prefix for feature access codes (FACs).
show stcapp feature codes	Displays all feature access codes (FACs).
stcapp feature access-code	Enables feature access codes (FACs) and enters STC application feature access-code configuration mode for changing values of the prefix and features codes from the default.

call-forward-to-voicemail

To configure forwarding of calls to voicemail so that all incoming calls to a directory number are forwarded to voicemail, use the **forward-to-voicemail**command. The **stcappfeatureaccess-code** command must be enabled on the Cisco voice gateway. To disable call forwarding, use the**no** form of this command.

forward-to-voicemail *forward-to-voicemail-code*
no forward-to-voicemail

Syntax Description

forward-to-voicemail-code	Default prefix and code is **7.
keypad-character	In Cisco IOS Release 15.0(1)M and later releases, the string can be either of the following: • Three digits (000-999) • Four digits (0000-9999)

Command Default

Call forwarding to voicemail is not set.

Command Modes

STC application feature access-code configuration (config-stcapp-fac).

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)T	Cisco Unified CME 4.0(3)	This command was introduced.
15.0(1)M	--	This command was modified. The default user behavior of the feature access code was modified.

Usage Guidelines

In Cisco IOS Release 15.0(1)M and later releases, if the length of the keypad-character argument is three or four digits, phone users are not required to dial a prefix or any special characters to access this feature. Typically, phone users dial a special feature access code (FAC) consisting of a prefix plus a feature code, for example **2. If the feature code is 788, the phone user dials only 788, without the FAC prefix, to access the corresponding feature.

The FAC for forward-to-voicemail follows the same rules as for other FAC, such as **callforwardall,** in terms of allowable string as its FAC code.

Examples

The following example show how to configure forward-to-voicemail using a four digit code:

```
VG224(config-stcapp-fac)# forward-to-voicemail 1234

do not use prefix. forward-to-voicemail is 1234
```

Related Commands

Command	Description
call-forward all	Configures call forwarding so that all incoming calls to a particular directory number are forwarded to another directory number.
call forward cancel	Defines a feature code for a FAC to cancel the call-forward-all condition.
show stcapp feature codes	Displays all FACs.
stcapp feature access-code	Enables FACs and enters STC application feature access-code configuration mode for changing values of the prefix and features codes from the default.

call history max

To retain call history information and to specify the number of call records to be retained, use the **callhistorymax** command in global configuration mode.

call history max *number*

Syntax Description

number	The maximum number of call history records to be retained in the history table. Values are from 0 to 1200. The default is 15.

Command Default

If this command is not configured, no call history is maintained for disconnected calls. If the command is configured, the default value for number of records is 15.

Command Modes

Global configuration (config)

Command History

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

Usage Guidelines

The number of disconnected calls displayed is the number specified in the number argument. This maximum number helps to reduce CPU usage in the storage and reporting of this information.

Examples

The following example configures the history table on the gatekeeper to retain 25 records:

```
Router# call history max 25
```

Related Commands

Command	Description
show call history voice	Displays historical information on disconnected calls.

call-history-mib

To define the history MIB parameters, use the **call-history-mib**command in global configuration mode. To disable the configured parameters, use the **no** form of this command.

call-history-mib {**max-size** *num-of-entries* | **retain-timer** *seconds*}
no call-history-mib {**max-size** *num-of-entries* | **retain-timer** *seconds*}

Syntax Description

max-size	Specifies the maximum size of the call history MIB table.
number-of-entries	Number of entries in the call history MIB table. The valid range is from 0 to 500. The default value is 100.
retain-timer	Specifies the timer for entries in the call history MIB table.
seconds	Time in minutes, for removing an entry. The valid range is from 0 to 500. The default time is 15 minutes.

Command Default

The default values are set if the command is not enabled.

Command Modes

Global configuration (config)

Command History

Release	Modification
15.0(1)M	This command was introduced in a release earlier than Cisco IOS Release 15.0(1)M.

Usage Guidelines

CISCO-CALL-HISTORY-MIB describes the objects defined and used for storing the call information for all calls. The MIB contains a table that stores the past call information. The call information will include the destination number, the call connect time, the call disconnect time and the disconnection cause. These calls could be circuit switched or they could be virtual circuits. The history of each call will be stored. An entry will be created when a call gets disconnected. At the time of creation, the entry will contain the connect time and the disconnect time and other call information.

The history table is characterized by two values, the maximum number (*number-of-entries*) of entries that could be stored in a period of time (*seconds*).

The **max-size** value specifies the maximum size of the call history MIB table.

The**retain-timer** value specifies the length of time, in minutes, that entries will remain in the call history MIB table. Setting the value to 0 prevents any call history from being retained.

Examples

The following examples shows how to set call history MIB parameters:

```
Router# configure terminal
Router(config)# call-history-mib max-size 250
Router# configure terminal
Router(config)# call-history-mib retain-timer 250
```

Related Commands

Command	Description
show startup-config	Displays the contents of the startup configuration file.

call-progress-analysis

To activate call progress analysis (CPA) for a digital signal processor (DSP) farm profile on the Cisco Unified Border Element (Cisco UBE), use the **call-progress-analysis** command in DSP farm profile configuration mode. To disable this command from your configuration, use the **no** form of this command.

call-progress-analysis
no call-progress-analysis

Syntax Description

This command has no arguments or keywords.

Command Default

Call progress analysis is disabled.

Command Modes

DSP farm profile configuration (config-dspfarm-profile)

Command History

Release	Modification
15.3(2)T	This command was introduced.
Cisco IOS XE Release 3.9S	This command was integrated into Cisco IOS XE Release 3.9S.

Usage Guidelines

Use the **call-progress-analysis** command to activate CPA on Cisco UBE. This command is applicable only for local transcoding interface (LTI)-based DSP farm profiles, which has associate application CUBE applied on the respective DSP farm profiles. This command is not available on Skinny Call Control Protocol (SCCP)-based DSP farm profiles. If CPA is not activated on the DSP farm profile, you cannot configure the CPA timing and threshold parameters for VoIP calls.

Examples

The following example shows how to activate CPA on a DSP farm profile:

```
Device> enable
Device# configure terminal
Device(config)# dspfarm profile 15 transcode universal
Device(config-dspfarm-profile)# call-progress-analysis
```

Related Commands

Command	Description
cpa	Enables the CPA algorithm for outbound VoIP calls and sets the CPA parameters.
dspfarm profile	Enters DSP farm profile configuration mode and defines a profile for DSP farm services.

call language voice

To configure an external Tool Command Language (Tcl) module for use with an interactive voice response (IVR) application, use the **calllanguagevoicecommandin**global configuration mode.

call language voice *language url*

Syntax Description

language	Two-character abbreviation for the language; for example, "**en**" for English or "**ru**" for Russian.
url	URL that points to the Tcl module.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(2)T	This command was introduced.
12.3(14)T	This is obsolete in Cisco IOS Release 12.3(14)T. Use the **paramlanguage**command in application parameter configuration mode.

Usage Guidelines

The built-in languages are English *(en)* , Chinese *(ch)* , and Spanish *(sp)* . If you specify "**en**" , "**ch**" , or **sp**" , the new Tcl module replaces the built-in language functionality. When you add a new Tcl module, you create your own prefix to identify the language. When you configure and load the new languages, any upper-layer application (Tcl IVR) can use the language.

You can use the language abbreviation in the *language* argument of any **callapplicationvoice** command. The language and the text-to-speech (TTS) notations are available for the IVR application to use after they are defined by the Tcl module.

Examples

The following example adds Russian (**ru**) as a Tcl module:

```
call language voice ru tftp://box/unix/scripts/multi-lang/ru_translate.tcl
```

Related Commands

Command	Description
call application voice	Configures an application.
debug voip ivr	Specifies the type of VoIP IVR debug output that you want to view.
param language	Configures the language parameter in a service or package on the gateway.
show language voice	Displays information about configured languages and applications.

call language voice load

To load or reload a Tool Command Language (Tcl) module from the configured URL location, use the **calllanguagevoiceload** command in EXEC mode.

call language voice load *language*

Syntax Description

language	The two-character prefix configured with the **calllanguagevoice** command in global configuration mode; for example, "en" for English or "ru" for Russian.

Command Default

No default behavior or values

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.2(2)T	This command was introduced.

Usage Guidelines

You cannot use this command if the interactive voice response (IVR) application using the language that you want to configure has an active call. A language that is configured under an IVR application is not necessarily in use. To determine if a call is active, use the **showcallapplicationvoice** command.

Examples

The following example loads French (fr) into memory:

```
call language voice load fr
```

Related Commands

Command	Description
call application voice load	Loads an application.
debug voip ivr	Specifies the type of VoIP IVR debug output that you want to view.
show language voice	Displays information about configured languages and applications.

call leg dump event-log

To flush the event log buffer for call legs to an external file, use the **calllegdumpevent-log**command in privileged EXEC mode.

call leg dump event-log

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

This command immediately writes the event log buffer to the external file whose location is defined with the **calllegevent-logdumpftp** command in global configuration mode.

Note The **calllegdumpevent-log** command and the **calllegevent-logdumpftp** command are two different commands.

Examples

The following example writes the event log buffer to an external file named leg_elogs:

```
Router(config)# call leg event-log dump ftp ftp-server/elogs/leg_elogs.log username myname
 password 0 mypass
Router(config)# exit
Router# call leg dump event-log
```

Related Commands

Command	Description
call leg event-log	Enables event logging for voice, fax, and modem call legs.
call leg event-log dump ftp	Enables the voice gateway to write the contents of the call-leg event log buffer to an external file.
call leg event-log max-buffer-size	Sets the maximum size of the event log buffer for each call leg.
monitor call leg event-log	Displays the event log for an active call leg in real-time.
show call leg	Displays event logs and statistics for voice call legs.

call leg event-log

To enable event logging for voice, fax, and modem call legs, use the **calllegevent-log** command in global configuration mode. To reset to the default, use the **no** form of this command.

call leg event-log
no call leg event-log

Syntax Description

This command has no arguments or keywords.

Command Default

Event logging for call legs is disabled.

Command Modes

Global configuration (config)

Command History

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

Usage Guidelines

This command enables event logging for telephony call legs. IP call legs are not supported.

Note To prevent event logging from adversely impacting system performance for production traffic, the system includes a throttling mechanism. When free processor memory drops below 20%, the gateway automatically disables all event logging. It resumes event logging when free memory rises above 30%. While throttling is occurring, the gateway does not capture any new event logs even if event logging is enabled. You should monitor free memory on the gateway and enable event logging only when necessary to isolate faults.

Examples

The following example enables event logging for all telephony call legs:

```
call leg event-log
```

Related Commands

Command	Description
call leg dump event-log	Flushes the event log buffer for call legs to an external file.
call leg event-log dump ftp	Enables the voice gateway to write the contents of the call-leg event log buffer to an external file.
call leg event-log error-only	Restricts event logging to error events only for voice call legs.
call leg event-log max-buffer-size	Sets the maximum size of the event log buffer for each call leg.
call leg history event-log save-exception-only	Saves to history only event logs for call legs that had at least one error.
monitor call leg event-log	Displays the event log for an active call leg in real-time.

Command	Description
show call leg	Displays event logs and statistics for voice call legs.

call leg event-log dump ftp

To enable the gateway to write the contents of the call-leg event log buffer to an external file, use the **calllegevent-logdumpftp**command in global configuration mode. To reset to the default, use the **no** form of this command.

call leg event-log dump ftp *server*[{*:port*}]*file* **username** *username* **password** [{*ecryption-type*}]*password*
no call leg event-log dump ftp

Syntax Description

server	Name or IP address of FTP server where the file is located.
: *port*	(Optional) Specific port number on the server.
/ *file*	Name and path of the file.
username *username*	Username required for accessing the file.
password *encryption-type*	(Optional) The Cisco proprietary algorithm used to encrypt the password. Values are **0** or **7**. **0** disables encryption; **7** enables encryption. If you specify **7**, you must enter an encrypted password (a password already encrypted by a Cisco router).
password	Password required for accessing the file.

Command Default

Event logs are not written to an external file.

Command Modes

Global configuration (config)

Command History

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

Usage Guidelines

This command enables the gateway to automatically write the event log buffer to the named file either after an active call leg terminates or when the event log buffer becomes full. The default buffer size is 4 KB. To modify the size of the buffer, use the **calllegevent-logmax-buffer-size** command. To manually flush the event log buffer, use the **calllegdumpevent-log**command in privileged EXEC mode.

Note The **calllegdumpevent-log** command and the **calllegevent-logdumpftp** command are two different commands.

Enabling the gateway to write event logs to FTP could adversely impact gateway memory resources in some scenarios, for example, when:

- The gateway is consuming high processor resources and FTP does not have enough processor resources to flush the logged buffers to the FTP server.
- The designated FTP server is not powerful enough to perform FTP transfers quickly.

- Bandwidth on the link between the gateway and the FTP server is not large enough.
- The gateway is receiving a high volume of short-duration calls or calls that are failing.

You should enable FTP dumping only when necessary and not enable it in situations where it might adversely impact system performance.

Examples

The following example enables the gateway to write call leg event logs to an external file named leg_elogs.log on a server named ftp-server:

```
call leg event-log dump ftp ftp-server/elogs/leg_elogs.log username myname password 0 mypass
```

The following example specifies that call leg event logs are written to an external file named leg_elogs.log on a server with the IP address 10.10.10.101:

```
call leg event-log dump ftp 10.10.10.101/elogs/leg_elogs.log username myname password 0
mypass
```

Related Commands

Command	Description
call leg dump event-log	Flushes the event log buffer for call legs to an external file.
call leg event-log	Enables event logging for voice, fax, and modem call legs.
call leg event-log error-only	Restricts event logging to error events only for voice call legs.
call leg event-log max-buffer-size	Sets the maximum size of the event log buffer for each call leg.
call leg history event-log save-exception-only	Saves to history only event logs for call legs that had at least one error.
monitor call leg event-log	Displays the event log for an active call leg in real-time.
show call leg	Displays event logs and statistics for voice call legs.

call leg event-log errors-only

To restrict event logging to error events only for voice call legs, use the **calllegevent-logerrors-only**command in global configuration mode. To reset to the default, use the **no** form of this command.

call leg event-log errors-only
no call leg event-log errors-only

Syntax Description

This command has no arguments or keywords.

Command Default

All call leg events are logged.

Command Modes

Global configuration (config)

Command History

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

Usage Guidelines

This command limits the severity level of the events that are logged; it does not enable logging. You must use this command with the **calllegevent-log** command, which enables event logging for call legs.

Examples

The following example shows how to capture event logs only for call legs with errors:

```
Router(config)# call leg event-log
Router(config)# call leg event-log errors-only
```

Related Commands

Command	Description
call leg event-log	Enables event logging for voice, fax, and modem call legs.
call leg event-log dump ftp	Enables the gateway to write the contents of the call-leg event log buffer to an external file.
call leg event-log max-buffer-size	Sets the maximum size of the event log buffer for each call leg.
call leg history event-log save-exception-only	Saves to history only event logs for call legs that had at least one error.
monitor call leg event-log	Displays the event log for an active call leg in real-time.
show call leg	Displays event logs and statistics for voice call legs.

call leg event-log max-buffer-size

To set the maximum size of the event log buffer for each call leg, use the **calllegevent-logmax-buffer-size**command in global configuration mode. To reset to the default, use the **no** form of this command.

call leg event-log max-buffer-size *kbytes*
no call leg event-log max-buffer-size

Syntax Description

kbytes	Maximum buffer size, in kilobytes (KB). Range is 1 to 20. Default is 4.

Command Default

4 KB

Command Modes

Global configuration (config)

Command History

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

Usage Guidelines

If the event log buffer reaches the limit set by this command, the gateway allocates a second buffer of equal size. The contents of both buffers is displayed when you use the **showcallleg** command. When the first event log buffer becomes full, the gateway automatically appends its contents to an external FTP location if the **calllegevent-logdumpftp**command is used.

A maximum of two buffers are allocated for an event log. If both buffers are filled, the first buffer is deleted and another buffer is allocated for new events (buffer wraps around). If the **calllegevent-logdumpftp** command is configured and the second buffer becomes full before the first buffer is dumped, event messages are dropped and are not recorded in the buffer.

Examples

The following example sets the maximum buffer size to 8 KB:

```
call leg event-log max-buffer-size 8
```

Related Commands

Command	Description
call leg dump event-log	Flushes the event log buffer for call legs to an external file.
call leg event-log dump ftp	Enables the voice gateway to write the contents of the call-leg event log buffer to an external file.
monitor call leg event-log	Displays the event log for an active call leg in real-time.
show call leg	Displays event logs and statistics for voice call legs.

call leg history event-log save-exception-only

To save to history only event logs for call legs that had at least one error, use the **callleghistoryevent-logsave-exception-only** command in global configuration mode. To reset to the default, use the **no** form of this command.

call leg history event-log save-exception-only
no call leg history event-log save-exception-only

Syntax Description

This command has no arguments or keywords.

Command Default

By default all the events will be logged.

Command Modes

Global configuration (config)

Command History

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

Usage Guidelines

Call leg event logs move from the active to the history table after the call leg terminates. If you use this command, event logs are saved only for those legs that had errors. Event logs for normal legs that do not contain any errors are not saved.

Note This command does not affect records saved to an FTP server by using the **calllegdumpevent-log** command.

Examples

The following example saves to history only call leg records that have errors:

```
call leg history event-log save-exception-only
```

Related Commands

Command	Description
call leg dump event-log	Flushes the event log buffer for call legs to an external file.
call leg event-log	Enables event logging for voice, fax, and modem call legs.
call leg event-log error-only	Restricts event logging to error events only for voice call legs.
call leg event-log max-buffer-size	Sets the maximum size of the event log buffer for each call leg.
show call leg	Displays event logs and statistics for voice call legs.

callmonitor

To enable call monitoring messaging functionality on a SIP endpoint in a VoIP network, use the **callmonitor** command in voice-service configuration mode. To return to the default, use the **no** form of this command.

callmonitor
no callmonitor

Syntax Description

This command has no arguments or keywords.

Command Default

Monitoring service is disabled.

Command Modes

Voice service VoIP configuration (config-voi-serv).

Voice class tenant configuration.

Command History

Cisco IOS Release	Modification
12.4(11)XW2	This command was introduced.
12.4(20)T	This command was integrated into Cisco IOS Release 12.4(20)T.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Use this command in voice service configuration mode to allow a SIP endpoint, such as an external feature server, to watch call activity on a VoIP network.

To view call activity, use the **showcallmon**command.

Examples

The following example enables call monitoring messaging functionality on a SIP endpoint:

```
Router(config-voi-serv)# callmonitor
```

Related Commands

Command	Description
show callmon	Displays call-monitor information.

call preserve

To enable the preservation of H.323 VoIP calls, use the**callpreserve** command in h323, voice-class, and voice-service configuration modes. To reset to the default, use the **no** form of this command.

call preserve [**limit-media-detection**]
no call preserve [**limit-media-detection**]

Syntax Description

limit-media-detection	Limits RTP and RTCP inactivity detection and bidirectional silence detection (if configured) to H.323 VoIP preserved calls only.

Command Default

H.323 VoIP call preservation is disabled.

Command Modes

h323
Voice-class configuration (config-voice-class)
Voice-service configuration (config-voi-serv)

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 **callpreserve**command activates H.323 VoIP call preservation for following types of failures and connections:

Failure Types

- WAN failures that include WAN links flapping or degraded WAN links
- Cisco Unified CallManager software failure, such as when the ccm.exe service crashes on a Cisco Unified CallManager server.
- LAN connectivity failure, except when a failure occurs at the local branch

Connection Types

- Calls between two Cisco Unified CallManager controlled endpoints
  - During Cisco Unified CallManager reloads
  - When a Transmission Control Protocol (TCP) connection between one or both endpoints and Cisco Unified CallManager used for signaling H.225.0 or H.245 messages is lost or flapping
  - Between endpoints that are registered to different Cisco Unified CallManagers in a cluster and the TCP connection between the two Cisco Unified CallManagers is lost
  - Between IP phones and the PSTN at the same site
- Calls between Cisco IOS gateway and an endpoint controlled by a softswitch where the signaling (H.225.0, H.245 or both) flows between the gateway and the softswitch and media flows between the gateway and the endpoint.
  - When the softswitch reloads.

  - When the H.225.0 or H.245 TCP connection between the gateway and the softswitch is lost, and the softswitch does not clear the call on the endpoint
  - When the H.225.0 or H.245 TCP connection between softswitch and the endpoint is lost, and the soft-switch does not clear the call on the gateway

- Call flows that involve a Cisco IP in IP (IPIP) gateway running in media flow-around mode that reload or lose connection with the rest of the network

When bidirectional silence and RTP and RTCP inactivity detection are configured, they are enabled for all calls by default. To enable them for H.323 VoIP preserved calls only, you must use the **callpreserve**command's **limit-media-detection** keyword.

H.323 VoIP call preservation can be applied globally to all calls and to a dial peer.

Examples

The following example enables H.323 VoIP call preservation for all calls.

```
voice service voip
 h323
  call preserve
```

The following configuration example enables H.323 VoIP call preservation for dial peer 1.

```
voice-class h323 4
 call preserve
dial-peer voice 1 voip
 voice-class h323 4
```

The following example enables H.323 VoIP call preservation and enables RTP and RTCP inactivity detection and bidirectional silence detection for preserved calls only:

```
voice service voip
 h323
  call preserve limit-media-detection
```

The following example enables RTP and RTCP inactivity detection. Note that for H.323 VoIP call preservation VAD must be set to off (**novad** command).

```
dial-peer voice 10 voip
 no vad
gateway
 timer receive-rtcp
ip rtcp report-interval
```

The following configuration example enables bidirectional silence detection:

```
gateway
 timer media-inactive
ip rtcp report interval
```

Related Commands

Command	Description
h323	Enables the H.323 voice service configuration commands.
show h323 calls preserved	Displays data about active H.323 VoIP preserved calls.

Command	Description
voice-class h323	Assigns an H.323 voice class to a VoIP dial peer.
voice service voip	Enters voice-service configuration mode.

call-route

To enable Header-Based routing, at the global configuration level, use the **call-route** command in voice service VoIP SIP configuration mode or voice class tenant configuration mode. To disable Header-Based routing, use the **no** form of this command.

call-route {**dest-route-string** | **p-called-party-id** | **history-info** | **url**} [system]
no call-route {**dest-route-string** | **p-called-party-id** | **history-info** | **url**}

Syntax Description

dest-route-string	Enables call routing based on the Destination-Route-String header.
p-called-party-id	Enables call routing based on the P-Called-Party-Id header.
history-info	Enables call routing based on the History-Info header.
url	Enables call routing based on the URL.
system	Use the global value of the header. This keyword is available only for the tenant configuration mode.

Command Default

Support for call routing based on the header in a received INVITE message is disabled.

Command Modes

Voice service VoIP SIP configuration (conf-serv-sip)

Voice class tenant configuration (config-class)

Command History

Release	Modification
12.4(22)YB	This command was introduced.
15.0(1)M	This command was integrated into Cisco IOS Release 15.0(1)M.
15.1(2)T	This command was modified. The **history-info** keyword was added.
Cisco IOS XE Release 3.3S	This command was integrated into Cisco IOS XE Release 3.3S.
15.2(1)T	This command was modified. The **url** keyword was added.
15.3(3)M	This command was modified. The **dest-route-string** keyword was added.
Cisco IOS XE Release 3.10S	This command was modified. The **dest-route-string** keyword was added.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: **system**. This command is now available under voice class tenants.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Use the **call-route** command to enable the Cisco Unified Border Element to route calls based on the Destination-Route-String, P-Called-Party-ID or History-Info header in a received INVITE message. If multiple call routes are configured, call routing enabled based on destination route string takes precedence over other header configurations. Destination route string configuration is applicable only for outbound dial-peer matching.

Examples

The following example shows how to enable call routing based on the header value:

```
Router> enable

Router# configure terminal
Router(config)# voiceservicevoip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# call-route dest-route-string
Router(conf-serv-sip)# call-route p-called-party-id
Router(conf-serv-sip)# call-route history-info
Router(conf-serv-sip)# call-route url
```

The following example shows how to route a call based on the History-Info header in the voice class tenant configuration mode:

```
Router(config-class)# call-route history-info system
```

Related Commands

Command	Description
voice-class sip call-route	Enables call routing based on the Destination-Route-String, P-called-party-id and History-Info header values at the dial-peer configuration level.

call-router h323-annexg

To enable the Annex G border element (BE) configuration commands by invoking H.323 Annex G configuration mode, use the **call-router** command in global configuration mode. To remove the definition of a BE, use the no form of this command.

call-router h323-annexg *border-element-id*
no call-router h323-annexg

Syntax Description

border -element-id	Identifier of the BE that you are provisioning. Possible values are any International Alphabet 5 (IA5) string, without spaces and up to 20 characters in length. This value must match the value that you specified for the BE ID in the **border-element** command.

Command Default

No default behaviors or values

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T. This command does not support the Cisco AS5300, Cisco AS5350, and Cisco AS5400 series in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

Use this command to enter Annex G configuration mode and to identify BEs.

Examples

The following example shows that Annex G configuration mode is being entered for a BE named "be20":

```
Router(config)# call-router h323-annexg be20
```

Related Commands

Command	Description
show call history	Displays the fax call history table for a fax transmission.
show call-router status	Displays the Annex G BE status.

call-routing hunt-scheme

To enable capacity based load-balancing, use the **call-routinghunt-scheme**command in gatekeeper configuration mode. To disable this function, use the **no** form of this command.

call-routing hunt-scheme percentage-capacity-util
no call-routing hunt-scheme

Syntax Description

percentage -capacity-util	Selects the one with least percentage capacity utilized among the gateways.

Command Default

This command is disabled.

Command Modes

Gatekeeper configuration (config-gk)

Command History

Release	Modification
12.4(11)T	This command was introduced.

Usage Guidelines

Use the **call-routinghunt-scheme**command to turn on load balancing based on capacity of gateway and verify that the gateway capacity reporting is enabled.

Examples

The following example shows the gateway with the with least percentage capacity being selected:

```
Router(gk-config)# call-routing hunt-scheme percentage-capacity-util
```

Related Commands

Command	Description
timer cluster-element	Sets the time between resource update messages to gatekeepers in local cluster.

call rscmon update-timer

To change the value of the resource monitor throttle timer, use the **callrscmonupdate-timer** command in privileged EXEC mode. To revert to the default value, use the **no** form of this command.

call rscmon update-timer *milliseconds*
no call rscmon update-timer

Syntax Description

milliseconds	Duration of the resource monitor throttle timer, in milliseconds (ms). Range is from 20 to 3500. The default is 2000.

Command Default

2000 ms

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T. This command does not support the Cisco AS5300, Cisco AS5350, and Cisco AS5400 series in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

This command specifies the duration of the resource monitor throttle timer. When events are delivered to the resource monitor process, the throttle timer is started and the event is processed after the timer expires (unless the event is a high-priority event). The timer ultimately affects the time it takes the gateway to send Resource Availability Indicator (RAI) messages to the gatekeeper. This command allows you to vary the timer according to your needs.

Examples

The following example shows how the timer is to be configured:

```
Router(config)# call rscmon update-timer 1000
```

Related Commands

Command	Description
resource threshold	Configures a gateway to report H.323 resource availability to its gatekeeper.

call rsvp-sync

To enable synchronization between Resource Reservation Protocol (RSVP) signaling and the voice signaling protocol, use the **callrsvp-sync** command in global configuration mode. To disable synchronization, use the **no** form of this command.

call rsvp-sync
no call rsvp-sync

Syntax Description

This command has no keywords or arguments.

Command Default

Synchronization is enabled between RSVP and the voice signaling protocol (for example, H.323).

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)XI	This command was introduced on the Cisco 2600 series, 3600 series, 7200 series, Cisco AS5300, Cisco AS5800, and Cisco MC3810.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

The **callrsvp-sync** command is enabled by default.

Examples

The following example enables synchronization between RSVP and the voice signaling protocol:

```
call rsvp-sync
```

Related Commands

Command	Description
call rsvp-sync resv-timer	Sets the timer for reservation requests.
call start	Forces the H.323 Version 2 gateway to use fast connect or slow connect procedures for a dial peer.
debug call rsvp-sync events	Displays the events that occur during RSVP synchronization.
h323 call start	Forces an H.323 Version 2 gateway to use fast connect or slow connect procedures for all VoIP services.
ip rsvp bandwidth	Enables the use of RSVP on an interface.
show call rsvp-sync conf	Displays the RSVP synchronization configuration.
show call rsvp-sync stats	Displays statistics for calls that have attempted RSVP reservation.

call rsvp-sync resv-timer

To set the timer on the terminating VoIP gateway for completing RSVP reservation setups, use the **callrsvp-syncresv-timer** command in global configuration mode. To restore the default value, use the **no** form of this command.

call rsvp-sync resv-timer *seconds*
no call rsvp-sync resv-timer

Syntax Description

seconds	Number of seconds in which the reservation setup must be completed, in both directions. Range is from 1 to 60. The default is 10.

Command Default

10 seconds

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)XI	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco AS5300, Cisco AS5800, and Cisco MC3810.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

The reservation timer is started on the terminating gateway when the session protocol receives an indication of the incoming call. This timer is not set on the originating gateway because the resource reservation is confirmed at the terminating gateway. If the reservation timer expires before the RSVP setup is complete, the outcome of the call depends on the acceptable quality of service (QoS) level configured in the dial peer; either the call proceeds without any bandwidth reservation or it is released. The timer must be set long enough to allow calls to complete but short enough to free up resources. The optimum number of seconds depends on the number of hops between the participating gateways and the delay characteristics of the network.

Examples

The following example sets the reservation timer to 30 seconds:

```
call rsvp-sync resv-timer 30
```

Related Commands

Command	Description
call rsvp -sync	Enables synchronization of RSVP and the H.323 voice signaling protocol.
debug call rsvp -syncevents	Displays the events that occur during RSVP synchronization.
show call rsvp -syncconf	Displays the RSVP synchronization configuration.

Command	Description
show call rsvp -syncstats	Displays statistics for calls that have attempted RSVP reservation.

call service stop

To shut down VoIP call service on a gateway, use the **call service stop** command in voice service SIP or voice service H.323 configuration mode. To enable VoIP call service, use the **no** form of this command. To set the command to its defaults, use the **default call service stop** command.

call service stop [**forced**] [**maintain-registration**]
no call service stop
default call service stop

Syntax Description

forced	(Optional) Forces the gateway to immediately terminate all in-progress calls.
maintain -registration	(Optional) Forces the gateway to remain registered with the gatekeeper.

Command Default

VoIP call service is enabled.

Command Modes

Voice service SIP configuration (conf-serv-sip)
Voice service H.323 configuration (conf-serv-h323)

Command History

Release	Modification
12.3(1)	This command was introduced.
12.4(22)T	Support for IPv6 was added.
12.4(23.08)T01	The default behavior was clarified for SIP and H.323 protocols.
Cisco IOS XE Amsterdam 17.2.1r	Introduced support for YANG models.

Usage Guidelines

Use the **call service stop** command to shut down the SIP or H.323 services regardless of whether the **shutdown** or **no shutdown** command was configured in voice service configuration mode.

Use the **no call service stop** command to enable SIP or H.323 services regardless of whether the **shutdown** or **no shutdown** command was configured in voice service configuration mode.

Use the **default call service stop** command to set the command to its defaults. The defaults are as follows:

- Shut down SIP or H.323 service, if the **shutdown** command was configured in voice service configuration mode.
- Enable SIP or H.323 service, if the **no shutdown** command was configured in voice service configuration mode.

Examples

The following example shows SIP call service being shut down on a Cisco gateway:

```
Router> enable
Router# configure terminal
Router(config)# voice service voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# call service stop
```

The following example shows H.323 call service being enabled on a Cisco gateway:

```
Router> enable
Router# configure terminal
Router(config)# voice service voip
Router(conf-voi-serv)# h323
Router(conf-serv-h323)# no call service stop
```

The following example shows SIP call service being enabled on a Cisco gateway because the **no shutdown** command was configured in voice service configuration mode:

```
Router> enable
Router# configure terminal
Router(config)#voice service voip
Router(conf-voi-serv)# no shutdown
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# default call service stop
```

The following example shows H.323 call service being shut down on a Cisco gateway because the **shutdown** command was configured in voice configuration mode:

```
Router> enable
Router# configure terminal
Router(config)# voice service voip
Router(conf-voi-serv)# shutdown
Router(conf-voi-serv)# h323
Router(conf-serv-h323)# default call service stop
```

Related Commands

Command	Description
bandwidth audio as-modifier	Allows SIP SDP bandwidth-related options.
billing b-channel	Enables the H.323 gateway to access B-channel information for all H.323 calls.
outbound-proxy	Configures an outbound proxy server.
telephony-service ccm-compatible	Enables the detection of a Cisco Unified Communications Manager system in the network and allows the exchange of calls.

call spike

To configure the limit on the number of incoming calls received in a short period of time (a call spike), use the **callspike** command in global or dial peer voice configuration mode. To disable this command, use the**no** form of this command.

call spike *call-number* [**steps** *number-of-steps* **size** *milliseconds*]
no call spike

Dial Peer Voice Configuration Mode
call spike *threshold* [**steps** *number-of-steps* **size** *milliseconds*]

Syntax Description

call -number	Incoming call count for the spiking threshold. Range is 1 to 2147483647.
steps *number -of-steps*	(Optional) Specifies the number of steps for the spiking sliding window. Range is from 3 to 10. The default is 5.steps for the spiking sliding window.
size *milliseconds*	(Optional) Specifies step size in milliseconds. Range is from 100 to 250. The default is 200.
threshold	Threshold for the incoming call count for spiking. Range is 1 to 2147483647.

Command Default

The limit on the number of incoming calls received during a specified period is not configured.

Command Modes

Global configuration (config)
Dial peer voice configuration (config-dial-peer)

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T. This release does not support the Cisco AS5300, Cisco AS5350, and Cisco AS5400 series.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on Cisco 1750 and Cisco 1751 routers. Support for other Cisco platforms was not included in this release.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on Cisco 7200 series routers. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 was not included in this release.
12.2(11)T	Support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.
15.1(3)T	This command was modified. Support for this command was added in the dial peer level.

Release	Modification
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

A call spike occurs when a large number of incoming calls arrive from the Public Switched Telephone Network (PSTN) in a short period of time (for example, 100 incoming calls in 10 milliseconds). Setting this command allows you to control the number of call requests that can be received in a configured time period. The sliding window buffers the number of calls that get through. The counter resets according to the specified step size.

The period of the sliding window is calculated by multiplying the number of steps by the size. If an incoming call exceeds the configured call number during the period of the sliding window the call is rejected.

If the **callspike** is configured at both the global and dial-peer levels, the dial-peer level takes precedence and the call spike is calculated. If the call spike threshold is exceeded the call gets rejected, and the call spike calculation is done at the global level.

Examples

The following example shows how to configure the **callspike** command with a call-number and the of 1, a sliding window of 10 steps, and a step size of 200 milliseconds. The period of the sliding window is 2 seconds. If the gateway receives more than 1 call within 2 seconds the call is rejected.

```
Router(config)# call spike 1 steps 10 size 200
```

The following example shows how to configure the **callspike** command with a call number of 30, a sliding window of 10 steps, and a step size of 2000 milliseconds:

```
Router(config)# call spike 30 steps 10 size 2000
```

The following example shows how to configure the **callspike** command in dial peer voice mode with threshold of 20, a sliding window of 7, and a step size of 2000 milliseconds:

```
Router(config)# dial-peer voice 400 voip
Router(config-dial-peer)# call spike 20 steps 7 size 2000
```

Related Commands

Command	Description
dtmf-relay (Voice over IP)	Specifies how an H.323 gateway relays DTMF tones between telephony interfaces and an IP network.
show call spike status	Displays the configuration of the threshold for incoming calls.

call start

To force an H.323 Version 2 gateway to use either fast connect or slow connect procedures for a dial peer, use the **callstart** command in H.323 voice-service configuration mode. To restore the default setting, use the **no** form of this command.

call start {**fast** | **slow** | **system** | **interwork**} [**sync-rsvp slow-start**]
no call start

Syntax Description

fast	Gateway uses H.323 Version 2 (fast connect) procedures.
slow	Gateway uses H.323 Version 1 (slow connect) procedures.
system	Gateway defaults to voice-service configuration mode.
interwork	Gateway interoperates between fast-connect and slow-connect procedures. **Note** The **interwork** keyword is applicable to IP-to-IP gateways only and supports basic audio calls Dual-tone multi-frequency (DTMF), fax, and audio transcoding calls are not supported).
sync-rsvp slow-start	(Optional) Gateway uses Resource Reservation Protocol (RSVP) synchronization for slow-start calls.

Command Default

The gateway defaults to voice-service configuration mode.

Command Modes

H.323 voice-service configuration (conf-serv-h323)

Command History

Release	Modification
12.1(3)XI	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco AS5300, Cisco AS5800, and Cisco MC3810.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.2(2)XA	This command was changed to use the H.323 voice-service configuration mode from the voice-class configuration mode.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command was implemented on the Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.

Release	Modification
12.3(4)T	The **synch-rsvpslow-start**keywords were added.
12.3(8)T	The **interwork** keyword was added.
Cisco IOS XE Release 3.3S	This command was integrated into Cisco IOS XE Release 3.3S.

Usage Guidelines

In Cisco IOS Release 12.1(3)XI and later releases, H.323 VoIP gateways by default use H.323 Version 2 (fast connect) for all calls, including those initiating RSVP. Previously, gateways used only slow-connect procedures for RSVP calls. To enable Cisco IOS Release 12.1(3)XI gateways to be backward-compatible with earlier releases of Cisco IOS Release 12.1T, the **callstart** command allows the originating gateway to initiate calls using slow connect.

The **callstart** command is configured as part of the voice class assigned to an individual VoIP dial peer. It takes precedence over the**h323callstart**command that is enabled globally to all VoIP calls, unless the **system** keyword is used, in which case the gateway defaults to Version 2.

The **sync-rsvpslow-start**keyword, when used in H.323 voice-class configuration mode, controls RSVP synchronization for all slow-start calls handled by the gateway. When the **sync-rsvpslow-start**keyword is used in an H.323 voice-class definition, the behavior can be specified for individual dial peers by invoking the voice class in dial-peer voice configuration mode. This command is enabled by default in some Cisco IOS images, and in this situation the **showrunning-config** command displays this information only when the **no**form of the command is used.

Note The **callstart** command supports only H.323 to H.323 calls.

The **interwork** keyword is only used with IP-to-IP gateways connecting fast connect from one side to slow connect on the other for basic audio calls. Configure the **interwork** keyword in voice-class H.323 configuration mode or on both the incoming and outgoing dial peers. Codecs must be specified on both dial peers for interworking to function. When the **interwork** keyword is configured, codecs need to be specified on both dial-peers and the **codectransparent** command should not be configured.

Examples

The following example shows slow connect for the voice class 1000 being selected:

```
voice service class h323 1000
 call start slow
!
dial-peer voice 210 voip
 voice-class h323 1000
```

The following example shows the gateway configured to use the H.323 Version 1 (slow connect) procedures:

```
h323
 call start slow
```

Related Commands

Command	Description
acc-qos	Selects the acceptable quality of service for a dial peer.

Command	Description
call rsvp-sync	Enables synchronization between RSVP and the H.323 voice signaling protocol.
call rsvp-sync resv-timer	Sets the timer for RSVP reservation setup.
codec transparent	Enables codec capabilities to be passed transparently between endpoints in a Cisco IPIPGW.
debug call rsvp-sync events	Displays the events that occur during RSVP synchronization.
h323	Enables H.323 voice service configuration commands.
req-qos	Selects the desired quality of service to use in reaching a dial peer.
show call rsvp-sync conf	Displays the RSVP synchronization configuration.
show call rsvp-sync stats	Displays statistics for calls that attempted RSVP reservation.
show running-config	Displays the contents of the currently running configuration file.
voice class h323	Enters voice-class configuration mode and creates a voice class for H.323 attributes.

call threshold global

To enable the global resources of a gateway, use the **callthresholdglobal**command in global configuration mode. To disable the global resources of the gateway, use the **no** form of this command.

call threshold global *trigger-name* **low** *percent* **high** *percent* [**busyout**] [**treatment**]
no call threshold global *trigger-name*

Syntax Description

trigger -name	Specifies the global resources on the gateway. The *trigger-name*argument can be one of the following: • **cpu-5sec** --CPU utilization in the last 5 seconds. • **cpu-avg** --Average CPU utilization. • **io-mem** --I/O memory utilization. • **proc-mem** --Processor memory utilization. • **total-calls** --Total number of calls. • **total-mem** --Total memory utilization.
low *percent*	Value of low threshold: Range is 1–100% for the utilization triggers; 1–10000 calls for the total-calls.
high *percent*	Value of high threshold: Range is 1–100% for the utilization triggers; 1–10000 calls for the total-calls.
busyout	(Optional) Busy out the T1/E1 channels if the resource is not available.
treatment	(Optional) Applies call treatment from the session application if the resource is not available.

Command Default

The default is **busyout**and **treatment** for global resource triggers.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T. Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 is not included in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on the Cisco 1750 and Cisco 1751 routers. Support for other Cisco platforms is not included in this release

Release	Modification
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on Cisco 7200 series routers. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command was implemented on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5800 in this release.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Note YANG model for bandwidth-based CAC configuration is not available.

For example:

call threshold interface type number int-bandwidth class-map name [l2-overhead percentage] | low low-threshold high high-threshold} [midcall-exceed]

Use this command to enable a trigger and define associated parameters to allow or disallow new calls on the router. Action is enabled when the trigger value goes above the value that is specified by the **high** keyword and is disabled when the trigger drops below the value that is specified by the **low**keyword.

You can configure these triggers to calculate Resource Availability Indicator (RAI) information. An RAI is forwarded to a gatekeeper so that it can make call admission decisions. You can configure a trigger that is global to a router or is specific to an interface.

Examples

The following example shows how to busy out the total calls when a low of 5 or a high of 5000 is reached:

```
call threshold global total-calls low 5 high 5000 busyout
```

The following example shows how to busy out the average CPU utilization if a low of 5 percent or a high of 65 percent is reached:

```
call threshold global cpu-avg low 5 high 65 busyout
```

Related Commands

Command	Description
call threshold (interface)	Enables interface resources of a gateway.
call threshold poll-interval	Enables a polling interval threshold for CPU or memory.
clear call threshold	Clears enabled triggers and their associated parameters.
show call threshold	Displays enabled triggers, current values for configured triggers, and number of API calls that were made to global and interface resources.

call threshold interface

To enable interface resources of a gateway, use the **call threshold interface** command in global configuration mode. To disable the interface resources of the gateway, use the **no** form of this command.

call threshold interface *type number* {**int-bandwidth** {**class-map** *name* [{**l2-overhead** *percentage*}] | **low** *low-threshold* **high** *high-threshold*} [{**midcall-exceed**}] | **int-calls low** *value* **high** *value*}
no call threshold interface *type number* {**int-bandwidth** | **int-calls**}

Syntax Description

type	Interface type. For more information, use the question mark (?) online help function.
number	Interface or subinterface number. For more information about the numbering syntax for your networking device, use the question mark (?) online help function.
int-bandwidth	Configures the threshold bandwidth for VoIP media through an interface.
class-map *name*	Specifies a traffic class for the VoIP media traffic that is configured through Modular Quality of Service (MQS).
l2-overhead *percentage*	(Optional) Configures the Layer 2 overhead as the percentage of the configured bandwidth. This is the value by which the configured bandwidth will be deducted to obtain the IP bandwidth. The default value is 10 percent.
low *low-threshold*	Specifies the low threshold for the aggregate interface bandwidth value in Kbps. The range is from 8 to 2000000.
high *high-threshold*	Specifies the high threshold for the aggregate interface bandwidth value in Kbps. The range is from 8 to 2000000.
midcall-exceed	(Optional) Allows the bandwidth that exceeds the configured threshold during midcall media renegotiation.
int-calls	Specifies the number of calls that are transmitted through the interface.
low *value*	Specifies the low threshold for the number of calls allowed. The range is from 1 to 10000.
high *value*	Specifies the high threshold for the number of calls allowed. The range is from 1 to 10000.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T. Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 is not included in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.

Release	Modification
12.2(4)XM	This command was implemented on Cisco 1750 and Cisco 1751 routers. This command does not support any other Cisco platforms in this release.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on Cisco 7200 series routers. This command is not supported on Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 routers in this release.
15.2(2)T	This command was modified. The **int-bandwidth**, **class-map** *name*, **l2-overhead** *percentage*, **low** *low-threshold*, **high** *high-threshold*, and **midcall-exceed** keywords and arguments were added.

Usage Guidelines

Use this command to specify thresholds that allow or disallow new calls on the router.

The Bandwidth-Based Call Admission Control feature is supported on the following interfaces:

- ATM
- Ethernet (Fast Ethernet, Gigabit Ethernet)
- Loopback
- Serial

Examples

The following example shows how to enable thresholds as low as 5 and as high as 2500 for interface calls on Ethernet interface 0/1:

```
Router> enable
Router# configure terminal
Router(config)# call threshold interface Ethernet 0/1 int-calls low 5 high
2500
```

The following example shows how to configure the Cisco Unified Border Element (Cisco UBE) to reject new SIP calls when the VoIP media bandwidth on Gigabit Ethernet interface 0/0 exceeds 400 kbps and continues to have 100 Kbps:

```
Router> enable
Router# configure terminal
Router(config)# call threshold interface GigabitEthernet 0/0 int-bandwidth
low 100 high 400
```

The following example shows how to configure Cisco UBE to reject new SIP calls when the VoIP media bandwidth on Gigabit Ethernet interface 0/0 exceeds the configured bandwidth for priority traffic in the “voip-traffic” class:

```
Router> enable
Router# configure terminal
Router(config)# class-map match-all voip-traffic

Router(config-cmap)# policy-map voip-policy
Router(config-pmap)# class  voip-traffic
Router(config-pmap-c)# priority 440
Router(config-pmap-c)# end
```

```
Router# configure terminal
Router(config)# call threshold interface GigabitEthernet 0/0 int-bandwidth
class-map voip-traffic l2-overhead 10
```

Related Commands

Command	Description
call threshold (global)	Enables global resources of a gateway.
call threshold poll-interval	Enables a polling interval threshold for CPU or memory.
clear call threshold	Clears enabled triggers and their associated parameters.
show call threshold	Displays enabled triggers, current values for configured triggers, and the number of API calls that were made to global and interface resources.

call threshold poll-interval

To enable a polling interval threshold for assessing CPU or memory thresholds, use the **callthresholdpoll-interval** command in global configuration mode. To disable this command, use the **no** form of this command.

call threshold poll-interval {**cpu-average** | **memory**} *seconds*
no call threshold poll-interval {**cpu-average** | **memory**}

Syntax Description

cpu -average	The CPU average interval, in seconds. The default is 60.
memory	The average polling interval for the memory, in seconds. The default is 5.
seconds	Window of polling interval, in seconds. Range is from 10 to 300 for the CPU average interval, and from 1 to 60 for the memory average polling interval.

Command Default

cpu -average: 60 seconds**memory**: 5 seconds

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T. Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 is not included in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on Cisco 1750 and Cisco 1751 routers. This release does not support any other Cisco platforms in this release.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series routers. This release does not support the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5800.

Examples

The following example shows how to specify that memory thresholds be polled every 10 seconds:

```
call threshold poll-interval memory 10
```

Related Commands

Command	Description
call threshold	Enables the global resources of the gateway.
clear call threshold	Clears enabled triggers and their associated parameters.

Command	Description
show call threshold	Displays enabled triggers, current values for configured triggers, and number of API calls that were made to global and interface resources.

call treatment action

To configure the action that the router takes when local resources are unavailable, use the **call treatment action** command in global configuration mode. To disable call treatment action, use the **no** form of this command.

call treatment action {**hairpin** | **playmsg** *url* | **reject**}
no call treatment action

Syntax Description

hairpin	Hairpins the calls through the POTS dial peer. **Note** The **hairpin** keyword is not available on Cisco 1750 and Cisco 1751 routers.
playmsg	Plays a specified message to the caller.
url	Specifies the URL of the audio file to play.
reject	Disconnects the call and pass-down cause code.

Command Default

No treatment is applied.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T. This command does not support the Cisco AS5300, Cisco AS5350, and Cisco AS5400 series in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on the Cisco 1750 and Cisco 1751 routers. This command does not support any other Cisco platforms in this release.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series routers. This command does not support the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T. Support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5800.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Use this command to define parameters to disconnect (with cause code), or hairpin, or whether a message or busy tone is played to the user.

Examples

The following example shows how to enable the call treatment feature with a "hairpin" action:

```
call treatment on
call treatment action hairpin
```

The following example shows how to enable the call treatment feature with a "playmsg" action. The file "congestion.au" plays to the caller when local resources are not available to handle the call.

```
call treatment on
call treatment action playmsg tftp://keyer/prompts/conjestion.au
```

Related Commands

Command	Description
call threshold	Clears enabled triggers and their associated parameters.
call treatment on	Enables call treatment to process calls when local resources are unavailable.
clear call treatment stats	Clears the call treatment statistics.
show call treatment	Displays the call treatment configuration and statistics for handling calls on the basis of resource availability.

call treatment cause-code

To specify the reason for the disconnection to the caller when local resources are unavailable, use the **call treatment cause-code** command in global configuration mode. To disable the call treatment cause-code specification, use the **no** form of this command.

call treatment cause-code {**busy** | **no-QoS** | **no-resource**}
no call treatment cause-code

Syntax Description

busy	Indicates that the gateway is busy.
no-QoS	Indicates that the gateway cannot provide quality of service (QoS).
no-resource	Indicates that the gateway has no resources available.

Command Default

Disconnect reason is not specified to the caller.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T. This command does not support the Cisco AS5300, Cisco AS5350, and Cisco AS5400 series in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on the Cisco 1750 and Cisco 1751 routers. This command does not support any other Cisco platforms in this release.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series routers. This command does not support the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T. Support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5800.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Use this command to associate a cause-code with a disconnect event.

Examples

The following example shows how to configure a call treatment cause code to reply with "no-Qos" when local resources are unavailable to process a call:

```
call treatment on
call treatment cause-code no-Qos
```

Related Commands

Command	Description
call threshold	Clears enabled triggers and their associated parameters.
call treatment on	Enables call treatment to process calls when local resources are unavailable.
clear call treatment stats	Clears the call treatment statistics.
show call treatment	Displays the call treatment configuration and statistics for handling calls on the basis of resource availability.

call treatment isdn-reject

To specify the rejection cause code for ISDN calls when all ISDN trunks are busied out and the switch ignores the busyout trunks and still sends ISDN calls into the gateway, use the **call treatment isdn-reject** command in global configuration mode. To disable call treatment, use the **no** form of this command.

call treatment isdn-reject *cause-code*
no call treatment isdn-reject

Syntax Description

cause-code **34**	No circuit/channel available—The connection cannot be established because no appropriate channel is available to take the call.
cause-code **38**	Network out of order—The destination cannot be reached because the network is not functioning correctly, and the condition might last for an extended period of time. An immediate reconnect attempt will probably be unsuccessful.
cause-code **41**	Temporary failure—An error occurred because the network is not functioning correctly. The problem will be resolved shortly.
cause-code **42**	Switching equipment congestion—The destination cannot be reached because the network switching equipment is temporarily overloaded.
cause-code **43**	Access information discarded—Discarded information element identifier. The network cannot provide the requested access information.
cause-code **44**	Requested circuit/channel not available—The remote equipment cannot provide the requested channel for an unknown reason. This might be a temporary problem.
cause-code **47**	Resources unavailable, unspecified—The requested channel or service is unavailable for an unknown reason. This might be a temporary problem.

Command Default

No value is specified.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T. This command does not support the Cisco AS5300, Cisco AS5350, and Cisco AS5400 series in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on the Cisco 1750 and Cisco 1751 routers. This command does not support any other Cisco platforms in this release.

Release	Modification
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series routers. This command does not support the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5800.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Use this command only when all ISDN trunks are busied out and the switch ignores the busyout trunks and still sends ISDN calls into the gateway. The gateway should reject the call in the ISDN stack using the configured cause code.

Under any other conditions, the command has no effect.

Examples

The following example shows how to configure the call treatment to reply to an ISDN call with an ISDN rejection code for "temporary failure" when local resources are unavailable to process a call:

```
call treatment on
call treatment isdn-reject 41
```

Related Commands

Command	Description
call threshold	Clears enabled triggers and their associated parameters.
call treatment on	Enables call treatment to process calls when local resources are unavailable.
clear call treatment stats	Clears the call treatment statistics.
show call treatment	Displays the call treatment configuration and statistics for handling calls on the basis of resource availability.

call treatment on

To enable call treatment to process calls when local resources are unavailable, use the **call treatment on**command in global configuration mode. To disable call treatment, use the **no** form of this command.

call treatment on
no call treatment on

Syntax Description

This command has no arguments or keywords.

Command Default

Treatment is inactive.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T. This command does not support the Cisco AS5300, Cisco AS5350, and Cisco AS5400 series in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on the Cisco 1750 and Cisco 1751 routers. This command does not support any other Cisco platforms in this release.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series routers. This command does not support the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T. Support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5800.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Use this command to enable a trigger and define associated parameters to disconnect (with cause code), or hairpin, or whether a message or busy tone is played to the user.

Examples

The following example shows how to enable the call treatment feature with a "hairpin" action:

```
call treatment on
call treatment action hairpin
```

The following example shows how to enable the call treatment feature with a "playmsg" action. The file "congestion.au" plays to the caller when local resources are not available to handle the call.

```
call treatment on
call treatment action playmsg tftp://keyer/prompts/conjestion.au
```

The following example shows how to configure a call treatment cause code to reply with "no-QoS" when local resources are unavailable to process a call:

```
call treatment on
call treatment cause-code no-QoS
```

Related Commands

Command	Description
call threshold	Clears enabled triggers and their associated parameters.
call treatment action	Configures the action that the router takes when local resources are unavailable.
call treatment cause-code	Specifies the reason for the disconnection to the caller when local resources are unavailable.
call treatment isdn-reject	Specifies the rejection cause-code for ISDN calls when local resources are unavailable.
clear call treatment stats	Clears the call treatment statistics.
show call treatment	Displays the call treatment configuration and statistics for handling calls on the basis of resource availability.

call-waiting

To enable call waiting, use the **call-waiting**command in interface configuration mode. To disable call waiting, use the **no** form of this command.

call-waiting
no call-waiting

Syntax Description

This command has no arguments or keywords.

Command Default

Call waiting is enabled.

Command Modes

Interface configuration (config-if)

Command History

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

Usage Guidelines

This command is applicable to Cisco 800 series routers.

You must specify this command when creating a dial peer. This command does not work if it is not specified within the context of a dial peer. For information on creating a dial peer, refer to the *Cisco800SeriesRoutersSoftwareConfigurationGuide*.

Examples

The following example disables call waiting:

```
no call-waiting
```

Related Commands

Command	Description
destination-pattern	Specifies either the prefix, the full E.164 telephone number, or an ISDN directory number (depending on the dial plan) to be used for a dial peer.
dial peer voice	Enters dial peer configuration mode, defines the type of dial peer, and defines the tag number associated with a dial peer.
port (dial peer)	Enables an interface on a PA-4R-DTR port adapter to operate as a concentrator port.
ring	Sets up a distinctive ring for telephones, fax machines, or modems connected to a Cisco 800 series router.
show dial peer voice	Displays configuration information and call statistics for dial peers.

called-number (dial peer)

To enable an incoming Voice over Frame Relay (VoFR) call leg to get bridged to the correct plain old telephone service (POTS) call leg when a static FRF.11 trunk connection is used, use the **callednumber**command in dial peer configuration mode. To disable a static trunk connection, use the **no** form of this command.

called-number *string*
no called-number

Syntax Description

string	A string of digits, including wildcards, that specifies the telephone number of the voice port dial peer.

Command Default

This command is disabled.

Command Modes

Dial peer configuration (config-dial-peer)

Command History

Release	Modification
12.0(4)T	This command was introduced on the Cisco 2600 series and Cisco 3600 series.

Usage Guidelines

The **callednumber**command is used only when the dial peer type is VoFR and you are using the frf11-trunk (FRF.11) session protocol. It is ignored at all times on all other platforms using the Cisco-switched session protocol.

Because FRF.11 does not provide any end-to-end messaging to manage a trunk, the **callednumber**command is necessary to allow the router to establish an incoming trunk connection. The E.164 number is used to find a matching dial peer during call setup.

Examples

The following example shows how to configure a static FRF.11 trunk connection to a specific telephone number (555-0150), beginning in global configuration mode:

```
voice-port 1/0/0
 connection trunk 55Router0
 exit
dial-peer voice 100 pots
 destination pattern 5550150
 exit
dial-peer voice 200 vofr
 session protocol frf11-trunk
 called-number 5550150
 destination pattern 55Router0
```

Related Commands

Command	Description
codec (dial peer)	Specifies the voice coder rate of speech for a VoFR dial peer.
connection	Specifies a connection mode for a voice port.

Command	Description
destination pattern	Specifies either the prefix, the full E.164 telephone number, or an ISDN directory number (depending on the dial plan) to be used for a dial peer.
dtmf relay (VoFR)	Enables the generation of FRF.11 Annex A frames for a dial peer.
fax rate	Establishes the rate at which a fax is sent to the specified dial peer.
preference	Indicates the preferred order of a dial peer within a rotary hunt group.
session protocol	Establishes a session protocol for calls between the local and remote routers via the packet network.
session target	Specifies a network-specific address for a specified dial peer or destination gatekeeper.
signal type	Sets the signaling type to be used when connecting to a dial peer.
vad (dial peer)	Enables VAD for the calls using a particular dial peer.

caller-id (dial peer) through ccm-manager switchover-to-backup

caller-id (dial peer)

To enable caller ID, use the caller - id command in dial peer configuration mode. To disable caller ID, use the **no** form of the command.

caller-id
no caller-id

Syntax Description

This command contains no arguments or keywords.

Command Default

Caller ID is disabled

Command Modes

Dial peer configuration (config-dial-peer)

Command History

Release	Modification
12.1.(2)XF	This command was introduced on the Cisco 800 series routers.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.

Usage Guidelines

This command is available on Cisco 800 series routers that have plain old telephone service (POTS) ports. The command is effective only if you subscribe to caller ID service. If you enable caller ID on a router without subscribing to the caller ID service, caller ID information does not appear on the telephone display.

The configuration of caller ID must match the device connected to the POTS port. That is, if a telephone supports the caller ID feature, use the **callerid** command to enable the feature. If the telephone does not support the caller ID feature, use the command default or disable the caller ID feature. Odd ringing behavior might occur if the caller ID feature is disabled when it is a supported telephone feature or enabled when it is not a supported telephone feature.

Note Specific hardware is required to provide full support for the caller ID features. To determine support for these features in your configuration, review the appropriate hardware documentation and data sheets. This information is available on Cisco.com.

Examples

The following example enables a router to use the caller ID feature:

```
dial-peer voice 1 pots
 caller-id
```

Related Commands

Command	Description
block caller	Configures call blocking on caller ID.
debug pots csm csm	Activates events from which an application can determine and display the status and progress of calls to and from POTS ports.

Command	Description
isdn i-number	Configures several terminal devices to use one subscriber line.
pots call waiting	Enables local call waiting on a router.
registered caller ring	Configures the Nariwake service-registered caller ring cadence.

caller-id alerting dsp-pre-alloc

To statically allocate a digital signal processor (DSP) resource for receiving caller ID information for on-hook (Type 1) caller ID at a receiving Foreign Exchange Office (FXO) voice port, use the **caller-idalertingdsp-pre-alloc** command in voice-port configuration mode. To disable the command's effect, use the **no** form of this command.

caller-id alerting dsp-pre-alloc
no caller-id alerting dsp-pre-alloc

Syntax Description

This command contains no arguments or keywords.

Command Default

No preallocation of DSP resources

Command Modes

Voice-port configuration (config-voiceport)

Command History

Release	Modification
12.1(2)XH	This command was introduced on the Cisco MC3810, Cisco 2600 series, and Cisco 3600 series.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T.

Usage Guidelines

The **calleridalertingdspprealloc** command may be required on an FXO port if the central office uses line polarity reversal to signal the start of caller-ID information transmission. Preallocating a DSP allows the DSP to listen for caller-ID information continuously without requiring an alerting signal from the central office (CO).

This command is the FXO counterpart to the **calleridalertinglinereversal**command, which is applied to the Foreign Exchange Station (sending) end of the caller-ID call.

Note Specific hardware is required to provide full support for the caller ID features. To determine support for these features in your configuration, review the appropriate hardware documentation and data sheets. This information is available on Cisco.com.

Examples

The following example configures a voice port where caller-ID information is received:

```
voice-port 1/0/1
  cptone US
  caller-id enable
  caller-id alerting line-reversal
  caller-id alerting dsp-pre-alloc
```

Related Commands

Command	Description
caller-id alerting line-reversal	Sets the line-reversal method of caller-ID call alerting.

caller-id alerting line-reversal

To set the line-reversal alerting method for caller-ID information for on-hook (Type 1) caller ID at a sending Foreign Exchange Station (FXS) voice port, use the **calleridalertinglinereversal** command in voice-port configuration mode. To disable the command's effect, use the **no** form of this command.

caller-id alerting line-reversal
no caller-id alerting line-reversal

Syntax Description

This command has no arguments or keywords.

Command Default

No line-reversal alert

Command Modes

Voice-port configuration (config-voiceport)

Command History

Release	Modification
12.1(2)XH	This command was introduced.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T.

Usage Guidelines

This command is required only when the telephone device attached to an FXS port requires the line-reversal method to signal the start of a caller-ID transmission. Use it on FXS voice ports that send caller-ID information.

This command is the FXS counterpart to the **calleridalertingdspprealloc** command, which is applied to the FXO (receiving) end of the caller-ID call with the line-reversal alerting method.

Note Specific hardware is required to provide full support for the caller ID features. To determine support for these features in your configuration, review the appropriate hardware documentation and data sheets. This information is available on Cisco.com.

Examples

The following example configures a voice port from which caller-ID information is sent:

```
voice-port 1/0/1
   cptone US
   station name  A. sample
   station number 4085550111
   caller-id alerting line-reversal
   caller-id alerting dsp-pre-alloc
```

Related Commands

Command	Description
caller-id alerting dsp-pre-alloc	At the receiving end of a line-reversal alerting caller-ID call, preallocates DSPs for caller ID calls.

caller-id alerting pre-ring

To set a 250-millisecond prering alerting method for caller ID information for on-hook (Type 1) caller ID at a sending Foreign Exchange Station (FXS) voice port, use the **caller-idalertingpre-ring** command in voice-port configuration mode. To disable the command, use the **no** form of this command.

caller-id alerting pre-ring
no caller-id alerting pre-ring

Syntax Description

This command has no arguments or keywords.

Command Default

No prering alert

Command Modes

Voice-port configuration (config-voiceport)

Command History

Release	Modification
12.1(2)XH	This command was introduced on the Cisco MC3810, Cisco 2600 series, and Cisco 3600 series.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T.

Usage Guidelines

This command is required only when the telephone device attached to an FXS port requires the prering (immediate ring) method to signal the start of caller ID transmission. Use it on FXS voice ports that send caller ID information. This command allows the FXS port to send a short prering preceding the normal ring cadence. On an FXO port, an incoming prering (immediate ring) is simply counted as a normal ring using the **caller-idalertingring** command.

Note Specific hardware is required to provide full support for the caller ID features. To determine support for these features in your configuration, review the appropriate hardware documentation and data sheets. This information is available on Cisco.com.

Examples

The following example configures a voice port from which caller ID information is sent:

```
voice-port 1/0/1
   cptone US
   station name  A. sample
   station number 4085550111
   caller-id alerting pre-ring
```

Related Commands

Command	Description
caller-id alerting line-reversal	Enables caller ID operation and sets the line-reversal alerting type at an FXS port.
caller-id alerting ring	Enables caller ID operation and sets an alerting ring type at an FXO or FXS port.

caller-id alerting ring

To set the ring-cycle method for receiving caller ID information for on-hook (Type 1) caller ID at a receiving Foreign Exchange Office (FXO) or a sending Foreign Exchange Station (FXS) voice port, use the **calleridalertingring** command in voice-port configuration mode. To set the command to the default, use the **no** form of this command.

caller-id alerting ring {**1** | **2**}
no caller-id alerting ring

Syntax Description

1	Use this setting if your telephone service provider specifies it to provide caller ID alerting (display) after the first ring at the receiving station. This is the most common setting.
2	Use this setting if your telephone service provider specifies it to provide caller ID alerting (display) after the second ring. This setting is used in Australia, where the caller ID information is sent following two short rings (double-pulse ring).

Command Default

1

Command Modes

Voice-port configuration (config-voiceport)

Command History

Release	Modification
12.1(2)XH	This command was introduced.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T.

Usage Guidelines

This setting is determined by the Bellcore/Telcordia or ETSI standard that your telephone service provider uses for caller ID. Use it on FXO loop-start and ground-start voice ports where caller ID information arrives and on FXS voice ports from which caller ID information is sent.

This setting must match on the sending and receiving ends of the telephone line connection.

Note Specific hardware is required to provide full support for the caller ID features. To determine support for these features in your configuration, review the appropriate hardware documentation and data sheets. This information is available on line.

Examples

The following example configures a voice port where caller ID information is received:

```
voice-port 1/0/1
   cptone US
   caller-id alerting ring 1
```

The following example configures a voice port from which caller ID information is sent:

```
voice-port 1/0/1
```

```
cptone northamerica
station name A. sample
station number 4085550111
caller-id alerting ring 1
```

Related Commands

Command	Description
caller-id alerting line-reversal	Enables caller ID operation and sets the line-reversal alerting type at an FXS port.
caller-id alerting pre-ring	Enables caller ID operation and sets the pre-ring alerting method at an FXS port.

caller-id attenuation

To set the attenuation for caller ID at a receiving Foreign Exchange Office (FXO) voice port, use the **caller-idattenuation** command in voice-port configuration mode. To set the command to the default, use the **no** form of this command.

caller-id attenuation [*attenuation*]
no caller-id attenuation

Syntax Description

attenuation	(Optional) specifies the attenuation, in decibels (dB). Range is from 0 to 64. The default is 14.

Command Default

The default value is 14 dB, signal level of -14 dBm.

Command Modes

Voice-port configuration (config-voiceport)

Command History

Release	Modification
12.1(2)XH	This command was introduced.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T.

Usage Guidelines

Use this setting to specify the attenuation for a caller ID FXO port. If the setting is not used, the attenuation is set to 14 dB, signal level of -14 dBm.

Note Specific hardware is required to provide full support for the caller ID features. To determine support for these features in your configuration, review the appropriate hardware documentation and data sheets. This information is available on line.

Examples

The following example configures a voice port where caller ID information is received:

```
voice-port 1/0/1
   cptone US
   caller-id attenuation 0
```

caller-id block

To request the blocking of the display of caller ID information at the far end of a call from calls originated at a Foreign Exchange Station (FXS) port, use the **caller-idblock** command in voice-port configuration mode at the originating FXS voice port. To allow the display of caller ID information, use the **no**form of this command.

caller-id block
no caller-id block

Syntax Description

This command has no arguments or keywords.

Command Default

No blocking of caller ID information

Command Modes

Voice-port configuration (config-voiceport)

Command History

Release	Modification
12.1(2)XH	This command was introduced.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T.

Usage Guidelines

This command is used on FXS voice ports that are used to originate on-net telephone calls. This command affects all calls sent to a far-end FXS station from the configured originating FXS station. Calling number and called number are provided in the H.225 setup message for VoIP, through the H.225 Octet 3A field. Calling name information is included in a display information element.

Note Cisco-switched calls using Voice over Frame Relay (VoFR) and Voice over ATM (VoATM) carry calling party information in the Cisco proprietary setup message. For standards-based, point-to-point VoFR (FRF.11) trunks where transparent signaling is applied for FXS-to-FXO calls, only pass-through of in-band automatic number identification (ANI) is supported. ANI information is always unblocked for these communications. Interface technology using transparent channel-associated signaling (CAS) can support only ANI through Feature Group D (in-band MF signaling). The Caller ID feature cannot be used with fixed point-to-point trunk connects created using the **connectiontrunk** command.

Note Specific hardware is required to provide full support for the caller ID features. To determine support for these features in your configuration, review the appropriate hardware documentation and data sheets. This information is available on Cisco.com.

Examples

The following example configures a voice port from which caller ID information is sent:

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

```
station name A. sample
station number 4085550111
caller-id block
```

Related Commands

Command	Description
caller-id enable	Enables caller ID operation.

caller-id enable

To allow the sending or receiving of caller-ID information, use the **caller-idenable** command in voice-port configuration mode at the sending foreign exchange station (FXS) voice port or the receiving foreign exchange office (FXO) voice port. To disable the sending and receiving of caller-ID information, use the **no** form of this command.

caller-id enable [**type** {**1** | **2**}]
no caller-id enable [**type** {**1** | **2**}]

Syntax Description

type	(Optional) Indicates that the following keyword is a caller-ID type. • **1** --Type I only. Type I transmits the signal when the receiving phone is on hook. • **2** --Type II only. Type II transmits the signal when the receiving phone is off hook, for instance to display the caller ID of an incoming call when the receiving phone is busy (call-waiting caller ID).

Command Default

The sending and receiving of caller-ID information is disabled.

Command Modes

Voice-port configuration (config-voiceport)

Command History

Release	Modification
12.1(2)XH	This command was introduced.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T.
12.3(7)T	The **type1** and **type2** keywords were added.

Usage Guidelines

This command applies to FXS voice ports that send caller-ID information and to FXO ports that receive caller-ID information. Calling number and called number are provided in the H.225.0 setup message for VoIP through the H.225.0 Octet 3A field. Calling name information is included in a display information element.

Some users that do not have caller ID type II support on their phones hear noise when type II caller ID is enabled. The **caller-idenabletype1** command allows only type I on the voice port and disables type II, so that the user does not hear this noise.

If this command is used without the optional **type** keyword, both type I and type II caller ID are enabled.

Note The **no** form of this command also clears all other caller-ID configuration settings for the voice port.

Note Cisco-switched calls using Voice over Frame Relay (VoFR) and Voice over ATM (VoATM) carry calling-party information in the Cisco-proprietary setup message. For standards-based, point-to-point VoFR (FRF.11) trunks where transparent signaling is applied for FXS-to-FXO calls, only pass-through of in-band automatic number identification (ANI) is supported. ANI information is always unblocked for these communications. Interface technology using transparent channel-associated signaling (CAS) can support only ANI through Feature Group D (in-band multifrequency signaling). Caller ID cannot be used with fixed point-to-point trunk connections created using the **connectiontrunk** command.

If the **stationname,stationnumber**, or a **caller-idalerting** command is configured on the voice port, caller ID is automatically enabled, and the **caller-idenable** command is not necessary.

Note Specific hardware is required to provide full support for the caller-ID features. To determine support for these features in your configuration, review the appropriate hardware documentation and data sheets. This information is available on line.

Examples

The following example configures a Cisco 2600 series or Cisco 3600 series router voice port at which caller-ID information is received:

```
voice-port 1/0/1
 cptone US
 caller-id enable
```

The following example configures a Cisco 2600 series or Cisco 3600 series router voice port from which caller-ID information is sent:

```
voice-port 1/0/1
 cptone northamerica
 station name A. sample
 station number 4085550111
 caller-id enable
```

The following example enables only type I caller ID on port 2/0:

```
voice-port 2/0
 caller-id enable type 1
```

Related Commands

Command	Description
caller-id alerting line-reversal	Enables caller ID operation and sets the line-reversal alerting type at an FXS port.
caller-id alerting pre-ring	Enables caller ID operation and sets the pre-ring alerting method at an FXS port.
caller-id alerting ring	Enables caller ID operation and sets an alerting ring type at an FXO or FXS port.
caller-id block	Disables the sending of caller ID information from an FXS port.

Command	Description
station name	Enables caller ID operation and sets the name sent from an FXS port.
station number	Enables caller ID operation and sets the number sent from an FXS port.

caller-id mode

To specify a noncountry, standard caller ID mode, use the **caller-id mode** command in voice port configuration mode at the sending Foreign Exchange Station (FXS) voice port or at the receiving Foreign Exchange Office (FXO) voice port. To allow the caller-ID mode to be country-specific, use the **no** form of this command.

caller-id mode {**BT** | **FSK** | **DTMF** {**start** | **end**}{**#** | ***** | **A** | **B** | **C** | **D**}}
no caller-id mode

Syntax Description

BT	Specifies Frequency-Shift Keying (FSK) with Dual Tone Alerting Signal (DTAS) used by British Telecom.
FSK	Specifies FSK before or during a call.
DTMF	Specifies dual tone multifrequency (DTMF) digits with the start and end digit codes.
start	Specifies the start digit code.
end	Specifies the end digit code.
#	Specifies the DTMF digit #.
*****	Specifies the DTMF digit *.
A	Specifies the DTMF digit A.
B	Specifies the DTMF digit B.
C	Specifies the DTMF digit C.
D	Specifies the DTMF digit D.

Command Default

The caller-ID mode is disabled.

Command Modes

Voice port configuration (config-voiceport)

Command History

Release	Modification
15.2(1)T	This command was introduced.

Usage Guidelines

This command applies to FXS voice ports that send caller ID information to FXO ports that receive the caller ID information. The start and end digit codes are applicable only for the DTMF mode.

The command default is based on the cptone setting that specifies a regional voice-interface-related tone, ring, and cadence setting. The **no** form of this command defaults to a country-specific setting.

Note Specific hardware is required to provide full support for the caller-ID features. To determine support for these features in your configuration, review the appropriate hardware documentation and data sheets. This information is available on Cisco.com.

Examples

The following example configures a noncountry, standard caller ID mode of DTMF with a start code and end code:

```
Device> enable
Device# configure terminal
Device(config)# voice-port 1/0/1
Device(config-voiceport)# caller-id mode DTMF start A end B
Device(config-voiceport)# end
```

Related Commands

Command	Description
caller-id alerting	Defines the caller ID alerting method.
caller-id attenuation	Configures the attenuation for a caller ID FXO voice port.
caller-id block	Blocks caller ID information.
caller-id enable	Enables caller ID operation.
caller-id format	Specifies the caller ID format.

cancel-call-waiting

To define a feature code for a Feature Access Code (FAC) to enable the Cancel Call Waiting feature, use the **cancel-call-waiting**command in STC application feature access-code configuration mode. To reset the feature code to its default, use the **no** form of this command.

cancel-call-waiting *keypad-character*
no cancel-call-waiting

Syntax Description

keypad-character	Character string that can be dialed on a telephone keypad (0-9, *, #). Default: 8. The string can be any of the following: • A single character (0-9, *, #) • Two digits (00-99) • Two to four characters (0-9, *, #) and the leading or ending character must be an asterisk (*) or number sign (#)

Command Default

Feature code for Cancel Call Waiting is 8.

Command Modes

STC application feature access-code configuration (config-stcapp-fac)

Command History

Release	Modification
15.0(1)XA	This command was introduced.
15.1(1)T	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command changes the default value of the feature code for Cancel Call Waiting (8).

If you attempt to configure this command with a value that is already configured for another FAC, speed-dial code, or the Redial FSD, you receive a message. If you configure a duplicate code, the system implements the first matching feature in the order of precedence shown in the output of the **showstcappfeaturecodes** command.

If you attempt to configure this command with a value that precludes or is precluded by another FAC, speed-dial code, or the Redial FSD, you receive a message. If you configure a feature code to a value that precludes or is precluded by another code, the system always executes the call feature with the shortest code and ignores the longer code. For example, #1 will always preclude #12 and #123. You must configure a new value for the precluded code in order to enable phone user access to that feature.

To display a list of all FACs, use the **showstcappfeaturecodes** command.

Examples

The following example shows how to change the value of the feature code for cancel call waiting. With this configuration, a phone user must press **9 on the phone keypad to cancel call waiting.

```
Router(config)# stcapp feature access-code
Router(config-stcapp-fac)# cancel-call-waiting **9
```

Related Commands

Command	Description
prefix (stcapp-fac)	Defines the prefix for FACs.
show stcapp feature codes	Displays all FACs.

caller-number (dial peer)

To associate a type of ring cadence with a specific caller ID, use the **callernumber**command in dial peer voice configuration mode. To disable the type of ring cadence for a specific caller ID, use the **no** form of this command.

caller-number *number* **ring** *cadence*
no caller-number *number* **ring** *cadence*

Syntax Description

number	Caller ID for which the user wants to set the cadence. Twenty numbers along with their respective cadences may be set for each of the plain old telephone service (POTS) ports.
ring *cadence*	Ring cadence level. The three cadence levels (0, 1, and 2), which differ in duration and cadence, are as follows: • **0** --The ring cadence is 1 second on and 2 seconds off (NTT-defined regular ring). • **1** --The ring cadence is 0.25 seconds on, 0.2 seconds off, 0.25 seconds on, and 2.3 seconds off (NTT-defined nonregular ring). • **2** --The ring cadence is 0.5 seconds on, 0.25 seconds off, 0.25 seconds on, and 2 seconds off (Cisco-defined nonregular ring).

Command Default

The router does not associate any caller ID with a cadence level. Therefore, there is no distinctive ring.

Command Modes

Dial peer voice configuration (config-dial-peer)

Command History

Release	Modification
12.2(8)T	This command was introduced on the Cisco 803, Cisco 804, and Cisco 813 routers.

Usage Guidelines

You can enter the **callernumber** command for each POTS port. A maximum of 20 caller IDs can be associated with distinct ring cadences. After 20 numbers per port have been set, you cannot set more numbers (and their ring cadences) for that port until you have removed any of the numbers that have already been set. To remove already-set numbers and their ring cadences, use the**no** form of the **callernumber**command.

The command must be set within each dial peer. Six dial peers are available, you can specify 20 caller IDs per port, for a maximum of 120 caller ID numbers.

Note If you have already subscribed to Nariwake service, the priority goes to the Nariwake caller ID cadence.

To disable distinctive ringing based on a caller ID number, configure the **nocallernumber**command. Disabling the ringing removes the specific cadence that has been set for that particular number. If you have set 20 numbers and their ring cadences, you need to set the **nocallernumber**command for each of the 20 numbers.

Use the **showrunningconfig** command to check distinctive ringing status.

Examples

The following output examples show that three caller ID numbers and their ring cadences have been set for POTS port 1 and that five caller ID numbers and their ring cadences have been set for POTS port 2:

```
dial-peer voice 1 pots
 destination-pattern 5550102
 port 1
 no call-waiting
 ring 0
 volume 4
 caller-number 1111111 ring 2
 caller-number 2222222 ring 1
 caller-number 3333333 ring 1
dial-peer voice 2 pots
 destination-pattern 5550110
 port 2
 no call-waiting
 ring 0
 volume 2
 caller-number 4444444 ring 1
 caller-number 6666666 ring 2
 caller-number 7777777 ring 0
 caller-number 8888888 ring 1
 caller-number 9999999 ring 2
```

Related Commands

Command	Description
call waiting	Enables call waiting.
volume	Configures the receiver volume level in the router.

calling-info pstn-to-sip

To specify calling information treatment for public switched telephone network (PSTN) to Session Initiation Protocol (SIP) calls, use the **calling-infopstn-to-sip** command in SIP user agent configuration mode. To disable calling information treatment for PSTN-to-SIP calls, use the **no** form of this command.

calling-info pstn-to-sip {**unscreened discard** | {**from** | **remote-party-id** | **asserted-id** {**name set** *name* | **number set** *number*}}}
no calling-info pstn-to-sip

Syntax Description

unscreened discard	(Optional) Specifies that the calling name and number be discarded.
from name set *name*	(Optional) Specifies that the display-name of the From header is unconditionally set to the configured ASCII string in the forwarded INVITE message.
from number set *number*	(Optional) Specifies that the user part of the From header is unconditionally set to the configured ASCII string in the forwarded INVITE message.
remote-party-id name set *name*	(Optional) Specifies that the display-name of the Remote-Party-ID header is unconditionally set to the configured ASCII string in the forwarded INVITE message.
remote-party-id number set *number*	(Optional) Specifies that the user part of the Remote-Party-ID header is unconditionally set to the configured ASCII string in the forwarded INVITE message.
asserted-id name *set name*	(Optional) Specifies that the display-name in the Asserted-ID header is unconditionally set to the configured ASCII string in the forwarded INVITE message.
asserted-id number *set number*	(Optional) Specifies that the user part in the Asserted-ID header is unconditionally set to the configured ASCII string in the forwarded INVITE message.

Command Default

This command is disabled.

Command Modes

SIP UA configuration (config-sip-ua)

Command History

Release	Modification
12.2(13)T	This command was introduced.
12.4(15)T	The **asserted-id** keyword was added.

Usage Guidelines

When a call exits the gateway, the **calling-infopstn-to-sip** treatments are applied.

Examples

The following example enables calling information treatment for PSTN-to-SIP calls and sets the company name and number:

```
Router(config-sip-ua)# calling-info pstn-to-sip from name set CompanyA
Router(config-sip-ua)# calling-info pstn-to-sip from number set 5550101
Router(config-sip-ua)# exit
Router(config)# exit
Router# show running-config
Building configuration...
.
.
.
!
sip-ua
 calling-info pstn-to-sip from name set CompanyA
 calling-info pstn-to-sip from number set 5550101
 no remote-party-id
!
.
.
.
```

Related Commands

Command	Description
asserted-id	Sets the privacy level and enables either P-Asserted-Identity (PAI) or P-Preferred-Identity (PPI) privacy headers in outgoing SIP requests or response messages.
calling-info sip-to-pstn	Specifies calling information treatment for SIP-to-PSTN calls.
debug ccsip events	Enables tracing of SIP SPI events.
debug ccsip messages	Enables tracing SIP messages exchanged between the SIP UA client and the access server.
debug isdn q931	Displays call setup and teardown of ISDN connections.
debug voice ccapi error	Enables tracing error logs in the call control API.
debug voip ccapi in out	Enables tracing the execution path through the call control API.

calling-info sip-to-pstn

To specify calling information treatment for Session Initiation Protocol (SIP) to public switched telephone network (PSTN) calls, use the **calling-infosip-to-pstn** command in SIP user agent configuration mode. To disable calling information treatment for SIP-to-PSTN calls, use the **no** form of this command.

calling-info sip-to-pstn {**unscreened discard** | **name set** *name* | **number set** *number*}
no calling-info sip-to-pstn

Syntax Description

unscreened *discard*	(Optional) Specifies that the calling name and number be discarded.
name set *name*	(Optional) Specifies that the calling name be unconditionally set to the configured ASCII string in the forwarded Setup mesage.
number set *number*	(Optional) Specifies that the calling number be unconditionally set to the configured ASCII string in the forwarded Setup message.

Command Default

This command is disabled.

Command Modes

SIP user agent configuration (config-sip-ua)

Command History

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

Usage Guidelines

When a call enters the gateway, the **calling-infosip-to-pstn** treatments are applied.

Examples

The following example enables calling information treatment for SIP-to-PSTN calls and sets the company name to CompanyA and the number to 5550100:

```
Router(config-sip-ua)# calling-info sip-to-pstn name set CompanyA
Router(config-sip-ua)# calling-info sip-to-pstn number set 5550100
Router(config-sip-ua)# exit
Router(config)# exit
Router# show running-config
Building configuration...
.
.
.
!
sip-ua
 calling-info sip-to-pstn name set CompanyA
  calling-info sip-to-pstn number set 5550100
!
.
.
.
```

Related Commands

Command	Description
debug ccsip events	Enables tracing of SIP SPI events.
debug ccsip messages	Enables SIP SPI message tracing.
debug isdn q931	Displays call setup and teardown of ISDN connections.
debug voip ccapi in out	Enables tracing the execution path through the call control API.
calling-info pstn-to-sip	Specifies calling information treatment for PSTN-to-SIP calls.

calling-number outbound

To specify automatic number identification (ANI) to be sent out when T1-channel-associated signaling (T1-CAS) Feature Group D-Exchange Access North American (FGD-EANA) is configured as the signaling type, use the **calling-numberoutbound** command in dial peer or voice-port configuration mode. To disable this command, use no form of this command.

calling-number outbound {**range** *string1 string2* | **sequence** *string1* . . . *string5* | **null**}
no calling-number outbound {**range** *string1 string2* | **sequence** *string1* . . . *string5* | **null**}

Syntax Description

range	Generates the sequence of ANI by rotating through the specified range (*string1* to *string2*).
sequence	Configures a sequence of discrete strings (*string1*...*string5*) to be passed out as ANI for successive calls using the peer **Note** The ellipses (**...**) is entered as shown above.
null	Suppresses ANI. If used, no ANI is passed when this dial peer is selected.
string# **...**	Valid E.164 telephone number strings. Strings must be of equal length and cannot be more than 32 digits long.

Command Default

No outbound calling number is specified.

Command Modes

Dial peer configuration (config-dial-peer)
Voice-port configuration (config-voiceport)

Command History

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

Usage Guidelines

This command is effective only for FGD-EANA signaling.

Examples

Use the **calling-numberoutbound** command to enable or disable the passing of ANI on a T1-CAS FGD-EANA configured T1 interface for outgoing calls. Syntax for this command is the same for both voice-port mode and dial peer mode. Examples are given for both modes.

calling-number outbound Range

```
calling-number outbound range
string1

string2
```

The values *string1* and *string2* are valid E.164 telephone number strings. Both strings must be of the same length and cannot be more than 32 digits long. Only the last four digits are used for specifying the range (*string1* to *string2*) and for generating the sequence of ANI by rotating through the range until *string2* is reached and then starting from *string1* again. If strings are fewer than four digits in length, then entire strings are used.

ANI is generated by using the 408555 prefix and by rotating through 0100 to 0101 for each call using this peer.

Dial peer configuration mode:

```
dial-peer voice 1 pots
 calling-number outbound range 4085550100 4085550101
 calling Number Outbound is effective only for fgd_eana signaling
```

Voice-port configuration mode:

```
voice-port 1:D
 calling-number outbound range 4085550100 4085550105
 Calling Number Outbound is effective only for fgd_eana signaling
```

calling-number outbound Sequence

```
calling-number outbound sequence
string1 string2 string3
string4 string5
```

This option configures a sequence of discrete strings (*string1... string5*) to be passed out as ANI for successive calls using the peer. The limit is five strings. All strings must be valid E.164 numbers, up to 32 digits in length.

Dial peer configuration mode:

```
dial-peer voice 1 pots
 calling-number outbound sequence 6000 6006 4000 5000 5025
 Calling Number Outbound is effective only for fgd_eana signaling
```

Voice-port configuration mode:

```
voice-port 1:D
 calling-number outbound sequence 6000 6006 4000 5000 5025
 Calling Number Outbound is effective only for fgd_eana signaling
```

calling-number outbound Null

```
calling-number outbound null
```

This option suppresses ANI. If used, no ANI is passed when this dial peer is selected.

Dial peer configuration mode:

```
dial-peer voice 1 pots
 calling-number outbound null
 Calling Number Outbound is effective only for fgd_eana signaling
```

Voice-port configuration mode:

```
voice-port 1:D
 calling-number outbound null
 Calling Number Outbound is effective only for fgd_eana signaling
```

Related Commands

Command	Description
info-digits string1	Configures two information digits to be prepended to the ANI string.

capacity update interval (dial peer)

To change the capacity update for prefixes associated with this dial peer, use the **capacityupdateinterval** command in dial peer configuration mode. To return to the default, use the **no** form of this command.

capacity update interval *seconds*
no capacity update interval *seconds*

Syntax Description

seconds	Interval, in seconds, between the sending of periodic capacity updates. This can be a number in the range 10 to 1000. The default value is 25 seconds.

Command Default

25 seconds

Command Modes

Dial peer configuration (config-dial-peer)

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

The update interval should be set depending how many updates that are sent. Updates are sent more often when more calls are coming in, which can lead to data getting out of synchrony. If the interval is too short for the number of updates, the location server can be overwhelmed.

If a dial peer gets too much traffic, set the *seconds* argument to a higher value.

Examples

The following example shows that POTS dial peer 10 is having the capacity update occur every 35 seconds:

```
Router(config)# dial-peer voice 10 pots
Router(config-dial-peer)# capacity update interval 35
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice-related encapsulation.

capacity update interval (trunk group)

To change the capacity update for carriers or trunk groups, use the **capacityupdateinterval** command in trunk group configuration mode. To return to the default, use the **no** form of this command.

capacity {**carrier** | **trunk-group**} **update interval** *seconds*
no capacity {**carrier** | **trunk-group**}

Syntax Description

carrier	Carrier capacity.
trunk-group	Trunk group capacity.
seconds	Interval, in seconds, between the sending of periodic capacity updates. This can be a number in the range 10 to 1000. The default value is 25 seconds.

Command Default

25 seconds

Command Modes

Trunk group configuration (config-trunkgroup)

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

The update interval should be set depending how many updates that are sent. Updates are sent more often when more calls are coming in, which can lead to data getting out of synchrony. If the interval is too short for the number of updates, the location server can be overwhelmed.

If a dial peer gets too much traffic, set the *seconds* argument to a higher value.

Examples

The following example sets the capacity update for trunk group 101 to occur every 45 seconds:

```
Router(config)# trunk group 101
Router(config-trunkgroup)# capacity trunk-group update interval 45
```

Related Commands

Command	Description
trunk group	Defines the trunk group and enters trunk group configuration mode.

cap-list vfc

To add a voice codec overlay file to the capability file list, use the **cap-listvfc**command in global configuration mode. To disable a particular codec overlay file that has been added to the capability list, use the **no** form of this command.

cap-list *filename* **vfc** *slot-number*
no **cap-list** *filename* **vfc** *slot-number*

Syntax Description

filename	Identifies the codec file stored in voice feature card (VFC) flash memory.
slot -number	Identifies the slot where the VFC is installed. Range is 0 to 2. There is no default value.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

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

Usage Guidelines

When VCWare is unbundled, it automatically adds DSPWare to flash memory, creates both the capability and default file lists, and populates these lists with the default files for the particular version of VCWare. The capability list defines the available voice codecs for H.323 capability negotiation. Use the **cap-listvfc**command to add the indicated voice codec overlay file (defined by*filename*) to the capability file list in flash memory.

Examples

The following example adds the following codec to the list included in flash memory:

```
config terminal
 cap-list cdc-g711-1.0.14.0.bin vfc 0
```

Related Commands

Command	Description
default-file vfc	Specifies an additional (or different) file from the ones in the default file list and stored in VFC Flash memory.

capf-address

To specify the Certificate Authority Proxy Function (CAPF) for a locally significant certificate (LSC) update, use the **capf-address** command in phone proxy configuration mode. To remove the CAPF for an LSC update, use the **no** form of the command.

capf-address ipv4 *capf-ipv4-address* **acc-addr ipv4** *access-ipv4-address*
no capf-address ipv4 *capf-ipv4-address* **acc-addr ipv4** *access-ipv4-address*

Syntax Description

capf-ipv4-address	Specifies the IPv4 address as the local address for the CAPF service.
acc-addr ipv4 *access-ipv4-address*	Specifies the access side address used as a CAPF server address.

Command Default

No CAPF address is specified.

Command Modes

Phone proxy configuration mode (config-phone-proxy)

Command History

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

Usage Guidelines

Example

The following example shows how to specify a CAPF address for an LSC update. The IPv4 address for the for the CAPF service is 198.51.100.101 and the access side address is 192.168.0.109:

```
Device(config)# voice-phone-proxy first-pp
Device(config-phone-proxy)# capf-addr ipv4 198.51.100.101 acc-addr ipv4 192.168.0.109
```

card type (T1-E1)

To configure a T1 or E1 card type, use the **cardtype** command in global configuration mode. To deselect the card type on non-SPA platforms, use the **no** form of this command. The no form of this command is not available on the SPA platforms.

card type {**t1** | **e1**} *slot* [*bay*]
no card type {**t1** | **e1**} *slot* [*bay*]

Channelized T1/E1 Shared Port Adapters
card type {**t1** | **e1**} *slot subslot*

Syntax Description

t1	Specifies T1 connectivity of 1.544 Mbps through the telephone switching network, using AMI or B8ZS coding.
e1	Specifies a wide-area digital transmission scheme used predominantly in Europe that carries data at a rate of 2.048 Mbps.
slot	Chassis slot number. Refer to the appropriate hardware manual for slot information. For SIPs, refer to the platform-specific SPA hardware installation guide or the corresponding "Identifying Slots and Subslots for SIPs and SPAs" topic in the platform-specific SPA software configuration guide.
bay	(Optional) Card interface bay number in a slot (route switch processor [RSP] platform only). This option is not available on other platforms.
subslot	(Channelized T/E1 Shared Port Adapters Only) Secondary slot number on a SPA interface processor (SIP) where a SPA is installed. Refer to the platform-specific SPA hardware installation guide and the corresponding "Specifying the Interface Address on a SPA" topic in the platform-specific SPA software configuration guide for subslot information.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

Release	Modification
12.0(5)XE	This command was introduced.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0(7)T.
12.3(1)	This command was integrated into Cisco IOS Release 12.3(1) and support was added for Cisco 2610XM, Cisco 2611XM, Cisco 2620XM, Cisco 2621XM, Cisco 2650XM, Cisco 2651XM, Cisco 2691, Cisco 3631, Cisco 3660, Cisco 3725, and Cisco 3745 platforms.
12.2S	This command was integrated into Cisco IOS Release 12.2S.

Release	Modification
12.2(18)SXE	This command was integrated into Cisco IOS Release 12.2(18)SXE to support SPAs on the Cisco 7600 series routers and Catalyst 6500 series switches.
12.0(31)S	This command was integrated into Cisco IOS Release 12.0(31)S to support SPAs on Cisco 12000 series routers.
12.2(33)SRA	This command was integrated into Cisco IOS Release 12.2(33)SRA.
XE 3.18SP	This command was integrated into Cisco NCS 4200 Series.

Usage Guidelines

Changes made using this command on non-SPA platforms, do not take effect unless the **reload** command is used or the router is rebooted.

Channelized T1/E1 Shared Port Adapters

There is no card type when the SPA is inserted for first time. The user must configure this command before they can configure individual ports.

The no form of this command is not available on the SPA platforms. To change an existing card type on SPA platforms, perform the following steps:

1. Remove the SPA from its subslot.
2. Save the configuration.
3. Reboot the router.
4. Insert the new SPA into the subslot.
5. Configure the new card using this command.

Examples

The following example configures T1 data transmission on slot 1 of the router:

```
Router(config)# card type t1 1
```

The following example configures all ports of an 8-Port Channelized T1/E1 SPA, seated in slot 5, subslot 2, in T1 mode:

```
Router(config)# card type t1 5 2
```

Related Commands

Command	Description
controller	Configures a T1 or E1 controller and enters controller configuration mode.
reload	Reloads the operating system.
show controller	Displays the controller state that is specific to controller hardware
show interface serial	Displays the serial interface type and other information.

card type (T3-E3)

To configure a T3 or E3 card type, use the **cardtype** command in global configuration mode. To deselect the card type, use the **no** form of this comand. The no form of this command is not supported on the 2-Port and 4-Port Clear Channel T3/E3 SPA on Cisco 12000 series routers.

T3 or E3 Controllers
card type {**t3** | **e3**} *slot*
no card type {**t3** | **e3**} *slot*

Clear Channel T3/E3 Shared Port Adapters
card type {**t3** | **e3**} *slot subslot*
no card type {**t3** | **e3**} *slot subslot*

Clear Channel T3/E3 Shared Port Adapters on Cisco 12000 Series Routers
card type {**t3** | **e3**} *slot subslot*

Syntax Description

t3	Specifies T3 connectivity of 44210 kbps through the network, using B8ZS coding.
e3	Specifies a wide-area digital transmission scheme used predominantly in Europe that carries data at a rate of 34010 kbps.
slot	Slot number of the interface.
subslot	(Clear Channel T3/E3 Shared Port Adapters Only) Secondary slot number on a SIP where a SPA is installed. Refer to the platform-specific SPA hardware installation guide and the corresponding "Specifying the Interface Address on a SPA" topic in the platform-specific SPA software configuration guide for subslot information.

Command Default

No default behavior or values.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(1)T	This command was introduced.
12.2(11)YT	This command was integrated into Cisco IOS Release 12.2(11)YT and implemented on the following platforms: Cisco 2650XM, Cisco 2651XM, Cisco 2691, Cisco 3660 series, Cisco 3725, and Cisco 3745 routers.
12.2(15)T	This command was integrated into Cisco IOS Release 12.2(15)T.
12.3(1)	This command was integrated into Cisco IOS Release 12.3(1) and support was added for Cisco 2610XM, Cisco 2611XM, Cisco 2620XM, Cisco 2621XM, Cisco 2650XM, Cisco 2651XM, Cisco 2691, Cisco 3631, Cisco 3660, Cisco 3725, and Cisco 3745 platforms.
12.2S	This command was integrated into Cisco IOS Release 12.2S.

Release	Modification
12.2(25)S3	This command was integrated into Cisco IOS Release 12.2(25)S3 to support SPAs on the Cisco 7304 routers.
12.2(18)SXE	This command was integrated into Cisco IOS Release 12.2(18)SXE to support SPAs on the Cisco 7600 series routers and Catalyst 6500 series switches.
12.0(31)S	This command was integrated into Cisco IOS Release 12.0(31)S to support SPAs on the Cisco 12000 series routers.
12.2(33)SRA	This command was integrated into Cisco IOS Release 12.2(33)SRA.

Usage Guidelines

Usage guidelines vary slightly from platform to platform as follows:

T3 or E3 Controllers

Once a card type is issued, you enter the **nocardtype** command and then another **cardtype** command to configure a new card type. You must save the configuration to the NVRAM and reboot the router in order for the new configuration to take effect.

When the router comes up, the software comes up with the new card type. Note that the software will reject the configuration associated with the old controller and old interface. You must configure the new controller and serial interface and save it.

Clear Channel T3/E3 Shared Port Adapters

To change all the SPA ports from T3 to E3, or vice versa, you enter the **nocardtype** command and then another **cardtype** command to configure a new card type.

When the router comes up, the software comes up with the new card type. Note that the software will reject the configuration associated with the old controller and old interface. You must configure the new controller and serial interface and save it.

Clear Channel T3/E3 Shared Port Adapters on Cisco 12000 Series Routers

The no form of this command is not available on the 2-Port and 4-Port Clear Channel T3/E3 SPA on Cisco 12000 series routers. To change an existing card type on Cisco 12000 series routers, perform the following steps:

1. Remove the SPA from its subslot.
2. Save the configuration.
3. Reboot the router.
4. Insert the new SPA into the subslot.
5. Configure the new card using this command.

Examples

The following example shows T3 data transmission configured in slot 1:

```
Router(config)# card type t3 1
```

The following example configures all ports of 2-Port and 4-Port Clear Channel T3/E3 SPA, seated in slot 5, subslot 2, in T3 mode:

```
Router(config)# card type t3 5 2
```

Related Commands

Command	Description
controller	Configures a T3 or E3 controller and enters controller configuration mode.
reload	Reloads the operating system.
show interface serial	Displays the serial interface type and other information.

carrier-id (dial peer)

To specify the carrier associated with a VoIP call in a dial peer, use the **carrier-id**command in dial peer configuration mode. To delete the source carrier ID, use the **no** form of this command.

carrier-id {**source** | **target**} *name*
no carrier-id {**source** | **target**} *name*

Syntax Description

source	Indicates the carrier that the dial peer uses as a matching key for inbound dial-peer matching.
target	Indicates the carrier that the dial peer uses as a matching key for outbound dial-peer matching.
name	Specifies the ID of the carrier to use for the call. Valid carrier IDs contain a maximum of 127 alphanumeric characters.

Command Default

No default behavior or values

Command Modes

Dial peer configuration (config-dial-peer)

Command History

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

Usage Guidelines

A Gatekeeper Transaction Message Protocol (GKTMP) route server-based application at the terminating gateway uses the source carrier ID to select a target carrier that routes the call over a plain old telephone service (POTS) line.

The terminating gateway uses the target carrier ID to select a dial peer for routing the call over a POTS line.

For IP-to-IP calls, the **carrier-id**command alone is not an oubound dialpeer match criterion.

Examples

The following example indicates that dial peer 112 should use carrier ID "east17" for outbound dial-peer matching in the terminating gateway:

```
Router(config)# dial-peer voice 112 pots
Router(config-dial-peer)# carrier-id target east17
```

The following example indicates that dial peer 111 should use carrier ID "beta23" for inbound dial-peer matching in the terminating gateway:

```
Router(config)# dial-peer voice 111 voip
Router(config-dial-peer)# carrier-id source beta23
```

Related Commands

Command	Description
translation-profile (dial peer)	Associates a translation profile with a dial peer.

Command	Description
trunkgroup (dial peer)	Assigns a trunk group to a source IP group or dial peer for trunk group label routing.

carrier-id (global)

To set the carrier ID for trunk groups when a local carrier ID is not configured, use the **carrier-id**command in global configuration mode. To disable the carrier ID, use the **no** form of this command.

carrier-id *name* [**cic**]
no carrier-id *name* [**cic**]

Syntax Description

name	Identifier for the carrier ID. Must be four-digit numeric carrier identification code to be advertised as a TRIP carrier family but can be alphanumeric if used otherwise.
cic	(Optional) Specifies that the carrier ID is a circuit identification code (CIC).

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

To advertise the carrier as a TRIP carrier family, the **cic** keyword must be used. When the **cic** keyword is used, only numeric values can be accepted for the *name* value. If the **cic**keyword is not used, the *name* value can be alphanumeric but is not advertised to TRIP location servers.

Examples

The following example shows a carrier ID using the circuit identification code:

```
Router(config)# carrier-id 1234 cic
```

Related Commands

Command	Description
carrier-id (trunk group)	Configures the carrier ID locally on the trunk group.

carrier-id (trunk group)

To specify the carrier associated with a trunk group, use the **carrier-id**command in trunk group configuration mode. To delete the source carrier ID, use the **no** form of this command.

carrier-id *name* [**cic**]
no carrier-id *name* [**cic**]

Syntax Description

name	The ID of the carrier to use for the call. Valid carrier IDs contain a maximum of 127 alphanumeric characters. To be advertised as a TRIP carrier family, this must be set to a four-digit numeric carrier identification code.
cic	(Optional) Specifies that the carrier ID is a circuit identification code.

Command Default

No default behavior or values

Command Modes

Trunk group configuration (config-trunkgroup)

Command History

Release	Modification
12.2(11)T	This command was introduced.
12.3(1)	The **cic** keyword was added.

Usage Guidelines

In a network, calls are routed over incoming trunk groups and outgoing trunk groups. The *name* arguments identifies the carrier that handles the calls for a specific trunk group. In some cases, the same trunk group may be used to carry both incoming calls and outgoing calls.

The carrier ID configured locally on the trunk group supersedes the globally configured carrier ID.

To advertise the carrier as a TRIP carrier family, the **cic** keyword must be used. When **cic** is used, only numeric values can be accepted for the *name*value. If **cic** is not used, the *name*value can be alphanumeric but is not advertised to TRIP location servers.

Examples

The following example indicates that carrier "alpha1" carries calls for trunk group 5:

```
Router(config)# trunk group 5
Router(config-trunk-group)# carrier-id alpha1
```

The following example shows that the carrier with circuit identification code 1234 carries calls for trunk group 101. This trunk group can carry TRIP advertisements.

```
Router(config)# trunk group 101
Router(config-trunk-group)# carrier-id 1234 cic
```

Related Commands

Command	Description
carrier-id (global)	Configures the carrier ID globally for all trunk groups.
translation-profile (trunk group)	Associates a translation profile with a trunk group.
trunk group	Initiates the definition of a trunk group.

carrier-id (voice source group)

To specify the carrier associated with a VoIP call, use the **carrier-id**command in voice source group configuration mode. To delete the source carrier ID, use the **no** form of this command.

carrier-id {**source** | **target**} *name*
no carrier-id {**source** | **target**} *name*

Syntax Description

source	Indicates the carrier ID associated with an incoming VoIP call at the terminating gateway.
target	Indicates the carrier ID used by the terminating gateway to match an outbound dial peer.
name	The ID of the carrier to use for the call. Valid carrier IDs contain a maximum of 127 alphanumeric characters.

Command Default

No default behavior or values

Command Modes

Voice source group configuration (cfg-source-grp)

Command History

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

Usage Guidelines

A Gatekeeper Transaction Message Protocol (GKTMP) server application at the terminating gateway uses the source carrier ID to select a target carrier that routes the call over a plain old telephone service (POTS) line. The terminating gateway uses the target carrier ID to select a dial peer for routing the call over a POTS line.

Note If an incoming H.323 VoIP call matches a source IP group that has a target carrier ID, the source IP group’s target carrier ID overrides the VoIP call’s H.323 setup message.

Examples

The following example indicates that voice source IP group "group1" should use carrier ID named "source3" for incoming VoIP calls and carrier ID named "target17" for outbound dial-peer matching in the terminating gateway:

```
Router(config)# voice source-group group1
Router(cfg-source-grp)# carrier-id source source3
Router(cfg-source-grp)# carrier-id target target17
```

Related Commands

Command	Description
voice source-group	Initiates the definition of a source IP group.

cause-code

To represent internal failures with former and nonstandard H.323 or Session Initiation Protocol (SIP) cause codes, use the **cause-code**command in voice service VoIP configuration mode. To use standard cause-code categories, use the **no** form of this command.

cause-code legacy
no cause-code legacy

Syntax Description

legacy	Sets the internal cause code to the former and nonstandard set of H.323 and SIP values.

Command Default

The default for SIP and H.323 is to use standard cause-code categories, so the command is disabled.

Command Modes

Voice service VoIP configuration (config-voi-srv)

Command History

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

Usage Guidelines

This command is used for backward compatibility purposes.

Examples

The following example sets the internal cause codes to the former and nonstandard set of SIP and H.323 values for backward compatibility:

```
Router(config)# voice service voip
Router(config-voi-srv)# cause-code legacy
```

Related Commands

Command	Description
show call history voice	Displays the call history table for voice calls.

cbarge

To enable idle phones to join an active call on a shared line on a Foreign Exchange Station (FXS) port by going offhook, use the **cbarge** command in supplementary-service voice-port configuration mode. To return to the command default, use the **no** form of this command.

cbarge
no cbarge

Syntax Description

This command has no arguments or keywords.

Command Default

cBarge is disabled and idle phones are unable to join an active call on a shared line.

Command Modes

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

Command History

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

Usage Guidelines

Use the **cbarge** command to allow one idle IP or analog phone that is connected to the same FXS port to automatically join an active call on the shared line by going offhook.

The **hold-resume**command must be configured on each port before the **cbarge** command is configured.

Only one analog phone is allowed to join an active call.

Examples

The following example shows how to enable idle phones to join active calls on ports 2/2, 2/3, and 2/4 on a Cisco VG224:

```
Router(config)# stcapp supplementary-services
Router(config-stcapp-suppl-serv)# port 2/2
Router(config-stcapp-suppl-serv-port)# hold-resume
Router(config-stcapp-suppl-serv-port)# cbarge
Router(config-stcapp-suppl-serv)# port 2/3
Router(config-stcapp-suppl-serv-port)# hold-resume
Router(config-stcapp-suppl-serv-port)# cbarge
Router(config-stcapp-suppl-serv)# port 2/4
Router(config-stcapp-suppl-serv-port)# hold-resume
Router(config-stcapp-suppl-serv-port)# cbarge
Router(config-stcapp-suppl-serv-port)# end
```

Related Commands

Command	Description
hold-resume	Turns the STCAPP supplementary-service features on and off using hookflash.
stcapp supplementary-services	Enters supplementary-service configuration mode for configuring STCAPP supplementary-service features on an FXS port.

ccm-manager application redundant-link port

To configure the port number for the redundant link application, use the **ccm-managerapplicationredundant-linkport** command in global configuration mode. To disable the configuration, use the **no** form of this command.

ccm-manager application redundant-link port *number*
no ccm-manager application redundant-link port

Syntax Description

port *number*	Port number for the transport protocol. The protocol may be User Data Protocol (UDP), Reliable User Datagram Protocol (RDUP), or TCP. Range is from 0 to 65535, and the specified value must not be a well-known reserved port number such as 1023. The default is 2428.

Command Default

Port number: 2428

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)T	This command was introduced with Cisco CallManager Version 3.0 and the Cisco Voice Gateway 200 (VG200).
12.2(2)XA	The command was implemented on the Cisco 2600 series and Cisco 3600 series.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T.

Usage Guidelines

Use this command only when defining an application-specific port other than the default.

Examples

In the following example, the port number of the redundant link application is 2429:

```
ccm-manager application redundant-link port 2429
```

Related Commands

Command	Description
ccm-manager redundant-host	Configures the IP address or the DNS name of up to two backup Cisco CallManagers.
ccm-manager switchback	Configures the switchback mode that determines when the primary Cisco CallManager is used if it becomes available again while a backup Cisco CallManager is being used.

ccm-manager config

To specify the TFTP server from which the Media Gateway Control Protocol (MGCP) gateway downloads Cisco Unified Communications Manager (Cisco UCM) Extensible Markup Language (XML) configuration files and to enable the download of the configuration, use the **ccm-managerconfig** command in global configuration mode. To disable the dial-peer and server configurations, use the **no** form of this command.

ccm-manager config [{**dialpeer-prefix** *prefix* | **server** {*ip-addressname*}}]
no ccm-manager config [{**dialpeer-prefix** *prefix* | **server**}]

Syntax Description

dialpeer -prefix*prefix*	(Optional) Specifies the prefix to use for autogenerated dial peers. Range is 1 to 2147483647. The default is 999. **Note** When manually adding a dial peers prefix, select a prefix number other than the default.
server {*ip-address* | *name*}	(Optional) Specifies the IP address or logical name of the TFTP server from which the XML configuration files are downloaded. The arguments are as follows: • *ip-address*-- IP address of the TFTP server from which to download the XML configuration files to the local MGCP voice gateway. • *name*-- Logical (symbolic) name of the TFTP server from which to download XML configuration files to the local MGCP voice gateway.

Command Default

The configuration download feature is disabled.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(2)XN	This command was introduced and implemented on the Cisco 2600 series, Cisco 3600 series, and the Cisco VG200.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco IAD2420 series.

Usage Guidelines

The **ccm-managerconfig** command is required to enable the download of Cisco UCM XML configuration files. If you separate the MGCP and H.323 dial peers under different dial-peer tags, ensure that the MGCP dial peers are configured before the H.323 dial peers. Direct-inward-dial (DID) is required for E1 PRI dial peers.

Note To keep manually added dial peers from being deleted from the running configuration when Cisco UCM downloads the configuration to the gateway, use a dial peer-prefix value other than the default (999).

Do not delete the POTS dial peer created by the automatic download process. However, if a dial peer has been deleted, you can restore the deleted dial peer by entering the following commands to repeat the download of the configuration file:

```
no mgcp
no ccm-manager config
ccm-manager config
mgcp
```

After you enter these commands, use the **showccm-managerconfig-download** command to display the the configuration file downloaded from the TFTP server via the interface specified. The following is an example of the output:

```
Loading sample.cnf.xml from 9.13.22.100 (via GigabitEthernet0/0): !
[OK - 12759 bytes]
```

Examples

The following example shows how to enable the automatic download of configuration files:

```
ccm-manager config
```

In the following example, the IP address of the TFTP server from which a configuration file is downloaded is identified:

```
ccm-manager config server 10.10.0.21
```

Related Commands

Command	Description
debug ccm-manager config-download	Displays dialog during configuration download from the Cisco UCM to the gateway.
show ccm-manager config-download	Displays whether the Cisco UCM configuration is enabled.

ccm-manager download-tones

To configure a Cisco IOS gateway to download a XML configuration file that contains custom tone information from a TFTP server at the time of gateway registration, use the **ccm-managerdownload-tones** command in global configuration mode. To disable this functionality, use the **no** form of this command.

ccm-manager download-tones
no ccm-manager download-tones

Syntax Description

This command has no arguments or keywords.

Command Default

Cisco CallManager download tones are disabled.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(15)ZJ	This command was introduced.
12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T.

Examples

The following example shows a Cisco IOS gateway being configured to download an XML configuration file that contains custom tone information from a TFTP server:

```
Router(config)# ccm-manager download-tones
```

Related Commands

Command	Description
cptone	Specifies a regional voice-interface-related tone, ring, and cadence setting.
debug ccm-manager	Displays debugging of Cisco CallManager.
show ccm-manager	Displays a list of Cisco CallManager servers and their current status and availability.

ccm-manager fallback-mgcp

To enable the gateway fallback feature and allow a Media Gateway Control Protocol (MGCP) voice gateway to provide call processing services when Cisco CallManager is unavailable, use the **ccm-managerfallback-mgcp** command in global configuration mode. To disable fallback on the MGCP voice gateway, use the **no** form of this command.

ccm-manager fallback-mgcp
no ccm-manager fallback-mgcp

Syntax Description

This command has no arguments or keywords.

Command Default

The gateway fallback feature is enabled

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(2)XN	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and the Cisco VG200.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and Cisco CallManager Version 3.2 and implemented on Cisco IAD2420 series.
12.2(15)ZJ	This command was integrated into Cisco IOS Release 12.2(15)ZJ.
12.3(2)T	This command was implemented on the Cisco 26xxXM, Cisco 2691, Cisco 3640, Cisco 3640A, Cisco 3660, and Cisco 37xx.

Usage Guidelines

This command causes the gateway to fall back and provide call processing services if connectivity is lost between the gateway and all Cisco CallManager servers. The mode and timing are set by default.

Examples

The following example enables fallback:

```
Router(config)# ccm-manager fallback-mgcp
```

Related Commands

Related Command	Purpose
ccm-manager config	Supplies the local MGCP voice gateway with the IP address or logical name of the TFTP server from which to download XML configuration files and enable the download of the configuration.
debug ccm-manager	Displays debugging information about the Cisco CallManager.
show ccm-manager fallback-mgcp	Displays the status of the MGCP gateway fallback feature.

ccm-manager fax protocol

To enable fax-relay protocol for endpoints on a gateway, use the **ccm-managerfaxprotocol** command in global configuration mode. To disable fax-relay protocol, use the **no** form of this command.

ccm-manager fax protocol cisco
no ccm-manager fax protocol cisco

Syntax Description

cisco	Cisco-proprietary fax-relay protocol. This is the only choice.

Command Default

Cisco-proprietary fax-relay protocol is enabled by default.

Command Default

Fax relay is enabled.

Command Modes

Global configuration

Command History

Release	Modification
12.2(9)T	This command was introduced.

Usage Guidelines

Use the **no** form of this command to disable fax relay.

Because fax relay is enabled by default, the **showrunning-config** command does not explicitly show it to be enabled.

Fax over IP enables interoperability of traditional analog fax machines with IP telephony networks. In its original form, fax data is digital. For transmission across a traditional public switched telephone network (PSTN), it is converted to analog form. For transmission across the IP (packet) network, it is reconverted to digital form, and then, at the destination fax machine, converted again to analog form.

Most Cisco voice gateways support two methods of transmitting fax traffic across the IP network:

- Cisco fax relay--The gateway terminates the T.30 fax signaling. This is the preferred method.
- Fax pass-through--The gateway does not distinguish a fax call from a voice call. All Cisco voice gateways support fax pass-through.

Examples

The following example configures a Media Gateway Control Protocol (MGCP) gateway for Cisco fax relay:

```
Router(config)# ccm-manager fax protocol cisco
Router(config)# mgcp fax t38 inhibit
```

The following example configures an MGCP gateway for fax pass-through:

```
Router(config)# ccm-manager fax protocol cisco
Router(config)# mgcp modem passthrough voip mode nse
Router(config)# mgcp modem passthrough voip codec g711ulaw
```

Related Commands

Command	Description
debug ccm-manager	Displays debugging of Cisco CallManager.
show ccm-manager	Displays a list of Cisco CallManager servers and their current status and availability.
show running-config	Displays the contents of the currently running configuration file.

ccm-manager mgcp

To enable the gateway to communicate with Cisco CallManager through the Media Gateway Control Protocol (MGCP) and to supply redundant control agent services, use the **ccm-managermgcp** command in global configuration mode. To disable communication with Cisco CallManager and redundant control agent services, use the **no** form of this command.

ccm-manager mgcp [**codec-all**]
no ccm-manager mgcp [**codec-all**]

Syntax Description

codec-all	(Optional) Enables all codec on the gateway for the Cisco CallManager.

Command Default

Cisco CallManager does not communicate with the gateway through MGCP.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)T	This command was introduced with Cisco CallManager Version 3.0 on the Cisco VG200.
12.2(2)XA	The command was integrated into Cisco IOS Release 12.2(2)XA and implemented on the Cisco 2600 series and Cisco 3600 series.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series, 3600 series, and Cisco VG200.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T.
12.2(11)T	This command was integrated into the Cisco IOS Release 12.2(11)T and Cisco CallManager Version 3.2 and was implemented on the Cisco IAD2420 series routers.
12.2(11)YU	This command was integrated into Cisco IOS Release 12.2(11)YU and implemented on the Cisco 1760 gateway.
15.0(1)M	This command was modified in a release earlier than Cisco IOS Release 15.0(1)M. The **codec-all** keyword was added.

Usage Guidelines

This command enables the gateway to communicate with Cisco CallManager through MGCP. This command also enables control agent redundancy when a backup Cisco CallManager server is available.

Examples

In the following example, support for Cisco CallManager and redundancy is enabled within MGCP:

```
Router# configure terminal
Router(config)# ccm-manager mgcp
```

Related Commands

Command	Description
ccm-manager redundant-host	Configures the IP address or the DNS name of up to two backup Cisco CallManagers.
ccm-manager switchback	Configures the switchback mode that determines when the primary Cisco CallManager is used if it becomes available again while a backup Cisco CallManager is being used.
mgcp	Enables Media Gateway Control Protocol mode.

ccm-manager music-on-hold

To enable the multicast music-on-hold (MOH) feature on a voice gateway, use the **ccm-managermusic-on-hold** command in global configuration mode. To disable the MOH feature, use the **no** form of this command.

ccm-manager music-on-hold
no ccm-manager music-on-hold

Syntax Description

This command has no arguments or keywords.

Command Default

Disabled

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(2)XN	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco VG200.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and Cisco CallManager Version 3.2 and implemented on the Cisco IAD 2420 series routers.

Examples

The following example shows multicast MOH configured for a MGCP voice gateway:

```
mgcp call-agent 10.0.0.21 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000
mgcp modem passthrough voip mode cisco
mgcp package-capability rtp-package
mgcp package-capability sst-package
no mgcp timer receive-rtcp
call rsvp-sync
!
ccm-manager redundant-host 10.0.0.21
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server 10.0.0.21
!
```

Related Commands

Command	Description
ccm-manager music-on-hold bind	Enables the multicast MOH feature on a voice gateways.
debug ccm-manager music-on-hold	Displays debugging information for MOH.
show ccm-manager music-on-hold	Displays MOH information.

ccm-manager music-on-hold bind

To bind the multicast music-on-hold (MOH) feature to an interface type, use the **ccm-managermusic-on-hold** bind command in global configuration mode. To unbind the MOH feature on the interface type, use the **no** form of this command.

ccm-manager music-on-hold bind *type slot/port*
no ccm-manager music-on-hold bind *type slot/port*

Syntax Description

type	Interface type to which the MOH feature is bound. The options follow: • **async** -- Asynchronous interface • **bvi** -- Bridge-Group Virtual Interface • **ctunnel** -- CTunnel interface • **dialer** -- Dialer interface • **ethernet** -- IEEE 802.3 • **lex** -- Lex interface • **loopback** -- Loopback interface • **mfr** -- Multilink Frame Relay bundle interface • **multilink** -- Multilink interface • **null** -- Null interface • **serial** -- Serial interface • **tunnel** -- Tunnel interface • **vif** -- PGM Multicast Host interface • **virtual -FrameRelay**--Virtual Frame Relay interface • **virtual -Template**-- Virtual template interface • **virtual -TokenRing**-- Virtual Token Ring
slot / port	Number of the slot being configured. See the appropriate hardware manual for slot and port information.

Command Default

This command is disabled by default, so the MOH feature is not bound to an interface type.

Command Modes

Global configuration (config)

Command History

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

Usage Guidelines

Use the **ccm-managermusic-on-hold** bind command to bind the multicast music-on-hold (MOH) feature to an interface type. Dynamic configuration of multicast MOH bind is not supported.

Examples

The following example shows multicast MOH bound to serial interface 0/0:

```
ccm-manager music-on-hold bind serial 0/0
```

Related Commands

Command	Description
ccm-manager music-on-hold	Enables the MOH feature.
debug ccm-manager music-on-hold	Displays debugging information for MOH.
show ccm-manager music-on-hold	Displays MOH information.

ccm-manager redundant-host

To configure the IP address or the Domain Name System (DNS) name of one or two backup Cisco CallManager servers, use the **ccm-managerredundant-host**command in global configuration mode. To disable the use of backup Cisco CallManager servers as call agents, use the **no** form of this command.

ccm-manager redundant-host {*ip-addressdns-name*} [{*ip-addressdns-name*}]
no ccm-manager redundant-host {*ip-addressdns-name*} [{*ip-addressdns-name*}]

Syntax Description

ip -address	IP address of the backup Cisco CallManager server.
dns -name	DNS name of the backup Cisco CallManager server.

Command Default

If you do not configure a backup Cisco CallManager, the redundancy is disabled.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)T	This command was introduced with Cisco CallManager Version 3.0 on the Cisco Voice Gateway 200 (VG200).
12.2(2)XA	The command was implemented on the Cisco 2600 series and Cisco 3600 series. The *dns-name* argument was added.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series, 3600 series, and the Cisco VG200.
12.2(11)T	This command was integrated into the Cisco IOS Release 12.2(11)T and Cisco CallManager Version 3.2 and implemented on the Cisco IAD2420 series routers.

Usage Guidelines

The list of IP addresses or DNS names is an ordered and prioritized list. The Cisco CallManager server that was defined with the **mgcpcall-agent** command has the highest priority--it is the primary Cisco CallManager server. The gateway selects a Cisco CallManager server on the basis of the order of its appearance in this list.

Examples

In the following example, the IP address 10.0.0.50 is configured as the backup Cisco CallManager
:

```
ccm-manager redundant-host 10.0.0.50
```

Related Commands

Command	Description
ccm-manager application	Configures the port number for the redundant link application.

Command	Description
ccm-manager switchback	Configures the switchback mode that determines when the primary Cisco CallManager is used if it becomes available again while a backup Cisco CallManager is being used.
ccm-manager switchover-to-backup	Redirects (manually and immediately) a Cisco 2600 series router or Cisco 3600 series router to the backup Cisco CallManager server.
mgcp call-agent	Defines the Cisco CallManager server as the highest priority.

ccm-manager sccp

To enable Cisco CallManager autoconfiguration of the Cisco IOS gateway, use the **ccmmanagersccp**command in global configuration mode. To disable autoconfiguration, use the **no**form of this command.

ccm-manager sccp
no ccm-manager sccp

Syntax Description

This command has no arguments or keywords.

Command Default

Autoconfiguration is disabled.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(14)T	This command was introduced.

Usage Guidelines

Use this command to trigger TFTP download of the eXtensible Markup Language (XML) configuration file. Issuing this command immediately triggers the download, and also enables the Skinny Client Control Protocol (SCCP) and SCCP Telephony Control Application (STCAPP), applications that enable Cisco CallManager control of gateway-connected telephony endpoints.

Examples

The following example enables autoconfiguration of gateway-connected endpoints:

```
Router(config)# ccm-manager sccp
```

Related Commands

Command	Description
ccm-manager config	Specifies the TFTP server from which the Cisco IOS gateway downloads Cisco CallManager XML configuration files.
ccm-manager sccp local	Selects the local interface for SCCP application use for Cisco CallManager registration.
show ccm-manager config-download	Displays information about the status of the Cisco IOS gateway configuration download.

ccm-manager sccp local

To select the local interface that the Skinny Client Control Protocol (SCCP) application uses to register with Cisco CallManager, use the **ccm-managersccplocal** command in global configuration mode. To deselect the interface, use the **no**form of this command.

ccm-manager sccp local *interface-type interface-number*
no ccm-manager sccp local *interface-type interface-number*

Syntax Description

interface-type	Interface type that the SCCP application uses for Cisco CallManager registration.
interface-number	Interface number that the SCCP application uses for Cisco CallManager registration.

Command Default

No local interface is selected.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.3(14)T	This command was introduced.

Usage Guidelines

You must specify this interface before enabling the Cisco CallManager autoconfiguration process. The MAC address of this interface is used to identify gateway endpoints.

Examples

The following example configures a FastEthernet interface for SCCP application use for Cisco CallManager registration:

```
Router(config)# ccm-manager sccp local fastethernet 0/0
```

Related Commands

Command	Description
show ccm-manager	Displays a list of Cisco CallManager servers and their current status and availability.

ccm-manager shut-backhaul-interfaces

To disable ISDN Layer 2 connectivity on a Cisco Call Manager Media Gateway Control Protocol (MGCP) PRI or BRI backhauled trunk when communication is lost between the Cisco Call Manager and the MGCP gateway, use the **ccm-managershut-backhaul-interfaces** command in global configuration mode. To restore the default behavior, where ISDN Layer 2 is maintained between the MGCP gateway and the ISDN switch even when no connectivity exists between the MGCP gateway and any Cisco Call Manager, use the **no** form of this command.

ccm-manager shut-backhaul-interfaces
no ccm-manager shut-backhaul-interfaces

Syntax Description

This command has no arguments or keywords.

Command Default

The default behavior is for the ISDN Layer 2 connection to be maintained (to make the Cisco Call Manager MGCP PRI or BRI backhaul continue to function) between the MGCP gateway and the ISDN switch even if no connectivity exists between the MGCP gateway and any Cisco Call Manager.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.4(8)	This command was introduced.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(3f)	This command was integrated into Cisco IOS Release 12.4(3f).
12.4(5c)	This command was integrated into Cisco IOS Release 12.4(5c).
12.4(7c)	This command was integrated into Cisco IOS Release 12.4(7c).
12.4(4)T5	This command was integrated into Cisco IOS Release 12.4(4)T5.
12.4(6)T4	This command was integrated into Cisco IOS Release 12.4(6)T4.

Usage Guidelines

Use this command on Cisco IOS voice routers configured for Cisco Call Manager MGCP PRI or BRI backhaul.

Prior to the introduction of the **ccm-managershut-backhaul-interfaces** command, a Cisco Call Manager MGCP PRI or BRI backhaul trunk would maintain ISDN Layer 2 connectivity between the MGCP gateway and the ISDN switch in a MULTIPLE_FRAMES_ESTABLISHED state even if Layer 3 Q.931 backhaul connectivity between the Cisco Call Manager and the MGCP gateway was unavailable. This causes problems because the ISDN switch interprets the PRI or BRI trunk as being active and continues to place calls to the MGCP gateway, even though all of the calls fail. After you enter the **ccm-managershut-backhaul-interfaces**command, Layer 2 is disabled when connectivity between the Cisco Call Manager and the MGCP gateway is unavailable.

Examples

The following example disables ISDN Layer 2 connectivity on a Cisco Call Manager MGCP PRI or BRI backhauled trunk when communication is lost between Cisco Call Manager and the MGCP gateway:

```
ccm-manager shut-backhaul-interfaces
```

The following example restores the default behavior (functionality of the **ccm-managershut-backhaul-interfaces**command is disabled) so that the ISDN Layer 2 connection is maintained between the MGCP gateway and the ISDN switch, even when no connectivity exists between the MGCP gateway and any Cisco Call Manager:

```
no ccm-manager mgcp
no ccm-manager shut-backhaul-interfaces
ccm-manager mgcp
```

Related Commands

Command	Description
ccm-manager mgcp	Enables the gateway to communicate with the Cisco Call Manager through the MGCP and to supply redundant control agent services.

ccm-manager shut-interfaces-tftp-fails

To configure the number of TFTP download failures allowed before the gateway shuts down ports, use the **ccm-managershut-interfaces-tftp-fails**command in global configuration mode. To return to the default configuration, use the **no** form of this command.

ccm-manager shut-interfaces-tftp-fails *retries*
no ccm-manager shut-interfaces-tftp-fails

Syntax Description

retries	Number or TFTP retries. Range is from 2 to 10. The default is 2.

Command Default

Ports shut down after the second TFTP retry. However TFTP download attempts continue.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.4(15)T2	This command was introduced.
12.4(20)T	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use the **ccm-managershut-interfaces-tftp-fails**command to configure the number of TFTP download failures allowed before the gateway put the port in a shutdown state.

Examples

The following example shows a gateway being configured to put the port in a shutdown state after four TFTP download failures:

```
Router(config)# ccm-manager shut-interfaces-tftp-fails 4
```

Related Commands

Command	Description
show ccm-manager	Displays a list of Cisco Unified Communications Manager servers and their current status and availability.

ccm-manager switchback

To specify the time when control is to be returned to the primary Cisco CallManager server once it becomes available, use the **ccm-managerswitchback** command in global configuration mode. To reset to the default, use the **no** form of this command.

ccm-manager switchback {**graceful** | **immediate** | **never** | **schedule-time** *hh* **:** *mm* | **uptime-delay** *minutes*}
no ccm-manager switchback

Syntax Description

graceful	Specifies that control is returned to the primary Cisco CallManager server after the last active call ends (when there is no voice call in active setup mode on the gateway). Default value.
immediate	Specifies an immediate switchback to the primary Cisco CallManager server when the TCP link to the primary Cisco CallManager server is established, regardless of current call conditions.
never	Specifies not to return control to the primary Cisco CallManager server, as long as the secondary is up and running. The gateway registers to primary if the secondary is down and when the primary is up and running.
schedule -time*hh***:***mm*	Specifies an hour and minute, based on a 24-hour clock, when control is returned to the primary Cisco CallManager server. If the specified time is earlier than the current time, the switchback occurs at the specified time on the following day.
uptime -delay*minutes*	Specifies the number of minutes the primary Cisco CallManager server must run after the TCP link to is reestablished and control is returned to that primary call agent. Valid values are from 1 to 1440 (1 minute to 24 hours).

Command Default

Graceful switchback

Command Modes

Global configuration (config)

Command History

Release	Modification
12.1(3)T	This command was modified. This command was introduced with Cisco CallManager Version 3.0 on the Cisco VG200.
12.2(2)XA	The command was implemented on the Cisco 2600 series and Cisco 3600 series.
12.2(2)XN	Support for enhanced Media Gateway Control Protocol (MGCP) voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series, 3600 series, and the Cisco VG200.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T.
12.2(11)T	This command was implemented on the Cisco IAD2420 series routers.

Release	Modification
15.0(1)M	This command was modified in a release earlier than Cisco IOS Release 15.0(1)M. The **never** keyword was added.

Usage Guidelines

This command allows you to configure switchback to the higher priority Cisco CallManager when it becomes available. Switchback allows call control to revert to the original (primary) Cisco CallManager once service has been restored.

Examples

In the following example, the primary Cisco CallManager is configured to be used as soon as it becomes available:

```
Router# configure terminal
Router(config)# ccm-manager switchback immediate
```

Related Commands

Command	Description
ccm-manager application	Configures the port number for the redundant link application.
ccm-manager redundant-host	Configures the IP address or the DNS name of up to two backup Cisco CallManagers.
ccm-manager switchover-to-backup	Redirects a Cisco 2600 series or Cisco 3600 series router to the backup Cisco CallManager.

ccm-manager switchover-to-backup

To manually redirect a gateway to the backup Cisco CallManager server, use the **ccm-managerswitchover-to-backup**command in privileged EXEC mode.

ccm-manager switchover-to-backup

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.2(2)XN	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco VG200.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and Cisco CallManager Version 3.2 and implemented on the Cisco IAD2420 series.

Usage Guidelines

Switchover to the backup Cisco CallManager server occurs immediately. This command does not switch the gateway to the backup Cisco CallManager server if you have the **ccm-managerswitchback** command option set to " immediate" and the primary Cisco CallManager server is still running.

Examples

In the following example, the backup Cisco CallManager server is configured to be used as soon as it becomes available:

```
ccm-manager switchover-to-backup
```

Related Commands

Command	Description
ccm-manager application redundant-link	Configures the port number for the redundant link application (that is, for the secondary Cisco CallManager server).
ccm-manager redundant-host	Configures the IP address or the DNS name of up to two backup Cisco CallManager servers.
ccm-manager switchback	Specifies the time at which control is returned to the primary Cisco CallManager server once the server is available.

ccs connect (controller) through clear vsp statistics

ccs connect (controller)

To configure a common channel signaling (CCS) connection on an interface configured to support CCS frame forwarding, use the **ccsconnect** command in controller configuration mode. To disable the CCS connection on the interface, use the**no**form of this command.

ccs connect{**serial** | **atm**}*number*[{[*dlci*] | **pvc** *vpi/vci*| **pvc***name*}][{*cid-number*}]
no ccs connect{**serial** | **atm**}*number*[{[*dlci*] | **pvc** *vpi/vci*| **pvc***name*}][{*cid-number*}]

Syntax Description

serial	Makes a serial CCS connection for Frame Relay.
atm	Makes an ATM CCS connection.
dlci	(Optional) Specifies the data-link connection identifier (DLCI) number.
pvc *vpi/vci*	(Optional) Specifies the permanent virtual circuit (PVC) virtual path identifier (VPI) or virtual channel identifier (VCI). Range is from 0 to 255; the slash is required.
pvc *name*	(Optional) Specifies the PVC string that names the PVC for recognition.
cid-number	(Optional) If you have executed the **ccsencapfrf11** command, the *cid-number* argument allows you to specify any channel identification (CID) number from 5 to 255.

Command Default

No CCS connection is made.

Command Modes

Controller configuration (config-controller)

Command History

Release	Modification
12.0(2)T	This command was introduced on the Cisco MC3810.
12.0(7)XK	The *cidnumber* argument was added; the **dlci** keyword and **vcd**options were removed.
12.1(2)T	The CID syntax addition and removal of the **dlci** keyword and **vcd** options were integrated into Cisco IOS Release 12.1(2)T.
12.1(2)XH	This command was implemented on the Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, and Cisco 7500 series.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T.

Usage Guidelines

Use this command to configure a CCS connection. If the CCS connection is over Frame Relay, specify a serial interface and the DLCI. If the CCS connection is over ATM, specify **atm**, the slot number, and the PVC.

If you have executed the **ccsencapfrf11** command, the *cidnumber* option of the **ccsconnect** command allows you to specify any CID from 5 to 255. If you do not issue the **ccsencapfrf11** command, Cisco encapsulation is used, and any CID value other than 254 is ignored.

Note CDP and keepalives are disabled by default on a D-channel interface.

Examples

To configure a Frame Relay CCS frame-forwarding connection on DLCI 100 by using the default CID of 254, enter the following command:

```
ccs connect serial 1 100
```

or:

```
ccs connect serial 1 100 10
```

To configure a CCS frame-forwarding connection over an ATM PVC, enter the following command:

```
ccs connect atm 0 pvc 100/10
```

or:

```
ccs connect atm 0 pvc 10/100 21
```

or:

```
ccs connect atm 0 pvc mypvc 10 21
```

To configure a Frame Relay CCS frame-forwarding connection on DLCI 100 using a CID of 110, enter the following command:

```
ccs connect serial 1 100 110
```

Related Commands

Command	Description
ccs encap frf11	Allows the specification of the standard Annex-C FRF.11 format.

ccs connect (interface)

To configure a common channel signaling (CCS) connection on an interface configured to support CCS frame forwarding, use the **ccsconnect** command in interface configuration mode. To disable the CCS connection on the interface, use the**no**form of this command.

ccs connect{**serial** | **atm**}[{*dlci* | **pvc** *vpi/vci* | **pvc** *name*}][{*cid-number*}]
no ccs connect{**serial** | **atm**}[{*dlci* | **pvc** *vpi/vci*| **pvc** *name*}][{*cid-number*}]

Syntax Description

serial	Makes a serial CCS connection for Frame Relay.
atm	Makes an ATM CCS connection.
dlci	(Optional) Data-link connection identifier (DLCI) number.
pvc *vpi* / *vci*	(Optional) Permanent virtual circuit (PVC) virtual path identifier or virtual channel identifier (VCI). Range is from 0 to 255; the slash is required.
pvc *name*	(Optional) PVC string that names the PVC for recognition.
cid -number	(Optional) If you have executed the**ccsencapfrf11** command, the *cid-number*argument allows you to specify any channel identification (CID) number from 5 to 255.

Command Default

No CCS connection is made.

Command Modes

Interface configuration (config-if)

Command History

Release	Modification
12.0(2)T	This command was introduced on the Cisco MC3810.
12.0(7)XK	The *cid-number* argument was added; the **dlci** keyword and **vcd** options were removed.
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 router and integrated into Cisco IOS Release 12.2(2)T.

Usage Guidelines

Use this command to configure a CCS connection. If the CCS connection is over Frame Relay, specify a serial interface and the DLCI. If the CCS connection is over ATM, specify **atm**, the interface number (0), and the PVC.

If you have executed the **ccsencapfrf11** command, the *cid-number* option of the **ccsconnnect** command allows you to specify any CID from 5 to 255. If you do not issue the **ccsencapfrf11** command, Cisco encapsulation is used, and any CID value other than 254 is ignored.

Note Cisco Discovery Protocol and keepalives are disabled by default on a D-channel interface.

Examples

To configure a Frame Relay CCS frame-forwarding connection on DLCI 100 by using the default CID of 254, enter the following command:

```
ccs connect serial 1 100
```

or

```
ccs connect serial 1 100 10
```

To configure a CCS frame-forwarding connection over an ATM PVC, enter the following command:

```
ccs connect atm 0 pvc 100/10
```

or

```
ccs connect atm 0 pvc 10/100 21
```

or

```
ccs connect atm 0 pvc mypvc 10 21
```

To configure a Frame Relay CCS frame-forwarding connection on DLCI 100 using a CID of 110, enter the following command:

```
ccs connect serial 1 100 110
```

Related Commands

Command	Description
ccs encap frf11	Allows the specification of the standard Annex-C FRF.11 format.

ccs encap frf11

To configure the common channel signaling (CCS) packet encapsulation format for FRF.11, use the **ccsencapfrf11** command in interface configuration mode. To disable CCS encapsulation for FRF11, use the **no** form of this command.

ccs encap frf11
no ccs encap frf11

Syntax Description

This command has no arguments or keywords.

Command Default

By default, the format is a Cisco packet format, using a channel ID (CID) of 254.

Command Modes

Interface configuration (config-if)

Command History

Release	Modification
12.0(7)XK	This command was introduced for the Cisco MC3810.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.1(2)XH	This command was implemented on the Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, and Cisco 7500 series.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T.

Usage Guidelines

This command allows the specification of the standard Annex-C format. Use this command to define the packet format for the CCS packet; it places the FRF.11 Annex-C (Data Transfer Syntax) standard header on the CCS packets only.

Once the **ccsencapfrf11** command is executed, you can use the **ccsconnect** command to specify a CID other than 254.

Examples

The following example shows how to configure a serial interface for Frame Relay:

```
interface Serial1:15
 ccs encap frf11
 ccs connect Serial0 990 100
```

Related Commands

Command	Description
mode ccs frame-forwarding	Set to forward frames on the controller.
ccs connect	Configures a CCS connection on an interface configured to support CCS frame forwarding.

cdr-format

To select the format of the call detail records (CDRs) generated for file accounting, use the **cdr-format** command in gateway accounting configuration mode. To reset to the default, use the **no** form of this command.

cdr-format {**compact** | **detailed**}
no cdr-format

Syntax Description

compact	Compact set of voice attributes is generated in CDRs.
detailed	Full set of voice attributes is generated in CDRs. Default value.

Command Default

Detailed (full version of CDRs is generated).

Command Modes

Gateway accounting file configuration (config-gw-accounting-file)

Command History

Release	Modification
12.4(15)XY	This command was introduced.
12.4(20)T	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command determines whether the CDRs generated by the file accounting process contain the complete set of voice attributes or a compact set of 17 voice attributes.

For a list of the complete set of voice attributes generated with the **detailed** keyword, see the "VSAs Supported by Cisco Voice Products" section in the *RADIUS VSA Voice Implementation Guide* .

The name and order of the attributes generated with the **compact** keyword are: CallLegType, ConnectionId, SetupTime, PeerAddress, PeerSubAddress, DisconnectCause, DisconnectText, ConnectTime, DisconnectTime, CallOrigin, ChargedUnits, InfoType, TransmitPackets, TransmitBytes, ReceivePackets, ReceiveBytes, feature_vsa.

Examples

The following example shows the CDR format set to compact:

```
gw-accounting file
 primary ftp server1/cdrtest1 username bob password temp
 maximum buffer-size 60
 maximum fileclose-timer 720
 cdr-format compact
```

Related Commands

Command	Description
acct-template	Selects a group of voice vendor-specific attributes to collect in accounting records.
maximum buffer-size	Sets the maximum size of the file accounting buffer.

Command	Description
maximum fileclose-timer	Sets the maximum time for saving records to an accounting file before closing the file and creating a new one.
primary	Sets the primary location for storing the CDRs generated for file accounting.

ces-clock

To configure the clock for the Circuit Emulation Services (CES) interface, use the **ces-clock** command in controller configuration mode. To disable the CES clock, use the no form of this command.

ces-clock {**adaptive** | **srts** | **synchronous**}
no **ces-clock** {**adaptive** | **srts** | **synchronous**}

Syntax Description

adaptive	Adjusts the output clock on a received ATM adaptation layer 1 (AAL1) on a first-in, first-out basis. Use in unstructured mode.
srts	Sets the clocking mode to synchronous residual time stamp.
synchronous	Configures the timing recovery to synchronous for structured mode.

Command Default

The default setting is synchronou.s

Command Modes

Controller configuration (config-controller)

Command History

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

Usage Guidelines

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

Examples

The following example configures the CES clock mode for synchronous residual time stamp:

```
ces-clock srts
```

Related Commands

Command	Description
controller	Configures the T1 or E1 controller.

cgma-agent

To enable the Cisco Gateway Management Agent (CGMA) on the Cisco IOS gateway, use the **cgma-agent** command in global configuration mode. To disable the CGMA, use the **no** form of this command.

cgma-agent [{**tcp-port** *number* | **time-period** *seconds*}]
no cgma-agent

Syntax Description

tcp -port*number*	(Optional) Specifies the TCP port number for the CGMA to use in communication with a third-party management system. Range is from 5000 to 65535. The default is 5000.
time -period *seconds*	(Optional) Specifies the maximum time period, in seconds ,for maintaining the link between the CGMA and the third-party management system during a period of inactivity. If twice the timeout value is met or exceeded with no message received from the client, the TCP connection is closed. Additionally, a 60-second timer is maintained in the CGMA, which closes the connection if no handshake query message is received from the third-party management system for 60 seconds. Range is from 45 to 300. The default is 45.

Command Default

Default *number* value is 5000. Default *seconds* value is 45.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.2(2)XB	This command was introduced on the Cisco 2600 series, Cisco 3600 series, Cisco AS5300, Cisco AS5350, and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5800 for Cisco IOS release 12.2(2)XB1 release only.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 2600 series, Cisco 3600 series, and Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5800 is not included in this release.

Usage Guidelines

Use this command to enable the CGMA on the Cisco IOS gateway. The CGMA communicates with the third-party management system to provide real-time information for gateway management, including the following:

- Handshake query, status query, and response messages between the CGMA and the third-party management system.
- Call information such as start and end of call from call detail records (CDRs) sent using eXtensible Markup Language (XML) over TCP/IP.
- Shows if T1 or E1 controllers and analog ports are up or down, and are also generated at the removal or addition of a "pri-group" or "ds0-group" under the T1 or E1 controller.

Examples

The following example shows that the CGMA is enabled on TCP port 5300 and that the CGMA times out after 300 seconds and closes its connection to the third-party management system because of inactivity in the link:

```
Router(config)# cgma-agent tcp-port
 5300 time-period 300
Router# show running-config
Building configuration...
Current configuration : 1797 bytes
!
version 12.2
service config
no service single-slot-reload-enable
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname gw1
!
.
.
.
resource-pool disable
!
ip subnet-zero
no ip domain-lookup
!
no ip dhcp-client network-discovery
isdn switch-type primary-ni
!
!
!
!
!
!
cgma-agent tcp-port 5300 time-period 300
fax interface-type modem
mta receive maximum-recipients 2
!
!
controller T1 0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
!
!
interface Ethernet0
 ip address 209.165.200.225 255.255.255.0
!
interface Serial0:23
 no ip address
 isdn switch-type primary-ni
 isdn protocol-emulate network
 isdn incoming-voice modem
 isdn T310 10000
 no cdp enable
!

voice-port 0:D
!
dial-peer voice 1213 voip
 destination-pattern 12135550100
 session target ipv4:209.165.200.229
!
dial-peer voice 1415 pots
 destination-pattern 14155550100
 direct-inward-dial
 port 0:D
!
dial-peer voice 12136 voip
 destination-pattern 12135550120
 session target ipv4:209.165.200.229
!
dial-peer voice 14156 pots
 incoming called-number .
 direct-inward-dial
!
gateway
!
end
```

channel-group

To configure serial WAN on a T1 or E1 interface, use the **channel-group** command in controller configuration mode. To clear a channel group, use the **no** form of this command.

Cisco 2600 Series

channel-group *channel-group-number* **timeslots** *range* [**speed** {**56** | **64**}] [**aim** *aim-slot-number*]
no channel-group *channel-group-number*

Cisco 2611 (Cisco Signaling Link Terminal [SLT])

channel-group *channel-number*
no channel-group *channel-number*

Cisco ASR 901 Series, Cisco 2600XM Series, Cisco 2691, Cisco 3631, Cisco 3660, Cisco 3725, and Cisco 3745

channel-group *channel-group-number* {**timeslots** *range* [**speed** {**56** | **64**}] | **unframed**} [**aim** *aim-slot-number*]
no channel-group [*channel-group-number* **timeslots** *range*]

Cisco AS5350 and Cisco AS5400 Series

channel-group *channel-group-number*
no channel-group *channel-group-number*

Cisco MC3810

channel-group *channel-number* **timeslots** *range* [**speed** {**56** | **64**}]
no channel-group [*channel-number* **timeslots** *range*]

Syntax Description

channel-group-number	Channel-group number on the Cisco 2600 series, Cisco 2600XM, Cisco 2691, Cisco 3631, Cisco 3660, Cisco 3725, and Cisco 3745 routers. When a T1 data line is configured, channel-group numbers can be values from 0 to 23. When an E1 data line is configured, channel-group numbers can be values from 0 to 30. Valid values can be 0 or 1 on the Cisco AS5350 and Cisco AS5400.
timeslots *range*	Specifies one or more time slots separated by commas, and spaces or ranges of time slots belonging to the channel group separated by a dash. The first time slot is numbered 1. • For a T1 controller, the time slots range from 1 to 24. • For an E1 controller, the time slots range from 1 to 31. You can specify a time slot range (for example, 1-29), individual time slots separated by commas (for example 1, 3, 5), or a combination of the two (for example 1-14, 15, 17-31). See the "Examples" section for samples of different timeslot ranges.

speed {**56**|**64**}	(Optional) Specifies the speed of the underlying DS0s in kilobits per second. Valid values are 56 and 64. The default line speed when configuring a T1 controller is 56 kbps on the Cisco 2600 series, Cisco 2600XM series, Cisco 2691, Cisco 3631, Cisco 3660, Cisco 3725, Cisco 3745, and Cisco MC3810. The default line speed when configuring an E1 controller is 64 kbps on the Cisco 2600 series, Cisco 2600XM series, Cisco 2691, Cisco 3631, Cisco 3660, Cisco 3725, Cisco 3745, and Cisco MC3810. The line speed controls real-time (VBR-RT) traffic shaping, and the maximum burst size (MBS) is 255 cells.
aim *aim-slot-number*	(Optional) Directs HDLC traffic from the T1/E1 interface to the AIM-ATM-VOICE-30 digital signaling processor (DSP) card on the Cisco 2600 series, Cisco 2600XM series, Cisco 2691, Cisco 3631, Cisco 3660, Cisco 3725, and Cisco 3745.
channel-number	Number of the channel. Valid values can be 0 or 1 on the Cisco SLT (Cisco 2611).
unframed	Specifies the use of all 32 time slots for data. None of the 32 time slots is used for framing signals on the Cisco ASR 901 Series, Cisco 2600XM series, Cisco 2691, Cisco 3631, Cisco 3660, Cisco 3725, and Cisco 3745. This keyword is applicable to E1 only.

Command Default

The T1/E1 line is connected to the Motorola MPC-860x processor serial communication controller (SCC) or network module with two voice or WAN interface card (VIC or WIC) slots and 0/1/2 FastEthernet ports DSCC4 by default on Cisco 2600 series, Cisco 2600XM, Cisco 2691, Cisco 3631, Cisco 3660, Cisco 3725, and Cisco 3745 routers.

There is no default behavior or values on the Cisco SLT (Cisco 2611).

The serial interface object encapsulation is set to HDLC on a network access server (NAS) (Cisco AS5350 and Cisco AS5400 series routers).

The default line speed is 56 kbps when a T1 controller is configured on the Cisco 2600 series, Cisco 2600XM series, Cisco 2691, Cisco 3631, Cisco 3660, Cisco 3725, Cisco 3745, and the Cisco MC3810.

The default line speed is 64 kbps when an E1 controller is configured on the Cisco 2600 series, Cisco 2600XM series, Cisco 2691, Cisco 3631, Cisco 3660, Cisco 3725, Cisco 3745, and the Cisco MC3810.

Command Modes

Controller configuration (config-controller)

Command History

Release	Modification
11.3MA	This command was introduced on the Cisco MC3810.
12.0	This command was integrated into Cisco IOS Release 12.0 on the Cisco MC3810.
12.0(7)XE	This command was implemented on the Catalyst 6000 family switches.
12.1(1)E	This command was integrated into Cisco IOS Release 12.1(1)E.

Release	Modification
12.1(1)T	This command was modified to accommodate two channel groups on a port on 1- and 2-port T1/E1 multiflex voice or WAN interface cards on the Cisco 2600 and Cisco 3600 series routers.
12.1(3a)E3	The number of valid values for the *kbps*argument was changed on the Cisco MC3810; see the "Usage Guidelines" section for valid values.
12.2(11)T	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(15)T	The **aim** keyword was added for use on the Cisco 2600 series (including the Cisco 2691), Cisco 2600XM, Cisco 3660, Cisco 3725, and Cisco 3745.
12.3(1)	The **unframed** keyword was added for use on the Cisco 2600XM series, Cisco 2691, Cisco 3631, Cisco 3660, Cisco 3725, and Cisco 3745.
12.2SX	This command is supported in the Cisco IOS Release 12.2SX train. Support in a specific 12.2SX release of this train depends on your feature set, platform, and platform hardware.
15.4(3)S	This command was implemented on the Cisco ASR 901 Series Aggregation Services Router.

Usage Guidelines

Use this command to direct HDLC traffic from the T1/E1 interface to the AIM-ATM-VOICE-30 DSP card. A channel group is created using Advanced Integration Module (AIM) HDLC resources when a **channel-group** command with the **aim** keyword is parsed during system initialization or when the command is entered during configuration. You must specify the **aim** keyword under a T1/E1 controller port to direct HDLC traffic from the T1/E1 interface to the AIM-ATM-VOICE-30 DSP card on the Cisco 2600 series, Cisco 2600XM series, Cisco 2691, Cisco 3631, Cisco 3660, Cisco 3725, and Cisco 3745.

Note Neither the Cisco AS5400 series NAS nor the Cisco MC3810 is supported with the integrated voice and data WAN on T1/E1 interfaces using the AIM-ATM-VOICE-30 module.

If previous **channel-group** commands are configured with the **aim** keyword, subsequent **channel-group** commands without the **aim** keyword are rejected. Similarly, if a regular **channel-group** command is followed by another **channel-group** command with the **aim** keyword implemented, the second command is rejected on the Cisco 2600 and Cisco 2600XM.

A channel group using AIM HDLC resources is deleted only when a **nochannel-group** command is entered.

By default, the**channel-group** command on a NAS sets the serial interface object encapsulation to HDLC. You must override the default by entering the **encapsulationss7** command for that serial interface object. Once you override the default, encapsulation cannot be changed again for that object. The SS7 encapsulation option is new to the Integrated Signaling Link Terminal feature and is available only for interface serial objects created by the**channel-group** command. The Integrated Signaling Link Terminal feature added SLT functionality on Cisco AS5350 and Cisco AS5400 platforms.

A digital SS7 link can be deleted by entering the **nochannel-group***channel-group-number* command on the associated T1/E1 controller. The link must first be stopped using the **noshutdown** command. It is not necessary to remove the channel ID association first.

Use the **channel-group** command in configurations where the router or access server must communicate with a T1 or E1 fractional data line. The channel group number may be arbitrarily assigned and must be unique for the controller. The time-slot range must match the time slots assigned to the channel group. The service provider defines the time slots that comprise a channel group.

Note Channel groups, channel-associated signaling (CAS) voice groups, DS0 groups, and time-division multiplexing (TDM) groups all use group numbers. All group numbers configured for channel groups, CAS voice groups, and TDM groups must be unique on the local Cisco MC3810 concentrator. For example, you cannot use the same group number for a channel group and for a TDM group. Furthermore, on the Cisco MC3810, only one channel group can be configured on a controller.

The channel group number can be 0 or 1 on the Cisco SLT (Cisco 2611).

The **channel-group** command also applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

Examples

The following example shows basic configuration directing HDLC traffic from the T1/E1 interface to the AIM-ATM-VOICE-30 DSP card, starting in global configuration mode:

```
Router(config)# controller e1 1/0
Router(config-controller)# clock source internal
Router(config-controller)# channel-group 0 timeslots 1-31 aim 0
```

The following example explicitly sets the encapsulation type to PPP to override the HDLC default:

```
Router# configure terminal
Router(config)# controller t1 6/0
Router(config-controller)# channel-group 2 timeslots 3 aim 0
Router(config-controller)# exit
Router(config)# interface serial 6/0:2
Router(config-if)# encapsulation ppp
Router(config-if)# ip address 10.0.0.1 255.0.0.0
Router(config-if)# no shutdown
Router(config-if)# end
```

The following example shows how to explicitly set the encapsulation type to SS7 to override the HDLC default using the Integrated Signaling Link Terminal feature. This example uses an 8PRI DFC card inserted into slot 7, and DS0-timeslot 3 on trunk 5 of that card is used as an SS7 link:

```
Router# configure terminal
Router(config)# controller t1 7/5
Router(config-controller)# channel-group 2 timeslots 3
Router(config-controller)# exit
Router(config)# interface serial 7/5:2
Router(config-if)# encapsulation ss7
Router(config-if)# channel-id 0
Router(config-if)# no shutdown
Router(config-if)# end
```

The following example defines three channel groups. Channel-group 0 consists of a single time slot, channel-group 8 consists of seven time slots and runs at a speed of 64 kbps per time slot, and channel-group 12 consists of two time slots.

```
Router(config-controller)# channel-group 0 timeslots 1
Router(config-controller)# channel-group 8 timeslots 5,7,12-15,20 speed 64
Router(config-controller)# channel-group 12 timeslots 2
```

The following example configures a channel group on controller T1 0 on a Cisco MC3810:

```
Router(config)# controller T1 0
Router(config-controller)# channel-group 10 timeslots 10-64
```

The following example configures a channel group on controller E1 1 and specifies that all time slots are used for data:

```
controller e1 1
channel-group 1 unframed
```

Note SS7 digital F-link support for the 8PRI line card requires use of a third onboard TDM stream to route trunk DS0 messages to the onboard MGCs.

Related Commands

Command	Description
framing	Specifies the frame type for the T1 or E1 data line.
invert data	Enables channel inversion.
linecode	Specifies the line code type for the T1 or E1 line.
voice-card	Configures a card with voice processing resources and enters voice card configuration mode.
encapsulation	Sets the encapsulation type.

channel-id

To assign a session channel ID to a SS7 serial link or assign an SS7 link to an SS7 session set on a Cisco AS5350 or Cisco AS5400, use the **channel-id** command in interface configuration mode. To disable a session channel ID link, use the **no** form of this command.

channel-id *channel-id* [**session-set** *session-set-id*]
no channel-id

Syntax Description

channel -id	A unique session channel ID. This session channel ID is needed when the link with a Reliable User Datagram Protocol (RUDP) session to the Media Gateway Controller (MGC) is associated.
session -set*session-set-id*	(Optional) Creates an SS7-link-to-SS7-session-set association on the Cisco AS5350- and Cisco AS5400-based Cisco Signaling Link Terminals (SLTs). The *session-set-id*argument represents the SS7 session ID. Valid values are 0 or 1. Default is 0.

Command Default

No default behavior or values

Command Modes

Interface configuration (config-if)

Command History

Release	Modification
12.2(11)T	This command was introduced on the Cisco AS5350 and Cisco AS5400.
12.2(15)T	The **session-set***session-set-id* keyword and argument were added.

Usage Guidelines

The **channel-id** command is visible only if the object's encapsulation type is changed to SS7.

Before an SS7 serial link can be enabled using the **noshutdown** command, you must enter the **channel-id** command in interface configuration mode to assign a session channel ID to the SS7 serial link. This ID is unique to the Cisco AS5350 and Cisco AS5400, and the command is visible only for provisioned objects whose encapsulation type is the new SS7 value.

The channel identifier is reserved when you explicitly assign an ID using the **channel-id** command for the associated serial interface object. This fails if the selected channel identifier is currently assigned to another link or if all channel identifiers are already assigned.

A channel identifier is released when the **nochannel-id** command is entered. The link must first be shut down to do this. If the **nochannel-id** command is used with the Mulitple OPC Support for the Cisco Signaling Link Terminal feature, the associated SS7 link has no channel ID. In this state the link is not fully configured and is incapable of supporting signaling traffic.

If the **session-set** keyword is omitted, the command is applied to SS7 session set 0, which is the default. Reissuing the **session-set** keyword with a different SS7 session ID is sufficient to remove the associated SS7 link from its existing SS7 session set and add it to the new one.

Examples

The following example shows a unique session channel ID zero being assigned to the Cisco AS5350 or Cisco AS5400:

```
Router(config-if)# channel-id 0
```

The following example assigns an SS7 link to an SS7 session set on a Cisco AS5350 or Cisco AS5400:

```
Router(config-if)# channel-id 0 session-set 1
```

Related Commands

Command	Description
channel-group	Assigns a channel group and selects the DS0 timeslot desired for SS7 links.
encapsulation ss7	Sets the encapsulation type to SS7.
no shutdown	Changes the administrative state of a port from out-of-service to in-service .
session-set	Creates an SS7-link-to-SS7-session-set association or to associate an SS7 link with an SS7 session set on the Cisco 2600-based SLT.
ss7 mtp2 variant bellcore	Configures the device for Telcordia (formerly Bellcore) standards.

cipher (voice class)

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

cipher { **ecdsa-cipher** [**curve-size 384**] | **strict-cipher** }]
no cipher

Syntax Description

ecdsa-cipher	(Optional) When the **ecdsa-cipher** keyword is not specified, the SIP TLS process uses larger set of ciphers depending on the support at the Secure Socket Layer (SSL). Following are the cipher suites that are supported: • TLS_ECDHE_ECDSA_WITH_AES_128_GCM_SHA256 • TLS_ECDHE_ECDSA_WITH_AES_256_GCM_SHA384 • TLS13_AES128_GCM_SHA256 • TLS13_AES256_GCM_SHA384 • TLS13_CHACHA20_POLY1305_SHA256
curve-size 384	(Optional) Configures specific size of elliptic curves to be used for a TLS session. You can configue an elliptic curve of size 384 bit.
strict-cipher	(Optional) The **strict-cipher** keyword supports only the TLS Rivest, Shamir, and Adelman (RSA) encryption with the Advanced Encryption Standard-128 (AES-128) cipher suite. Following are the cipher suites that are supported: • TLS_RSA_WITH_AES_128_CBC_SHA • TLS_ECDHE_RSA_WITH_AES_128_CBC_SHA256 • TLS_ECDHE_RSA_WITH_AES_256_CBC_SHA256 • TLS_ECDHE_RSA_WITH_AES_128_GCM_SHA256 • TLS_ECDHE_RSA_WITH_AES_256_GCM_SHA384 • TLS13_AES128_GCM_SHA256 • TLS13_AES256_GCM_SHA384 • TLS13_CHACHA20_POLY1305_SHA256 **Note** When the **strict-cipher** keyword is not specified, the SIP TLS process uses the default set of ciphers depending on the support at the Secure Socket Layer (SSL).

Command Default

No default behavior or values

Command Modes

Voice class configuration (config-class)

Command History

Release	Modification
Cisco IOS XE Amsterdam 17.3.1a	This command was introduced under voice class configuration mode. Introduced support for Yang Model.
Cisco IOS XE 17.14.1a	This command is modified to support TLS version 1.3 ciphers. Introduced support for TLS version 1.3 ciphers Yang Model.

Usage Guidelines

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

By default, the Secure Socket Layer (SSL) on CUBE supports the following cipher suites:

- TLS_ECDHE_ECDSA_AES256_GCM_SHA384
- TLS_ECDHE_ECDSA_AES128_GCM_SHA256
- TLS_ECDHE_RSA_AES128_GCM_SHA256
- TLS_ECDHE_RSA_AES256_GCM_SHA384
- TLS_DHE_RSA_AES256_GCM_SHA384
- TLS_DHE_RSA_AES128_GCM_SHA256
- TLS_DHE_RSA_AES256_CBC_SHA256
- TLS_DHE_RSA_AES128_CBC_SHA256
- TLS_RSA_WITH_AES_128_CBC_SHA
- TLS_RSA_WITH_AES_256_CBC_SHA
- TLS_DHE_RSA_WITH_AES_128_CBC_SHA
- TLS_DHE_RSA_WITH_AES_256_CBC_SHA
- TLS_1_3_AES128_GCM_SHA256
- TLS_1_3_AES256_GCM_SHA384
- TLS_1_3_CHACHA20_POLY1305_SHA256

Examples

The following example illustrates how to create a **voice class tls-profile** and associate elliptic curve settings required for a TLS session:

```
Router(config)#voice class tls-profile 2
Router(config-class)#cipher ecdsa-cipher curve-size 384
```

Related Commands

Command	Description
voice class tls-profile	Provides sub-options to configure the commands that are required for a TLS session.

Command	Description
crypto signaling	Identifies the trustpoint or the **tls-profile** *tag* that is used during the TLS handshake process.

cipher preference (voice class tls-cipher)

To configure preference TLS cipher suite in the list, use the **cipher** *preference* command in voice class configuration mode. To delete the cipher preference, use the **no** form of this command.

cipher *preference cipher-name*

no cipher *preference*

Syntax Description

preference	Specifies list of 13 supported ciphers.
cipher-name	The following lists the names of the cipher suite: • AES128_GCM_SHA256 • AES256_GCM_SHA384 • CHACHA20_POLY1305_SHA256 • DHE_RSA_AES128_GCM_SHA256 • DHE_RSA_AES256_GCM_SHA384 • DHE_RSA_WITH_AES_128_CBC_SHA • DHE_RSA_WITH_AES_256_CBC_SHA • ECDHE_RSA_AES128_GCM_SHA256 • ECDHE_RSA_AES256_GCM_SHA384 • RSA_WITH_AES_128_CBC_SHA • RSA_WITH_AES_256_CBC_SHA • ECDHE_ECDSA_AES128_GCM_SHA256 • ECDHE_ECDSA_AES256_GCM_SHA384

Command Default

No default behavior or values

Command Modes

voice class configuration (config-class)

Command History

Release	Modification
Cisco IOS XE Cupertino 17.8.1a	This command was introduced.

Release	Modification
Cisco IOS XE 17.14.1a	Introduced support for TLS version 1.3 ciphers. Following are the three ciphers supported with TLS version 1.3: • AES128_GCM_SHA256 • AES256_GCM_SHA384 • CHACHA20_POLY1305_SHA256

Usage Guidelines

Use the **cipher** *preference* command in voice class configuration mode.

Examples

The following example illustrates how to create a voice class preference:

```
Device(config)# voice class tls-cipher 100
Device(config-class)# cipher 1 DHE_RSA_AES128_GCM_SHA256
```

Related Commands

Command	Description
voice class tls-cipher	Configures an ordered set of TLS cipher suites
crypto signaling	Identifies the trustpoint or the **tls-profile** *tag* that is used during the TLS handshake process.

clear backhaul-session-manager group stats

To reset the statistics or traffic counters for a specified session group, use the clear **backhaul-session-managergroupstats** command in privileged EXEC mode.

clear backhaul-session-manager group stats {**all** | **name** *group-name*}

Syntax Description

all	All available session groups.
name *group -name*	A specified session group.

Command Default

The statistical information accumulates.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.1(1)T	This command was introduced.
12.2(2)T	This command was implemented on the Cisco 7200.
12.2(4)T	This command was implemented on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco IAD2420 series.
12.2(11)T	This command was implemented on the Cisco AS5350, Cisco AS5400, and Cisco AS5850.

Usage Guidelines

A session is the connection between a client and a server, and a session group is a collection of sessions in a group to implement switchover in case of a session failure. This command clears all statistics that pertain to the backhaul session manager group.

Examples

The following example clears all statistics for all available session groups:

```
Router(config)# clear backhaul-session-manager group stats all
```

Related Commands

Command	Description
show backhaul-session-manager group	Displays status, statistics, or configuration of a specified group or all session groups.

clear call application interface

To clear application interface statistics and event logs, use the **clearcallapplicationinterface**command in privileged EXEC mode.

clear call application interface [[{**aaa** | **asr** | **flash** | **http** | **ram** | **rtsp** | **smtp** | **tftp** | **tts**} [**server** *server*]] [{**event-log** | **stats**}]]

Syntax Description

aaa	Authentication, authorization, and accounting (AAA) interface type.
asr	Automatic speech recognition (ASR) interface type.
flash	Flash memory of the Cisco gateway.
http	HTTP interface type.
ram	Memory of the Cisco gateway.
rtsp	Real-Time Streaming Protocol (RTSP) interface type.
smtp	Simple Mail Transfer Protocol (SMTP) interface type.
tftp	TFTP interface type.
tts	Text-to-speech (TTS) interface type.
server *server*	(Optional) Clears statistics or event logs for the specified server.
event-log	(Optional) Clears event logs.
stats	(Optional) Clears statistic counters.

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

This command resets statistic counters to zero and clears event logs for application interfaces. If you do not use any keywords or arguments, this command clears statistics and event logs for all application interfaces.

Examples

The following example clears statistics and event logs for all application interfaces:

```
Router# clear call application interface
```

Related Commands

Command	Description
call application interface event-log	Enables event logging for external interfaces used by voice applications.
call application interface stats	Enables statistics collection for application interfaces.
clear call application stats	Clears application-level statistics in history and subtracts the statistics from the gateway-level statistics.
show call application interface	Displays event logs and statistics for application interfaces.

clear call application stats

To clear application-level statistics in history and subtract the statistics from the gateway-level statistics, use the **clearcallapplicationstats** command in privileged EXEC mode.

clear call application [**app-tag** *application-name*] **stats**

Syntax Description

app-tag *application-name*	(Optional) Clears statistics for the specified voice application.

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

This command resets application-level counters in history to zero and subtracts the counters from the gateway-level history. If you do not specify an application name, this command clears statistics for all applications at the application level and gateway level.

Note Statistic counters are automatically cleared for an application if the application is deleted with the **nocallapplicationvoice** command or the script is reloaded with the **callapplicationvoiceload** command.

Examples

The following example clears statistics for the application named sample_app:

```
Router# clear call application app-tag sample_app stats
```

Related Commands

Command	Description
call application stats	Enables statistics collection for voice applications.
call application voice load	Reloads the designated Tcl script.
clear call application interface	Clears application interface statistics and event logs.
call application voice	Reloads the designated Tcl script or VoiceXML document.
show call application app-level	Displays application-level statistics for voice applications.
show call application gateway-level	Displays gateway-level statistics for voice application instances.

clear call fallback cache

To clear the cache of the current Calculated Planning Impairment Factor (ICPIF) loss/delay busyout threshold estimates for all IP addresses or a specific IP address, use the **clearcallfallbackcache** command in privileged EXEC mode.

clear call fallback cache [*ip-address*] [**codec** *codec-type*]

Syntax Description

ip -address	(Optional) The target IP address. If no IP address is specified, all IP addresses are cleared.
codec *codec-type*	(Optional) Specifies the associated codec type.

Command Default

If no IP address is specified, all IP addresses are cleared.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810 series routers.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)T	This command was modified. The Public Switching Telephone Network (PSTN) Fallback feature and enhancements were implemented on the Cisco 7200 series routers and integrated into Cisco IOS Release 12.2(4)T.
12.2(4)T2	This command was implemented on the Cisco 7500 series routers.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.
15.0(1)M	This command was modified in a release earlier than Cisco IOS Release 15.0(1)M. The **codec** keyword and *codec-type* argument were added.

Usage Guidelines

If no IP address is specified, the **clearcallfallbackcache** command clears the cache of all CPIF estimates for all IP addresses. The available codec types are, g711alaw, g711ulaw, g723ar53, g723ar63, g723r53, g723r63, g726r16, g726r24, g726r32, g728, g729abr8, g729ar8, g729br8, g729r8, g729r8 pre-ietf, gsmamr-nb, gsmefr, gsmfr, and None.

Examples

The following example clears the cache of the ICPIF estimate for IP address 10.0.0.0:

```
Router#
clear call fallback cache 10.0.0.0
```

Related Commands

Command	Description
show call fallback cache	Displays the current ICPIF estimates for all IP addresses in the call fallback cache.

clear call fallback stats

To clear the call fallback statistics, use the **clearcallfallbackstats** command in privileged EXEC mode.

clear call fallback stats

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.2(2)XB1	This command was implemented on the Cisco AS5850 platform.
12.2(4)T	The PSTN Fallback feature and enhancements were implemented on the Cisco 7200 series and integrated into Cisco IOS Release 12.2(4)T.
12.2(4)T2	This command was implemented on the Cisco 7500 series.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Examples

The following example clears the call fallback statistics:

```
Router# clear call fallback stats
```

Related Commands

Command	Description
show call fallback stats	Displays the call fallback statistics.

clear callmon

To clear call monitor logs, use the **clearcallmon** command in privileged EXEC mode.

clear callmon {**dead-memory** | **trace**}

Syntax Description

dead-memory	Clears unreleased Communication Media Module (CMM) line card memory.
trace	Clears CMM trace buffers.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
15.0(1)M	This command was introduced in a release earlier than Cisco IOS Release 15.0(1)M.

Examples

The following example shows how to clear unreleased CMM memory:

```
Router# clear callmon dead-memory
```

The following example shows how to clear CMM trace buffers:

```
Router# clear callmon trace
```

Related Commands

Command	Description
clear tgrep neighbor	Clears TGREP counters and sessions.

clear call threshold

To clear enabled call threshold statistics, use the **clear call threshold** command in privileged EXEC mode.

clear call threshold interface *type number* {**stats** | **total-calls** [*value*] | **int-calls** [*value*]}

Syntax Description

interface -	Specifies the interface through which calls arrive. Types of interfaces and their numbers depends upon the configured interfaces.
type	Interface type. Values include: • ethernet • fastethernet • GigabitEthernet • serial
number	Interface or subinterface number. For more information about the numbering syntax for your networking device, use the question mark (?) online help function.
stats	Resets all call threshold statistics.
total -calls *value*	Resets the counter when the call volume reaches the specified number. The *value* argument represents call volume. Range is from 0 to 10000 calls. The default is 0.
int-calls *value*	Number of calls transmitted through the interface. The *value*argument clears calls when they reach a specified volume through the interface. Range is from 0 to 10000 calls. The default is 0.

Command Default

The default setting of 0 for the **total-calls**and**int-calls** keywords reset all threshold statistics immediately. **stats** is the default keyword.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T. Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 is not included in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on the Cisco 1750 and Cisco 1751 routers. Support for other Cisco platforms is not included in this release.
12.2(8)T	This command was integrated into the Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series routers. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.

Release	Modification
12.2(11)T	This command was implemented on the Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.

Examples

The following example resets all call threshold statistics:

```
clear call threshold stats
```

The following example resets the counter for all call volume in the gateway:

```
clear call threshold total-calls
```

The following example resets the counter when the call volume on Ethernet interface 0/1 reaches 5000 calls:

```
clear call threshold interface ethernet 0/1 int-calls 5000
```

The following example resets the counter for all call threshold statistics on a GigabitEthernet interface:

```
Device# clear call threshold interface GigabitEthernet stats
```

Related Commands

Command	Description
call threshold	Enables the global resources of a gateway.
call threshold poll-interval	Enables a polling interval threshold for CPU or memory.
show call treatment	Displays the call treatment configuration and statistics for handling the calls on the basis of resource availability.

clear call treatment stats

To clear call treatment statistics, use the **clearcalltreatmentstats** command in privileged EXEC mode.

clear call treatment stats

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	The command was integrated into Cisco IOS Release 12.2(4)T. Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 series is not included in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on the Cisco 1750 and Cisco 1751 routers. Support for other Cisco platforms is not included in this release.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series routers. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This feature was integrated into Cisco IOS Release 12.2(11)T and support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5800.

Examples

The following example clears the call treatment statistics:

```
clear call treatment stats
```

Related Commands

Command	Description
call treatment on	Enables call treatment to process calls when local resources are unavailable.
call treatment action	Configures the action that the router takes when local resources are unavailable.
call treatment cause-code	Specifies the reason for the disconnection to the caller when local resources are unavailable.
call treatment isdn-reject	Specifies the rejection cause-code for ISDN calls when local resources are unavailable.
show call treatment	Displays the call treatment configuration and statistics for handling calls on the basis of resource availability.

clear call voice

To clear one or more voice calls detected as inactive because there is no RTP or RTCP activity, use the **clearcallvoice** command in User EXEC or privileged EXEC mode.

clear call voice causecode *identifier* {**id** *identifier* | **media-inactive** | **calling-number** *number* | **called-number** *number* | **fpi-correlator***correlator-id*}

Syntax Description

causecode	Specifies a Q.850 disconnect cause code.
identifier	Numeric cause code identifier; a number 1 through 127.
id *identifier*	Clears one specific call with the ID specified. The identifier argument is the call identifier as shown in brief format.
media -inactive	Clears calls wherever a status of media inactive is detected and notified.
calling-number *number*	Clears a call with a specific calling number pattern. The *number* argument is the specific call number pattern of the calling number.
called-number *number*	Clears a call with a specific called number pattern. The *number* argument is the specific call number pattern of a called number.
fpi-correlator*correlator-id*	Clears calls based on VoIP FPI correlator.

Command Default

This command is disabled, and no calls are cleared.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Release	Modification
12.1(3)T	This command was introduced.
12.2	This command was integrated into Cisco IOS Release 12.2.
12.3(4)T	The **voice**keyword was added.
12.4(4)T	The **calling-number**and **called-number**keywords were added.
15.5(2)T	The **fpi-correlator** keyword was added.

Usage Guidelines

This command can be used to clear call resources at all the layers if there is a hung call. There is no no form of this command.

You can obtain the FPI correlator ID by executing the **show voip fpi calls** command.

Examples

The following example clears inactive voice calls with the cause code ID 16:

```
Router# clear call voice causecode 16 fpi-correlator 2
```

Related Commands

Command	Description
show call active voice	Displays active voice calls, based on specified parameters.

clear call-router routes

To remove the dynamic routes cached in the border element (BE), use the **clearcall-routerroutes** command in privileged EXEC mode.

clear call-router routes

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T. Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 is not included in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T. and implemented on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850.

Examples

The following example shows how to remove dynamic routes cached in the BE:

```
Router# clear call-router routes
```

Related Commands

Command	Description
call-router	Enables the Annex G BE configuration commands.
show call history	Displays the fax history table for a fax transmission.

clear controller call-counters

To clear the system DS0 high water marks (HWMs) and all individual controller statistics, use the **clearcontrollercall-counters** command in privileged EXEC mode.

clear controller call-counters {**system-hwm** | **all**}

Syntax Description

system -hwm	Clears the system HWMs only.
all	Clears all controller call counters including the individual controller time slots in use and the number of calls on those time slots since the last reset was done. The HWMs are set to 0.

Command Default

No default behavior or values

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.0(1)T	This command was introduced.
12.1(1)T	This command was implemented on the voice/WAN interface cards (VWICs) for the Cisco 2600 series and Cisco 3600 series.
12.1(2)T	This command was implemented on the Cisco AS5300, Cisco AS5400, and Cisco AS5800.

Usage Guidelines

The **clearcontrollercall-countersall**command clears the system DS0 HWMs and all individual controller statistics, including TotalCalls and Total Duration. The**clearcontrollercall-counterssystem-hwm** command clears the system DS0 HWMs and leaves all other call-counter statistics untouched.

Refer to the following comments for the meaning of call counters displayed before and after executing the **clearcontrollercall-counters** and **clearcontrollert1call-counters** related commands:

- The numbers displayed under TotalCalls for each time slot represent total calls that were connected successfully. If a call comes into time slot 10, then the **showcontrollerst1call-counters** command displays 1 under the TotalCalls column for time slot 10. A value of 20 displayed under TotalCalls for time slot 10 indicates a total of 20 calls connected on time slot 10 since the last time call counters were cleared.
- The DS0s Active field indicates the number of active calls on the specified controller. This number indicates the current number of calls on the controller at any given time.
- The DS0s Active High Water Mark field indicates the peak number of calls on the controller since the last time HWMs or calls were cleared. If the number of active calls "DS0s Active" is less than the DS0s HWM, then the HWM remains untouched. If new calls come in and the active DS0s are more than the HWM, then the HWM is incremented to reflect the new peak number of calls on that controller.

This value is reset to the current and active DS0s when call counters are cleared. For example, initially the HWM is 0. When a new call comes in, the HWM is 1. When the next call comes in, the HWM is 2.

If 20 calls come in, the HWM is 20 and the active DS0s are 20. If 5 calls get disconnected, the DS0 active is 15, but the HWM is 20. When a **clearcontroller** command is input for the specified controller, the HWM is reset to 15, which is the current and active DS0s also. If 10 calls get disconnected, the Active DS0s is set to 5 and the HWM remains at 15 until another **clearcontroller** command is input. If Active DS0s exceed 15, then the HWM is updated.

- The System DS0s High Water Mark field reflects the HWM at a system level including all DS0s controllers.

Examples

The following sample output shows what happens after the HWMs are cleared:

```
Router# clear controller call-counters system-hwm
!
Router# show controllers t1 call-counters
T1 1/3/0:3:
  DS0's Active: 2
  DS0's Active High Water Mark: 2
  TimeSlot   Type   TotalCalls   TotalDuration
      1       pri           0       00:00:00
      2       pri           0       00:00:00
      3       pri           0       00:00:00
      4       pri           0       00:00:00
      5       pri           0       00:00:00
      6       pri           0       00:00:00
      7       pri           0       00:00:00
      8       pri           0       00:00:00
      9       pri           0       00:00:00
     10       pri           0       00:00:00
     11       pri           0       00:00:00
     12       pri           0       00:00:00
     13       pri           0       00:00:00
     14       pri           0       00:00:00
     15       pri           0       00:00:00
     16       pri           0       00:00:00
     17       pri           0       00:00:00
     18       pri           0       00:00:00
     19       pri           0       00:00:00
     20       pri           0       00:00:00
     21       pri           0       00:00:00
     22       pri           1       00:08:51
     23       pri           1       00:09:21
T1 1/3/0:8:
  DS0's Active: 1
  DS0's Active High Water Mark: 1
  TimeSlot   Type   TotalCalls   TotalDuration
      1       pri           0       00:00:00
      2       pri           0       00:00:00
      3       pri           0       00:00:00
      4       pri           0       00:00:00
      5       pri           0       00:00:00
      6       pri           0       00:00:00
      7       pri           0       00:00:00
      8       pri           0       00:00:00
      9       pri           0       00:00:00
     10       pri           0       00:00:00
     11       pri           0       00:00:00
     12       pri           0       00:00:00
     13       pri           0       00:00:00
     14       pri           0       00:00:00
     15       pri           0       00:00:00
     16       pri           0       00:00:00
     17       pri           0       00:00:00
     18       pri           0       00:00:00
```

```
      19       pri           0       00:00:00
      20       pri           0       00:00:00
      21       pri           0       00:00:00
      22       pri           0       00:01:39
      23       pri           0       00:00:00
System's DS0's Active High Water Mark: 3
```

In the example above, the system HWM is reset to the total number of active calls in the system, which is 3. The number was 4. When a call goes down, HWM values are untouched. Only the DS0 Active value changes. Above, there is only one call on 1/3/0:3. Observe the HWM for individual controllers. Total number of active calls is 1.

The following is sample output when the **clearcontrollercall-counterssystem-hwm** command is used:

```
Router# clear controller call-counters system-hwm
!
Router# show controllers t1 call-counters
T1 1/3/0:3:
  DS0's Active: 1
  DS0's Active High Water Mark: 2
  TimeSlot   Type   TotalCalls   TotalDuration
       1       pri           0       00:00:00
       2       pri           0       00:00:00
       3       pri           0       00:00:00
       4       pri           0       00:00:00
       5       pri           0       00:00:00
       6       pri           0       00:00:00
       7       pri           0       00:00:00
       8       pri           0       00:00:00
       9       pri           0       00:00:00
      10       pri           0       00:00:00
      11       pri           0       00:00:00
      12       pri           0       00:00:00
      13       pri           0       00:00:00
      14       pri           0       00:00:00
      15       pri           0       00:00:00
      16       pri           0       00:00:00
      17       pri           0       00:00:00
      18       pri           0       00:00:00
      19       pri           0       00:00:00
      20       pri           0       00:00:00
      21       pri           0       00:00:00
      22       pri           1       00:12:16
      23       pri           1       00:10:20
T1 1/3/0:8:
  DS0's Active: 0
  DS0's Active High Water Mark: 1
  TimeSlot   Type   TotalCalls   TotalDuration
       1       pri           0       00:00:00
       2       pri           0       00:00:00
       3       pri           0       00:00:00
       4       pri           0       00:00:00
       5       pri           0       00:00:00
       6       pri           0       00:00:00
       7       pri           0       00:00:00
       8       pri           0       00:00:00
       9       pri           0       00:00:00
      10       pri           0       00:00:00
      11       pri           0       00:00:00
      12       pri           0       00:00:00
      13       pri           0       00:00:00
      14       pri           0       00:00:00
```

```
     15       pri           0       00:00:00
     16       pri           0       00:00:00
     17       pri           0       00:00:00
     18       pri           0       00:00:00
     19       pri           0       00:00:00
     20       pri           0       00:00:00
     21       pri           0       00:00:00
     22       pri           0       00:02:50
     23       pri           0       00:00:00
System's DS0's Active High Water Mark: 1
```

In the preceding example, only the system HWM is reset to active. For controllers 1/3/0:3 and 1/3/0:8, the HWMs are untouched.

The following is sample output when the **all**keyword is used, clearing at the system level:

```
Router# clear controller call-counters all
!
Router# show controllers t1 call-counters
T1 1/3/0:3:
  DS0's Active: 0
  DS0's Active High Water Mark: 0
  TimeSlot   Type   TotalCalls   TotalDuration
      1       pri           0       00:00:00
      2       pri           0       00:00:00
      3       pri           0       00:00:00
      4       pri           0       00:00:00
      5       pri           0       00:00:00
      6       pri           0       00:00:00
      7       pri           0       00:00:00
      8       pri           0       00:00:00
      9       pri           0       00:00:00
     10       pri           0       00:00:00
     11       pri           0       00:00:00
     12       pri           0       00:00:00
     13       pri           0       00:00:00
     14       pri           0       00:00:00
     15       pri           0       00:00:00
     16       pri           0       00:00:00
     17       pri           0       00:00:00
     18       pri           0       00:00:00
     19       pri           0       00:00:00
     20       pri           0       00:00:00
     21       pri           0       00:00:00
     22       pri           0       00:00:00
     23       pri           0       00:00:00
T1 1/3/0:8:
  DS0's Active: 0
  DS0's Active High Water Mark: 0
  TimeSlot   Type   TotalCalls   TotalDuration
      1       pri           0       00:00:00
      2       pri           0       00:00:00
      3       pri           0       00:00:00
      4       pri           0       00:00:00
      5       pri           0       00:00:00
      6       pri           0       00:00:00
      7       pri           0       00:00:00
      8       pri           0       00:00:00
      9       pri           0       00:00:00
     10       pri           0       00:00:00
     11       pri           0       00:00:00
     12       pri           0       00:00:00
     13       pri           0       00:00:00
     14       pri           0       00:00:00
     15       pri           0       00:00:00
     16       pri           0       00:00:00
     17       pri           0       00:00:00
     18       pri           0       00:00:00
     19       pri           0       00:00:00
     20       pri           0       00:00:00
     21       pri           0       00:00:00
     22       pri           0       00:00:00
     23       pri           0       00:00:00
System's DS0's Active High Water Mark: 0
```

In the preceding example, clearing at the system level using the **clearcontrollercall-counters**command clears all DS0 controllers in the system and also clears the system HWMs.

Related Commands

Command	Description
clear controller t1 call-counters	Clears call statistics on a specific T1 controller.
controller	Enters controller configuration mode.
show controllers t1 call-counters	Displays the total number of calls and call durations on a T1 controller.

clear controller t1

To clear the system DS0 high water marks (HWM) and all individual controller statistics, use the **clearcontrollert1** command in privileged EXEC mode.

clear controller t1 [*slot*] **call-counters** *timeslots* **firmware-status**

Syntax Description

slot	(Optional) Clears an individual T1 controller.
call -counters*timeslots*	Clears the call counters in the specified T1 time slots.
firmware -status	Clears the Neat crash history.

Command Default

No default behavior or values

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.0(1)T	This command was introduced.
12.1(1)T	This command was implemented on the voice and WAN interface cards (VWICs) for the Cisco 2600 series and Cisco 3600 series.
12.1(2)T	This command was implemented on the Cisco AS5300, Cisco AS5400, and Cisco AS5800.

Usage Guidelines

Refer to the following comments for the meaning of call counters displayed before and after executing **clearcontrollert1** related commands:

- The numbers displayed under TotalCalls for each time slot represent total calls that were connected successfully. If a call comes into time slot 10, then the **showcontrollerst1call-counters** command displays 1 under the TotalCalls column for time slot 10. A value of 20 displayed under TotalCalls for time slot 10 indicates a total of 20 calls connected on time slot 10 since the last time call counters were cleared.

If a time slot or time slot range is specified, only the counters for those channels are cleared. The TotalCalls field shows the time slots that have calls connected since the last clear was done and does not show the number of active calls in the controller. The TotalDuration field shows the same information as the TotalCalls field.

- The DS0's Active field indicates the number of active calls on the specified controller. This number indicates the current number of calls on the controller at any given time.
- The DS0's Active High Water Mark field indicates the peak number of calls on the controller since the last **clearcontrollert11/0/0call-counters** command was entered. If the number of active calls "DS0's Active" is less than DS0s HWM, then HWM remains untouched. If new calls come in and the active DS0s are more than the HWM, then the HWM is incremented to reflect the new peak number of calls on that controller.

This value is reset to the current and active DS0s when the **clearcontrollert11/3/0call-counters**command is entered. For example, initially the HWM is 0. When a new call comes in, the HWM is 1. When the next call comes in, the HWM is 2.

If 20 calls come in, the HWM is 20 and the active DS0s are 20. If 5 calls get disconnected, the DS0 active is 15, but the HWM is 20. When a **clearcontroller** command is input for the specified controller, the HWM is reset to 15, which is the current and active DS0s also. If 10 calls get disconnected, the Active DS0s value is set to 5 and the HWM remains at 15 until another **clearcontroller**command is input. If Active DS0s exceed 15, then the HWM is updated.

- The System DS0s High Water Mark field reflects the HWM at a system level including all DS0s controllers.

Examples

The following is sample output that shows two controllers numbered 1/3/0:3 and 1/3/0:8. Note the differences in the output shown by the **showcontrollerst1call-counters** command and how the **clearcontrollert1call-counters** command affects the output:

```
Router#  show controllers t1 call-counters
T1 1/3/0:3:
  DS0's Active: 0
  DS0's Active High Water Mark: 0
  TimeSlot   Type   TotalCalls   TotalDuration
      1       pri           0       00:00:00
      2       pri           0       00:00:00
      3       pri           0       00:00:00
      4       pri           0       00:00:00
      5       pri           0       00:00:00
      6       pri           0       00:00:00
      7       pri           0       00:00:00
      8       pri           0       00:00:00
      9       pri           0       00:00:00
     10       pri           0       00:00:00
     11       pri           0       00:00:00
     12       pri           0       00:00:00
     13       pri           0       00:00:00
     14       pri           0       00:00:00
     15       pri           0       00:00:00
     16       pri           0       00:00:00
     17       pri           0       00:00:00
     18       pri           0       00:00:00
     19       pri           0       00:00:00
     20       pri           0       00:00:00
     21       pri           0       00:00:00
     22       pri           0       00:00:00
     23       pri           0       00:00:00
T1 1/3/0:8:
  DS0's Active: 0
  DS0's Active High Water Mark: 0
  TimeSlot   Type   TotalCalls   TotalDuration
      1       pri           0       00:00:00
      2       pri           0       00:00:00
      3       pri           0       00:00:00
      4       pri           0       00:00:00
      5       pri           0       00:00:00
      6       pri           0       00:00:00
      7       pri           0       00:00:00
      8       pri           0       00:00:00
      9       pri           0       00:00:00
     10       pri           0       00:00:00
     11       pri           0       00:00:00
     12       pri           0       00:00:00
     13       pri           0       00:00:00
     14       pri           0       00:00:00
     15       pri           0       00:00:00
     16       pri           0       00:00:00
     17       pri           0       00:00:00
     18       pri           0       00:00:00
     19       pri           0       00:00:00
     20       pri           0       00:00:00
     21       pri           0       00:00:00
     22       pri           0       00:00:00
     23       pri           0       00:00:00
System's DS0's Active High Water Mark: 0
```

Note In the preceding example, all the fields are zero, indicating that no calls have come in since system startup or since the last clear was made by the **clearcontroller**command.

The following is sample output that shows that four calls have been initiated on the 1/5/12, 1/5/13, 1/5/14, and 1/5/15 controllers:

```
Router# show users
    Line       User       Host(s)              Idle       Location
*  0 con 0                idle                 00:00:00
  tty 1/5/12   Router Async interface      00:01:05   PPP: 55.61.1.1
  tty 1/5/13   Router Async interface      00:00:48   PPP: 55.62.1.1
  tty 1/5/14   Router Async interface      00:00:33   PPP: 55.54.1.1
  tty 1/5/15   Router Async interface      00:00:19   PPP: 55.52.1.1
  Interface  User      Mode                     Idle Peer Address
Router# show controllers t1 call-counters
T1 1/3/0:3:
  DS0's Active: 2
  DS0's Active High Water Mark: 2
  TimeSlot   Type   TotalCalls   TotalDuration
      1       pri           0       00:00:00
      2       pri           0       00:00:00
      3       pri           0       00:00:00
      4       pri           0       00:00:00
      5       pri           0       00:00:00
      6       pri           0       00:00:00
      7       pri           0       00:00:00
      8       pri           0       00:00:00
      9       pri           0       00:00:00
     10       pri           0       00:00:00
     11       pri           0       00:00:00
     12       pri           0       00:00:00
     13       pri           0       00:00:00
     14       pri           0       00:00:00
     15       pri           0       00:00:00
     16       pri           0       00:00:00
     17       pri           0       00:00:00
     18       pri           0       00:00:00
     19       pri           0       00:00:00
     20       pri           0       00:00:00
     21       pri           0       00:00:00
     22       pri           1       00:01:58
     23       pri           1       00:02:27
T1 1/3/0:8:
  DS0's Active: 2
  DS0's Active High Water Mark: 2
  TimeSlot   Type   TotalCalls   TotalDuration
      1       pri          0       00:00:00
      2       pri          0       00:00:00
      3       pri          0       00:00:00
      4       pri          0       00:00:00
      5       pri          0       00:00:00
      6       pri          0       00:00:00
      7       pri          0       00:00:00
      8       pri          0       00:00:00
      9       pri          0       00:00:00
     10       pri          0       00:00:00
     11       pri          0       00:00:00
     12       pri          0       00:00:00
     13       pri          0       00:00:00
     14       pri          0       00:00:00
     15       pri          0       00:00:00
     16       pri          0       00:00:00
     17       pri          0       00:00:00
     18       pri          0       00:00:00
     19       pri          0       00:00:00
     20       pri          0       00:00:00
     21       pri          0       00:00:00
     22       pri          1       00:02:14
     23       pri          1       00:02:46
System's DS0's Active High Water Mark: 4
```

In the preceding example , if a**clearcontroller** command is entered for a controller that has active calls, which have been connected during the last 30 minutes, the TotalCalls and TotalDuration fields are reset to zero.

The following is sample output that shows controller 1/3/0:3, with time slots 22 and 23 connected and active. When the**clearcontrollert11/3/0:3call-counters**command is entered, the corresponding fields are set to zero.

```
Router# clear controller t1 1/3/0:3 call-counters
!
Router# show controllers t1 call-counters
T1 1/3/0:3:
  DS0's Active: 2
  DS0's Active High Water Mark: 2
  TimeSlot   Type   TotalCalls   TotalDuration
      1       pri          0       00:00:00
      2       pri          0       00:00:00
      3       pri          0       00:00:00
      4       pri          0       00:00:00
      5       pri          0       00:00:00
      6       pri          0       00:00:00
      7       pri          0       00:00:00
      8       pri          0       00:00:00
      9       pri          0       00:00:00
     10       pri          0       00:00:00
     11       pri          0       00:00:00
     12       pri          0       00:00:00
     13       pri          0       00:00:00
     14       pri          0       00:00:00
     15       pri          0       00:00:00
     16       pri          0       00:00:00
     17       pri          0       00:00:00
     18       pri          0       00:00:00
     19       pri          0       00:00:00
     20       pri          0       00:00:00
     21       pri          0       00:00:00
     22       pri          1       00:29:14
     23       pri          1       00:29:47
Router# clear controller t1 1/3/0:3 call-counters
Router# show controllers t1 call-counters
T1 1/3/0:3:
  DS0's Active: 2
  DS0's Active High Water Mark: 2
  TimeSlot   Type   TotalCalls   TotalDuration
      1       pri          0       00:00:00
      2       pri          0       00:00:00
      3       pri          0       00:00:00
      4       pri          0       00:00:00
      5       pri          0       00:00:00
      6       pri          0       00:00:00
      7       pri          0       00:00:00
      8       pri          0       00:00:00
      9       pri          0       00:00:00
     10       pri          0       00:00:00
     11       pri          0       00:00:00
     12       pri          0       00:00:00
     13       pri          0       00:00:00
     14       pri          0       00:00:00
     15       pri          0       00:00:00
     16       pri          0       00:00:00
     17       pri          0       00:00:00
     18       pri          0       00:00:00
     19       pri          0       00:00:00
     20       pri          0       00:00:00
     21       pri          0       00:00:00
     22       pri          0       00:00:10   <<<<<<
     23       pri          0       00:00:10   <<<<<<
```

The following is sample output when a call is cleared on 1/5/12:

```
Router# clear line 1/5/12
[confirm]
 [OK]
!
Router# show users
    Line       User       Host(s)              Idle       Location
*  0 con 0                idle                 00:00:00
  tty 1/5/13   Router Async interface      00:03:04   PPP: 55.62.1.1
  tty 1/5/14   Router Async interface      00:02:49   PPP: 55.54.1.1
  tty 1/5/15   Router Async interface      00:02:35   PPP: 55.52.1.1
  Interface  User      Mode                     Idle Peer Address
Router# show controllers t1 call-counters
T1 1/3/0:3:
  DS0's Active: 2
  DS0's Active High Water Mark: 2
  TimeSlot   Type   TotalCalls   TotalDuration
      1       pri          0       00:00:00
      2       pri          0       00:00:00
      3       pri          0       00:00:00
      4       pri          0       00:00:00
      5       pri          0       00:00:00
      6       pri          0       00:00:00
      7       pri          0       00:00:00
      8       pri          0       00:00:00
      9       pri          0       00:00:00
     10       pri          0       00:00:00
     11       pri          0       00:00:00
     12       pri          0       00:00:00
     13       pri          0       00:00:00
     14       pri          0       00:00:00
     15       pri          0       00:00:00
```

```
      16       pri           0       00:00:00
      17       pri           0       00:00:00
      18       pri           0       00:00:00
      19       pri           0       00:00:00
      20       pri           0       00:00:00
      21       pri           0       00:00:00
      22       pri           1       00:03:44
      23       pri           1       00:04:14
T1 1/3/0:8:
  DS0's Active: 1
  DS0's Active High Water Mark: 2
  TimeSlot   Type   TotalCalls   TotalDuration
       1       pri           0       00:00:00
       2       pri           0       00:00:00
       3       pri           0       00:00:00
       4       pri           0       00:00:00
       5       pri           0       00:00:00
       6       pri           0       00:00:00
       7       pri           0       00:00:00
       8       pri           0       00:00:00
       9       pri           0       00:00:00
      10       pri           0       00:00:00
      11       pri           0       00:00:00
      12       pri           0       00:00:00
      13       pri           0       00:00:00
      14       pri           0       00:00:00
      15       pri           0       00:00:00
      16       pri           0       00:00:00
      17       pri           0       00:00:00
      18       pri           0       00:00:00
      19       pri           0       00:00:00
      20       pri           0       00:00:00
      21       pri           0       00:00:00
      22       pri           1       00:04:00
      23       pri           1       00:03:34
System's DS0's Active High Water Mark: 4
```

After a call gets disconnected, only the DS0 Active field changes to reflect the current active call on the controller. In the above example, 1/3/0:8 DS0 Active is changed to 1.

The following is sample output that shows call counters are cleared for an individual controller on 1/3/0:8:

```
Router# clear controller t1 1/3/0:8 call-counters
!
Router# show controllers t1 call-counters
T1 1/3/0:3:
  DS0's Active: 2
  DS0's Active High Water Mark: 2
  TimeSlot   Type   TotalCalls   TotalDuration
       1       pri           0       00:00:00
       2       pri           0       00:00:00
       3       pri           0       00:00:00
       4       pri           0       00:00:00
       5       pri           0       00:00:00
       6       pri           0       00:00:00
       7       pri           0       00:00:00
       8       pri           0       00:00:00
       9       pri           0       00:00:00
      10       pri           0       00:00:00
      11       pri           0       00:00:00
      12       pri           0       00:00:00
      13       pri           0       00:00:00
```

```
     14       pri           0       00:00:00
     15       pri           0       00:00:00
     16       pri           0       00:00:00
     17       pri           0       00:00:00
     18       pri           0       00:00:00
     19       pri           0       00:00:00
     20       pri           0       00:00:00
     21       pri           0       00:00:00
     22       pri           1       00:07:46
     23       pri           1       00:08:15
T1 1/3/0:8:
  DS0's Active: 1
  DS0's Active High Water Mark: 1
  TimeSlot   Type   TotalCalls   TotalDuration
      1       pri           0       00:00:00
      2       pri           0       00:00:00
      3       pri           0       00:00:00
      4       pri           0       00:00:00
      5       pri           0       00:00:00
      6       pri           0       00:00:00
      7       pri           0       00:00:00
      8       pri           0       00:00:00
      9       pri           0       00:00:00
     10       pri           0       00:00:00
     11       pri           0       00:00:00
     12       pri           0       00:00:00
     13       pri           0       00:00:00
     14       pri           0       00:00:00
     15       pri           0       00:00:00
     16       pri           0       00:00:00
     17       pri           0       00:00:00
     18       pri           0       00:00:00
     19       pri           0       00:00:00
     20       pri           0       00:00:00
     21       pri           0       00:00:00
     22       pri           0       00:00:35
     23       pri           0       00:00:00
System's DS0's Active High Water Mark: 4
```

In the previous example, after clearing call counters for controller 1/3/0:8, TotalCalls and TotalDuration reset. In addition the DS0 HWM is also cleared to the number of active DS0s. Whenever the DS0 HWM is cleared, it does not reset to zero, but rather it is set to Active DS0s. For 1/3/0:8, the HWM is 1 after clearing because DS0 Active is 1 (1 active call). TotalDuration is 35 seconds for time slot 22, and TotalCall is 0 because they got reset when the **clearcontrollercall-counters** command was entered. Total calls on this time slot is incremented when a new call comes in on this time slot.

The following is sample output when controller 1/5/15 is cleared:

```
Router# clear line 1/5/15
[confirm]
 [OK]
Router# show controllers t1 call-counters
T1 1/3/0:3:
  DS0's Active: 0
  DS0's Active High Water Mark: 2
  TimeSlot   Type   TotalCalls   TotalDuration
      1       pri           0       00:00:00
      2       pri           0       00:00:00
      3       pri           0       00:00:00
      4       pri           0       00:00:00
      5       pri           0       00:00:00
      6       pri           0       00:00:00
      7       pri           0       00:00:00
      8       pri           0       00:00:00
      9       pri           0       00:00:00
     10       pri           0       00:00:00
     11       pri           0       00:00:00
     12       pri           0       00:00:00
     13       pri           0       00:00:00
     14       pri           0       00:00:00
     15       pri           0       00:00:00
     16       pri           0       00:00:00
     17       pri           0       00:00:00
     18       pri           0       00:00:00
     19       pri           0       00:00:00
     20       pri           0       00:00:00
     21       pri           0       00:00:00
     22       pri           1       00:12:40
     23       pri           1       00:10:20
T1 1/3/0:8:
  DS0's Active: 0
  DS0's Active High Water Mark: 1
  TimeSlot   Type   TotalCalls   TotalDuration
      1       pri           0       00:00:00
      2       pri           0       00:00:00
      3       pri           0       00:00:00
      4       pri           0       00:00:00
      5       pri           0       00:00:00
      6       pri           0       00:00:00
      7       pri           0       00:00:00
      8       pri           0       00:00:00
      9       pri           0       00:00:00
     10       pri           0       00:00:00
     11       pri           0       00:00:00
     12       pri           0       00:00:00
     13       pri           0       00:00:00
     14       pri           0       00:00:00
     15       pri           0       00:00:00
     16       pri           0       00:00:00
     17       pri           0       00:00:00
     18       pri           0       00:00:00
     19       pri           0       00:00:00
     20       pri           0       00:00:00
     21       pri           0       00:00:00
     22       pri           0       00:02:50
     23       pri           0       00:00:00
System's DS0's Active High Water Mark: 1
```

The following is sample output showing four active calls:

```
Router# show users
Line       User       Host(s)              Idle       Location
*  0 con 0                idle                 00:00:00
  tty 1/5/16   Router Async interface      00:01:01   PPP: 55.1.1.1
  tty 1/5/17   Router Async interface      00:00:47   PPP: 55.2.1.1
  tty 1/5/18   Router Async interface      00:00:28   PPP: 55.3.1.1
  tty 1/5/19   Router Async interface      00:00:14   PPP: 55.4.1.1
  Interface  User      Mode                     Idle Peer Address
Router# show controllers t1 call-counters
T1 1/3/0:3:
  DS0's Active: 2
  DS0's Active High Water Mark: 2
  TimeSlot   Type   TotalCalls   TotalDuration
      1       pri           0       00:00:00
      2       pri           0       00:00:00
      3       pri           0       00:00:00
      4       pri           0       00:00:00
      5       pri           0       00:00:00
      6       pri           0       00:00:00
      7       pri           0       00:00:00
      8       pri           0       00:00:00
      9       pri           0       00:00:00
     10       pri           0       00:00:00
     11       pri           0       00:00:00
     12       pri           0       00:00:00
     13       pri           0       00:00:00
     14       pri           0       00:00:00
     15       pri           0       00:00:00
     16       pri           0       00:00:00
     17       pri           0       00:00:00
     18       pri           0       00:00:00
     19       pri           0       00:00:00
     20       pri           0       00:00:00
     21       pri           0       00:00:00
     22       pri           1       00:00:57
     23       pri           1       00:01:30
T1 1/3/0:8:
  DS0's Active: 2
  DS0's Active High Water Mark: 2
  TimeSlot   Type   TotalCalls   TotalDuration
      1       pri           0       00:00:00
      2       pri           0       00:00:00
      3       pri           0       00:00:00
      4       pri           0       00:00:00
      5       pri           0       00:00:00
      6       pri           0       00:00:00
      7       pri           0       00:00:00
      8       pri           0       00:00:00
      9       pri           0       00:00:00
     10       pri           0       00:00:00
     11       pri           0       00:00:00
     12       pri           0       00:00:00
     13       pri           0       00:00:00
     14       pri           0       00:00:00
     15       pri           0       00:00:00
     16       pri           0       00:00:00
     17       pri           0       00:00:00
     18       pri           0       00:00:00
     19       pri           0       00:00:00
     20       pri           0       00:00:00
     21       pri           0       00:00:00
     22       pri           1       00:01:12
     23       pri           1       00:01:45
System's DS0's Active High Water Mark: 4
```

Related Commands

Command	Description
clear controller call-counters	Clears all call statistics or system HWMs on a router.
controller	Enters controller configuration mode.
show controllers t1 call-counters	Displays the total number of calls and call durations on a T1 controller.

clear csm-statistics modem

To clear the call switching module (CSM) statistics for a modem or group of modems, use the **clearcsm-statisticsmodem** command in privileged EXEC mode.

clear csm-statistics modem[{*slot/portmodem-group-number*}]

Syntax Description

slot /port	(Optional) Identifies the location (and thereby the identity) of a specific modem.
modem -group-number	(Optional) Designates a defined modem group.

Command Default

No default behaviors or values

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
11.3NA	This command was introduced.

Usage Guidelines

Use the **clearcsm-statisticsmodem**command to clear CSM statistics for a particular modem or group of modems. If the *slot*/*port* argument is specified, the CSM call statistics for calls using the identified modem is cleared. If a modem group number is specified, then the CSM call statistics for calls using the modems associated with that group are cleared. If no argument is specified, all CSM call statistics for all modems are cleared.

Examples

The following example clears CSM call statistics for calls coming in on modems associated with modem group 2:

```
Router# clear csm-statistics modem 2
```

Related Commands

Command	Description
clear csm-statistics voice	Clears the CSM statistics for a particular or for all DSP channels.

clear csm-statistics voice

To clear the call switching module (CSM) statistics for a particular channel or for all digital signal processor (DSP) channels, use the **clearcsm-statisticsvoice** command in privileged EXEC mode.

clear csm-statistics voice[{*slot/dspm/dsp/dsp-channel*}]

Syntax Description

slot / *dspm* / *dsp* / *dsp -channel*	(Optional) Identifies the location of a particular DSP channel.

Command Default

No default behaviors or values

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3NA	This command was introduced.

Usage Guidelines

Use the **clearcsm-statisticsvoice**command to clear CSM statistics for a particular DSP channel. If the *slot*/*dspm*/*dsp*/*dsp-channel* argument is specified, the CSM call statistics for calls using the identified DSP channel are cleared. If no argument is specified, all CSM call statistics for all DSP channels are cleared.

Examples

The following example clears CSM call statistics for calls coming in on all DSP channels:

```
Router#
clear csm-statistics voice
```

Related Commands

Command	Description
clear csm -statisticsmodem	Clears the CSM statistics for a modem or group of modems.

clear h323 gatekeeper call

To force the disconnection of a specific call or of all calls active on a particular gatekeeper, use the **clearh323gatekeepercall** command in privileged EXEC mode.

clear h323 gatekeeper call {**all** | **local-callID** *local-callID*}

Syntax Description

all	Forces all active calls currently associated with this gatekeeper to be disconnected.
local -callID	Forces a single active call associated with this gatekeeper to be disconnected.
local-callID	Specifies the local call identification number (CallID) that identifies the call to be disconnected.

Command Default

No default behaviors or values

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.0(5)T	This command was introduced on the Cisco 2600 series, the Cisco 3600 series, and on the Cisco MC3810.
12.1(5)XM2	The command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T and implemented on the Cisco AS5300. Support for the Cisco AS5350 and Cisco AS5400 is not included in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.

Usage Guidelines

If you want to force a particular call to be disconnected (as opposed to all active calls on the gatekeeper), use the CallID number to identify that specific call. You can find the local CallID number for a specific call by using the **showgatekeepercalls** command; the ID number is displayed in the LocalCallID column.

Examples

The following example shows that an active call on the gatekeeper is being forced to disconnect. The local ID number of the active call is 12-3339.

```
Router# clear h323 gatekeeper call local-callID 12-3339
```

The following example shows that all active calls on the gatekeeper are being forced to disconnect:

```
Router# clear h323 gatekeeper call all
```

The following sample output from the **showgatekeepercalls** command displays information about a specific active call having a call ID of 12-3339:

```
Router# show gatekeeper calls
Total number of active calls =1
                    Gatekeeper Call Info
                    ====================
LocalCallID                    Age (secs)      BW
12-3339                        94              768 (Kbps)
 Endpt(s): Alias    E.164Addr     CallSignalAddr   Port   RASSignalAddr    Port
   src EP: epA                    10.0.0.11        1720   10.0.0.11        1700
   dst EP: epB2zoneB.com
   src PX: pxA                    10.0.0.1         1720   10.0.0.11        24999
   dst PX: pxB                    172.21.139.90    1720   172.21.139.90    24999
```

Related Commands

Command	Description
show gatekeeper calls	Displays the status of each ongoing call of which a gatekeeper is aware.

clear h323 gatekeeper endpoint

To unregister endpoints, use the **clearh323gatekeeperendpoint** command in privileged EXEC mode.

clear h323 gatekeeper endpoint {**alias e164** *digits* | **alias h323id** *name* | **all** | **id** *number* | **ipaddr** *address* [*port*]}

Syntax Description

alias e164 *digits*	E.164 alphanumeric address that is specified in the local alias table.
alias h323id *name*	H.323 ID name that is specified in the local alias table and is an alternate way to reach an endpoint.
all	All endpoints.
id *number*	ID of the endpoint.
ipaddr *address* [*port*]	Call signaling address and port (optional) of the endpoint. If a value for the *port* argument is not specified, the default is 1720.

Command Default

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(11)T	This command was introduced 12.2(11)T on the Cisco 3660 and Cisco MC3810.

Usage Guidelines

Using this command forces the gatekeeper to send an unregistration request (URQ) message to the specified endpoint or all endpoints and removes the endpoint from the gatekeeper registration database.

For gatekeeper cluster configurations, this command must be entered on the gatekeeper where the endpoint is registered. Use the **showgatekeeperendpoints** command to locate the endpoint in a gatekeeper cluster.

Note The endpoint that was unregistered using this command can come back if it sends the registration request (RRQ) back to the gatekeeper after the unregistration.

Examples

The following example shows how to unregister all endpoints:

```
Router# clear h323 gatekeeper endpoint all
Router# show gatekeeper endpoints
                    GATEKEEPER ENDPOINT REGISTRATION
                    ================================
CallSignalAddr  Port  RASSignalAddr   Port  Zone Name         Type    Flags
--------------- ----- --------------- ----- ---------         ----    -----
Total number of active registrations = 0
```

Related Commands

Command	Description
show gatekeeper endpoints	Locates the endpoint in a gatekeeper cluster.

clear h323 gatekeeper stats

To clear statistics about gatekeeper performance, use the **clearh323gatekeeperstats** command in privileged EXEC mode.

clear h323 gatekeeper stats

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.1(5)XM	This command was introduced.
12.2(2)T	This command was integrated into Cisco IOS Release 12.2(2)T.
12.2(2)XB1	This command was implemented on the Cisco AS5850.

Usage Guidelines

The **clearh323gatekeeperstats** command resets the gatekeeper performance counters to zero and records the time at which the last clear was performed.

Examples

The following is sample output from the**showgatekeeperperformancestats**command that shows the counters have been reset to zero after entering the **clearh323gatekeeperstats**command.

```
clear h323 gatekeeper stats
show gatekeeper performance stats
RAS inbound message counters:
Originating ARQ: 0 Terminating ARQ: 0 LRQ: 0
RAS outbound message counters:
ACF: 2  ARJ: 0 LCF: 2  LRJ: 0
ARJ due to overload: 0
LRJ due to overload: 0
Load balancing events: 0
Real endpoints: 2
```

Related Commands

Command	Description
show gatekeeper performance statistics	Displays information about the number of calls accepted and rejected by the gatekeeper.

clear h323 gateway

To clear the H.323 gateway counters, use the **clearh323gateway** command in privileged EXEC mode.

clear h323 gateway [{**cause-codes** | **h225** | **ras**}]

Syntax Description

cause -codes	(Optional) Clears only the disconnected cause code counters.
h225	(Optional) Clears only the H.225 counters.
ras	(Optional) Clears only the Registration, Admission, and Status (RAS) counters.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.2(4)T	This command was introduced on all Cisco H.323 platforms except for the Cisco AS5300, Cisco AS5350, and Cisco AS5400.

Usage Guidelines

To clear all H.323 counters, use the **clearh323gateway** command without any optional keywords. After you have used the **clearh323gateway** command, the respective counters are set to zero.

Examples

In the following example from a Cisco 3640 router, the **clearh323gateway** command is used without keywords to clear all H.323 counters:

```
Router# clear h323 gateway
All H.323 stats cleared at 01:54:38
```

In the following example from a Cisco 3640 router, the **clearh323gateway** command is used with the **cause-codes**keyword to clear the disconnect cause code counters:

```
Router# clear h323 gateway cause-codes
Cause code stats cleared at 01:54:08
```

In the following example from a Cisco 3640 router, the **clearh323gateway** command is used with the **h225** keyword to clear the H.225 counters:

```
Router# clear h323 gateway h225
H.225 stats cleared at 01:53:18
```

In the following example from a Cisco 3640 router, the **clearh323gateway** command is used with the **ras** keyword to clear the RAS counters:

```
Router# clear h323 gateway ras
RAS stats cleared at 01:53:25
```

Related Commands

Command	Description
debug cch323	Provides debug output for various components within the H.323 subsystem.
show h323 gateway	Displays the statistics for H.323 gateway messages that have been sent and received and displays the reasons for which H.323 calls have been disconnected.

clear http client statistics

To reset to zero all the counters that collect the information about the communication between the HTTP server and the client displayed in the output from the **showhttpclientstatistics**command, use the **clearhttpclientstatistics** command in user EXEC or privileged EXEC mode.

clear http client statistics

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

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

Usage Guidelines

Use the **showhttpclientstatistics** command to display the data collected by the counters the **clearhttpclientstatistics** command resets to zero.

Examples

The following example resets the counters to zero:

```
Router# clear http client statistics
```

Related Commands

Command	Description
show http client statistics	Displays information about the communication between the HTTP server and the client.

clear interface cable-modem

To reset the controller for a specified cable modem daughter card, use the **clearinterfacecable-modem**command in privileged EXEC mode. This command does not have a **no** version.

clear interface cable-modem

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.4(11)T	This command was introduced.

Examples

The following example shows how the **clearinterfacecable-modem** command clears the interface on the selected slot and port:

```
Router# clear interface cable-modem
*May 17 16:36:57.344: %CABLE_MODEM_HWIC-6-RESET: Interface Cable-Modem0/2/0 has been reset:
 clear command
*May 17 16:37:05.348: %LINK-3-UPDOWN: Interface Cable-Modem0/2/0, changed state to down
*May 17 16:37:06.348: %LINEPROTO-5-UPDOWN: Line protocol on Interface Cable-Modem0/2/0,
changed state to down
*May 17 16:37:19.740: %LINK-3-UPDOWN: Interface Cable-Modem0/2/0, changed state to up
*May 17 16:37:27.996: %LINEPROTO-5-UPDOWN: Line protocol on Interface Cable-Modem0/2/0,
changed state to up
```

Related Commands

Command	Description
show interfaces	Displays statistics for all interfaces configured.
show interfaces cable-modem	Displays statistics for all interfaces configured on the port.

clear media-proxy sessions summary history

To clear the history data for CUBE Media Proxy recording sessions, use the **clear media-proxy sessions summary history** command in privileged EXEC mode.

clear media-proxy sessions summary history

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
IOS XE 16.10.1	This command was introduced.

Usage Guidelines

Use the command **clear media-proxy sessions summary history** to clear the history data for CUBE Media Proxy recording sessions, that are displayed by the command **show media-proxy sessions summary history**.

Examples

```
Device# clear media-proxy sessions summary history
```

Related Commands

Command	Description
show media-proxy sessions summary history	Displays the summary of the completed CUBE MediaProxy SIP recording sessions.

clear mgcp src-stats

To clear the statistics gathered for Media Gateway Control Protocol (MGCP) System Resource Check (SRC) Call Admission Control (CAC) on an MGCP gateway, use the **clearmgcpsrc-stats**command in privileged EXEC mode.

clear mgcp src-stats

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.2(2)XB	This command was introduced.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(11)T	This command was implemented on the Cisco AS5350, Cisco AS5400, and Cisco AS5850.

Usage Guidelines

Use the **clearmgcpsrc-stats**commandto clear the MGCP gateway buffer that holds SRC CAC statistics gathered during the most recent inspection interval.

Examples

The following example clears MGCP VoIP SRC CAC statistics:

```
Router# clear mgcp src-stats
```

Related Commands

Command	Description
show mgcp statistics	Displays MGCP statistics regarding received and transmitted network messages.

clear mgcp statistics

To reset the Media Gateway Control Protocol (MGCP) statistical counters, use the **clearmgcpstatistics** command in privileged EXEC mode.

clear mgcp statistics

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.1(1)T	This command was introduced on the Cisco AS5300.
12.1(3)T	This command was implemented on the Cisco 3660, Cisco UBR924, and Cisco 2600 series.
12.2(11)T	This command was implemented on the Cisco AS5850.

Examples

The following is an example shows the MGCP statistical counters being reset:

```
Router# clear mgcp statistics
```

Related Commands

Command	Description
mgcp	Starts the MGCP daemon.
show mgcp statistics	Displays statistics for received and transmitted packets.

clear mrcp client statistics

To clear all Media Resource Control Protocol (MRCP) statistics, use the **clearmrcpclientstatistics** command in privileged EXEC mode.

clear mrcp client statistics {**all** | **hostname** {*hostnameip-address*}}

Syntax Description

all	Clears the accumulated MRCP session statistics for all hosts.
hostname	Clears the accumulated MRCP session statistics for the specified host.
hostname	Host name of the MRCP server. Format uses host name only or *hostname***:***port*.
ip -address	IP address of the MRCP server.

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

This command resets all MRCP session statistics to 0. Use the **showmrcpclientstatisticshostname**command to display the current statistics.

Examples

The following example resets the statistics for the host called "asr_server":

```
Router# clear mrcp client statistics hostname asr_server
```

Related Commands

Command	Description
show mrcp client statistics hostname	Displays cumulative information about MRCP sessions.

clear rlm group

To reset all Redundant Link Manager (RLM) time stamps to zero, use the **clearrlmgroup**command in privileged EXEC mode.

clear rlm group [*group-number*] [{**link** | **statistics**}]

Syntax Description

group-number	(Optional) RLM group number. Range is from 0 to 255. There is no default value.
link	(Optional) Specifies the RLM group link.
statistics	(Optional) Specifies the RLM group statistics.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
11.3(7)	This command was introduced.
15.0(1)M	This command was modified in a release earlier than Cisco IOS Release 15.0(1)M. The **statistics** keyword was added.

Examples

The following example resets the time stamps on RLM group 1:

```
Router# clear rlm group 1 link
!
02:48:17: rlm 1: [State_Up, rx ACTIVE_LINK_BROKEN] over link [10.1.1.1(Loopback1), 10.1.4.1]
02:48:17: rlm 1: link [10.1.1.2(Loopback2), 10.1.4.2] requests activation
02:48:17: rlm 1: link [10.1.1.1(Loopback1), 10.1.4.1] is deactivated
02:48:17: rlm 1: [State_Recover, rx LINK_BROKEN] over link [10.1.1.2(Loopback2), 10.1.4.2]
02:48:17: rlm 1: link [10.1.1.1(Loopback1), 10.1.4.1] = socket[10.1.1.1, 10.1.4.1]
02:48:17: rlm 1: [State_Recover, rx USER_SOCKET_OPENED] over link [10.1.1.1(Loopback1),
10.1.4.1] for user RLM_MGR
02:48:17: rlm 1: link [10.1.1.1(Loopback1), 10.1.4.1] is opened
02:48:17: rlm 1: link [10.1.1.2(Loopback2), 10.1.4.2] = socket[10.1.1.2, 10.1.4.2]
02:48:17: rlm 1: [State_Recover, rx USER_SOCKET_OPENED] over link [10.1.1.2(Loopback2),
10.1.4.2] for user RLM_MGR
02:48:17: rlm 1: link [10.1.1.2(Loopback2), 10.1.4.2] is opened
02:48:17: rlm 1: link [10.1.1.1(Loopback1), 10.1.5.1] = socket[10.1.1.1, 10.1.5.1]
02:48:17: rlm 1: [State_Recover, rx USER_SOCKET_OPENED] over link [10.1.1.1(Loopback1),
10.1.5.1] for user RLM_MGR
02:48:17: rlm 1: link [10.1.1.1(Loopback1), 10.1.5.1] is opened
02:48:17: rlm 1: link [10.1.1.2(Loopback2), 10.1.5.2] = socket[10.1.1.2, 10.1.5.2]
02:48:17: rlm 1: [State_Recover, rx USER_SOCKET_OPENED] over link [10.1.1.2(Loopback2),
10.1.5.2] for user RLM_MGR
02:48:17: rlm 1: link [10.1.1.2(Loopback2), 10.1.5.2] is opened
02:48:17: rlm 1: [State_Recover, rx LINK_OPENED] over link [10.1.1.1(Loopback1), 10.1.4.1]
02:48:17: rlm 1: link [10.1.1.1(Loopback1), 10.1.4.1] requests activation
02:48:17: rlm 1: [State_Recover, rx LINK_OPENED] over link [10.1.1.2(Loopback2), 10.1.4.2]
02:48:17: rlm 1: [State_Recover, rx START_ACK] over link [10.1.1.1(Loopback1), 10.1.4.1]
02:48:17: rlm 1: link [10.1.1.1(Loopback1), 10.1.4.1] is activated
```

Related Commands

Command	Description
clear interface	Resets the hardware logic on an interface.
interface	Defines the IP addresses of the server, configures an interface type, and enters interface configuration mode.
link (RLM)	Specifies the link preference.
protocol rlm port	Reconfigures the port number for the basic RLM connection for the whole RLM group.
retry keepalive	Allows consecutive keepalive failures a certain amount of time before the link is declared down.
server (RLM)	Defines the IP addresses of the server.
show rlm group statistics	Displays the network latency of the RLM group.
show rlm group status	Displays the status of the RLM group.
show rlm group timer	Displays the current RLM group timer values.
timer	Overwrites the default setting of timeout values.

clear rpms-proc counters

To clear Resource Policy Management System (RPMS) statistics counters for the number of leg 3 authentication, authorization, and accounting (AAA) preauthentication requests, successes, and rejects, use the **clearrpms-proccounters** command in privileged EXEC mode.

clear rpms-proc counters

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

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

Examples

The following example clears statistics counters for leg 3 AAA preauthentication requests, successes, and rejects:

```
Router# clear rpms-proc counters
```

Related Commands

Command	Description
show rpms-proc counters	Displays statistics for the number of leg 3 AAA preauthentication requests, successes, and rejects.

clear rudpv0 statistics

To clear the counters that track Reliable User Datagram Protocol (RUDP) statistics, enter the **clearrudpv0statistics**command in privileged EXEC mode.

clear rudpv0 statistics

Syntax Description

This command has no arguments or keywords.

Command Default

The statistical information accumulates.

Command Modes

Privileged EXEC (#)

Command History

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

Examples

The following example shows how to clear RUDP statistics on a Cisco 2611:

```
Router(config)#clear rudpv0 statistics
```

Related Commands

Command	Description
show rudpv0 failures	Displays RUDP information about failed connections and the reasons for them.
show rudpv0 statistics	Displays RUDP information about number of packets sent, received, and so forth.

clear rudpv1 statistics

To clear the counters that track Reliable User Datagram Protocol (RUDP) statistics, use the **clearrudpv1statistics**command in privileged EXEC mode.

clear rudpv1 statistics

Syntax Description

This command has no arguments or keywords.

Command Default

The statistical information accumulates.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.1(1)T	This command was introduced.
12.2(2)T	This command was implemented on Cisco 7200.
12.2(4)T	This command was implemented on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco IAD2420 series.
12.2(11)T	This command was implemented on the Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 in this release.

Examples

The following example clears all RUDP statistics for all available session groups:

```
Router# clear rudpv1 statistics
```

Related Commands

Command	Description
debug rudpv1	Displays debugging information for RUDP.
show rudpv1	Displays RUDP information.

clear sccp server statistics

To clear the counts displayed under the **showsccpserverstatistics** command, use the **clearsccpserverstatistics** command in privileged EXEC mode.

clear sccp server statistics

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.4(15)XY	This command was introduced.
15.0(1)M	This command was integrated into a release earlier than Cisco IOS Release 15.0(1)M.

Examples

The following example shows the Skinny Client Control Protocol (SCCP) server statistics counts being cleared, followed by verification that the counters are reset to zero with the**showsccpserverstatistics**command. The field descriptions are self-explanatory.

```
Router# show sccp server statistics
Failure type              Error count
------------------------  -----------
Send queue enqueue        0
Socket send               0
Msg discarded upon error  0
```

Related Commands

Command	Description
show sccp server statistics	Displays the number of SCCP messages sent and received by the SCCP server.

clear sdspfarm counters

To reset the server counts of the digital signal processor farms that are registered to the Skinny Client Control Protocol (sdspfarm) displayed under the **servershowsdspfarmmessagestatistics** command to zero, use the **clearsdspfarmcounters** command in privileged EXEC mode.

clear sdspfarm counters

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.4(15)XY	This command was introduced.

Examples

The following example shows the sdspfarm counters being cleared and verification that the counters are reset to zero with theshowsdspfarmsessionsstate command:

```
Router# clear sdspfarm counters
Router# show sdspfarm sessions state

Call state      Num of sessions
----------      ---------------
IDLE            1022
ALERTING        0
SEIZE           0
PROGRESS        0
CONNECTED       0
DIGITS          0
BUSY            0
RINGING         0
ERROR           0
HOLD            0
END             0
STOP            0
START           2
RESTART         0
UNKNOWN         0
DELAYED-SMT     0
```

Field descriptions should be self-explanatory.

Related Commands

Command	Description
show sdspfarm message statistics	Displays the number of SCCP messages sent and received by the SCCP server.
show sdspfarm sessions state	Displays the number of sessions in each SCCP call state.

clear sgcp statistics

To clear all Simple Gateway Control Protocol (SGCP) statistics, use the **clearsgcpstatistics** command in privileged EXEC mode.

clear sgcp statistics

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.0(5)T	This command was introduced in a private release on the Cisco AS5300 only and was not generally available.
12.0(7)XK	This command was implemented on the Cisco MC3810 and the Cisco 3600 series (except for the Cisco 3620) in a private release that was not generally available.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Examples

The following example shows all SGCP statistics being cleared:

```
Router# clear sgcp statistics
```

Related Commands

Command	Description
show sgcp statistics	Displays global statistics for SGCP packet counts.

clear sip-ua registration

To clear the registration records of SIP user agents, use the **clear sip-ua registration** command in privileged EXEC mode.

clear sip-ua registration passthrough {**all** | **call-id** *call-id* | **dial-peer** *dial-peer* | **dn** *dn*}

Syntax Description

passthrough	Clears SIP registration passthrough status
all	Clears SIP registration records of all user agents
call-id *call-id*	Clears SIP registration records of the user agent with the specified call ID
dial-peer *dial-peer*	Clears SIP registration records of user agents on the specified dial peer
dn *dn*	Clears SIP registration records of user agents on the specified directory number

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
IOS XE Fuji Release 16.8.1	This command was introduced.

Usage Guidelines

This command clears registration records of SIP User Agents based on the call-id, dial-peer, and directory number. If you use the keyword “all”, this command clears the registration records of all SIP user agents.

Example

The following example clears registration records of the SIP user agent with call-id 2147483647:

```
Router# clear sip-ua registration passthrough call-id 2147483647
```

Related Commands

clear sip ua-statistics	Clears all SIP statistics counters.
show sip-ua registration passthrough status	Displays the SIP user agent pass-through status information .

clear sip-ua statistics

To reset the Session Initiation Protocol (SIP) user-agent (UA) statistical counters, use the **clearsip-uastatistics** command in privileged EXEC mode.

clear sip-ua statistics

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

Use this command to clear all SIP statistics counters that are displayed by the **showsip-uastatistics** command.

Examples

The following example shows all SIP-UA statistics being cleared:

```
Router# clear sip-ua statistics
```

Related Commands

Command	Description
show sip-ua statistics	Displays response, traffic, and retry SIP statistics.

clear sip-ua tcp connection

To clear a session initiation protocol (SIP) TCP connection, use the **clearsip-uatcpconnection**command in privileged EXEC mode.

clear sip-ua tcp connection{**id***connection-id*[{**target ipv4***:address:port* | **id***connection-id*}]}**target ipv4***:address:port*

Syntax Description

id *connection-id*	Specifies the ID of the connection that needs to be closed in the SIP TCP process. The *connection-id*argument represents the connection ID. The range is from 1 to 2048.
target ipv4: *address* **:** *port*	Specifies the target address for the connection that needs to be closed in the SIP transport layer.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.4(6)T	This command was replaced by the **clearsip-ua**command.

Usage Guidelines

Inappropriate usage of the **clearsip-uatcpconnection** command can lead to erroneous call behavior, inappropriate usage of connections, and failure of calls.

Examples

To cear the connection entry only at the upper transport layer, assign the target IP address and port:

```
Router# clear sip-ua tcp connection target ipv4:172.18.194.183:5060
```

To clear the connection entry only at the lower TCP or User Datagram Protocol (UDP) layer, specify the connection:

```
Router# clear sip-ua tcp connection id 1
```

To completely clear a valid connection to target 172.18.194.183, port 5060, consider the following output example from the **showsip-uaconnections**command:

```
Router# show sip-ua connections tcp detail

Total active connections : 1
No. of send failures : 0
No. of remote closures : 0
No. of conn. failures : 0
No. of inactive conn. ageouts : 0
Max. tcp send msg queue size of 1, recorded for 172.18.194.183:5060
---------Printing Detailed Connection Report---------
Note:
** Tuples with no matching socket entry
- Do 'clear sip <tcp/udp> conn t ipv4:<addr>:<port>'
```

```
to overcome this error condition
++ Tuples with mismatched address/port entry
- Do 'clear sip <tcp/udp> conn t ipv4:<addr>:<port> id <connid>'
to overcome this error condition
Remote-Agent:172.18.194.183, Connections-Count:1
Remote-Port Conn-Id Conn-State WriteQ-Size
=========== ======= =========== ===========
5060 1 Established 0
```

Then execute the **clearsip-uatcpconnection** command:

```
Router# clear sip-ua tcp connection id 1 target ipv4:172.18.194.183:5060

Purging the entry from sip tcp process
Purging the entry from reusable global connection table
```

The result is that all connections are cleared after inputting the **clearsip-uatcpconnection**command:

```
Router# show sip-ua connections tcp detail
Total active connections : 0
No. of send failures : 0
No. of remote closures : 0
No. of conn. failures : 0
No. of inactive conn. ageouts : 0
Max. tcp send msg queue size of 1, recorded for 172.18.194.183:5060
---------Printing Detailed Connection Report---------
Note:
** Tuples with no matching socket entry
- Do 'clear sip <tcp/udp> conn t ipv4:<addr>:<port>'
to overcome this error condition
++ Tuples with mismatched address/port entry
- Do 'clear sip <tcp/udp> conn t ipv4:<addr>:<port> id <connid>'
to overcome this error condition
Remote-Agent:172.18.194.183, Connections-Count:0
```

Related Commands

Command	Description
clear sip-ua udp connection	Clears a SIP UDP connection.
show sip-ua connections	Displays SIP UA transport connection tabless.
timers connection aging	Sets the time before the SIP UA ages out a TCP and UDP connection.

clear sip-ua tcp tls connection

To clear a session initiation protocol (SIP) TCP connection, use the **clearsip-uatcptlsconnection**command in privileged EXEC mode.

clear sip-ua tcp tls connection{**id***connection-id*[{**target ipv4**:*address:port* | **id***connection-id*}]}**target ipv4**:*address:port*

Syntax Description

id *connection-id*	Specifies the ID of the connection that needs to be closed in the SIP TCP process. The *connection-id*argument represents the connection ID. The range is from 1 to 2048.
target ipv4: *address* **:** *port*	Specifies the target address for the connection that needs to be closed in the SIP transport layer.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.4(6)T	This command was replaced by the **clearsip-ua**command.

Usage Guidelines

Inappropriate usage of the **clearsip-uatcptlsconnection** command can lead to erroneous call behavior, inappropriate usage of connections, and failure of calls.

Examples

To cear the connection entry only at the upper transport layer, assign the target IP address and port:

```
Router# clear sip-ua tcp tls connection target ipv4:172.18.194.183:5060
```

To clear the connection entry only at the lower TCP or User Datagram Protocol (UDP) layer, specify the connection:

```
Router# clear sip-ua tcp tls connection id 1
```

To completely clear a valid connection to target 172.18.194.183, port 5060, consider the following output example from the **showsip-uaconnections**command:

```
Router# show sip-ua connections tcp tls detail

Total active connections : 1
No. of send failures : 0
No. of remote closures : 0
No. of conn. failures : 0
No. of inactive conn. ageouts : 0
Max. tcp send msg queue size of 1, recorded for 172.18.194.183:5060
---------Printing Detailed Connection Report---------
Note:
** Tuples with no matching socket entry
- Do 'clear sip <tcp/udp> conn t ipv4:<addr>:<port>'
to overcome this error condition
++ Tuples with mismatched address/port entry
```

```
- Do 'clear sip <tcp/udp> conn t ipv4:<addr>:<port> id <connid>'
to overcome this error condition
Remote-Agent:172.18.194.183, Connections-Count:1
Remote-Port Conn-Id Conn-State WriteQ-Size
=========== ======= =========== ===========
5060 1 Established 0
```

Then execute the **clearsip-uatcpconnection** command:

```
Router# clear sip-ua tcp tls connection id 1 target ipv4:172.18.194.183:5060

Purging the entry from sip tcp process
Purging the entry from reusable global connection table
```

The result is that all connections are cleared after inputting the **clearsip-uatcpconnection**command:

```
Router# show sip-ua connections tcp tls detail
Total active connections : 0
No. of send failures : 0
No. of remote closures : 0
No. of conn. failures : 0
No. of inactive conn. ageouts : 0
Max. tcp send msg queue size of 1, recorded for 172.18.194.183:5060
---------Printing Detailed Connection Report---------
Note:
** Tuples with no matching socket entry
- Do 'clear sip <tcp/udp> conn t ipv4:<addr>:<port>'
to overcome this error condition
++ Tuples with mismatched address/port entry
- Do 'clear sip <tcp/udp> conn t ipv4:<addr>:<port> id <connid>'
to overcome this error condition
Remote-Agent:172.18.194.183, Connections-Count:0
```

Related Commands

Command	Description
clear sip-ua udp connection	Clears a SIP UDP connection.
show sip-ua connections	Displays SIP UA transport connection tabless.
timers connection aging	Sets the time before the SIP UA ages out a TCP and UDP connection.

clear sip-ua udp connection

To clear a SIP UDP connection, use the **clearsip-uaudpconnection**command in privileged EXEC mode.

clear sip-ua udp connection {**id** *value* [**target** *ip-address*] | [**id** *value*] **target** *ip-address*}

Syntax Description

id *value*	Specifies the ID of the connection that needs to be closed in the SIP UDP process. The *value* argument represents the value of the connection ID. The range is from 1 to 2048.
target *ip -address*	Specifies the target address for the connection that needs to be closed in the SIP transport layer. The *ip-address* argument is the target address in the form of **ipv4:***address***:***port*.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.4(6)T	This command was replaced by the **clearsip-ua**command.

Usage Guidelines

Inappropriate usage of the **clearsip-uaudpconnection** command without understanding the issue or the implications can lead to erroneous call behavior, inappropriate usage of connections, and failure of calls.

Examples

To purge the connection entry only at the upper transport layer, assign the target IP address and port.

```
Router# clear sip-ua udp connection target ipv4:172.18.194.183:5060
```

To purge the connection entry only at the lower TCP/UDP layer, assign the connection ID.

```
Router# clear sip-ua udp connection id 1
```

Note Inappropriate usage of the **clear** command without understanding the issue or the implications would lead to erroneous call behavior, inappropriate usage of connections, and failure of calls.

To completely purge a valid connection to target 172.18.194.183, port 5060, consider the following example.

Before executing the **clearsip-uaudpconnection** command, running the **showsip-uaconnections** command gave the following output.

```
Router# show sip-ua connections udp detail

Total active connections : 1
No. of send failures : 0
No. of remote closures : 0
No. of conn. failures : 0
No. of inactive conn. ageouts : 0
```

```
Max. udp send msg queue size of 1, recorded for 172.18.194.183:5060
---------Printing Detailed Connection Report---------
Note:
** Tuples with no matching socket entry
- Do 'clear sip <tcp/udp> conn t ipv4:<addr>:<port>'
to overcome this error condition
++ Tuples with mismatched address/port entry
- Do 'clear sip <tcp/udp> conn t ipv4:<addr>:<port> id <connid>'
to overcome this error condition
Remote-Agent:172.18.194.183, Connections-Count:1
Remote-Port Conn-Id Conn-State WriteQ-Size
=========== ======= =========== ===========
5060 1 Established 0
```

Then execute the **clearsip-uaudpconnection** command:

```
Router# clear sip-ua udp connection id 1 target ipv4:172.18.194.183:5060

Purging the entry from sip udp process
Purging the entry from reusable global connection table
```

The final result is that all connections are cleared after executing the **clearsip-uaudpconnection**command:

```
Router# show sip-ua connections udp detail
Total active connections : 0
No. of send failures : 0
No. of remote closures : 0
No. of conn. failures : 0
No. of inactive conn. ageouts : 0
Max. udp send msg queue size of 1, recorded for 172.18.194.183:5060
---------Printing Detailed Connection Report---------
Note:
** Tuples with no matching socket entry
- Do 'clear sip <tcp/udp> conn t ipv4:<addr>:<port>'
to overcome this error condition
++ Tuples with mismatched address/port entry
- Do 'clear sip <tcp/udp> conn t ipv4:<addr>:<port> id <connid>'
to overcome this error condition
Remote-Agent:172.18.194.183, Connections-Count:0
```

Related Commands

Command	Description
clear sip-ua tcp connection	Clears a SIP TCP connection.
show sip-ua connections	Displays SIP UA transport connections.
timers connection aging	Sets the time before the SIP UA ages out a TCP and UDP connection.

clear ss7 sm-stats

To clear the counters that track session manager statistics, use the **clearss7sm-stats** command in privileged EXEC mode.

clear ss7 sm-stats [**session-set** *number*]

Syntax Description

session-set	(Optional) Specifies the session set.
number	(Optional) Specifies the session-set number. The range is from 0 to 3.

Command Default

The statistical information accumulates.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.0(7)XR	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.
15.0(1)M	This command was modified in a release earlier than Cisco IOS Release 15.0(1)M. The **session-set** keyword and *number* argument were added.

Examples

The following example shows how to clear session manager statistics:

```
Router# clear ss7 sm-stats session-set 2
```

Related Commands

Command	Description
show ss7 sm stats	Displays session manager information about number of packets queued, received, and so forth.

clear statistics dial-peer voice

To reset voice call counters and recent call details stored in a dial peer, use the **clearstatisticsdial-peervoice** command in privileged EXEC mode.

clear statistics dial-peer voice{*tag* | **busy-trigger-counter**}

Syntax Description

tag	(Optional) Identification tag number of a specific dial peer. A valid entry is any integer that identifies a specific dial peer. Range is from 1 to 2147483647.
busy-trigger-counter	(Optional) Specifies to clear the dial peer busy trigger call counter.

Command Default

If the *tag* argument is not used, counters in all the configured voice dial peers are cleared.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.2(8)T	This command was introduced on the Cisco AS5300.
15.0(1)M	This command was modified in a release earlier than Cisco IOS Release 15.0(1)M. The **busy-trigger-counter** keyword was added.
Cisco IOS XE Release 2.1	This command was integrated into Cisco IOS XE Release 2.1 and implemented on the Cisco ASR 1000 Series Aggregation Services Routers.

Usage Guidelines

The **clearstatisticsdial-peervoice** command resets the following statistical information about calls:

- Time elapsed since last clearing of statistics
- Connect time
- Charged units
- Accepted calls
- Refused calls
- Successful calls
- Failed calls
- Incomplete calls
- Last disconnect cause
- Last disconnect text
- Last setup time

Examples

The following example shows how to clear voice dial peer statistics using tag 1234:

```
Router# clear statistics dial-peer voice 1234
Clear voice call statistics stored in this voice dial-peer [confirm]y
```

The following example shows how to clear statistics in all the configured voice dial peers:

```
Router# clear statistics dial-peer voice
Clear voice call statistics stored in all voice dial-peers [confirm]y
```

Related Commands

Command	Description
dial-peer voice	Enters dial peer configuration mode and specifies the method of voice encapsulation.
show call history voice record	Displays CDR events in the call history table.
show dial-peer voice	Displays configuration information for dial peers.

clear stcapp statistics

To clear SCCP Telephony Control Application (STCAPP) statistics, use the **clearstcappstatistics**command in privileged EXEC mode.

clear stcapp statistics {**all** | **port** *slot-number*}

Syntax Description

all	Clears all STCAPP statistics.
port	Clears port-level STCAPP statistics.
slot-number	Voice interface slot number. The range is from 0 to 2147483647.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
15.0(1)M	This command was introduced in a release earlier than Cisco IOS Release 15.0(1)M.

Examples

The following example show how to clear all STCAPP statistics:

```
Router# clear stcapp statistics all
```

Related Commands

Command	Description
stcapp	Enables the STCAPP.

clear subscription

To clear all active subscriptions or a specific subscription, use the **clearsubscription** command in privileged EXEC mode.

clear **subscription** {**all** | **session-id** *session-id* | **statistics**}

Syntax Description

all	All active subscriptions.
session-id *session-id*	Subscription session to be cleared.
statistics	Global subscription statistics and all subscription history records.

Command Default

No default behavior or values

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

To cancel a specific subscription, use the *session-id* argument. The session ID can be found in the display frm from the**showsubscriptions** command. When this command is used, the applications associated with subscriptions receive the ev_subscribe_cleanup event. On receiving this event, the script closes the subscription.

Examples

The following example shows global statistics and history records being cleared:

```
Router# clear subscription statistics
```

Related Commands

Command	Description
retry subscribe	Configures the number of retries for SUBSCRIBE messages.
show subscription sip	Displays active SIP subscriptions.
subscription maximum	Specifies the maximum number of outstanding subscriptions to be accepted or originated by the gateway.

clear tgrep counters

To clear Telephony Gateway Registration Protocol (TGREP) counters, use the **cleartgrepcounters**command in privileged EXEC mode.

clear tgrep counters {***** | **carrier** *string* | **csr** | **dial-peer** *tag* | **trunk-group** *label*} [**csr**] [**ac**]

Syntax Description

*	Clears all TGREP counters.
carrier	Clears available circuit counters.
string	Carrier ID.
dial-peer	Clears dial-peer.
tag	Dial peer tag. The range is from 1 to 2147483647.
trunk-group *label*	Clears the trunk-group counters.
csr	(Optional) Clears the call success rate counters.
ac	(Optional) Clears all the available circuit counters.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
15.0(1)M	This command was introduced in a release earlier than Cisco IOS Release 15.0(1)M.

Examples

The following example show how to clear all tgrep counter information:

```
Router# clear tgrep counters *
```

Related Commands

Command	Description
clear tgrep neighbor	Clears all neighbor sessions.

clear tgrep neighbor

To clear Telephony Gateway Registration Protocol (TGREP) neighbor sessions, use the **cleartgrepneighbor** command in privileged EXEC mode.

clear tgrep neighbor {*****ip-address*}

Syntax Description

*****	Clears all neighbor sessions.
ip-address	IP addresses of neighbor sessions.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
15.0(1)M	This command was introduced in a release earlier than Cisco IOS Release 15.0(1)M.

Examples

The following example shows how to clear neighbor sessions:

```
Router# clear tgrep neighbor *
```

Related Commands

Command	Description
clear tgrep counters	Clears TGREP counters.

clear voice accounting method

To clear VoIP AAA accounting statistics for a specific accounting method on the gateway, use the**clearvoiceaccountingmethod** command in privileged EXEC mode.

clear voice accounting method *method-list-name*

Syntax Description

method-list-name	Name of the method list.

Command Modes

Privileged EXEC

Command History

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

Examples

The following example clears accounting statistics for method list "h323":

```
Router# clear voice accounting method h323
```

Related Commands

Command	Description
voice statistics type csr	Configures the collection of signaling and VoIP AAA accounting statistics.

clear voice dsp

To "cold-start" one or more digital signal processor (DSP) voice channels, use the **clearvoicedsp**command in privileged EXEC mode.

clear voice dsp{**channels** | **error**}[{[*slot*]}][{*/dsp*}]{*/channel*}

channels	Clears DSP calls on a specific channel or a range of channels.
error	Clears DSP error statistics.
slot	(Optional) Specifies either a single slot or the first slot in a range. To specify a range of slots, you can enter a **secondslotinthesyntaxofthisargument.Thesecondslotspecifiestheendoftherange.** All slots in the range are affected by the command.
/ *dsp*	(Optional) Specifies either a single DSP on the slot or the first DSP in a range. To specify a range of DSPs, you can enter a **secondDSPinthesyntaxofthisargument.ThesecondDSPspecifiestheendoftherange.** All DSPs in the range are affected by the command.
/ *channel*	(Optional) Specifies either a single channel on the DSP or the first channel in a range. To specify a range of channels, you can enter a **secondchannelinthesyntaxofthisargument.Thesecondchannelspecifiestheendoftherange.** All channels in the range are affected by the command.

Command Default

If this command is not used, active calls continue on the DSP voice channels.

Command Modes

Privileged EXEC (#)

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 **clearvoicedsp** command allows you to cold-start DSPs. Execution of this command causes the configured firmware to be downloaded to the specified DSP or a range of DSPs. This command can be executed irrespective of the state of the DSPs. All the active channels of the DSPs are prematurely terminated.

Examples

The following example clears all active calls on slot 2, DSP 1:

```
Router# clear voice dsp 2/1
```

The following example clears the active calls on slot 2, DSP 1, channel 1:

```
Router# clear voice dsp 2/1/1
```

Related Commands

Command	Description
show voice dsp	Displays the current status or selective statistics of DSP voice channels

clear voice phone-proxy all-sessions

To clear all phone-proxy sessions use the **clear voice phone-proxy all-sessions** command in privileged EXEC mode.

clear voice phone-proxy all-sessions

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

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

Example

The following example shows how to clear all phone-proxy sessions:

```
Device# clear voice phone-proxy all-sessions
```

clear voice statistics

To clear voice-statistic collection settings on the gateway to reset the statistics collection, use the **clearvoicestatistics** command in privileged EXEC mode.

{**clear voice statistics** [**csr** [{**accounting** | **signaling**}]] | [**iec**]}

Syntax Description

csr	(Optional) All accounting and signaling statistics are cleared, but Cisco VoIP internal error codes (IECs) are not cleared.
accounting	(Optional) Only accounting statistics are cleared.
signaling	(Optional) Only signaling statistics are cleared.
iec	(Optional) Only Cisco VoIP IECs are cleared.

Command Default

If no keywords are specified, all accounting and signaling statistics, and all IECs are cleared.

Command Modes

Privileged EXEC (#)

Command History

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

Examples

The following example clears all accounting and signaling statistics, and all Cisco VoIP IECs:

```
Router# clear voice statistics
```

The following example clears all accounting and signaling statistics:

```
Router# clear voice statistics csr
```

The following example clears only accounting statistics:

```
Router# clear voice statistics csr accounting
```

The following example clears only signaling statistics:

```
Router# clear voice statistics csr signaling
```

The following example clears only Cisco VoIP IECs:

```
Router# clear voice statistics iec
```

Related Commands

Command	Description
voice statistics type csr	Configures the collection of signaling and VoIP AAA accounting statistics.

clear voip fpi rtts

To clear the Voice over IP (VoIP) forwarding plane interface (FPI) round-trip time counter, use the **clear voip fpi rtts** command in privileged EXEC mode.

clear voip fpi rtts

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
Cisco IOS XE Release 3.9S	This command was introduced.

The following example shows how to clear the VoIP FPI round-trip time counter.

```
Router# clear voip fpi rtts
```

clear voip fpi stats

To clear the Voice over IP (VoIP) forwarding plane interface (FPI) statistics counter, use the **clear voip fpi stats** command in privileged EXEC mode.

clear voip fpi stat

Syntax Description

This command has no arguments or keywords.

Command History

Release	Modification
Cisco IOS XE Release 3.9S	This command was introduced.

The following example shows how to clear the VoIP FPI statistics counter.

```
Router# clear voip fpi stats
```

clear voip rtp port

In some cases, Voice over IP (VoIP) Real Time Protocol (RTP) ports can remain assigned after a call ends. Use this command to clear such hung ports.

clear voip rtp port *table-id ports*

Syntax Description

table-id	Use the 'show voip rtp stats' command to establish the table identifier for the hung port that needs to be cleared.
ports	List of up to 32 comma separated port numbers in the range 5500 to 65498.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
Cisco IOS XE Bengaluru 17.4.1a	This command is introduced.

When you try to clear a hung port, the following confirmation message is displayed:

```
Router#clear voip rtp port 1 8002
Any port(s) associated with an active call will not be cleared.[confirm]
Cleared port 8002
```

If you try to clear a port associated with an active call, the following error is displayed:

```
Router#clear voip rtp port 2 9020,9022
Any port(s) associated with an active call will not be cleared.[confirm]
The following port(s) are associated with active calls and have not been cleared:
9020 9022
An active call may be cleared using the "clear call voice" command.
```

If you try to clear multiple invalid ports (odd port, out of range, not allocated from table), the following error is displayed:

```
Router#clear voip rtp port 1 9008,9009,9010,9011
Any port(s) associated with an active call will not be cleared.[confirm]
Error: Port(s) 9008 9009 9010 9011  invalid for table 1.
Use 'show voip rtp stats' command to view the ports allocated to each table
```

If more than 32 ports are specified, the following error is displayed:

```
Router# clear voip rtp port 1 8000, 8002, 8004 ,8006 ,8008 , 8010, 8012, 8014, 8016, 8018,
 8020, 8022, 8024, 8026, 8028, 8030, 8032, 8034, 8036, 8038, 8040, 8042, 8044, 8046, 8048,
 8050, 8052, 8054, 8056, 8058, 8060, 8062, 8064
Any port(s) associated with an active call will not be cleared.[confirm]
Error: A maximum of 32 ports may be cleared.
```

Note If you want to clear an active call, use the 'clear call voice' command.

clear voip stream-service connection

To delete a WebSocket connection in CUBE, use the **clear voip stream-service connection** *id* **forced** command in User EXEC or privileged EXEC mode.

clear voip stream-service connection *id* **forced**

Syntax Description

id	The ID associated with a WebSocket connection.

Command Default

This command is not enabled, and no connections are cleared.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Release	Modification
Cisco IOS XE Bengaluru 17.6.1a	This command was introduced.

Usage Guidelines

Use this command to send close message on the WebSocket associated with the WebSocket ID. There are two versions of this command:

- **clear voip stream-service connection** *id* —The command configuration for a scenario in which there are no active calls on the WebSocket connection.
- **clear voip stream-service connection** *id* **forced**—The command configuration for a scenario in which there are active calls on the WebSocket connection.

Note If there are active calls on the WebSocket connection, the command **clear voip stream-service connection** *id* displays an error message that the connection cannot be cleared.

Examples

The following is a sample output for clearing the WebSocket connection with ID 17 which has an active call.

To verify if there are active calls:

```
router#show voip stream-service connection
ID   Local IP:Port        Remote IP:Port     Active Calls   Total Calls
17   10.65.125.206:22377  10.64.86.215:8066  1              1
```

To clear the WebSocket connection:

```
router#clear voip stream-service connection 17
WARNING: There are active fork sessions on this WebSocket connection.
Use clear voip stream-service connection <id> forced to delete this WebSocket connection.

router#clear voip stream-service connection 17 forced
1 active fork sessions will be deleted. Continue? [confirm]
```

To verify if the WebSocket connection with active calls is cleared:

```
router#show voip stream-service conn history
ID  Local IP:Port        Remote IP:Port     Total Calls  Disconnect Cause
6   10.65.125.206:21811  10.64.86.215:8062  0            WS_ACTIVE
8   10.65.125.206:29867  10.64.86.215:8063  1            WS_IDLE_TIMEOUT_CLOSURE
11  10.65.125.206:51108  10.64.86.215:8064  1            WS_IDLE_TIMEOUT_CLOSURE
14  10.65.125.206:29918  10.64.86.215:8065  1            WS_IDLE_TIMEOUT_CLOSURE
17  10.65.125.206:22377  10.64.86.215:8066  1            WS_CLEAR_COMMAND_CLOSURE
```

Related Commands

Command	Description
show voip stream-service connection	Displays information about the active WebSocket connections in Unified Border Element.
show voip stream-service connection history	Displays information about all the closed WebSocket connections in Unified Border Element.
show voip stream-service server <ip:port>	Displays information about the WebSocket connection based on WebSocket server IP and port.

clear voip stream-service statistics

To reset the global WebSocket statistics on your CUBE, use the **clear voip stream-service statistics** command in User EXEC or privileged EXEC mode.

clear voip stream-service statistics

Command Default

This command is not enabled by default, and no statistics are cleared.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Release	Modification
Cisco IOS XE Bengaluru 17.6.1a	This command was introduced.

Usage Guidelines

Use this command to clear the global WebSocket statistics on a CUBE router.

Examples

The following is a sample output for **show voip stream-service statistics** after the statistics are cleared by **clear voip stream-service statistics** .

```
router#clear voip stream-service statistics
router#show voip stream-service statistics
Active connections:             0
Active forked calls:            0
Total connections created:      0
Total forked calls:             0

Connection failures:
HTTP failures:                  0
TCP failures:                   0
Remote WebSocket closures:      0
Remote TCP closures:            0
Idle age-outs:                  0

Message statistics:
WS_CREATE_REQ:                  0
WS_CREATE_RSP_OK:               0
WS_CREATE_RSP_FAIL:             0
WS_CLOSE_REQ:                   0
WS_CLOSE_RSP:                   0
WS_DOWN:                        0
WS_STATS_REQ:                   0
WS_STATS_RSP:                   0
```

Related Commands

Command	Description
show voip stream-service statistics	Displays statistical information about WebSocket connections in Unified Border Element.
show voip stream-service connection history	Displays information about all the closed WebSocket connections in Unified Border Element.

Command	Description
show voip stream-service server <ip:port>	Displays information about the WebSocket connection based on WebSocket server IP and port.

clear vsp statistics

To clear all Voice Streaming Processing (VSP) statistics that are displayed when the **showvsp** command is used, use the **clearvspstatistics** command in privileged EXEC mode.

clear vsp statistics

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

This command resets all cumulative VSP statistics to 0. Use the **showvspstatistics** command to display the current statistics.

Examples

The following example resets the statistics for VSP sessions:

```
Router# clear vsp statistics
```

Related Commands

Command	Description
show vsp	Displays cumulative information about VSP sessions.

clid through credentials (sip-ua)

clid

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

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

Syntax Description

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

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**.

Command Modes

AAA preauthentication configuration

Command History

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

Usage Guidelines

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

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

Examples

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

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

Related Commands

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

clid (dial peer)

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

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

Syntax Description

network-number *number*	Network number. Establishes the calling-party network number in the CLID for this router.
network-provided	Allows you to set the screening indicator to reflect the number that was provided by the network.
override rdnis	Supported for POTS dial peers only Overrides the CLID with the redirected dialed number identification service (RDNIS) if available.
pi-restrict	Restricted progress indicator (PI). Causes removal of the calling-party number from the CLID when the PI is restricted.
restrict	Restricts presentation of the caller ID in the CLID.
second-number strip	(Optional) Removes a previously configured second network number from the CLID.
strip	Strips the calling-party number from the CLID. • **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	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 (config-dial-peer)

Command History

Release	Modification
12.2(11)T	This command was introduced.
12.2(13)T	The **overriderdnis** keywords were added.
12.4(4)T	The following keywords were added: **network-provided**, **pi-restrictall**, and **substitutename**.

Usage Guidelines

The **overriderdnis** 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 **clidnetwork-number** command sets the presentation indicator to "y" and the screening indicator to "network-provided." The **second-numberstrip** 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 **clidoverriderdnis** command overrides the CLID with the RDNIS if it is available.

The **clidrestrict** 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 **clidstrip** 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-numberstrip** keywords as part of the **clidnetwork-number** command. The following example sets the calling-party network number to 56789 for VoIP dial peer 1234 and also prevents the second network number from being sent:

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

Related Commands

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

clid (voice service voip)

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

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

Syntax Description

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

Command Default

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

Command Modes

Voice service VoIP configuration (config-voi-srv)

Command History

Release	Modification
12.4(4)T	This command was introduced.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

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

Use the **clidstrippi-restrictall** keyword to remove the Calling Party Name and Calling Party Number from the CLID.

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

Examples

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

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

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

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

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

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

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

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

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

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

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

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

Related Commands

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

clid strip

To remove the calling-party number from calling-line-ID (CLID) information and to prevent the calling-party number from being presented to the called party, use the **clidstrip** 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 (config-dial-peer)

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	This command was modified. 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 **clidstrip** command is issued, the calling-party number is null in the information element, and the presentation indicator is set to "n" to prevent the presentation of the number to the called party.

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

Examples

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

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

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

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

Related Commands

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

clid strip reason

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

clid strip reason
no clid strip reason

Syntax Description

This command has no arguments or keywords.

Command Default

The CLID reason code is not removed.

Command Modes

Dial peer voice configuration (config-dial-peer)

Command History

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

Usage Guidelines

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

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

Examples

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

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

Related Commands

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

client-vtp (voice class)

To configure a client verification trustpoint, and associate it to a TLS profile, use the command **client-vtp** in voice class configuration mode. To delete the client verification trustpoint, use **no** form of this command.

client-vtp *verification trustpoint*
no client-vtp

Syntax Description

verification trustpoint	Assigns a client verification trustpoint.

Command Default

No default behavior or values

Command Modes

Voice class configuration (config-class)

Command History

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

Usage Guidelines

The client verification truspoint is associated to a TLS profile through the command **voice class tls-profile** *tag*. The *tag* associates the client verification trustpoint configuration to the command **crypto signaling**.

Examples

The following example illustrates how to create a voice class tls-profile and associate a client verification trustpoint:

```
Router(config)#voice class tls-profile 2
Router(config-class)#client-vtp TPname
```

Related Commands

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

clock-rate (codec-profile)

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

clock-rate *clock-rate*
no clock-rate

Syntax Description

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

Command Default

The default clock rate is 0.

Command Modes

Codec-profile configuration (config-codec-profile)

Command History

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

Usage Guidelines

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

Examples

The following example shows:

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

Related Commands

Command	Description
codec profile	Defines video capabilities needed for video endpoints.

clock-select

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

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

Syntax Description

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

Command Default

No default behavior or values

Command Modes

CES configuration (config-ces)

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 **showcesclock-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.

cm-current-enhance

To improve immunity to extreme levels of longitudinal noise present in wiring that includes long cable lengths, use the **cm-current-enhance** command in Voice-port configuration mode. To return to the default configuration, use the **no** form of this command.

cm−current−enhance
no cm−current−enhance

Syntax Description

This command has no arguments or keywords.

Command Default

The **cm-current-enhance** command is not configured.

Command Modes

Voice-port configuration (config-voiceport)

Command History

Release	Modification
15.2(1)T	This command was introduced.

Usage Guidelines

This command should not be used under normal conditions. It should be used only to improve immunity to noise in cases of extreme levels of longitudinal noise on the wiring.

The command is available on the following platforms, in the modes indicated:

- VIC3-2FXS-E/DID (FXS and DID mode)
- VIC3-2FXS/DID, VIC3-4FXS/DID, and EM3-HDA-8FXS/DID (DID mode only)

Mode of action: When the cm-current-enhance mode is activated, REG 73 of the Silab chip (Si324x) is programmed to 1 to enhance the immunity to common-mode current noise.

Change of signaling type: The command is effective for the current signaling type value. The command state is not saved and applied after a change of signaling type.

Examples

The following example indicates the usage:

```
Device# config t
Enter configuration commands, one per line.  End with CNTL/Z.
Device(config)# voice-port 0/1/0
Device(config-voiceport)# cm-current-enhance
```

cn-san validate (voice class tls-profile)

To enable server, client, or bidirectional identity validation of a peer certificate during TLS handshake, use the command **cn-sanvalidate** in voice class tls-profile configuration mode. To disable certificate identity validation, use **no** form of this command.

cn-san validate {**server** | **client** | **bidirectional**}

no cn-san

Syntax Description

validate server	Enables server identity validation through Common Name (CN) and Subject Alternate Name (SAN) fields in the server certificate during client-side SIP/TLS connections.
validate client	Enables client identity validation through CN and SAN fields in the client certificate during server side SIP/TLS connections.
validate bidirectional	Enables both client and server identity validation through CN-SAN fields.

Command Default

Identity validation is disabled.

Command Modes

Voice class configuration (config-class)

Command History

Release	Modification
Cisco IOS XE Cupertino 17.8.1a	**client** and **bidirectional** options were introduced under voice class tls-profile configuration mode.
Cisco IOS XE Amsterdam 17.3.1a	**validate server** command was introduced under voice class tls-profile configuration mode. Introduced support for YANG Model.

Usage Guidelines

Server identity validation is associated with a secure signaling connection through the global **crypto signaling** and **voice class tls-profile** configurations.

From Cisco IOS XE Amsterdam 17.3.1a release, **cn-san validate server** allows a server certificate to be validated while establishing a SIP TLS connection. For this validation, CUBE checks that the domain name configured in the session target matches one of the names included in either the CN or SAN fields. The session is established only if these match.

From Cisco IOS XE Cupertino 17.8.1a release, the command is enhanced to include the **client** and **biderectional** keywords. The client option allows a server to validate the identity of a client by checking CN and SAN hostnames included in the provided certificate against a trusted list of cn-san FQDNs. The connection will only be established if a match is found. This list of cn-san FQDNs is also now used to validate a server certificate, in addition to the session target host name. The **biderectional** option validates peer identity for both client and server connections by combining both **server** and **client** modes. Once you configure **cn-san validate**, the identity of the peer certificate is validated for every new TLS connection.

From Cisco IOS XE Cupertino 17.8.1a onwards, the **voice class tls-profile** *tag* can be associated to a **voice-class tenant** also. For CN-SAN validation of the client certificate, define a list of allowed hostnames and patterns using the command **cn-san** *tag san-name*.

Examples

The following example illustrates how to configure a voice class tls-profile and associate server identity validation functionality:

```
Router(config)#voice class tls-profile 2
Router(config-class)#cn-san validate server
```

```
Router(config)#voice class tls-profile 3
Router(config-class)#cn-san validate client


Router(config)#voice class tls-profile 4
Router(config-class)#cn-san validate bidirectional
```

Related Commands

Command	Description
voice class tls-profile	Provides suboptions to configure the commands that are required for a TLS session.
cn-san *tag san-name*	List of CN-SAN names used to validate the peer certificate for inbound or outbound TLS connections.

cn-san (voice class tls-profile)

To configure a list of Fully Qualified Domain Names (FQDN) names to validate against the peer certificate for inbound or outbound TLS connections, use the **cn-san** command in voice class tls-profile configuration mode.

For inbound connections, the list is used to validate CN and SAN fields in the client certificate. For outbound connections, the list is used along with the session target hostname to validate CN and SAN fields in the server certificate.

To delete a certificate validation **cn-san** entry, use the **no** form of this command.

cn-san *{1-10} fqdn*
no cn-san *{1-10} fqdn*

Syntax Description

1-10	Specifies the tag of **cn-san** FQDN list entry.
fqdn	Specifies the FQDN or a domain wildcard in the form of *.domain-name.

Command Default

no cn-san names are configured.

Command Modes

Voice class tls-profile configuration mode

Command History

Release	Modification
Cisco IOS XE Cupertino 17.8.1a	This command is introduced.

Usage Guidelines

FQDN used for peer certificate validation are assigned to a TLS profile with up to ten **cn-san** entries. At least one of these entries must be matched to an FQDN in either of the certificate Common Name (CN) or Subject-Alternate-Name (SAN) fields before a TLS connection is established. To match any domain host used in an CN or SAN field, a **cn-san** entry may be configured with a domain wildcard, strictly in the form *.domain-name (e.g. *.cisco.com). No other use of wildcards is permitted.

Note Server certificates may also be verified by matching the SIP session target FQDN to a CN or SAN field.

Examples

The following example globally enables cn-san names:

```
Router(config)# voice class tls-profile 1
Router(config-class)# cn-san 2 *.webex.com
```

Related Commands

Command	Description
voice class tls-profile	Provides suboptions to configure the commands that are required for a TLS session.

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 command settings to the default value, use the **no** form of this command.

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

Syntax Description

codec	Specifies the voice coder rate for speech. Codec options available for various platforms are described in the following (first) table.
bytes	(Optional) Precedes the argument that specifies the number of bytes in the voice payload of each frame.
payload-size	(Optional) Number of bytes in the voice payload of each frame. See the second table below for valid entries and default values.
transparent	Enables codec capabilities to be passed transparently between endpoints in a Cisco Unified Border Element. **Note** The **transparent** keyword is available only on the Cisco 2600, 3600, 7200, and 7500 series router platforms.
fixed-bytes	(Optional) Indicates that the codec byte size is fixed and nonnegotiable.
mode	(Optional) For Cisco internet Speech Audio Codec (iSAC) codec only. Specifies the iSAC operating frame mode that is encapsulated in each packet.
independent	(Optional) For iSAC codec only. Specifies that the configuration mode variable bit rate is independent (value 1).
adaptive	(Optional) For iSAC codec only. Specifies that the configuration mode variable bit rate is adaptive (value 0).
bit rate *value*	(Optional) For iSAC codec only. Configures the target bit rate in kilobits per second. Range is 10–32.
frame-size	(Optional) For iSAC codec only. Specifies the operating frame in milliseconds (ms). Valid entries are: • **30** --30-ms frames • **60** --60-ms frames • **fixed** --This keyword is applicable only for adaptive mode.
profile	(Optional) Defines the profile that is associated with the codec.
tag	(Optional) Specifies the codec profile tag that is associated with the codec. Range: 1–1000000.

Command Default

g729r8, 30-byte payload for Voice over Frame Relay (VoFR) and Voice over ATM (VoATM). g729r8, 20-byte payload for Voice over IP (VoIP). See the second table for valid entries and default values for codecs.

Command Modes

Dial peer configuration (config-dial-peer)

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
11.3(3)T	This command was implemented on the Cisco 2600 series.
12.0(3)T	This command was implemented on the Cisco AS5300. This release does not support the **clear-channel** keyword.
12.0(4)T	This command was implemented on the Cisco 3600 series, Cisco 7200 series, and Cisco MC3810, and the command was modified for VoFR dial peers.
12.0(5)XE	More *codec* choices and other options were implemented.
12.0(5)XK	The **g729br8** and **pre-ietf** codec keywords were added for the Cisco 2600 and Cisco 3600 series.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0.(7)T and implemented on the Cisco AS5800. Voice coder rates of speech were added. This release does not support the **clear-channel** keyword, so it is no longer available in the command syntax.
12.0(7)XK	The**g729abr8** and **g729ar8** codec keywords were added for the Cisco MC3810, and the **pre-ietf** keyword was deleted.
12.1(1)T	This command was integrated in Cisco IOS Release 12.1(1)T.
12.1(5)T	The**gsmefr** and **gsmfr** codec keywords were added.
12.2(8)T	The command was implemented on the Cisco 1750 and Cisco 1751.
12.2(13)T3	The **transparent** keyword was added for use with H.323 to H.323 connections. This keyword is available only in js2 images.
12.4(11)XJ2	The**gsmefr**and **gsmfr**keywords were removed as configurable codec options for all platforms except the **gsmfr** codec on the Cisco AS5400 and AS5350 with MSAv6 DSPs. The **transparent** keyword now supports H.323 to SIP connections.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.
12.4(15)XY	The **g722-64** keyword was added.
12.4(20)T	This command was integrated into Cisco IOS Release 12.4(20)T.
15.0(1)M	The **fixed-bytes** keyword was added.

Release	Modification
15.1(1)T	This command was modified. The**isac** keyword was added as a codec type, and the **mode**, **independent**, **adaptive**, **bitrate**, and **fixed** keywords were added as configurable parameters.
Cisco IOS XE Amsterdam 17.3.1a	The command was modified to support Opus codec in Cisco Unified Border Element. The keyword **profile** and the variable **tag** were added as configurable parameters for Opus codec.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Note In YANG, only **codec transparent** can be configured under dial-peer. For all other codec configurations, use 'voice class codec' configuration.

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 that is used with the **codeccomplexity** voice-card configuration command. The **codeccomplexity** command is voice-card specific and platform specific. The **codeccomplexity** voice-card configuration command is set to either high or medium.

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

If the **codeccomplexity** 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 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.

The table below describes the voice payload options and default values for the codecs and packet voice protocols.

Table 11: Voice Payload-per-Frame Options and Defaults

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

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

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

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

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

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

Examples

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

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

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

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

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

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

Related Commands

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

codec (dsp)

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

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

Syntax Description

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

Command Default

Medium complexity

Command Modes

DSP interface DSP farm

Command History

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

Usage Guidelines

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

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

The DSPinterface dspfarm codec complexity setting affects the options available for the **codecdialpeerconfiguration** 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 **codeccomplexity** command in voice-card configuration mode.

Examples

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

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

Related Commands

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

codec (DSP farm profile)

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

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

Syntax Description

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

Command Default

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

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

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

- **g711ulaw**

Command Modes

DSP farm profile configuration (config-dspfarm-profile)

Command History

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

Usage Guidelines

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

For homogeneous video profiles, only one video format is supported

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

To change the configured codec in the profile, you must first enter a **nomaximumsession**command.

The table below shows the relationship between DSP farm functions and codecs.

Table 12: DSP Farm Functions and Codec Relationships

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

Hardware MTPs support only G.711 a-law and G.711 mu-law. If you configure a profile as a hardware MTP and you want to change the codec to other than G.711, you must first remove the hardware MTP by using the**nomaximumsessionshardware** command.

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

Examples

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

```
Router(config)# dspfarm profile 123 transcode
Router(config-dspfarm-profile)# codec g729abr8
The following example shows how to set up a video conference with guaranteed-audio.
Router(config)# dspfarm profile 99 conference video guaranteed-audio
Router(config-dspfarm-profile)# codec h264 4cif
Router(config-dspfarm-profile)# codec h264 cif
Router(config-dspfarm-profile)# maximum conference-participants 8
```

Related Commands

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

codec (voice-card)

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

codec {**complexity** {**flex** [**reservation-fixed** {**high** | **medium**}] | **high** | **medium** | **secure**} | **sub-sample**}
no codec complexity

Syntax Description

complexity	Manages the complexity and density of codecs used in voice processing.
flex	When the **flex** keyword is used, up to 16 calls can be completed per digital signal processor (DSP). The number of supported calls varies from 6 to 16, depending on the codec used for a call. In this mode, reservation for analog voice interface cards (VICs) may be needed for certain applications such as Central Automatic Message Accounting (CAMA) E-911 calls because oversubscription of DSPs is possible. If this is true, enable the **reservation-fixed** keyword. There is no reservation by default.
reservation-fixed	(Optional) If you have specified the **flex** keyword, the **reservation-fixed** keyword ensures that sufficient DSP resources are available to handle a call. If you enter the **reservation-fixed** keyword, set the complexity for **high** or **medium**. (See the guidelines following to understand the effects of the keywords.) This option appears only when there is an analog VIC present.
high	If you specify the **high** keyword to define the complexity, each DSP supports two voice channels encoded in any of the following formats: • 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--G726 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,4800bps,7200bps,9600bps,12kbps,and14.4kbps.** **Note** Codecs G.723.1 and G.728 are not supported on Cisco 1750 and Cisco 1751 modular access routers for Cisco Hoot and Holler over IP applications.

medium	If you specify the **medium**keyword to define the complexity, each DSP supports four voice channels encoded in any of the following formats: • g711alaw--G.711 a-law 64,000 bps. • g711ulaw--G.711 mu-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 b**ps,4800bps,7200bps,9600bps,12kbps,and14.4kbps.Faxrelayisthedefault.**
secure	If you specify the **secure**keyword to define complexity, each DSP on an NM-HDV network module supports two voice channels encoded in any of the following formats: • g711alaw--G.711 a-law 64,000 bps. • g711ulaw--G.711 mu-law 64,000 bps. • g729--G.729 8000 bps. • g729A--G.729 8000 bps.
sub-sample	Increases the processing frequency for the G.711 codec with reduced 5510 DSP density.

Command Default

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

Command Modes

Voice-card configuration (config-voice-card)

Command History

Release	Modification
12.0(5)XK	This command was introduced as the codec complexity on the Cisco 2600 and Cisco 3600 series.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0(7)T.
12.0(7)XK	This command was implemented on the Cisco MC3810 for use with the high-performance compression module (HCM).
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.2(8)T	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(13)T	The **ecan-extended**keyword was added.

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

Usage Guidelines

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

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

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

The **high** keyword selects a higher codec complexity if that is required to support a particular codec or combination of codecs. When you use the **codeccomplexityhigh** 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 **mgcppackage-capability***srtp-package*command to enable MGCP gateway capability to process SRTP packages. Use the **showvoicedsp** command to display codec complexity status.

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

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

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

Examples

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

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

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

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

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

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

Related Commands

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

codec aal2-profile

To set the codec profile for a digital signal processor (DSP) on a per-call basis, use the **codecaal2-profile** command in dial peer configuration mode. To restore the default codec profile, use the **no** form of this command.

codec aal2-profile {**itut** | **custom** | **atmf**} *profile-number codec*
no codec aal2-profile

Syntax Description

itut	The*profile-number* as an ITU-T type.
custom	The *profile-number* as a custom type.
atmf	The *profile-number* as an Asynchronous Transfer Mode Forum (ATMF) type.
profile -number	The available*profile-number* selections depend on the profile type. For ITU-T: • **1** = G.711 mu-law • **2** = G.711 mu-law with silence insertion descriptor (SID) • **7** = G.711 mu-law and G.729ar8 For ATMF: **9** = Broadband Loop Emulation Services (BLES) support for VoAAL2 For custom: • **100** = G.711 mu-law and G.726r32 • **110** = G.711 mu-law, G.726r32, and G.729ar8
codec	Enter one codec for the DSP. The possible *codec*entries depend on the *profile-number*value.The valid entries are as follows: • For ITU 1--**g711mu-law** • For ITU 2--**g711mu-law** • For ITU 7--**g711mu-law** or **g729ar8** • For ATMF--**g711mu-law** • For custom 100--**g711mu-law** or **g726r32** • For custom 110--**g711mu-law** or **g726r32** or **g729ar8** • For lossless compression--**llcc**

Command Default

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

Command Modes

Dial peer configuration

Command History

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

Usage Guidelines

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

You must enter the **sessionprotocolaal2-trunk** command before configuring the codec ATM adaptation Layer 2 (AAL2) profile.

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

Examples

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

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

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

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

Related Commands

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

codec gsmamr-nb

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

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

Syntax Description

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

Command Default

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

Command Modes

Dial peer configuration (config-dial-peer)

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

Examples

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

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

Related Commands

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

codec ilbc

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

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

Syntax Description

mode	(Optional) Specifies the iLBC operating frame mode that is encapsulated in each packet.
frame_size	(Optional) iLBC operating frame in milliseconds (ms). Valid entries are: • 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.
payload_size	(Optional) Number of bytes in the voice payload of each frame. Valid entries are: • For **mode20--38**, **76**, **114**, **152**, **190**, **228**. Default is **38**. • For **mode30--50**, **100**, **150**, **200**. Default is **50**.

Command Default

20ms frames with a 15.2kbps bit rate.

Command Modes

Dial-peer configuration

Command History

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

Usage Guidelines

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

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

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

Examples

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

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

Related Commands

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

codec preference

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

codec preference *value codec-type* [**mode** {**independent** | **adaptive**}] [**frame-size** {**20** | **30** | **60** | **fixed**}] [**bit rate** *value*] [**bytes** *payload-size*] [**packetization-period 20**] [**encap rfc3267**] [**profile** *profile-tag*][**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. • **g722r-64** --G.722-64 at 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-NB codec capability. • **gsmfr** --Global System for Mobile Communications Full Rate (GSMFR) 13,200 bps. • **opus** --Opus upto 510 kbps. **Note** The **gsmfr** keyword is configurable only on the Cisco AS5350 and AS5400 with MSAv6 digital signal processors (DSPs). • **ilbc** --internet Low Bitrate Codec (iLBC) at 13,330 bps or 15,200 bps. • **isac** --Cisco internet Speech Audio Codec (iSAC) codec. • **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 and iSAC codecs only. Specifies the iLBC or iSAC operating frame mode that is encapsulated in each packet.
independent	(Optional) For iSAC codec only. Specifies that the configuration mode variable bit rate (VBR) is independent (value 1).

adaptive	(Optional) For iSAC codec only. Specifies that the configuration mode VBR is adaptive (value 0).
frame-size	(Optional) For iLBC and iSAC codecs only. Specifies the operating frame in milliseconds (ms). Valid entries are: • **20** --20-ms frames (iLBC only) • **30** --30-ms frames (iLBC or iSAC) • **60** --60-ms frames (iLBC or iSAC) • **fixed** --This keyword is applicable only for adaptive mode.
bit rate *value*	(Optional) Configures the target bit rate in kilobits per second. The range is 10 to 32.
bytes	(Optional) Specifies that the size of the voice frame is in bytes.
payload-size	(Optional) Number of bytes that you specify as the voice payload of each frame. Values depend on the codec type and the packet voice protocol.
packetization-period 20	(Optional) Sets the packetization period at 20 ms. This keyword is applicable only to GSMAMR-NB codec support.
encap rfc3267	(Optional) Sets the encapsulation value to comply with RFC 3267. This keyword is applicable only to GSMAMR-NB codec support.
frame-format	(Optional) Specifies a frame format. Supported values are **octet-aligned** and **bandwidth-efficient**. The default is **octet-aligned**. This keyword is applicable only to GSMAMR-NB codec support.
crc | **no-crc**	(Optional) Cyclic Redundancy Check (CRC) is applicable only for octet-aligned frame format. If you enter bandwidth-efficient frame format, the **crc | no-crc**options are not available because they are inapplicable. This keyword is applicable only to GSMAMR-NB codec support.
modes *modes-values*	(Optional) Valid values are from 0 to 7. You can specify modes as a range (for example, 0-2), or individual modes separated by commas (for example, 2,4,6), or a combination of the two (for example, 0-2,4,6-7). This argument is applicable only to GSMAMR-NB codec support.
profile *profile-tag*	(Optional) Specifies the codec profile for which preference is set within the voice class codec configuration mode. The range for *profile-tag* is 1 to 1000000.

Command Default

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

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

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

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

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

Command Modes

voice class configuration (config-class)

Command History

Release	Modification
12.0(2)XH	This command was introduced on the Cisco AS5300.
12.0(7)T	This command was implemented on the Cisco 2600 series and Cisco 3600 series.
12.0(7)XK	This command was implemented on the Cisco MC3810.
12.1(2)T	This command was integrated into Cisco Release IOS Release 12.1(2)T.
12.1(5)T	This command was modified. The **gsmefr** and **gsmfr** keywords were added.
12.2(13)T3	This command was modified.The **transparent** keyword was added.
12.4(4)XC	This command was extended to include GSMAMR-NB codec parameters on the Cisco AS5350XM and Cisco AS5400XM platforms.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(11)T	This command was modified. The **ilbc** and **mode** keywords were added.
12.4(11)XJ2	This command was modified. The **gsmefr**and **gsmfr**keywords were removed as configurable codec options for all platforms with the exception of the **gsmfr** codec on the Cisco AS5400 and AS5350 with MSAv6 dsps.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.
12.4(15)XY	This command was modified. The **g722r-64** keyword was added.
12.4(20)T	This command was integrated into Cisco IOS Release 12.4(20)T.
IOS Release XE 2.5	This command was integrated into Cisco IOS XE Release 2.5.
15.1(1)T	This command was modified. The**isac** keyword was added as a codec type, and the **independent**, **adaptive**, **bitrate**, and **fixed** keywords were added as configurable parameters.
Cisco IOS XE Amsterdam 17.2.1r	Introduced support for YANG models.
Cisco IOS XE Amsterdam 17.3.1a	This command was modified. Opus was added as a supported codec type.
Cisco IOS XE Dublin 17.10.1a	Introduced support for the following YANG model: • **video codec [h261 | mpeg4]**

Usage Guidelines

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

Note The **transparent** keyword is not supported when the **callstart** 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 VoFR VoATM	80, 160 40 to 240 in multiples of 40 40 to 240 in multiples of 40	160 240 240
g722r-64	VoIP	80, 160, 240	160
g723ar53 g723r53	VoIP VoFR VoATM	20 to 220 in multiples of 20 20 to 240 in multiples of 20 20 to 240 in multiples of 20	20 20 20
g723ar63 g723r63	VoIP VoFR VoATM	24 to 216 in multiples of 24 24 to 240 in multiples of 24 24 to 240 in multiples of 24	24 24 24
g726r16	VoIP VoFR VoATM	20 to 220 in multiples of 20 10 to 240 in multiples of 10 10 to 240 in multiples of 10	40 60 60
g726r24	VoIP VoFR VoATM	30 to 210 in multiples of 30 15 to 240 in multiples of 15 30 to 240 in multiples of 15	60 90 90
g726r32	VoIP VoFR VoATM	40 to 200 in multiples of 40 20 to 240 in multiples of 20 40 to 240 in multiples of 20	80 120 120
g728	VoIP VoFR VoATM	10 to 230 in multiples of 10 10 to 240 in multiples of 10 10 to 240 in multiples of 10	40 60 60
g729abr8 g729ar8 g729br8 g729r8	VoIP VoFR VoATM	10 to 230 in multiples of 10 10 to 240 in multiples of 10 10 to 240 in multiples of 10	20 30 30
ilbc	VoIP	For the **mode20** keyword, **38**,**76**, **114**, **152**, 190, 228 For the **mode30** keyword, **50**, **100**, **150**, **200**	38 50
iSAC	VoIP	--	--
opus	VoIP	Variable	--

Examples

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

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

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

```
voice class codec 99
codec preference 1 g711alaw
codec preference 2 g711ulaw bytes 80
codec preference 3 g723ar53
codec preference 4 g723ar63 bytes 144
codec preference 5 g723r53
codec preference 6 g723r63 bytes 120
codec preference 7 g726r16
codec preference 8 g726r24
codec preference 9 g726r32 bytes 80
codec preference 10 g729br8
codec preference 11 g729r8 bytes 50
end
dial-peer voice 1919 voip
 voice-class codec 99
```

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

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

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

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

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

The following example shows how to configure the codec profile, codec preference and apply it to a dial peer:

```
Device(config)#codec profile 79 opus
Device(conf-codec-profile)#fmtp "fmtp:114 maxplaybackrate=16000; sprop-maxcapturerate=16000;
 maxaveragebitrate=20000; stereo=1; sprop-stereo=0; useinbandfec=0; usedtx=0"
Device(conf-codec-profile)#exit

Device(config)#voice class codec 80
Device(config-class)#codec preference 1 opus profile 79
Device(config-class)#exit

Device(config)#dial-peer voice 604 voip
Device(config-dial-peer)#rtp payload-type opus 126
Device(config-dial-peer)#voice-class codec 80 offer-all
Device(config-dial-peer)#exit
```

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.

Command	Description
voice-class codec (dial peer)	Assigns a previously configured codec selection preference list to a dial peer.

codec profile

To define audio and video capabilities needed for video endpoints, use the **codec profile** command in global configuration mode. To disable the codec profile, use the **no** form of this command.

codec profile *tag profile*
no codec profile

Syntax Description

tag	A number in the range of 1 to 1000000.
profile	The name of the audio or video codec profile: • aacld • h263 • h263+ • h264 • opus

Command Default

No codec profile is configured.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.4(22)T	This command was introduced.
Cisco IOS XE Amsterdam 17.3.1a	Introduced support for the codec **opus**.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

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

Examples

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

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

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

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

The following example shows the codec tagged 2 assigned to the **opus** profile.

```
codec profile 2 opus
      fmtp "fmtp:114 maxplaybackrate=16000;
sprop-maxcapturerate=16000;maxaveragebitrate=20000; stereo=1; useinbandfec=1; usedtx=0"
```

Related Commands

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

codec transparent

Syntax Description

Command Default

Command Modes

Command History

Release	Modification

Usage Guidelines

Examples

The following example globally enables ANAT on a SIP trunk:

```
Router(config-serv-sip)# voice-class sip anat system
```

The following example enables ANAT on a specified dial peer:

```
Router(config-dial-peer)# voice-class sip anat
```

Related Commands

Command	Description

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 (config-voiceport)

Command History

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

Usage Guidelines

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

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

Examples

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

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

Related Commands

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

compand-type

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

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

Syntax Description

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

Command Default

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

Command Modes

Voice-port configuration (config-voiceport)

Command History

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

Usage Guidelines

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

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

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

Examples

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

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

Related Commands

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

complete (ctl file)

To complete the configuration of the Certificate Trust List (CTL) file use the **complete** command in CTL file configuration mode. To deactivate the CTL file use the **no** form of the command.

complete
no complete

This command has no arguments or keywords.

Command Default

The CTl file instance is not activated.

Command Modes

CTL file configuration mode (config-ctl-file)

Command History

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

Usage Guidelines

Example

The following example shows how to activate the CTL file called “myctl”. The specific configurations of myctl are entered before using the **complete** command:

```
Device(config)# voice-ctl-file myctl
Device(config-ctl-file)# record-entry capf trustpoint trustpoint_1
Device(config-ctl-file)# complete
```

complete (phone proxy)

To activate the phone proxy instance, use the **complete** command in phone proxy configuration mode. To deactivate the phone proxy instance, use the **no** form of the command.

complete
no complete

This command has no arguments or keywords.

Command Default

The phone proxy instance is not activated.

Command Modes

Phone proxy configuration mode (config-phone-proxy)

Command History

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

Usage Guidelines

If the phone proxy has been configured in any adjacency, and the adjacency's admin-status is attach, then you cannot deactivate it with the **no complete** command.

Example

The following example shows how to activate the specific phone proxy called “first-pp”. The specific configurations of first-pp are entered before using the **complete** command:

```
Device(config)# voice-phone-proxy first-pp
Device(config-phone-proxy)# description cluster-test
Device(config-phone-proxy)# tftp-server address ipv4 198.51.100.10 local-addr ipv4
192.168.0.109 acc-addr ipv4 10.0.0.8
Device(config-phone-proxy)# ctl-file myctl (config-phone-proxy)# access-secure
Device(config-phone-proxy)# disable-service-settings
Device(config-phone-proxy)# capf-addr ipv4 198.51.100.12 acc-addr ipv4 10.0.0.8
Device(config-phone-proxy)# service-map server-addr ipv4 198.51.100.12 port 8080 acc-addr
ipv4 10.0.0.8 port 1234
Device(config-phone-proxy)# session-timer 200
Device(config-phone-proxy)# complete
```

conference

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

conference *keypad-character*
no conference

Syntax Description

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

Command Default

The default value is #3.

Command Modes

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

Command History

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

Usage Guidelines

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

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

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

Examples

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

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

Related Commands

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

conference-join custom-cptone

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

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

Syntax Description

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

Command Default

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

Command Modes

DSP farm profile configuration (config-dspfarm-profile)

Command History

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

Usage Guidelines

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

- Use the **voiceclasscustom-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-joincustom-cptone** command to associate the join tone to the DSP farm profile for that conference. Use the **showdspfarmprofilecommand** to display the DSP farm profile.

Examples

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

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

Related Commands

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

conference-leave custom-cptone

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

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

Syntax Description

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

Command Default

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

Command Modes

DSP farm profile configuration (config-dspfarm-profile)

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 **voiceclasscustom-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-joincustom-cptone** command to associate the leave tone to the DSP farm profile for that conference. Use the **showdspfarmprofilecommand** 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 **voiceclasscustom-cptone** and **conference-joincustom-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.

Command	Description
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. (except for **tx -b-bit**)
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 (config-voiceport)

Command History

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

Usage Guidelines

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

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

Examples

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

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

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

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

Related Commands

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

connect (channel bank)

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

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

Syntax Description

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

Command Default

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

Command Modes

Global configuration (config)

Command History

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

Usage Guidelines

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

Examples

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

```
Router(config)# controller t1 1/0
Router(config-controller)# ds0-group 1 timeslot 0 type fxo-loop-start
Router(config-controller)# exit
Router(config)# voice-port 1/1/0
Router(config-voiceport)# signal-type fxs-loop-start
Router(config-voiceport)# exit
Router(config)# connect connection1 voice-port 1/1/0 t1 1/0 0
```

Related Commands

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

connect (drop-and-insert)

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

connect *connection-id* {**t1** | **e1**} *slotport-1* *tdm-group-no-1* {**t1** | **e1**} *slotport-2* *tdm-group-no-2*
no **connect** *connection-id* {**t1** | **e1**} *slotport-1* *tdm-group-no-1* {**t1** | **e1**} *slotport-2* *tdm-group-no-2*

Syntax Description

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

Command Default

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

Command Modes

Global configuration (config)

Command History

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

Usage Guidelines

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

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

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

Examples

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

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

Related Commands

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

connect atm

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

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

Syntax Description

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

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

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

Usage Guidelines

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

After TDM groups are created on two different physical ports, you can use the **connectatm**command to start the passage of data between the ports. If a crosspoint switch is provided in the advanced integration module

(AIM) slot, the connections can extend between ports on different cards. Otherwise, the connection is restricted to ports on the same VWIC card.

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

Examples

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

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

Related Commands

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

connect interval

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

connect interval *seconds*
no connect interval

Syntax Description

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

Command Default

60 seconds

Command Modes

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

Command History

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

Usage Guidelines

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

Examples

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

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

Related Commands

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

connect retries

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

connect retries *number*
no connect retries

Syntax Description

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

Command Default

3 connection attempts

Command Modes

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

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.

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 five attempts to connect to the Cisco Unified CallManager before giving up and attempting to connect to the next Cisco Unified CallManager specified in the group:

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

Related Commands

Command	Description
associate ccm	Associates a Cisco Unified CallManager with a Cisco Unified CallManager group and establishes its priority within the group.
associate profile	Associates a DSP farm profile with a Cisco Unified CallManager group.
bind interface	Binds an interface to a Cisco Unified CallManager group.
connect interval	Specifies how many times a given profile attempts to connect to the specific Cisco Unified CallManager.

Command	Description
sccp ccm group	Creates a Cisco Unified CallManager group and enters SCCP Cisco Unified CallManager configuration mode.

connection

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

{**connection** {**plar** | **tie-line** | **plar opx** [{**cut-through-wait** | **immediate**}]} *phone-number* | **trunk** *phone-number* [**answer-mode**]}
no {**connection** {**plar** | **tie-line** | **plar opx** [{**cut-through-wait** | **immediate**}]} *phone-number* | **trunk** *phone-number* [**answer-mode**]}

Syntax Description

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

Command Default

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

Command Modes

Voice-port configuration Router (config-voiceport)

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
11.3(1)MA1	This command was implemented on the Cisco MC3810, and the **tie-line** keyword added.
11.3(1)MA5	This command was modified. The**plaropx** 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	This command was modified. The **trunk** keyword was implemented on the Cisco MC3810. The **trunkanswer-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	This command was modified. The **cut-through-wait** keyword was added.
12.4(11)XW	This command was modified. The **immediate**keyword was added.
12.4(20)T	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

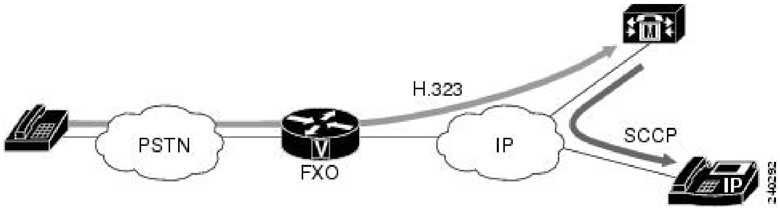
Use the **connection** command to specify a connection mode for a specific interface. For example, use the **connection plar** command to specify a PLAR interface. The string you configure for this command is used as the called number for all incoming calls over this connection. The destination peer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

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

Note **immediate** keyword is not recommended to configure on FXO ports that are managed by Cisco Unified Communications Manager (SCCP or MGCP) with caller ID enabled under voice-port. If **immediate** keyword is configured, then Cisco Unified Communications Manager could instruct FXO port immediately connected to destination port, close the loop as answer signal, stop collecting the caller ID and enter answer stage while the first ring is still on.

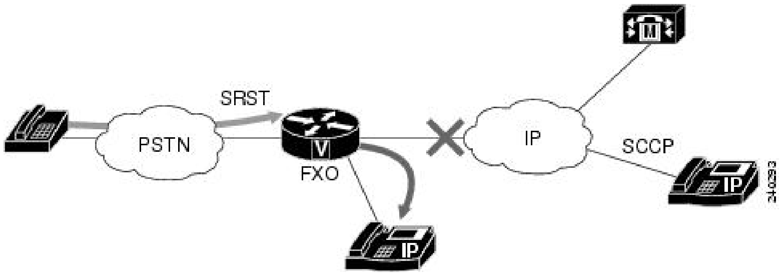
The two figures below show the network topology and call flow for the FXO Delayed Caller ID feature. The caller is in the PSTN, and the call arrives via an FXO port at the gateway. In the figure below, the gateway is connected via H.323 to Cisco CallManager. Cisco CallManager extends the call to the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

Figure 11: Network Topology for FXO Delayed Caller ID - H.323



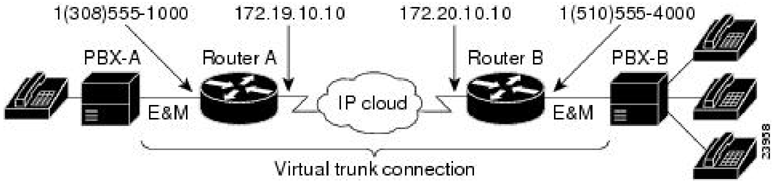
In the figure below, the gateway is on the same router as the figure above, and Survivable Remote Site Telephony (SRST) is active. SRST extends the call to the called party, which is a Skinny Client Control Protocol (SCCP)-based IP phone (Cisco 7941).

Figure 12: Network Topology for FXO Delayed Caller ID - SRST



Use the **connectiontrunk** 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 Virtual Trunk Connection Figure). 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 13: Virtual Trunk Connection



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

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

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

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

Note When using the **connectiontrunk** command, you must enter the **shutdown** command followed by the **noshutdown** 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 **connectiontie-line** command when the dial plan requires you to add digits in front of any digits dialed by the PBX, and the combined set of digits is used to route the call onto the network. The operation is similar to the **connectionplar** command operation, but in this case, the tie-line port waits to collect thedigits from the PBX. Tie-line digits are automatically stripped by a terminating port.

Examples

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

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

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

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

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

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

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

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

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

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

The following examples show configuration of the routers on both sides of a VoIP connection (as illustrated in the figure above) 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.

conn-reuse

To reuse the TCP connection of a SIP registration for an endpoint behind a firewall, use **conn-reuse** command in voice service SIP or voice class tenant configuration mode. To disable, use the **no** form of this command.

conn-reuse { **system** }
no conn-reuse

Syntax Description

system	Specifies that the conn-reuse requests use the global voice service voip value. This keyword is available only for the voice class tenant mode to allow it to fallback to the global configuration.

Command Default

This command is disabled by default.

Command Modes

Voice service SIP configuration (conf-serv-sip)

Voice class tenant configuration (config-class)

Command History

Release	Modification
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.
Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models under voice class tenant configuration.

Usage Guidelines

Running this command enables you to reuse the TCP connection of a SIP registration for an endpoint behind a firewall.

Examples

In voice service sip mode:

```
Router> enable
Router# configure terminal
Router(config)#voice service voip
Router(conf-voi-serv)#sip
Router(conf-serv-sip)#conn-reuse ?
  <cr>  <cr>
Router(conf-serv-sip)#conn-reuse
```

In voice class tenant mode:

```
Router> enable
Router# configure terminal
Router(config)#voice class tenant 222
Router(config-class)#conn-reuse ?
  system  Use global config for conn-reuse
  <cr>    <cr>
Router(config-class)#conn-reuse
```

Related Commands

Command	Description
connection-reuse	Uses global listener port for sending requests over UDP.

connection-reuse

To use global listener port for sending requests over UDP, use **connection-reuse** command in sip-ua mode or voice class tenant configuration mode. To disable, use **no** form of this command.

connection-reuse {**via-port** | **system**}
no connection-reuse

Syntax Description

via-port	Sends responses to the port present in via header.
system	Specifies that the connection-reuse requests use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

Command Default

CUBE or Local Gateway will use an ephemeral UDP port for sending requests over UDP.

Command Modes

SIP UA configuration

voice class tenant configuration

Command History

Release	Modification
Cisco IOS 15.6(2)T and Cisco IOS XE Denali 16.3.1	This command was modified to include the keyword: **system**. This command is now available under voice class tenants.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.
Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models under voice class tenant configuration.

Usage Guidelines

Executing this command enables the use listener port for sending requests over UDP. Default listener port for regular non-secure SIP is 5060 and secure SIP is 5061. Configure **listen-port [non-secure | secure]** *port* command in voice service voip > sip configuration mode to change the global UDP port.

Examples

In sip-ua mode:

```
Device> enable
Device# configure terminal
Device(config)# sip-ua
Device(config-sip-ua)# connection-reuse via-port
```

In voice class tenant mode:

```
Device> enable
Device# configure terminal
Device(config)# voice class tenant 1
Device(config-class)# connection-reuse via-port
```

Related Commands

Command	Description
listen-port	Changes UDP/TCP/TLS SIP listen Port.

connection-timeout

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

connection-timeout *seconds*
no connection-timeout *seconds*

Syntax Description

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

Command Default

3600 seconds (1 hour)

Command Modes

Settlement configuration (config-settlement)

Command History

Release	Modification
12.0(4)XH1	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco AS5300.
12.0(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.

connection (media-profile)

To configure idle timeout and call threshold for a media profile in CUBE, use the **connection** command in media profile configuration mode. To remove the configuration, use the **no** form of this command.

connection { **calls-threshold** *calls* | **idle-timeout** *minutes* }
no connection { **calls-threshold** *calls* | **idle-timeout** *minutes* }

Syntax Description

calls	Number of calls allowed per WebSocket connection. Range is 1–20. Default is 5.
minutes	Idle timeout period for a connection in minutes. Range: 1–60 minutes.

Command Default

Disabled by default.

Command Modes

Media Profile configuration mode (cfg-mediaprofile)

Command History

Release	Modification
Cisco IOS XE Bengaluru 17.6.1a	This command was introduced on Cisco Unified Border Element.

Usage Guidelines

The **connection** command configures the parameters associated with a media profile. You can configure the threshold for the number of calls supported per WebSocket connection. Also, you can configure the timeout interval for an idle connection using this command.

Examples

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

```
router(cfg-mediaprofile)#connection ?
calls-threshold number of calls per connection
idle-timeout idle timeout in minutes

router(cfg-mediaprofile)#connection calls-threshold ?
<1-20> number of calls per connection
router(cfg-mediaprofile)#connection calls-threshold 50

router(cfg-mediaprofile)#connection idle-timeout ?
<1-60> idle-timeout in minutes
router(cfg-mediaprofile)#connection idle-timeout 45
```

Related Commands

Command	Description
media profile stream-service	Enables stream service on CUBE.
proxy (media-profile)	Configures IP address or hostname of proxy in media profile.
source-ip (media-profile)	Configures local source IP address of a WebSocket connection.
media class	Applies the media class at the dial peer level.

contact-passing

To configure pass-through of the contact header from one leg to the other leg for 302 pass-through, use the **contact-passing** command in voice service SIP configuration mode. To disable this configuration, use the **no** form of the command.

contact-passing
no contact-passing

Syntax Description

This command has no arguments or keywords.

Command Default

Pass-through of the contact header from one leg to the other leg for 302 pass-through is not enabled.

Command Modes

Voice service SIP configuration mode (conf-serv-sip).

Voice class tenant configuration (config-class).

Command History

Release	Modification
15.4(1)T	This command was introduced.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Example

The following example shows how to configure pass-through of the contact header from one leg to the other leg for 302 pass-through using the **contact-passing** command:

```
Device> enable
Device# configure terminal
Device(config)# voice service voip
Device(conf-voi-serv)# sip
Device(conf-serv-sip)# contact-passing
Device(config-class)# end
```

Related Commands

Command	Description
requri-passing	Enables pass through of the host part of the Request-URI and To SIP headers.
session target sip-uri	Derives session target from incoming URI.
voice-class sip requri-passing	Enables the pass through of SIP URI headers.

content sdp version increment

To increment the SDP version for any RE-INVITE with SDP change even if the previous offer sent by CUBE was rejected, use **content sdp version increment** command in voice service voip sip configuration mode.

content sdp version increment

Syntax Description

This command has no arguments or keywords.

Command Default

SDP version will not be incremented for any RE-INVITE with SDP change even if the previous offer sent by CUBE was rejected.

Command Modes

voice service voip sip configuration mode (conf-serv-sip)

Command History

Release	Modification
Cisco IOS 15.5(2)T Cisco IOS XE 3.15	This command was introduced.

Usage Guidelines

Use **content sdp version increment** command to increment the SDP version for any RE-INVITE with SDP change even if the previous offer sent by CUBE was rejected.

Example

```
Device> enable
Device# configure terminal
Device(config)# voice service voip
Device(conf-voi-serv)# sip
Devoce(conf-serv-sip)# content sdp version increment
```

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 **copyflashvfc**command inprivileged EXEC mode.

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

Syntax Description

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

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

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

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

Examples

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

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

Related Commands

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

copy tftp vfc

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

copy tftp vfc *slot-number*

Syntax Description

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

Command Default

No default behavior or values

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

Use the **copytftpvfc**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 **unbundlevfc** 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 **corlistincoming** command in dial peer configuration mode. To clear the previously defined incoming COR list in preparation for redefining the incoming COR list, use the **no** form of this command.

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

Syntax Description

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

Command Default

No default behavior or values.

Command Modes

Dial peer configuration (config-dial-peer)

Command History

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

Usage Guidelines

The **dial-peercorlist** 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 **corlistincoming** 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 **corlistoutgoing** command). For example, if dial peer 100 is the incoming dial peer and its incoming COR list name is list100, dial peer 200 has list200 as the outgoing COR list name. If list100 does not include all the members of list200 (that is, if list100 is not a superset of list200), it is not possible to have a call from dial peer 100 that uses dial peer 200 as the outgoing dial peer.

Examples

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

```
dial-peer list list1
 member 900call
dial-peer list list2
 member 800call
 member othercall
dial-peer voice 526 pots
 answer-address 408555....
 corlist incoming list2
 direct-inward-dial
dial-peer voice 900 pots
 destination pattern 1900.......
 direct-inward-dial
 trunkgroup 101
 prefix 333
 corlist outgoing list1
```

Related Commands

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

corlist outgoing

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

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

Syntax Description

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

Command Default

No default behavior or values.

Command Modes

Dial peer configuration (config-dial-peer)

Command History

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

Usage Guidelines

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

Examples

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

```
dial-peer list list1
member 900call
dial-peer list list2
 member 800call
 member othercall
dial-peer voice 526 pots
 answer-address 408555....
 corlist incoming list2
 direct-inward-dial
dial-peer voice 900 pots
 destination pattern 1900.......
 direct-inward-dial
 trunk group 101
 prefix 333
 corlist outgoing list1
```

cpa

To enable the call progress analysis (CPA) algorithm for outbound VoIP calls and to set CPA parameters, use the **cpa** command in voice service configuration mode. To disable the CPA algorithm, use the **no** form of this command.

cpa [{**threshold** {**active-signal** {**9db** | **12db** | **15db** | **18db** | **21db**} | **noise-level** {**max** {**-45dBm0** | **-50dBm0** | **-55dBm0** | **-60dBm0**} | **min** {**-55dBm0** | **-60dBm0** | **-65dBm0** | **-70dBm0**}}} | **timing** {**live-person** *max-duration* | **noise-period** *max-duration* | **silent** *min-duration* | **term-tone** *max-duration* | **timeout** *max-duration* | **valid-speech** *min-duration*}}]
no cpa

Syntax Description

threshold	(Optional) Sets the CPA thresholds, in decibels (dB).
active-signal	(Optional) Sets the active signal threshold that is related to the measured noise floor level.
9dB | **12dB** | **15dB** | **18dB** | **21dB**	(Optional) Specifies active signal thresholds above the measured noise floor level (in dB). The default value is 15 dB.
noise-level	(Optional) Sets the CPA noise floor level limits.
max	(Optional) Sets the maximum noise floor level.
-45dBm0 | **-50dBm0** | **-55dBm0** | **-60dBm0**	(Optional) Specifies maximum noise floor level values (root mean square), in dBm0, where dBm0 is decibels referred to one milliwatt and corrected to a 0-dBm effective power level. The default value is -50 dBm0.
min	(Optional) Sets the minimum noise floor level.
-55dBm0 | **-60dBm0** | **-65dBm0** | **-70dBm0**	(Optional) Minimum noise floor level values, in dBm0, where dBm0 is decibels referred to one milliwatt and corrected to a 0-dBm effective power level. The default value is -60 dBm0. Note that this value must be less than or equal to the value configured by the **cpa threshold noise-level max** command.
timing	(Optional) Sets the CPA timing parameters.
live-person *max-duration*	(Optional) Sets the maximum waiting time (in milliseconds) that the CPA algorithm uses to determine if the call is answered by a living person. The range is from 1 to 60000. The default value is 2500.
noise-period *max-duration*	(Optional) Sets the maximum waiting time (in milliseconds) that the CPA algorithm uses to measure the noise floor level at the beginning of the call. The range is from 1 to 60000. The default value is 100.
silent *min-duration*	(Optional) Sets the minimum silent duration (in milliseconds) afer active speech is detected for the CPA algorithm to declare that the call is answered by a live human. The range is from 1 to 60000. The default value is 375.

term-tone *max-duration*	(Optional) Sets the maximum waiting time (in milliseconds) that the CPA algorithm uses to wait for the answering machine termination tone after the answering machine is detected. The range is from 1 to 60000. The default value is 15000.
timeout *max-duration*	(Optional) Sets the maximum waiting time (in milliseconds) that the CPA algorithm uses to timeout if it does not detect any voice signal. The range is from 1 to 60000. The default value is 3000.
valid-speech *min-duration*	(Optional) Sets the minimum voice duration (in milliseconds) for the CPA algorithm to consider it as a valid speech signal. The range is from 1 to 60000. The default value is 112.

Command Default

The CPA algorithm is enabled for outbound VoIP calls.

Command Modes

Voice service configuration (conf-voi-serv)

Command History

Release	Modification
12.4(24)T	This command was introduced.
Cisco IOS XE Release 3.9S	This command was integrated into Cisco IOS XE Release 3.9S.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Use the **cpa** command to enable the call progress analysis algorithm for outbound VoIP calls. You must activate the CPA capability using the **call-progress-analysis** command in digital signal processor (DSP) farm profile configuration mode before you use the **cpa** command to configure values for threshold and timing parameters.

Note With VCC codec configured on the dial-peer, the list of codecs in the VCC should match with the list of codec provisioned in DSP transcoder profile when CPA is enabled.

Examples

The following example shows how to enable CPA and configure the timing and threshold parameters:

```
Device> enable
Device# configure terminal
Device(config)# voice service voip
Device(conf-voi-serv)# cpa
Device(conf-voi-serv)# cpa threshold active-signal 12dB
Device(conf-voi-serv)# cpa threshold noise-level max -55dBm0
Device(conf-voi-serv)# cpa threshold noise-level min -65dBm0
Device(conf-voi-serv)# cpa timing live-person 5000
Device(conf-voi-serv)# cpa timing timeout 2000
Device(conf-voi-serv)# cpa timing term-tone 7000
Device(conf-voi-serv)# cpa timing silent 380
Device(conf-voi-serv)# cpa timing valid-speech 113
Device(conf-voi-serv)# cpa timing noise-period 101
Device(conf-voi-serv)# exit
```

Related Commands

Command	Description
call-progress-analysis	Activates CPA for a DSP farm profile on the Cisco UBE.
dspfarm profile	Enters DSP farm profile configuration mode and defines a profile for DSP farm services.
noisefloor	Configures the noise level, in dBm, above which noise reduction (NR) will operate.

cptone

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

cptone *locale*
no cptone *locale*

Syntax Description

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

Command Default

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

Command Modes

Voice-port configuration (config-voiceport)

Command History

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

Usage Guidelines

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

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

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

The table below 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**	Malta	**mt**
Brazil	**br**	Mexico	**mx**
Canada	**ca**	Nepal	**np**
Chile	**cl**	Netherlands	**nl**
China	**cn**	New Zealand	**nz**
Colombia	**co**	Nigeria	**ng**
Custom 1 [1]	**c1**	Norway	**no**
Custom 2 [2]	**c2**	Oman	**om**
Czech Republic	**cz**	Pakistan	**pk**
Denmark	**dk**	Panama	**pa**
Egypt	**eg**	Peru	**pe**
Finland	**fi**	Philippines	**ph**
France	**fr**	Poland	**pl**
Germany	**de**	Portugal	**pt**
Ghana	**gh**	Russian Federation	**ru**
Great Britain	**gb**	Saudi Arabia	**sa**
Greece	**gr**	Singapore	**sg**

Country	cptone *locale* Command Entry	Country	cptone *locale* Command Entry
Hong Kong	**hk**	Slovakia	**sk**
Hungary	**hu**	Slovenia	**si**
Iceland	**is**	South Africa	**za**
India	**in**	Spain	**es**
Indonesia	**id**	Sweden	**se**
Ireland	**ie**	Switzerland	**ch**
Israel	**il**	Taiwan	**tw**
Italy	**it**	Thailand	**th**
Japan	**jp**	Turkey	**tr**
Jordan	**jo**	United Arab Emirates	**ae**
Kenya	**ke**	United States	**us**
Korea Republic	**kr**	Venezuela	**ve**
Kuwait	**kw**	Zimbabwe	**zw**

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

Examples

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

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

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

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

Related Commands

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

cptone call-waiting repetition interval

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

cptone call-waiting repetition interval *second*
no cptone call-waiting repetition interval

Syntax Description

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

Command Default

A single-beep tone is the default behavior.

Command Modes

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

Command History

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

Usage Guidelines

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

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

Examples

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

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

Related Commands

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

credential load

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

credential load *tag*

Syntax Description

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

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

Examples

The following example shows how to reload credential file 3:

```
credential load 3
```

Related Commands

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

credentials (SIP UA)

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

credentials { **dhcp** | **number** *number* **username** *username* } **password** { **0** | **6** | **7** } *password* **realm** *realm*
no credentials { **dhcp** | **number** *number* **username** *username* } **password** { **0** | **6** | **7** } *password* **realm** *realm*

Syntax Description

dhcp	(Optional) Specifies the Dynamic Host Configuration Protocol (DHCP) is to be used to send the SIP message.
number *number*	(Optional) A string representing the registrar with which the SIP trunk will register (must be at least four characters).
username *username*	A string representing the username for the user who is providing authentication (must be at least four characters). This option is only valid when configuring a specific registrar using the **number** keyword.
password	Specifies password settings for authentication.
0	Specifies the encryption type as cleartext (no encryption).
6	Specifies secure reversible encryption for passwords using type **6** Advanced Encryption Scheme (AES). **Note** Requires AES primary key to be preconfigured.
7	Specifies the encryption type as encrypted.
password	A string representing the password for authentication. If no encryption type is specified, the password will be cleartext format. The string must be between 4 and 128 characters.
realm *realm*	(Optional) A string representing the domain where the credentials are applicable.

Command Default

SIP digest credentials are disabled.

Command Modes

SIP UA configuration (config-sip-ua)

Voice class tenant configuration (config-class)

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.4(22)T	This command was integrated into Cisco IOS Release 12.4(22)T.

Release	Modification
12.4(22)YB	This command was modified. The **dhcp** keyword was added and the **username** keyword and *username* argument were removed.
15.0(1)M	This command was integrated into Cisco IOS Release 15.0(1)M.
15.0(1)XA	This command was modified. The **number** keyword and *number* argument were added and the **username** keyword and *username* argument reintroduced to configure credentials for a given registrar when multiple registrars are configured.
15.1(1)T	This command was integrated into Cisco IOS Release 15.1(1)T.
15.6(2)T and IOS XE Denali 16.3.1	This command is now available under voice class tenants.
IOS XE 16.11.1a	Secure reversible encryption for passwords using type **6** Advanced Encryption Scheme (AES) was introduced.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

The following configuration rules are applicable when credentials are enabled:

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

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

It is mandatory to specify the encryption type for the password. If a clear text password (type **0**) is configured, it is encrypted as type **6** before saving it to the running configuration.

If you specify the encryption type as **6** or **7**, the entered password is checked against a valid type **6** or **7** password format and saved as type **6** or **7** respectively.

Type-6 passwords are encrypted using AES cipher and a user-defined primary key. These passwords are comparatively more secure. The primary key is never displayed in the configuration. Without the knowledge of the primary key, type **6** passwords are unusable. If the primary key is modified, the password that is saved as type 6 is re-encrypted with the new primary key. If the primary key configuration is removed, the type **6** passwords cannot be decrypted, which may result in the authentication failure for calls and registrations.

Note When backing up a configuration or migrating the configuration to another device, the primary key is not dumped. Hence the primary key must be configured again manually.

To configure an encrypted preshared key, see Configuring an Encrypted Preshared Key.

Note The password type **7** is supported in IOS XE Release 16.11.1a, but will be deprecated in the later releases. Following warning message is displayed when encryption type **7** is configured.

```
Warning: Command has been added to the configuration using a type 7
password. However, type 7 passwords will soon be deprecated. Migrate to
a supported password type 6.
```

Note In YANG, you cannot configure the same username across two different realms.

Examples

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

```
Router> enable
Router# configure terminal
Router(config)# sip-ua
Router(config-sip-ua)# credentials dhcp password 6 095FB01AA000401 realm example.com
```

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

```
Router> enable
Router# configure terminal
Router(config)# sip-ua
Router(config-sip-ua)# no credentials dhcp password 6 095FB01AA000401 realm example.com
```

Related Commands

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

crypto

To specify the preference for a SRTP cipher-suite that will be offered by Cisco Unified Border Element (CUBE) in the SDP in offer and answer, use the **crypto** command in voice class configuration mode. To disable this functionality, use the **no** form of this command.

crypto *preference cipher-suite*
no crypto *preference*

Syntax Description

preference	Specifies the preference for a cipher-suite. The range is from 1 to 4, where 1 is the highest.
cipher-suite	Associates the cipher-suite with the preference. The following cipher-suites are supported: • AEAD_AES_256_GCM • AEAD_AES_128_GCM • AES_CM_128_HMAC_SHA1_80 • AES_CM_128_HMAC_SHA1_32

Command Default

If this command is not configured, the default behavior is to offer the srtp-cipher suites in the following preference order:

- AEAD_AES_256_GCM
- AEAD_AES_128_GCM
- AES_CM_128_HMAC_SHA1_80
- AES_CM_128_HMAC_SHA1_32

Command Modes

voice class srtp-crypto (config-class)

Command History

Release	Modification
Cisco IOS XE Everest 16.5.1b	This command was introduced.

Usage Guidelines

If you change the preference of an already configured cipher-suite, the preference is overwritten.

Examples

Specify preference for SRTP cipher-suites

The following is an example for specifying the preference for SRTP cipher-suites:

```
Device> enable
Device# configure terminal
Device(config)# voice class srtp-crypto 100
Device(config-class)# crypto 1 AEAD_AES_256_GCM
Device(config-class)# crypto 2 AEAD_AES_128_GCM
Device(config-class)# crypto 4 AES_CM_128_HMAC_SHA1_32
```

Overwrite a cipher-suite preference

Specify SRTP cipher-suite preference:

```
Device> enable
Device# configure terminal
Device(config)# voice class srtp-crypto 100
Device(config-class)# crypto 1 AEAD_AES_256_GCM
Device(config-class)# crypto 2 AEAD_AES_128_GCM
Device(config-class)# crypto 4 AES_CM_128_HMAC_SHA1_32
```

The following is the snippet of **show running-config** command output showing the cipher-suite preference:

```
Device# show running-config
voice class srtp-crypto 100
crypto 1 AEAD_AES_256_GCM
crypto 2 AEAD_AES_128_GCM
crypto 4 AES_CM_128_HMAC_SHA1_32
```

If you want to change the preference 4 to AES_CM_128_HMAC_SHA1_80, execute the following command:

```
Device(config-class)# crypto 4 AES_CM_128_HMAC_SHA1_80
```

The following is the snippet of **show running-config** command output showing the change in cipher-suite:

```
Device# show running-config
voice class srtp-crypto 100
crypto 1 AEAD_AES_256_GCM
crypto 2 AEAD_AES_128_GCM
crypto 4 AES_CM_128_HMAC_SHA1_80
```

If you want to change the preference of AES_CM_128_HMAC_SHA1_80 to 3, execute the following commands:

```
Device(config-class)# no crypto 4
Device(config-class)# crypto 3 AES_CM_128_HMAC_SHA1_80
```

The following is the snippet of **show running-config** command output showing the cipher-suite preference overwritten:

```
Device# show running-config
voice class srtp-crypto 100
crypto 1 AEAD_AES_256_GCM
crypto 2 AEAD_AES_128_GCM
crypto 3 AES_CM_128_HMAC_SHA1_80
```

Related Commands

Command	Description
srtp-crypto	Assigns a previously configured crypto-suite selection preference list globally or to a voice class tenant.

Command	Description
voice class sip srtp-crypto	Enters voice class configuration mode and assigns an identification tag for a srtp-crypto voice class.
show sip-ua calls	Displays active user agent client (UAC) and user agent server (UAS) information on Session Initiation Protocol (SIP) calls.
show sip-ua srtp	Displays Session Initiation Protocol (SIP) user-agent (UA) Secure Real-time Transport Protocol (SRTP) information.

crypto signaling

To identify the **trustpoint** *trustpoint-name* keyword and argument used during the Transport Layer Security (TLS) handshake that corresponds to the remote device address, use the **crypto signaling** command in SIP user agent (UA) configuration mode. To reset to the default **trustpoint** string, use the **no** form of this command.

crypto signaling {**default** | **remote-addr** *ip address subnet-mask*}[**tls-profile** *tag* |**trustpoint** *trustpoint-name* [**cn-san-validate server**][**client-vtp** *trustpoint-name*] [{**ecdsa-cipher** [**curve-size 384**] | **strict-cipher**}]]

no crypto signaling{**remote-addr** ip-address subnet-mask | **default**}

Syntax Description

default	(Optional) Configures the default trustpoint.
remote-addr ip-address subnet-mask	(Optional) Associates an Internet Protocol (IP) address to a trustpoint.
tls-profile *tag*	(Optional) Associates TLS profile configuration to the command **crypto signaling**.
trustpoint *trustpoint-name*	(Optional) **trustpoint** *trustpoint-name* name refers to the device's certificate generated as part of the enrollment process using Cisco IOS public-key infrastructure (PKI) commands.
cn-san-validate server	(Optional) Enables the server identity validation through Common Name (CN) and Subject Alternate Name (SAN) fields in the server certificate during client-side SIP/TLS connections.
client-vtp *trustpoint-name*	(Optional) Assigns a client verification trustpoint to SIP-UA.
ecdsa-cipher	(Optional) When the **ecdsa-cipher** keyword is not specified, the SIP TLS process uses the larger set of ciphers depending on the support at the Secure Socket Layer (SSL). Following are the cipher suites supported: • TLS_ECDHE_ECDSA_WITH_AES_128_GCM_SHA256 • TLS_ECDHE_ECDSA_WITH_AES_256_GCM_SHA384
curve-size 384	(Optional) Configures the specific size of elliptic curves to be used for a TLS session.

strict-cipher	(Optional) The **strict-cipher** keyword supports only the TLS Rivest, Shamir, and Adelman (RSA) encryption with the Advanced Encryption Standard-128 (AES-128) cipher suite. Following are the cipher suites supported: • TLS_RSA_WITH_AES_128_CBC_SHA • TLS_DHE_RSA_WITH_AES_128_CBC_SHA1 • TLS_ECDHE_RSA_WITH_AES_128_GCM_SHA256 • TLS_ECDHE_RSA_WITH_AES_256_GCM_SHA384 **Note** When the **strict-cipher** keyword is not specified, the SIP TLS process uses the default set of ciphers depending on the support at the Secure Socket Layer (SSL).

Command Default

The crypto signaling command is disabled.

Command Modes

SIP UA configuration (sip-ua)

Command History

Release	Modification
12.4(6)T	This command was introduced.
15.6(1)T and 3.17S	This command was modified to include the keyword: **ecdsa-cipher**.
16.9.1	This command was modified to include the keyword: **client-vtp**.
16.10.1a	This command was modified to include the keyword: **curve-size 384**.
16.11.1a	This command was modified to include the keyword: **cn-san-validateserver**.
Cisco IOS XE Amsterdam 17.3.1a	This comand was modified to include the keyword: **tls-profile***tag*.
Cisco IOS XE Cupertino 17.7.1a	Introduced Yang Model support for this command.

Usage Guidelines

The **trustpoint** *trustpoint-name* keyword and argument refers to the CUBE certificate generated as part of the enrollment process using Cisco IOS PKI commands.

When a single certificate is configured, it is used by all the remote devices and is configured by the **default** keyword.

When multiple certificates are used, they may be associated with remote services using the **remote-addr** argument for each trustpoint. The **remote-addr** and default arguments may be used together to cover all services as required.

Note The default cipher suite in this case is the following set that is supported by the SSL layer on CUBE:

- TLS_RSA_WITH_RC4_128_MD5
- TLS_RSA_WITH_AES_128_CBC_SHA
- TLS_DHE_RSA_WITH_AES_128_CBC_SHA1
- TLS_ECDHE_RSA_WITH_AES_128_GCM_SHA256
- TLS_ECDHE_RSA_WITH_AES_256_GCM_SHA384
- TLS_ECDHE_ECDSA_WITH_AES_128_GCM_SHA256
- TLS_ECDHE_ECDSA_WITH_AES_256_GCM_SHA384

The keyword **cn-san-validate server** enables server identity validation through the CN and SAN fields in the certificate when establishing client-side SIP/TLS connections. Validation of the CN and SAN fields of the server certificate ensures that the server-side domain is a valid entity. When creating a secure connection with a SIP server, CUBE validates the configured session target domain name against the CN/SAN fields in the server's certificate before establishing a TLS session. Once you configure **cn-san-validateserver**, validation of the server identity happens for every new TLS connection.

The **tls-profile** option associates the TLS policy configurations made through the associated **voice class tls-profile**configuration. In addition to the TLS policy options available directly with the **crypto signaling** command, a **tls-profile** also includes the **sni send** option.

sni send enables Server Name Indication (SNI), a TLS extension that allows a TLS client to indicate the name of the server it is trying to connect to during the initial TLS handshake process. Only the fully qualified DNS hostname of the server is sent in the client hello. SNI does not support IPv4 and IPv6 addresses in the client hello extension. After receiving a "hello" with the server name from the TLS client, the server uses the appropriate certificate in the subsequent TLS handshake process. SNI requires TLS version 1.2.

Note From Cisco IOS XE Amsterdam 17.3.1a onwards, new TLS policy features will only be available through a **voice class tls-profile** configuration.

The **crypto signaling** command continues to support previously existing TLS crypto options. You can use either the **voice class tls-profile** *tag* or **crypto signaling** command to configure a trustpoint. From Cisco IOS XE Amsterdam 17.3.1a onwards, we recommend that you use the command **voice class tls-profile** *tag* to perform TLS profile configurations.

Examples

The following example configures the CUBE to use the **trustpoint** *trustpoint-name* keyword and argument when it establishes or accepts the TLS connection with a remote device with IP address 172.16.0.0:

```
configure terminal
sip-ua
 crypto signaling remote-addr 172.16.0.0 trustpoint user1
```

The following example configures the CUBE to use **trustpoint** *trustpoint-name* keyword and argument when it establishes or accepts the TLS connection with any remote devices:

```
configure terminal
sip-ua
 crypto signaling default trustpoint cube
```

The following example configures the CUBE to use its **trustpoint** *trustpoint-name* keyword and argument when it establishes or accepts the TLS connection with any remote devices with IP address 172.16.0.0:

```
configure terminal
sip-ua
 crypto signaling remote-addr 172.16.0.0 trustpoint cube ecdsa-cipher
```

The following example configures the specific size of elliptic curves to be used for a TLS session:

```
configure terminal
sip-ua
 crypto signaling default trustpoint cubeTP ecdsa-cipher curve-size 384
```

The following example configures the CUBE to perform the server identity validation through Common Name (CN) and Subject Alternate Name (SAN) fields in the server certificate:

```
configure terminal
sip-ua
 crypto signaling default trustpoint cubeTP cn-san-validate server
```

The following example, associates voice class configurations done using the command **voice class tls-profile** *tag* to the command **crypto signaling**:

```
/* Configure TLS Profile Tag */
Router#configure terminal
Router(config)#voice class tls-profile 2
Router(config-class)#trustpoint TP1
exit
/* Associate TLS Profile Tag to Crypto Signaling */
Router(config)#sip-ua
Router(config-sip-ua)#crypto signaling default tls-profile 2
Router(config-sip-ua)#crypto signaling remote-addr 192.0.2.1 255.255.255.255 tls-profile 2
```

Related Commands

Command	Description
sip-ua	Enables the SIP user agent configuration commands.
voice class tls-profile *tag*	Enables configuration of voice class commands required for a TLS session.