voice-class sip error-code-override through vxml version 2.0

voice-class sip error-code-override

To configure the Session Initiation Protocol (SIP) error code that a dial peer uses for options-keepalive failures, call spike, or cac-bandwidth failures, use the **voice-class sip error-code-override** command in dial peer voice configuration mode. To disable the SIP error code configuration, use the **no** form of this command.

voice-class sip error-code-override {**options-keepalive failure**| **call spike failure**| **cac-bandwidth failure**} {*sip-status-code-number*| **system**}

no voice-class sip error-code-override {**options-keepalive failure**| **call spike failure**| **cac-bandwidth failure**}

Syntax Description

options-keepalive failure	Configures the SIP error code for options-keepalive failures.
call spike failure	Configures the SIP error code for call spike failures.
cac-bandwidth failure	Configures the SIP error code for Call Admission Control bandwidth failures.
sip-status-code-number	The SIP status code that is sent for the options keepalive, call spike, or cac-bandwidth failure. The range is from 400 to 699. The default value is 503. The table below in the "Usage Guidelines" section describes these error codes.
system	Specifies the system configuration used for keepalive, call spike, or cac-bandwidth failures.

Command Default

By default the SIP error code is not configured.

Command Modes

Dial peer voice configuration (config-dial-peer)

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.
15.1(3)T	This command was modified. The **call spike failure** keyword was added.
15.2(2)T	This command was modified. The **cac-bandwidth failure** keyword was added.

Usage Guidelines

The **voice-class sip error-code-override** command in dial peer voice configuration mode configures the error code response for keepalive options, call spike, or cac-bandwidth failures at the dial peer level. The **error-code-override** command in voice service SIP configuration mode configures the error code responses for options-keepalive, call spike, or cac-bandwidth failures globally.

The table below describes the SIP error codes.

Table 1: SIP Error Codes

Error Code Number	Description
400	Bad request
401	Unauthorized
402	Payment required
403	Forbidden
404	Not found
408	Request timed out
416	Unsupported Uniform Resource Identifier (URI)
480	Temporarily unavailable
482	Loop detected
484	Address incomplete
486	Busy here
487	Request terminated
488	Not acceptable here
500–599	SIP 5xx—server/service failure
500	Internal server error
502	Bad gateway
503	Service unavailable
600–699	SIP 6xx—global failure

Examples

The following example shows how to configure the SIP error code for options-keepalive failures using the **voice-class sip error-code-override** command:

```
Router(config)# dial-peer voice 432 voip system
Router(config-dial-peer)# voice-class sip error-code-override options-keepalive failure 502
```

The following example shows how to configure the SIP error code for call spike failures using the **voice-class sip error-code-override** command:

```
Router(config)# dial-peer voice 432 voip system
Router(config-dial-peer)# voice-class sip error-code-override call spike failure 502
```

The following example shows how to configure the SIP error code for Call Admission Control bandwidth failures:

```
Router(config)# dial-peer voice 432 voip system
Router(config-dial-peer)# voice-class sip error-code-override cac-bandwidth failure 502
```

Related Commands

Command	Description
error-code-override	Configures the SIP error code for options-keepalive, call spike, or cac-bandwidth failures in voice service SIP and dial peer voice configuration mode, respectively.

voice-class sip g729 annexb-all

To configure settings on a Cisco IOS Session Initiation Protocol (SIP) gateway that determine if a specific dial peer on the gateway treats the G.729br8 codec as superset of G.729r8 and G.729br8 codecs for interoperation with Cisco Unified Communications Manager, use the **voice-class sip g729 annexb-all** command in dial peer voice configuration mode. To prevent a dial peer from treating the G.729br8 codec as a superset of the G.729r8 and G.729br8 codecs, use the **no** form of this command.

voice-class sip g729 annexb-all [system]

no voice-class sip g729 annexb-all

Syntax Description

annexb-all	Specifies that the G.729br8 codec is treated as a superset of G.729r8 and G.729br8 codecs to communicate with Cisco Unified Communications Manager.
system	(Optional) Specifies that the dial peer allow communication between incompatible G.729 codecs according to global settings configured for this feature on the Cisco IOS SIP gateway.

Command Default

The dial peer defers to global (system) settings for the Cisco IOS gateway.

Command Modes

Dial peer voice configuration (config-dial-peer)

Command History

Release	Modification
12.4(15)XZ	This command was introduced.
12.4(20)T	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

There are four variations of the G.729 coder-decoder (codec), which fall into two categories:

High Complexity

- G.729 (g729r8)--a high complexity algorithm codec on which all other G.729 codec variations are based.
- G.729 Annex-B (g729br8 or G.729B)--a variation of the G.729 codec that allows the DSP to detect and measure voice activity and convey suppressed noise levels for re-creation at the other end. Additionally, the Annex-B codec includes Internet Engineering Task Force (IETF) voice activity detection (VAD) and comfort noise generation (CNG) functionality.

Medium Complexity

- G.729 Annex-A (g729ar8 or G.729A)--a variation of the G.729 codec that sacrifices some voice quality to lessen the load on the DSP. All platforms that support G.729 also support G.729A.
- G.729A Annex-B (g729abr8 or G.729AB)--a variation of the G.729 Annex-B codec that, like G.729B, sacrifices voice quality to lessen the load on the DSP. Additionally, the G.729AB codec also includes IETF VAD and CNG functionality.

The VAD and CNG functionality is what causes the instability during communication attempts between two DSPs where one DSP is configured with Annex-B (G.729B or G.729AB) and the other without (G.729 or G.729A). All other combinations interoperate. To configure a dial peer on a Cisco IOS SIP gateway for interoperation with Cisco Unified Communications Manager (formerly known as the Cisco CallManager, or CCM), use the **voice-class sip g729 annexb-all** command in dial peer voice configuration mode to do one of the following:

- Override global settings for a Cisco IOS gateway and configure the dial peer to accept and connect calls between two DSPs with incompatible G.729 codecs.
- Specify that an individual dial peer use the global (**system**) settings on the Cisco IOS SIP gateway.
- Use the no form of the command to override global settings for the Cisco IOS gateway and specify that the dial peer does not treat the G.729br8 codec as a superset of G.729r8 and G.729br8 codecs.

Use the **g729 annexb-all** command in voice service SIP configuration mode to configure the global settings for the Cisco IOS SIP gateway.

Examples

The following example shows how to configure a dial peer on a Cisco IOS SIP gateway to connect calls between two DSPs using incompatible G.729 codecs, overriding global gateway settings for this feature:

```
Router> enable
Router# configure
 terminal
Router(config)# dial-peer
 voice 1
Router(config-dial-peer)# voice-class sip g729 annexb-all
```

Related Commands

Command	Description
g729 annexb-all	Configure global settings that determine if a Cisco IOS SIP gateway treats the G.729br8 codec as superset of G.729r8 and G.729br8 codecs.

voice-class sip history-info

To enable Session Initiation Protocol (SIP) history-info header support on the Cisco IOS gateway at the dial-peer level, use the **voice-class sip history-info** command in dial peer configuration mode. To disable SIP history-info header support, use the **no** form of this command.

voice-class sip history-info [system]

no voice-class sip history-info

Syntax Description

system	(Optional) Enables history-info support using global configuration settings.

Command Default

History-info header support is disabled.

Command Modes

Dial peer configuration (conf-dial-peer)

Command History

Release	Modification
12.4(22)T	This command was introduced.
Cisco IOS XE Release 3.1S	This command was integrated into Cisco IOS XE Release 3.1S

Usage Guidelines

Use this command to enable history-info header support at the dial-peer level. The history-info header (as defined in RFC 4244) records the call or dialog history. The receiving application uses the history-info header information to determine how and why the call has reached it.

Note The Cisco IOS SIP gateway cannot use the information in the history-info header to make routing decisions.

Examples

The following example enables SIP history-info header support at the dial-peer level:

```
Router(config)# dial-peer voice 2 voip
Router(config-dial-peer)# voice-class sip history-info
```

The following example enables SIP history-info header support at the dial-peer level using the global configuration settings:

```
Router(config)# dial-peer voice 2 voip
Router(config-dial-peer)# voice-class sip history-info system
```

Related Commands

Command	Description
history-info	Enables SIP history-info header support on Cisco IOS gateway at a global level.

voice-class sip localhost

To configure individual dial peers to override global settings on Cisco IOS voice gateways, Cisco Unified Border Element (Cisco UBE), or Cisco Unified Communications Manager Express (Cisco Unified CME) and substitute a Domain Name System (DNS) hostname or domain as the localhost name in place of the physical IP address in the From, Call-ID, and Remote-Party-ID headers in outgoing messages, use the **voice-class sip localhost** command in dial peer voice configuration mode. To disable substitution of a localhost name on a specific dial peer, use the **no** form of this command. To configure a specific dial peer to defer to global settings for localhost name substitution, use the **default** form of this command.

voice-class sip localhost dns:[*hostname*]*domain*[**preferred**]

no voice-class sip localhost

default voice-class sip localhost

Syntax Description

dns: [*hostname.*]*domain*	Alphanumeric value representing the DNS domain (consisting of the domain name with or without a specific hostname) in place of the physical IP address that is used in the host portion of the From, Call-ID, and Remote-Party-ID headers in outgoing messages. This value can be the hostname and the domain separated by a period (**dns:** *hostname.domain*) or just the domain name (**dns:** *domain*). In both case, the **dns:** delimiter must be included as the first four characters.
preferred	(Optional) Designates the specified DNS hostname as preferred.

Command Default

The dial peer uses the global configuration setting to determine whether a DNS localhost name is substituted in place of the physical IP address in the From, Call-ID, and Remote-Party-ID headers of outgoing messages.

Command Modes

Dial peer voice configuration (config-dial-peer)

Command History

Release	Modification
12.4(2)T	This command was introduced.
15.0(1)XA	This command was modified. The **preferred** keyword was added to specify the preferred localhost if multiple registrars are configured on a SIP trunk.
IOS Release XE 2.5	This command was integrated into Cisco IOS XE Release 2.5.

Release	Modification
15.1(1)T	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

Use the **voice-class sip localhost** command in dial peer voice configuration mode to override the global configuration on Cisco IOS voice gateways, Cisco UBEs, or Cisco Unified CME and configure a DNS localhost name to be used in place of the physical IP address in the From, Call-ID, and Remote-Party-ID headers of outgoing messages on a specific dial peer. When multiple registrars are configured for an individual dial peer you can then use the **voice-class sip localhost preferred** command to specify which host is preferred for that dial peer.

To globally configure a localhost name on a Cisco IOS voice gateway, Cisco UBE, or Cisco Unified CME, use the **localhost** command in voice service SIP configuration mode. Use the **no voice-class sip localhost** command to remove localhost name configurations for the dial peer and to force the dial peer to use the physical IP address in the host portion of the From, Call-ID, and Remote-Party-ID headers regardless of the global configuration.

Examples

The following example shows how to configure dial peer 1 (overriding any global configuration) to substitute a domain (no hostname specified) as the preferred localhost name in place of the physical IP address in the From, Call-ID, and Remote-Party-ID headers of outgoing messages:

```
Router> enable
Router# configure
 terminal
Router(config)# dial-peer voice 1 voip
Router(config-dial-peer)# voice-class sip localhost dns:example.com preferred
```

The following example shows how to configure dial peer 1 (overriding any global configuration) to substitute a specific hostname on a domain as the preferred localhost name in place of the physical IP address in the From, Call-ID, and Remote-Party-ID headers of outgoing messages:

```
Router> enable
Router# configure
 terminal
Router(config)# dial-peer voice 1 voip
Router(config-dial-peer)# voice-class sip localhost dns:MyHost.example.com preferred
```

The following example shows how to force dial peer 1 (overriding any global configuration) to use the physical IP address in the From, Call-ID, and Remote-Party-ID headers of outgoing messages:

```
Router> enable
Router# configure
 terminal
Router(config)# dial-peer voice 1 voip
Router(config-dial-peer)# no voice-class sip localhost
```

Related Commands

Command	Description
authentication (dial peer)	Enables SIP digest authentication on an individual dial peer.
authentication (SIP UA)	Enables SIP digest authentication.

Command	Description
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 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 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 map resp-code

To configure an individual dial peer on a Cisco Unified Border Element (Cisco UBE) to map specific received Session Initiation Protocol (SIP) provisional response messages to a different SIP provisional response message on the outgoing SIP dial peer, use the **voice-class sip map resp-code** command in dial peer voice configuration mode. To disable mapping of received SIP provisional response messages on an individual dial peer, use the **no** form of this command. To configure a specific dial peer to defer to global settings for mapping of incoming SIP provisional response messages, use the **default** form of this command.

voice-class sip map resp-code 181 to 183

no voice-class sip map resp-code 181 to 183

default voice-class sip map resp-code 181 to 183

Syntax Description

181	The code representing the specific incoming SIP provisional response messages to be mapped and replaced.
to	The designator for specifying that the specified incoming SIP provisional response message should be mapped to and replaced with a different SIP provisional response message on the outgoing SIP dial peer.
183	The code representing the specific SIP provisional response message on the outgoing dial peer to which incoming SIP message responses should be mapped.

Command Default

Mapping behavior is determined by the global configuration setting, which, if not specifically configured, means that incoming SIP provisional responses are passed, as is to the outbound SIP dial peer.

Command Modes

Dial peer voice configuration (config-dial-peer)

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.
Cisco IOS XE Release 3.1S	This command was integrated into Cisco IOS XE Release 3.1S.

Usage Guidelines

Use the **voice-class sip map resp-code**command in dial peer voice configuration mode to configure an individual dial peer on a Cisco UBE to map incoming SIP 181 provisional response messages to SIP 183 provisional response messages on the outgoing SIP dial peer.

Note If the **block** command is configured for incoming SIP 181 messages, either globally or at the dial-peer level, the messages may be dropped before they can be passed or mapped to a different message--even when the **voice-class sip map resp-code** command is enabled. To globally configure whether and when incoming SIP 181 messages are dropped, use the **block** command in voice service SIP configuration mode (or use the **voice-class sip block** command in dial peer voice configuration mode to configure drop settings on individual dial peers).

To configure mapping of SIP provisional response messages globally on a Cisco UBE, use the **map resp-code** command in voice service SIP configuration mode. To disable mapping of SIP 181 message for an individual dial peer on a Cisco UBE, use the **no voice-class sip map resp-code** command in voice service SIP configuration mode.

As an example, to enable interworking of SIP endpoints that do not support the handling of SIP 181 provisional response messages, you could use the **block** command to configure a Cisco UBE to drop SIP 181 provisional response messages received on the SIP trunk or you can use the **map resp-code** command to configure the Cisco UBE to map the incoming messages to and send out, instead, SIP 183 provisional response messages to the SIP line in Cisco Unified Communications Manager Express (Cisco Unified CME).

Note This command is supported only for SIP-to-SIP calls and will have no effect on H.323-to-SIP or time-division multiplexing (TDM)-to-SIP calls.

Examples

The following example shows how to configure dial peer 1 to map incoming SIP 181 provisional response messages to SIP 183 provisional response messages on the outbound dial peer:

```
Router> enable
Router# configure
 terminal
Router(config)# dial-peer voice 1 voip
Router(config-dial-peer)# voice-class sip map resp-code 181 to 183
```

Related Commands

Command	Description
block	Configures global settings for dropping specific SIP provisional response messages on a Cisco IOS voice gateway or Cisco UBE.
map resp-code	Configures global settings on a Cisco UBE for mapping specific incoming SIP provisional response messages to a different SIP response message.

Command	Description
voice-class sip block	Configures an individual dial peer on a Cisco IOS voice gateway or Cisco UBE to drop specified SIP provisional response messages.

voice-class sip midcall-signaling

To configure the method used for signaling messages, use the **voice-class sip midcall-signaling** command in SIP configuration mode or dial peer configuration mode. To disable the mid-call signaling feature, use the **no** form of this command.

voice-class sip midcall-signaling {**passthru media-change | block | preserve-codec**}

no voice-class sip midcall-signaling

Syntax Description

passthru media-change	Passes SIP messages that inolve media-change from one IP leg to another IP leg.
block	Blocks all SIP messages during mid-call.
preserve-codec	Preserves codec negotiated during call initialization. Mid-call codec change is disabled.

Command Default

Mid call-signaling is disabled. Codec negotiation in the middle of a call is enabled.

Command Modes

Dial peer configuration mode (config-dial-peer)

Command History

Release	Modification
12.4(15)XZ	This command was introduced.
12.4(20)T	This command was integrated into Cisco IOS Release 12.4(20)T.
Cisco IOS XE Release 2.5	This command was integrated into Cisco IOS XE Release 2.5.
15.2(1)T	This command was integrated into Cisco IOS Release 15.2(1)T. The **media-change** and **block** keywords were added.
15.3(2)S, 15.3(1)T	This command was modified. The **preserve-codec** keyword was added.

Usage Guidelines

The **voice-class sip midcall-signaling** command distinguishes between the way Cisco Unified Communications Express and Cisco Unified Border Element handle signaling messages. Most SIP-to-SIP video and SIP-to-SIP reinvite based supplementary services require the **voice-class sip midcall-signaling** command to be configured before configuring other supplementary services. Supplementary service features that are functional without configuring **voice-class sip midcall-signaling** include: session refresh, fax, and refer-based supplementary services. The **voice-class sip midcall-signaling** command is for SIP-to-SIP calls only. All other calls (H323-to-SIP, and H323-to-H323) do not require the **voice-class sip midcall-signaling**command be configured.

The **allow-connections sip-to-sip** command must be configured before the **voice-class sip midcall-signaling** command.

Configuring the Session Refresh with Reinvites feature on a dial-peer basis is not supported.

Examples

The following example shows SIP messages configured to passthrough from one IP leg to another IP leg:

```
Router(config)#voice service voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# voice-class sip midcall-signaling passthru
```

The following example shows SIP messages configured to media passthru from one IP leg to another IP leg:

```
Router(config)#voice service voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# voice-class sip midcall-signaling passthru media-change
```

The following example shows how to block SIP messages.

```
Router(config)#voice service voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# voice-class sip midcall-signaling block
```

The following example shows how to disable codec negotiation in the middle of a call and retains the codec negotiated at the start of the call.

```
Router(config)#voice service voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# voice-class sip midcall-signaling preserve-codec
```

Related Commands

Command	Description
allow-connections	Allows connections between specific types of endpoints in a Cisco Unified BE.

voice-class sip options-keepalive

To monitor connectivity between Cisco Unified Border Element VoIP dial-peers and SIP servers to, use the **voice-class sip options-keepalive** command in dial peer configuration mode. To disable monitoring connectivity, use the **no** form of this command.

voice-class sip options-keepalive{**up-interval** *seconds* | **down-interval** *seconds* | **retry** *retries*}

no voice-class sip options-keepalive

Syntax Description

up-interval *seconds*	Number of up-interval seconds allowed to pass before marking the UA as unavailable.The range is 5-1200. The default is 60.
down-interval *seconds*	Number of down-interval seconds allowed to pass before marking the UA as unavailable.The range is 5-1200. The default is 30.
retry *retries*	Number of retry attempts before marking the UA as unavailable. The range is 1 to 10. The default is 5 attempts.

Command Default

The dial-peer is active (UP).

Command Modes

Dial peer configuration mode (config-dial-peer).

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.

Usage Guidelines

Use the **voice-class sip options-keepalive**command to configure a out-of-dialog (OOD) Options Ping mechanism between any number of destinations. When monitored endpoint heartbeat responses fails, the configured dial-peer is busied out. If there is a alternate dial-peer configured for the same destination pattern, the call is failed over to the next preference dial peer or the on call is rejected with an error cause code.

The response to options ping will be considered unsuccessful and dial-peer will be busied out for following scenarios:

Table 2: Error Codes that busyout the endpoint

Error Code	Description
503	service unavailable
505	sip version not supported
no response	i.e. request timeout

All other error codes, including 400 are considered a valid response and the dial peer is not busied out.

Examples

The following example shows a sample configuration of dial peer 100 configured to reset:

```
dial-peer voice 100 voip
  voice-class sip options-keepalive up-interval 12 down-interval 65 retry 3
```

Related Commands

Command	Description
dial-peer voice	Defines a particular dial peer and specifies the method of voice encapsulation

voice-class sip outbound-proxy

To configure an outbound proxy, use the **voice-class sip outbound-proxy** command in dial peer configuration mode. To reset the outbound proxy value to its default, use the **no** form of this command.

voice-class sip outbound-proxy {**dhcp**| **ipv4:** *ipv4-address*| **ipv6:** [*ipv6-address*]| **dns:** *host: domain*} [*:port-number*]

no voice-class sip outbound-proxy

Syntax Description

dhcp	Specifies that the outbound-proxy IP address is retrieved from a DHCP server.
ipv4: *ipv4-address*	Configures proxy on the server, sending all initiating requests to the specified IPv4 address destination. The colon is required.
ipv6:[*ipv6- address* **]**	Configures proxy on the server, sending all initiating requests to the specified IPv6 address destination. Brackets must be entered around the IPv6 address. The colon is required.
dns: *host:domain*	Configures proxy on the server, sending all initiating requests to the specified domain destination. The colons are required.
: *port-number*	(Optional) Port number for the Session Initiation Protocol (SIP) server. The colon is required.

Command Default

An outbound proxy is not configured.

Command Modes

Dial peer configuration (config-dial-peer)

Command History

Release	Modification
12.4(15)T	This command was introduced.
12.4(22)T	This command was modified. Support for IPv6 was added.
12.4(22)YB	This command was modified. The dhcp keyword was added.
15.0(1)M	This command was integrated in Cisco IOS Release 15.0(1)M.

Usage Guidelines

The **voice-class sip outbound-proxy** command, in dial peer configuration mode, takes precedence over the command in SIP global configuration mode.

Brackets must be entered around the IPv6 address.

Examples

The following example shows how to configure the **voice-class sip outbound-proxy**command on a dial peer to generate an IPv4 address (10.1.1.1) as an outbound proxy:

```
Router> enable
Router# configure
 terminal
Router(config)# dial
-peer
 voice
 111
 voip
Router(config-dial-peer)# voice-class sip outbound-proxy ipv4:10.1.1.1
```

The following example shows how to configure the **voice-class sip outbound-proxy**command on a dial peer to generate a domain (sipproxy:cisco.com) as an outbound proxy:

```
Router> enable
Router# configure
 terminal
Router(config)# dial
-peer
 voice
 111
 voip
Router(config-dial-peer)# voice-class sip outbound-proxy dns:sipproxy:cisco.com
```

The following example shows how to configure the **voice-class sip outbound-proxy**command on a dial peer to generate an outbound proxy using DHCP:

```
Router> enable
Router# configure
 terminal
Router(config)# dial
-peer
 voice
 111
 voip
Router(config-dial-peer)# voice-class sip outbound-proxy dhcp
```

Related Commands

Command	Description
dial -peer voice	Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial peer configuration mode.
voice service	Enters voice-service configuration mode and specifies a voice encapsulation type.

voice-class sip preloaded-route

To enable preloaded route support for dial-peer Session Initiation Protocol (SIP) calls, use the **voice-class sip preloaded-route**command in dial peer voice configuration mode. To reset to the default value, use the **no** form of this command.

voice-class sip preloaded-route {**[sip-server] service-route| system**}

no voice-class sip preloaded-route

Syntax Description

sip-server	(Optional) Adds SIP server information to the Route header.
service-route	Adds the Service-Route information to the Route header.
system	Uses the global system value. This is the default.

Command Default

SIP calls at the dial-peer level use the global configuration level settings.

Command Modes

Dial peer voice configuration (config-dial-peer)

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.

Usage Guidelines

The **voice-class sip preloaded-route** command takes precedence over the **preloaded-route** command configured in SIP configuration mode. However, if the **voice-class sip preloaded-route** command is used with the **system** keyword, the gateway uses the global settings configured by the **preloaded-route** command.

Examples

The following example shows how to configure the dial peer to include SIP server and Service-Route information in the Route header:

```
dial-peer voice 102 voip
 voice-class sip preloaded-route sip-server service-route
```

The following example shows how to configure the dial peer to include only Service-Route information in the Route header:

```
dial-peer voice 102 voip
 voice-class sip preloaded-route service-route
```

Related Commands

Command	Description
preloaded-route	Enables preloaded route support for VoIP SIP calls.

voice-class sip privacy

To set privacy support at the dial-peer level as defined in RFC 3323, use the **voice-class sip privacy** command in dial peer configuration mode. To remove privacy support as defined in RFC 3323, use the **no** form of this command.

voice-class sip privacy {**disable| pstn| system|** *privacy-option* **[critical]**}

no voice-class sip privacy

Syntax Description

disable	Disables the privacy service for this dial peer regardless of prior implementations. When selected, this becomes the only valid option.
pstn	Requests that the privacy service implements a privacy header using the default Public Switched Telephone Network (PSTN) rules for privacy (based on information in Octet 3a). When selected, this becomes the only valid option.
system	Uses the global configuration settings to enable the privacy service on this dial peer. When selected, this becomes the only valid option.

privacy-option	The privacy support options to be set at the dial-peer level. The following keywords can be specified for the *privacy-option* argument: • **header** -- Requests that privacy be enforced for all headers in the Session Initiation Protocol (SIP) message that might identify information about the subscriber. • **history** -- Requests that the information held in the history-info header is hidden outside the trust domain. • **id** -- Requests that the Network Asserted Identity that authenticated the user be kept private with respect to SIP entities outside the trusted domain. • **session** -- Requests that the information held in the session description is hidden outside the trust domain. • **user** -- Requests that privacy services provide a user-level privacy function. **Note** The keywords can be used alone, altogether, or in any combination with each other, but each keyword can be used only once.
critical	(Optional) Requests that the privacy service performs the specified service or fail the request. **Note** This optional keyword is only available after at least one of the *privacy-option* keywords (**header**, **history**, **id**, **session**, or **user**) has been specified and can be used only once per command.

Command Default

Privacy support is disabled.

Command Modes

Dial peer configuration (config-dial-peer)

Command History

Release	Modification
12.4(15)T	This command was introduced.
12.4(22)T	The **history** keyword was added to provide support for the history-info header information.

Usage Guidelines

Use the **voice-class sip privacy** command to instruct the gateway to add a Proxy-Require header, set to a value supported by RFC 3323, in outgoing SIP request messages at the dial-peer level.

Use the **voice-class sip privacy critical**command to instruct the gateway to add a Proxy-Require header with the value set to critical. If a user agent sends a request to an intermediary that does not support privacy extensions, the request fails.

The **voice-class sip privacy** command takes precedence over the **privacy** command in voice service voip sip configuration mode. However, if the **voice-class sip privacy** command is used with the **system** keyword, the gateway uses the settings configured globally by the **privacy** command.

Examples

The following example shows how to disable the privacy on dial peer 2:

```
Router> enable

Router# configure
 terminal
Router(config)# dial-peer voice 2 voip

Router(config-dial-peer)# voice-class sip privacy disable
```

The following example shows how to configure the **voice-class sip privacy** command so that the information held in the history-info header is hidden outside the trust domain:

```
Router> enable

Router# configure
 terminal
Router(config)# dial-peer voice 2 voip

Router(config-dial-peer)# voice-class sip privacy history
```

Related Commands

Command	Description
asserted-id	Sets the privacy level and enables either PAI or PPI privacy headers in outgoing SIP requests or response messages.
calling-info pstn-to-sip	Specifies calling information treatment for PSTN-to-SIP calls.
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.
privacy	Sets privacy support at the global level as defined in RFC 3323.

voice-class sip privacy-policy

To configure the privacy header policy options at the dial-peer level, use the **voice-class sip privacy-policy** command in dial peer voice configuration mode. To disable privacy-policy options, use the **no** form of this command.

voice-class sip privacy-policy {**passthru**| **send-always**| **strip** {**diversion**| **history-info**}} **[system]**

no voice-class sip privacy-policy {**passthru**| **send-always**| **strip** {**diversion**| **history-info**}}

Syntax Description

passthru	Passes the privacy values from the received message to the next call leg.
send-always	Passes a privacy header with a value of None to the next call leg, if the received message does not contain privacy values but a privacy header is required.
strip	Strip the diversion or history-info headers received from the next call leg.
diversion	Strip the diversion header received from the next call leg.
history-info	Strip the history-info header received from the next call leg.
system	(Optional) Uses the global configuration settings to configure the dial peer.

Command Default

No privacy-policy settings are configured.

Command Modes

Dial peer voice configuration (config-dial-peer)

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 integrated into Cisco IOS Release 15.1(2)T. The **strip**, **diversion**, and **history-info** keywords were added.

Usage Guidelines

If a received message contains privacy values, use the **voice-class sip privacy-policy passthru** command to ensure that the privacy values are passed from one call leg to the next. If a received message does not contain privacy values but the privacy header is required, use the **voice-class sip privacy-policy send-always** command to set the privacy header to None and forward the message to the next call leg. You can configure the system to support both options at the same time.

The **voice-class sip privacy-policy** command takes precedence over the **privacy-policy** command in voice service voip sip configuration mode. However, if the **voice-class sip privacy-policy** command is used with the **system** keyword, the gateway uses the settings configured globally by the **privacy-policy** command.

Examples

The following example shows how to enable the pass-through privacy policy on the dial peer:

```
Router> enable

Router# configure
 terminal
Router(config)# dial-peer voice 2611 voip
Router(config-dial-peer)# voice-class sip privacy-policy passthru
```

The following example shows how to enable the pass-through, send-always, and strip policies on the dial peer:

```
Router> enable

Router# configure
 terminal
Router(config)# dial-peer voice 2611 voip
Router(config-dial-peer)# voice-class sip privacy-policy passthru
Router(config-dial-peer)# voice-class sip privacy-policy send-always
Router(config-dial-peer)# voice-class sip privacy-policy strip diversion
Router(config-dial-peer)# voice-class sip privacy-policy strip history-info
```

The following example shows how to enable the send-always privacy policy on the dial peer:

```
Router> enable

Router# configure
 terminal
Router(config)# dial-peer voice 2611 voip
Router(config-dial-peer)# voice-class sip privacy-policy send-always
```

The following example shows how to enable both the pass-through privacy policy and send-always privacy policies on the dial peer:

```
Router> enable

Router# configure
 terminal
Router(config)# dial-peer voice 2611 voip
Router(config-dial-peer)# voice-class sip privacy-policy passthru
Router(config-dial-peer)# voice-class sip privacy-policy send-always
```

Related Commands

Command	Description
asserted-id	Sets the privacy level and enables either PAID or PPID privacy headers in outgoing SIP requests or response messages.

Command	Description
privacy-policy	Configures the privacy header policy options at the global configuration level.

voice-class sip random-contact

To populate the outgoing INVITE message with random-contact information (instead of clear contact information) at the dial-peer level, use the **voice-class sip random-contact** command in dial peer voice configuration mode. To disable random contact information, use the **no** form of this command.

voice-class sip random-contact [system]

no voice-class sip random-contact

Syntax Description

system	(Optional) Uses the global configuration settings to populate the INVITE message with random contact information.

Command Default

Support for random contact at the dial-peer level uses the the global configuration level settings.

Command Modes

Dial peer voice configuration (config-dial-peer)

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.

Usage Guidelines

To populate outbound INVITE messages (from the Cisco Unified Border Element) with random-contact information instead of clear-contact information at the dial-peer level, use the **voice-class sip random-contact** command. This functionality will work only when the Cisco Unified Border Element is configured for SIP registration with random-contact, using the **credentials** and **registrar** commands.

The **voice-class sip random-contact** command takes precedence over the **random-contact** command in voice service voip sip configuration mode. However, if the **voice-class sip random-contact** command is used with the **system** keyword, the gateway uses the settings configured globally by the **random-contact** command.

Examples

The following example shows how to populate outbound INVITE messages, at the dial-peer level, with random-contact information:

```
Router> enable

Router# configure
 terminal
Router(config)# dial-peer voice 2611 voip
Router(config-dial-peer)# voice-class sip random-contact
```

Related Commands

Command	Description
credentials (sip ua)	Sends a SIP registration message from a Cisco Unified Border Element in the UP state.
registrar	Enables SIP gateways to register E.164 numbers on behalf of FXS, EFXS, and SCCP phones with an external SIP proxy or SIP registrar.
random-contact	Populates the outgoing INVITE message with random contact information at the global level.

voice-class sip random-request-uri validate

To enable the validation of the called-number based on the random value generated during the registration of the number, at dial-peer configuration level, use the **voice-class sip random-request-uri validate** command in dial peer voice configuration mode. To disable validation, use the **no** form of this command.

voice-class sip random-request-uri validate [system]

no voice-class sip random-request-uri validate

Syntax Description

system	(Optional) Uses the global configuration settings to enable called-number validation on this dial peer.

Command Default

Validation is disabled.

Command Modes

Dial peer voice configuration (config-dial-peer)

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.

Usage Guidelines

The system generates a random string when registering a new number. An INVITE message with the P-Called-Party-ID value can have the Request-URI set to this random number. To enable the system to identify the called number from the random number in the Request-URI, use the **voice-class sip random-request-uri validate** command on the inbound dial peer.

If the P-Called-Party-ID is not set in the INVITE message, the Request URI for that message must contain the called party information (and cannot contain a random number). Therefore validation is performed only on INVITE messages with a P-Called-Party-ID.

The **voice-class sip random-request-uri validate** command takes precedence over the **random-request-uri validate** command in voice service voip sip configuration mode. However, if the **voice-class sip random-request-uri validate** command is used with the **system** keyword, the gateway uses the settings configured globally by the **random-request-uri validate** command.

Examples

The following example shows how to enable call routing based on the P-Called-Party-ID header value at the dial-peer configuration level:

```
Router> enable
Router# configure
```

```
 terminal
Router(config)# dial-peer voice 2611 voip

Router(config-dial-peer)# voice-class sip random-request-uri validate
```

Related Commands

Command	Description
credentials (sip ua)	Sends a SIP registration message from a Cisco Unified Border Element in the UP state.
random-request-uri validate	Validates the called number based on the random value generated during the registration of the number at the global configuration level.
registrar	Enables 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 referto-passing

To disable the modification of the Refer-To header during REFER message pass-through on the Cisco Unified Border Element (UBE) on the specified dial peer, use the **voice-class sip referto-passing** command in dial peer voice configuration mode. To allow the modification of the Refer-To header during REFER message pass-through on the Cisco UBE, use the **no** form of this command.

voice-class sip referto-passing [**system**]

no voice-class sip referto-passing

Syntax Description

system	(Optional) Enables the **referto-passing** command configured in global configuration mode.

Command Default

The Refer-To header modification is enabled.

Command Modes

Dial peer voice configuration (config-dial-peer)

Command History

Release	Modification
15.2(1)T	This command was introduced.

Usage Guidelines

The dial peer configuration setting of the **voice-class sip referto-passing** command takes precedence over the global configuration setting of the **referto-passing** command. You can use the **system** keyword to toggle the precedence.

Examples

The following example shows how to enable REFER message pass-through on the Cisco UBE for dial peer 22:

```
Router(config)# dial-peer voice 22 voip
Router(config-dial-peer)# voice-class sip referto-passing
```

Related Commands

Command	Description
dial-peer voice	Defines a particular dial peer, specifies the method of encapsulation, and enters dial peer voice configuration mode.

Command	Description
referto-passing	Disables dial peer lookup and modification of the Refer-To header when the Cisco UBE passes across a REFER message during a call transfer

voice-class sip registration passthrough

To configure Session Initiation Protocol (SIP) registration pass-through options on a dial peer, use the **voice-class sip registration passthrough**command in dial peer voice configuration mode. To disable the configuration, use the **no** form of this command.

voice-class sip registration passthrough [[**static**] [**rate-limit** [**expires** *value*] [**fail-count** *value*]] [**registrar-index** [*index*]]| **system**]

no voice-class sip registration passthrough

Syntax Description

static	(Optional) Configures Cisco Unified Border Element (UBE) to use static registrar details for SIP registration. Cisco UBE works in point-to-point mode when the **static** keyword is used.
rate-limit	(Optional) Configures SIP registration pass-through rate-limiting options.
expires *value*	(Optional) Sets the expiry value for rate limiting, in seconds. The range is from 60 to 65535. The default is 3600.
fail-count *value*	(Optional) Sets the fail-count value for rate limiting. The range is from 2 to 20. The default is 0.
registrar-index	(Optional) Configures the registrar index used for registration pass-through.
index	(Optional) Registration index value. The range is from 1 to 6.
system	(Optional) Uses global registration pass-through configuration to configure the SIP registration pass-through options.

Command Default

SIP registration pass-through options that are configured at the global level are configured.

Command Modes

Dial peer voice configuration (config-dial-peer)

Command History

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

Usage Guidelines

You can use the **voice-class sip registration passthrough** command to configure the following SIP pass-through functionalities on a dial peer:

- Back-to-back registration facility to register phones for call routing.
- Options to configure the rate-limiting values, such as the expiry time, fail-count, and a list of registrars to be used for registration.

Examples

The following example shows how to set the registrar index of 1 for the SIP registration pass-through rate limiting:

```
Router# configure terminal
Router(config)# dial-peer voice 444 voip
Router(config-dial-peer)# voice-class sip registration passthrough static rate-limit
registrar-index 1
```

Related Commands

Command	Description
registration passthrough	Configures SIP registration pass-through options at the global level.

voice-class sip rel1xx

To enable all Session Initiation Protocol (SIP) provisional responses (other than 100 Trying) to be sent reliably to the remote SIP endpoint, use the **voice-class sip rel1xx** command in dial-peer configuration mode. To reset to the default, use the **no** form of this command.

voice-class sip rel1xx {**supported** *value*| **require** *value*| **system**| **disable**}

no sip rel1xx

Syntax Description

supported *value*	Supports reliable provisional responses. The *value* argument may have any value, as long as both the user-agent client (UAC) and user-agent server (UAS) configure it the same.
require *value*	Requires reliable provisional responses. The *value* argument may have any value, as long as both the UAC and UAS configure it the same.
system	Uses the value configured in voice service mode. This is the default.
disable	Disables the use of reliable provisional responses.

Command Default

system

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.2(2)XB	This command was introduced.
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 applicable to the Cisco AS5300, Cisco AS5350, and Cisco AS5400 in this release.

Usage Guidelines

There are two ways to configure reliable provisional responses:

- Dial-peer mode. You can configure reliable provisional responses for the specific dial peer only by using the **voice-class sip rel1xx** command.
- SIP mode. You can configure reliable provisional responses globally by using the **rel1xx**command.

The use of resource reservation with SIP requires that the reliable provisional feature for SIP be enabled either at the VoIP dial-peer level or globally on the router.

This command applies to the dial peer under which it is used or points to the global configuration for reliable provisional responses. If the command is used with the **supported** keyword, the SIP gateway uses the Supported header in outgoing SIP INVITE requests. If it is used with the **require** keyword, the gateway uses the Required header.

This command, in dial-peer configuration mode, takes precedence over the **rel1xx** command in global configuration mode with one exception: If this command is used with the system keyword, the gateway uses what was configured under the **rel1xx** command in global configuration mode.

Examples

The following example shows how to use this command on either an originating or a terminating SIP gateway:

- On an originating gateway, all outgoing SIP INVITE requests matching this dial peer contain the Supported header where *value* is 100rel.
- On a terminating gateway, all received SIP INVITE requests matching this dial peer support reliable provisional responses.

```
Router(config)# dial-peer voice 102 voip
Router(config-dial-peer)# voice-class sip rel1xx supported 100rel
```

Related Commands

Command	Description
rel1xx	Provides provisional responses for calls on all VoIP calls.

voice-class sip reset timer expires

To configure an individual dial peer on Cisco Unified Communications Manager Express (Cisco Unified CME), a Cisco IOS voice gateway, or a Cisco Unified Border Element (Cisco UBE) to reset the expires timer upon receipt of a Session Initiation Protocol (SIP) 183 Session In Progress message, use the **voice-class sip reset timer expires** command in dial peer voice configuration mode. To globally disable resetting of the expires timer upon receipt of SIP 183 messages, use the **no** form of this command.

voice-class sip reset timer expires 183

no voice-class sip reset timer expires 183

Syntax Description

183	Specifies resetting of the expires timer upon receipt of SIP 183 Session In Progress messages.

Command Default

The expires timer is not reset after receipt of SIP 183 Session In Progress messages and a session or call that is not connected within the default expiration time (three minutes) is dropped.

Command Modes

Dial peer voice configuration (config-dial-peer)

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

In some scenarios, early media cut-through calls (such as emergency calls) rely on SIP 183 with session description protocol (SDP) Session In Progress messages to keep the session or call alive until receiving a FINAL SIP 200 OK message, which indicates that the call is connected. In these scenarios, the call can time out and be dropped if it does not get connected within the default expiration time (three minutes).

Note The expires timer default is three minutes. However, you can configure the expiration time to a maximum of 30 minutes using the **timers expires** command in SIP user agent (UA) configuration mode.

To prevent early media cut-through calls from being dropped on a specific dial peer because they reach the expires timer limit, use the **voice-class sip reset timer expires** command in dial peer voice configuration mode.

To globally configure all dial peers on Cisco Unified CME, a Cisco IOS voice gateway, or a Cisco UBE so that the expires timer is reset upon receipt of any SIP 183 message, use the **reset timer expires** command in voice service SIP configuration mode. To disable resetting of the expires timer on receipt of SIP 183 messages for an individual dial peer, use the **no voice-class sip reset timer expires** command in dial peer voice configuration mode.

Examples

The following example shows how to configure dial peer 1 on Cisco Unified CME, a Cisco IOS voice gateway, or a Cisco UBE to reset the expires timer each time a SIP 183 message is received:

```
Router> enable
Router# configure
 terminal
Router(config)# dial-peer voice 1 voip
Router(config-dial-peer)# voice-class sip reset timer expires 183
```

Related Commands

Command	Description
reset timer expires	Globally configures Cisco Unified CME, a Cisco IOS voice gateway, or a Cisco UBE to reset the expires timer upon receipt of a SIP 183 message.
timers expires	Specifies how long a SIP INVITE request remains valid before it times out if no appropriate response is received for keeping the session alive.

voice-class sip resource priority dscp-profile

To apply a differentiated services code point (DSCP) profile to a dial peer, use the **voice-class sip resource priority dscp-profile** in dial peer voice configuration mode. To disable the configuration, use the **no** form of this command.

voice-class sip resource priority dscp-profile *tag*

no voice-class sip resource priority dscp-profile

Syntax Description

tag	DSCP profile group tag number. The range is from 1 to 10000.

Command Default

A DSCP profile is not applied.

Command Modes

Dial peer voice configuration (config-dial-peer)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

You can use the **voice-class sip resource priority dscp-profile** command to apply the DSCP profile that is configured using the **dscp media** command for a dial peer.

Examples

The following example shows how to configure a DSCP profile for a dial peer:

```
Router> enable
Router# configure terminal
Router(config)# dial-peer voice 4 voip
Router(config-dial-peer)# voice-class sip resource priority dscp-profile 1
```

Related Commands

Command	Description
dial-peer voice	Configures a dial peer and enters dial peer voice configuration mode.
dscp media	Specifies the RPH to DSCP mapping.

voice-class sip resource priority mode (dial-peer)

To push the user access server (UAS) to operate in a loose or strict mode, use the **voice-class sip resource priority mode** command in dial peer voice configuration mode. To disable the **voice-class sip resource priority mode**, use the **no** form of this command.

voice-class sip resource priority mode [**loose**| **strict**]

no voice-class sip resource priority mode [**loose**| **strict**]

Syntax Description

loose	(Optional) In the loose mode, unknown values of name space or priority values received in the Resource-Priority header in Session Initiation Protocol (SIP) requests are ignored by the gateway. The request is processed as if the Resource-Priority header was not present.
strict	(Optional) In the strict mode, unknown values of name space or priority values received in the Resource-Priority header in SIP requests are rejected by the gateway using a SIP response code 417 (Unknown Resource-Priority) message response. An Accept-Resource-Priority header enumerating the supported name space and values is included in the 417 message response.

Command Default

The default value is **loose mode**.

Command Modes

Dial peer voice configuration

Command History

Release	Modification
12.4(2)T	This command was introduced.

Usage Guidelines

When the no version of this command is executed, the call operates in the **loose** mode.

Examples

The following example shows how to set up the **voice-class sip resource priority mode** command in loose mode:

```
Router(config)# dial-peer voice 102 voip
Router(config-dial-peer)# voice-class sip resource priority mode loose
```

The following example shows how to set up the **voice-class sip resource priority mode** command in strict mode:

```
Router(config)# dial-peer voice 102 voip
Router(config-dial-peer)# voice-class sip resource priority mode strict
```

Related Commands

Command	Description
voice-class sip resource priority namespace	Priorities mandatory call prioritization handling for initial original INVITE message requests.

voice-class sip resource priority namespace (dial-peer)

To prioritize mandatory call prioritization handling for initial original INVITE message requests, use the **voice-class sip resource priority namespace** command in dial peer voice configuration mode. To disable the **voice-class sip resource priority namespace** command, use the **no** form of this command.

voice-class sip resource priority namespace [**drsn**| **dsn**| **q735**]

no voice-class sip resource priority namespace [**drsn**| **dsn**| **q735**]

Syntax Description

drsn	(Optional) U. S. Defense Red Switched Network (DRSN).
dsn	(Optional) U. S. Defense Switched Network (DSN).
q735	(Optional) International Telecommunications Union, *Stage 3 description for community of interest supplementary services using Signaling System No. 7: Multilevel precedence and preemption, Recommendation Q.735.3* , March 1993.

Command Default

When the no version of this command is executed using namespace, the Cisco IOS gateway transparently passes the multilevel precedence and preemption (MLPP) values that were received on the PSTN side.

Command Modes

Dial peer voice configuration

Command History

Release	Modification
12.4(2)T	This command was introduced.

Usage Guidelines

When the no version of this command is executed using the namespace, the Cisco IOS gateway transparently passes the multilevel precedence and preemption (MLPP) values that were received on the PSTN side.

Examples

The following example shows how to set up the **voice-class sip resource priority namespace** command in the U. S. DSN format name space:

```
Router(config)# dial-peer voice 102 voip
Router(config-dial-peer)# voice-class sip resource priority namespace dsn
```

The following example shows how to set up the **voice-class sip resource priority namespace** command in the U. S. DRSN format name space:

```
Router(config)# dial-peer voice 102 voip
Router(config-dial-peer)# voice-class sip resource priority namespace drsn
```

The following example shows how to set up the **voice-class sip resource priority namespace** command in the Public SS7 Network format name space:

```
Router(config)# dial-peer voice 102 voip
Router(config-dial-peer)# voice-class sip resource priority namespace q735
```

Related Commands

Command	Description
voice-class sip resource priority mode	Pushes the UAS to operate in a loose or strict mode.

voice-class sip rsvp-fail-policy

To specify the action that takes place at the dial peer level on a Cisco IOS Session Initiation Protocol (SIP) gateway when Resource Reservation Protocol (RSVP) negotiation fails, use the **voice-class sip rsvp-fail-policy** command in dial peer configuration mode. To reset failure behavior to the default settings, use the **no** form of this command.

voice-class sip rsvp-fail-policy {**video**| **voice**} **post-alert** {**optional keep-alive**| **mandatory** {**keep-alive**| **disconnect retry** *retry-attempts*}} **interval** *seconds*

no voice-class sip rsvp-fail-policy {**video**| **voice**} **post-alert** {**optional [keep-alive]**| **mandatory** [**keep-alive**| **disconnect retry** *retry-attempts*]} [**interval** *seconds*]

Syntax Description

video	Specifies the video RSVP stream type.
voice	Specifies the audio or fax RSVP stream type.
post-alert	Specifies that behavior takes place only when the call state is post alert.
optional	Specifies that behavior takes place when RSVP fails even if RSVP negotiation is optional.
mandatory	Specifies that behavior takes place when RSVP fails only if RSVP negotiation is mandatory.
keep-alive	Specifies the sending of keepalive messages when RSVP fails.
disconnect	Specifies that the call is disconnected if RSVP fails after the specified number of retry settings.
retry	Specifies the number of reconnection attempts before disconnecting the call.
retry-attempts	The number of retry attempts. Valid entries are from 1 to 100.
interval	Specifies the interval between keepalive or retry attempts.
seconds	The retry interval in seconds. Valid entries are from 5 to 3600.

Command Default

Keepalive messages are sent at 30-second intervals when a post alert voice or video call fails to negotiate RSVP regardless of the RSVP negotiation setting (mandatory or optional).

Command Modes

Dial peer configuration (config-dial-peer)

Command History

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

Usage Guidelines

Use this command to configure call handling behavior when a call fails RSVP negotiation. You can configure the behavior that takes place for either optional or mandatory RSVP negotiation but the behavior will apply only to calls in a post alert call state. To configure the behavior that takes place when RSVP negotiation fails, use the **voice-class sip rsvp-fail-policy** command in dial peer configuration mode.

If a call fails RSVP negotiation where negotiation is optional, then RSVP negotiation should be retried using the keepalive function at specified intervals until RSVP negotiation is successful.

If a call fails RSVP negotiation where negotiation is mandatory, then RSVP negotiation should be configured in one of two ways:

- The call that failed RSVP negotiation is disconnected after a specified number of attempts to renegotiate RSVP with each retry taking place at a specified interval. If negotiation succeeds during these retry attempts, counters and timers are reset to zero.
- The call that failed RSVP negotiation is kept alive with keepalive messages sent at specified intervals until negotiation is successful.

Examples

The following example shows how to specify sending of keepalive messages at 60-second intervals for a call that fails RSVP negotiation when negotiation is optional:

```
Router(config)# dial-peer voice 102 voip
Router(config-dial-peer)# voice-class sip rsvp-fail-policy voice post-alert optional
keep-alive interval 60
```

Related Commands

Command	Description
acc-qos	Defines the acceptable QoS for inbound and outbound calls on a VoIP dial peer.
handle-replaces	Configures fallback to legacy handling of SIP INVITE.
ip qos defending-priority	Configures the RSVP defending priority value.
ip qos dscp	Sets the DSCP value for QoS.
ip qos policy-locator	Configures application-specific reservations (application IDs) used for specifying bandwidth reservations.

Command	Description
ip qos preemption-priority	Configures the RSVP preemption priority value.
req-qos	Requests a particular QoS using RSVP to be used in reaching a specified dial peer in VoIP.
show-sip-ua calls	Displays the active UAC and UAS information on SIP calls.

voice-class sip send 180 sdp

To configure a Cisco Unified Border Element (Cisco UBE) to map an incoming 180 Session Description Protocol (SDP) message to a 180 SDP message, use the **voice-class sip send 180 sdp** command in dial peer voice configuration mode or SIP configuration mode. To disable this functionality, use the **no** form of this command.

voice-class sip send 180 sdp

novoice-class sip send 180 sdp

Syntax Description

This command has no arguments or keywords.

Command Default

This command is disabled. Cisco UBE converts an incoming 180 SDP message to a 183 SDP message.

Command Modes

Dial peer voice configuration (config-dialpeer)

SIP configuration (conf-serv-sip)

Command History

Release	Modification
15.2(4)M	This command was introduced.

Usage Guidelines

This command must be enabled at the inbound dial peer. Enable the **voice-class sip send 180 sdp** command to map a 180 SDP message to a 180 SDP message. When this command is disabled, an incoming 180 SDP (Ringing) message is mapped to a 183 SDP (Session in Progress) message.

Examples

The following example shows how to configure the **voice-class sip send 180 sdp** command at dial peer level:

```
Device> enable
Device# configure terminal
Device(config)# dial peer voice
Device(config-dialpeer)# voice-class sip send 180 sdp
Device(config-dialpeer)# exit
```

Related Commands

Command	Description
voice-class sip block	Configures an individual dial peer on a Cisco IOS voice gateway or Cisco UBE to drop (not pass) specific incoming Session Initiation Protocol (SIP) provisional response messages.

voice-class sip srtp negotiate

To enable Secure Real-Time Transport Protocol (SRTP) negotiation so that an individual dial peer on a Cisco IOS Session Initiation Protocol (SIP) gateway can accept and send an RTP Audio/Video Profile (AVP) in response to an RTP Secure AVP offer (also known as an SRTP profile), use the **voice-class sip srtp negotiate** command in dial peer voice configuration mode. To return to the default (global) SRTP negotiation setting on a dial peer, use the **system** keyword. To disable SRTP negotiation on a dial peer, use the **no** form of this command.

voice-class sip srtp negotiate {**cisco**| **system**}

no voice-class sip srtp negotiate

Syntax Description

cisco	Enables an individual dial peer on a Cisco IOS SIP gateway to negotiate the sending and accepting of RTP profiles in response to SRTP offers, overriding the global setting for the gateway.
system	Specifies that the individual dial peer use global (system) SRTP negotiation settings for the Cisco IOS SIP gateway. This is the default setting.

Command Default

SRTP negotiation is determined by global settings for the Cisco IOS gateway (**voice-class sip srtp negotiate system**).

Command Modes

Dial peer voice configuration (config-dial-peer)

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.
12.4(22)T	Support was extended to the Cisco Unified Border Element.

Usage Guidelines

The **srtp fallback** command enables a SIP gateway (or individual dial peer on a SIP gateway) to allow SRTP fallback using SIP 4*xx* message responses. With the **srtp negotiate** command, a SIP gateway can be configured to accept and send an RTP (nonsecure) profile in response to an SRTP profile.

Use the **voice-class sip srtp negotiate** command in dial peer voice configuration mode to enable SRTP negotiation for an individual dial peer on a Cisco IOS SIP gateway, overriding the global settings on the gateway. Enabling SRTP negotiation allows a dial peer to accept and send nonsecure RTP profiles in response to SRTP offers. To configure global SRTP negotiation settings for a SIP gateway, use the **srtp negotiate** command in voice service SIP configuration mode.

There are two scenarios for SRTP negotiation when the **voice-class sip srtp negotiate** command is enabled:

- On a SIP dial peer with the **srtp fallback** command enabled, the dial peer accepts RTP answers to SRTP offers.
- On a SIP dial peer with the **srtp fallback** command disabled, the dial peer allows incoming SRTP calls and responds with an RTP answer.

These behaviors are accomplished using the "X-cisco-srtp-fallback" extension in the supported header of initial SIP messages involved in establishment of the session.

Examples

The following example shows SRTP negotiation being enabled on a dial peer, overriding global settings:

```
Device(config)# dial-peer voice 1
Device(config-dial-peer)# voice-class sip srtp negotiate cisco
```

Related Commands

Command	Description
srtp (dial peer)	Specifies that an individual dial peer use SRTP to enable secure calls and, optionally, enables fallback to RTP (overriding global settings).
srtp (voice)	Specifies use of SRTP to enable secure calls and, optionally, enables fallback to RTP globally on a Cisco IOS SIP gateway.
srtp negotiate	Enables SRTP negotiation globally on a Cisco IOS SIP gateway.

voice-class sip tel-config to-hdr

To configure the To: Header (to hdr) request Uniform Resource Identifier (URI) to telephone (TEL) format for dial-peer VoIP Session Initiation Protocol (SIP) calls, use the **voice-class sip tel-config to-hdr**command in dial peer voice configuration mode. To reset to the default, use the **no** form of this command.

voice-class sip tel-config to-hdr {**phone-context**| **system**}

no voice-class sip tel-config to-hdr

Syntax Description

phone-context	Appends the phone context parameter to the TEL URL on a dial-peer basis.
system	Uses the system value. This is the default.

Command Default

The To: Header request URIs at the dial-peer level use the global configuration level settings.

Command Modes

Dial peer voice configuration (config-dial-peer)

Command History

Release	Modification
12.4(22)YB	This command was introduced.
12.4(24)T	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

The **voice-class sip tel-config to-hdr** command takes precedence over the **tel-config to-hdr** command configured in SIP configuration mode. However, if the **voice-class sip tel-config to-hdr** command is used with the **system** keyword, the gateway uses the global settings configured by the **tel-config to-hdr** command.

Examples

The following example configures the To: header in TEL format for a dial peer VoIP SIP call, and appends the phone-context parameter:

```
dial-peer voice 102 voip
 voice-class sip tel-config to-hdr phone-context
```

Related Commands

Command	Description
tel-config to-hdr	Configures the To: Header Request URI to telephone format for VoIP SIP calls.

voice-class sip transport switch

To enable switching between UDP and TCP transport mechanisms for large Session Initiation Protocol (SIP) messages for a specific dial peer, use the **voice-class sip transport switch**command in dial-peer configuration mode. To disable switching between UDP and TCP transport mechanisms for large SIP messages for a specific dial peer, use the **no** form of this command.

voice-class sip transport switch udp tcp

no voice-class sip transport switch udp tcp

Syntax Description

udp	Enables switching transport from UDP on the basis of the size of the SIP request being greater than the MTU size.
tcp	Enables switching transport to TCP.

Command Default

Disabled.

Command Modes

Dial-peer configuration

Command History

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

Usage Guidelines

The **voice-class sip transport switch**command takes precedence over the global **transport switch**command.

Examples

The following example shows how to set up the **voice-class sip transport switch**command:

```
Router(config)# dial-peer voice 102 voip
Router(config-dial-peer)# voice-class sip transport switch udp tcp
```

Related Commands

Command	Description
debug ccsip transport	Enables tracing of the SIP transport handler and the TCP or UDP process.
transport switch	Enables switching between transport mechanisms globally if the SIP message is larger than 1300 bytes.

voice-class sip url

To configure URLs to either the Session Initiation Protocol (SIP), SIP security (SIPS), or telephone (TEL) format for your dial-peer SIP calls, use the **voice-class sip url** command in dial peer voice configuration mode. To reset to the default value use the **no** form of this command.

voice-class sip url {**sip| sips| tel [phone-context]|| system**}

no voice-class sip url

Syntax Description

sip	Generates URLs in the SIP format for calls on a dial-peer basis.
sips	Generates URLs in the SIPS format for calls on a dial-peer basis.
tel	Generates URLs in the TEL format for calls on a dial-peer basis.
phone-context	(Optional) Appends the phone context parameter to the TEL URL on a dial-peer basis.
system	Uses the system value. This is the default.

Command Default

SIP calls at the dial-peer level use the global configuration level settings.

Command Modes

Dial peer voice configuration (config-dial-peer)

Command History

Release	Modification
12.2(2)XB	This command was introduced.
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 was not included in this release.
12.2(11)T	This command was implemented on the Cisco AS5300, Cisco AS5350, and Cisco AS5400 platforms.
12.4(6)T	The **sips** keyword was added.
12.4(22)YB	The **phone-context** keyword was added.

Release	Modification
15.0(1)M	This command was integrated into Cisco IOS Release 15.0(1)M.

Usage Guidelines

This command affects only user-agent clients (UACs), because it causes the use of a SIP, SIPS, or TEL URL in the request line of outgoing SIP INVITE requests. SIP URLs indicate the originator, recipient, and destination of the SIP request; TEL URLs indicate voice-call connections.

The **voice-class sip url** command takes precedence over the **url** command configured in SIP configuration mode. However, if the **voice-class sip url** command is used with the **system** keyword, the gateway uses what was globally configured with the **url** command.

Examples

The following example shows how to configure the **voice-class sip url**command to generate URLs in the SIP format:

```
dial-peer voice 102 voip
 voice-class sip url sip
```

The following example shows how to configure the **voice-class sip url**command to generate URLs in the SIPS format:

```
dial-peer voice 102 voip
 voice-class sip url sips
```

The following example shows how to configure the **voice-class sip url**command to generate URLs in the TEL format:

```
dial-peer voice 102 voip
 voice-class sip url tel
```

The following example shows how to configure the **voice-class sip url**command to generate URLs in the TEL format, and append the phone-context parameter:

```
dial-peer voice 102 voip
 voice-class sip url tel phone-context
```

Related Commands

Command	Description
sip url	Generates URLs in the SIP, SIPS, or TEL format.
url	Configures URLs to either SIP, SIPS, or TEL format.

voice-class source interface

To allow a loopback interface to be associated with a VoIP or VoIPv6 dial-peer profile, use the **voice-class source interface** command in dial peer configuration mode. To disable this association, use the **no** form of this command.

voice-class source interface loopback *interface-id* [*ipv4-address*| *ipv6-address*]

no voice-class source interface loopback *interface-id* [*ipv4-address*| *ipv6-address*]

Syntax Description

loopback	Specifies the loopback interface address.
interface-id	Specifies the interface on which the address is to be configured.
ipv4-address	(Optional) IPv4 address used in the loopback interface address.
ipv6-address	(Optional) IPv6 address used in the loopback interface address.

Command Default

No loopback interface is associated with a VoIPv6 dial-peer profile.

Command Modes

Dial peer configuration (config-dial-peer)

Command History

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

Usage Guidelines

When the **voice-class source interface**command is configured, the source address of Routing Table Protocol (RTP) generated by the gateway is taken from the address configured under the loopback interface. This command is used for policy-based routing (PBR) of voice packets originated by the gateway. The policy route map is configured under the loopback interface, and then the loopback interface is specified under the VoIP or VoIPv6 dial peer.

Examples

The following example associates a loopback interface with a VoIPv6 dial-peer profile:

```
Router(config)# dial-peer voice 1 voip
Router (config-dial-peer)# voice-class source interface loopback0
```

Related Commands

Command	Description
dial-peer voice	Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial peer configuration mode.

voice-class stun-usage

To configure voice class, enter voice class configuration mode called stun-usage and use the **voice-class stun-usage** command in global, dial-peer, ephone, ephone template, voice register pool, or voice register pool template configuration mode. To disable the voice class, use the **no** form of this command.

voice-class stun-usage *tag*

no voice-class stun-usage *tag*

Syntax Description

tag	Unique identifier in the range 1 to 10000.

Command Default

The voice class is not defined.

Command Modes

Global configuration (config) Dial peer configuration (config-dial-peer) Ephone configuration (config-ephone) Ephone template configuration (config-ephone-template) Voice register pool configuration (config-register-pool) Voice register pool template configuration (config-register-pool)

Command History

Release	Cisco Product	Modification
12.4(22)T	Cisco Unified CME 7.0	This command was introduced.
15.1(2)T	Cisco Unified CME 8.1	This command was modified. This command can be enabled in ephone summary, ephone template, voice register pool, or voice register pool template configuration mode.

Usage Guidelines

When the voice-class stun-usage is removed, the same is removed automatically from the dial-peer, ephone, ephone template, voice register pool, or voice register pool template configurations.

Examples

The following example shows how to set the **voice class stun-usage**tag to 10000:

```
Router(config)# voice class stun-usage 10000
Router(config-ephone)# voice class stun-usage 10000
Router(config-voice-register-pool)# voice class stun-usage 10000
```

Related Commands

Command	Description
stun usage firewall-traversal flowdata	Enables firewall traversal using STUN.

Command	Description
stun flowdata agent-id	Configures the agent ID.

voice-class tone-signal

To assign a previously configured tone-signal voice class to a voice port, use the **voice-class tone-signal** command in voice-port configuration mode. To delete a tone-signal voice class, use the **no** form of this command.

voice-class tone-signal *tag*

no voice-class tone-signal *tag*

Syntax Description

tag	Unique label assigned to the voice class. The *tag* label maps to the tag label created using the **voice class tone-signal** global configuration command. Can be up to 32 alphanumeric characters.

Command Default

Voice ports have no tone-signal voice class assigned.

Command Modes

Voice-port 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

The **voice-class tone-signal** command is available on an ear and mouth (E&M) voice port only if the signal type for that port is Land Mobile Radio (LMR). Note that the hyphenation in this command differs from the hyphenation used in a similar command, **voice class tone-signal**, which is used in global configuration mode.

Examples

The following example assigns a previously configured voice class to voice port 1/1/0:

```
voice-port 1/0/0
 voice-class tone-signal mytones
```

Related Commands

Command	Description
voice class tone-signal	Enters voice-class configuration mode and assigns an identification tag number for a tone-signal voice class.

voice confirmation-tone

To disable the two-beep confirmation tone for private line, automatic ringdown (PLAR), or PLAR off-premises extension (OPX) connections, use the **voice confirmation-tone** command in voice-port configuration mode. To enable the two-beep confirmation tone, use the **no** form of this command.

voice confirmation-tone

no voice confirmation-tone

Syntax Description

This command has no arguments or keywords.

Command Default

The two-beep confirmation tone is heard on PLAR and PLAR OPX connections.

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

Use this command to disable the two-beep confirmation tone that a caller hears when picking up the handset for PLAR and PLAR OPX connections. This command is valid only if the voice-port **connection** command is set to PLAR or PLAR OPX.

Examples

The following example disables the two-beep confirmation tone on voice port 1/0/0:

```
voice-port 1/0/0
 connection plar-opx
 voice confirmation-tone
```

Related Commands

Command	Description
connection	Specifies a connection mode for a voice port.

voice dnis-map

To create or modify a Digital Number Identification Service (DNIS) map, use the **voice dnis-map** command in global configuration mode. To delete a DNIS map, use the **no** form of this command.

voice dnis-map *map-name* [*url*]

no voice dnis-map map-name

Syntax Description

map-name	Name of the DNIS map.
url	(Optional) URL of an external text file that contains a list of DNIS entries.

Command Default

No default behavior or values

Command Modes

Global configuration

Command History

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

Usage Guidelines

A DNIS map is a table of DNIS numbers associated with a single dial peer. For applications such as VoiceXML, using a DNIS map makes it possible to configure a single dial peer for all DNIS numbers used to refer to VoiceXML documents. Keep the following considerations in mind when using voice DNIS maps.

- A separate entry must be made for each DNIS entry in a DNIS map. Wildcards are not supported.
- If a URL is not supplied, the command enters DNIS-map configuration mode, permitting the entry of DNIS numbers by using the **dnis** command.
- The URL argument points to the location of an external text file containing a list of DNIS entries **(**forexample: tftp://dnismap.txt). This allows the administrator to maintain a single master file of all DNIS map entries, if desired, rather than configuring the DNIS entries on each gateway.

The name of the text file extension is not significant; .doc, .txt, or .cfg are all acceptable because the extension is not checked. The entries in the file should look the same as a DNIS entry configured in Cisco IOS software (for example: dnis 5553305 url tftp://global/tickets/movies.vxml).

- External text files used for DNIS maps must be stored on TFTP servers; they cannot be stored on HTTP servers.
- To associate a DNIS map with a dial peer, use the **dnis-map** command.
- To view the configuration information for DNIS maps, use the **show voice dnis-map** command.

Examples

The following example shows how the voice dnis-map command is used to create a DNIS map:

```
voice dnis-map dmap1
```

The following example shows the voice dnis-map command used with a URL that specifies the location of a text file containing the DNIS entries:

```
voice dnis-map dmap2 tftp://keyer/dmap2/dmap2.txt
```

Following is an example of the contents of a text file comprising a DNIS map:

```
!Example dnis-map with 8 entries.
!
dnis 5550112 url tftp://global/ticket/vapptest1.vxml
dnis 5550111 url tftp://global/ticket/vapptest2.vxml
dnis 5550134 url tftp://global/ticket/vapptest3.vxml
dnis 5550178
dnis 5550100
dnis 5550101
dnis 5550102
dnis 5550103
```

Related Commands

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

voice dnis-map load

To reload a DNIS map that has been modified, use the **voice dnis-map load** command in privileged EXEC mode. This command does not have a **no** form.

voice dnis-map load *map-name*

Syntax Description

map-name	Name of the DNIS map to reload.

Command Default

No default behavior or values

Command Modes

Privileged EXEC

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.

Usage Guidelines

This command reloads a DNIS map residing on an external server. Use this command when the DNIS map file has changed since the previous load.

To create or modify a DNIS map, use the **voice dnis-map** command.

Examples

The following example reloads a DNIS map named "mapfile1":

```
Router# voice dnis-map load mapfile1
```

Related Commands

Command	Description
dnis	Adds a DNIS number to a DNIS map.
dnis-map	Associates a DNIS map with a dial peer.
show voice dnis-map	Displays configuration information about DNIS maps.

Command	Description
voice dnis-map	Enters DNIS map configuration mode to create a DNIS map.

voice dsp crash-dump

To enable the crash dump feature and to specify the destination file and the file limit, enter the **voice dsp crash-dump**command in global configuration mode. To disable the feature, use the **no** form of the command.

voice dsp crash-dump [**destination** *url*| **file-limit** *limit-number*]

no voice dsp crash-dump

Syntax Description

destination *url*	Designates a valid file system where crash dump analysis is stored. The *url* argument must be set to a valid file system. The destination url can be one of the following • The file on a TFTP server with the following format: tftp://x.x.x.x/subfolder/filename. The x.x.x.x value is the IP address of the TFTP server • The file on the flashcard of the router, with the following format: slot0:filename **Note** The digital signal processor (DSP) crash dump feature is disabled when either the crash-dump destination is not specified.
file-limit *limit-number*	The crash dump file-limit keyword must be set to a non-zero value. The default is that the crash dump capability is turned off, as the url argument is empty, and the file-number argument is zero. The limit-number argument may range from 0 (no file will be written) to 99. **Note** The DSP crash dump feature is disabled when the crash-dump file limit is set to 0.

Command Default

Crash dump capability is turned off.

Command Modes

Global configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Usage Guidelines

To configure the router to write a crash dump file, the destination url in the **voice dsp crash-dump** command must be set to a valid file system, and the crash dump file limit must be set to a non-zero value. The default is that the crash dump capability is turned off, as the url field is empty, and the file limit is zero.

As each crash-dump file is created, the name of the file has a number appended to the end. This number is incremented from 1 to up to the file limit for each subsequent crash dump file written. If the router reloads, the number is reset back to 1, and so file number 1 is written again. After the file number reaches the maximum file limit, no more files are written.

The file count can be manually reset by setting the file limit to zero and then setting it to a non-zero limit. This has the effect of restarting the count of files written, causing the files 1 to the file limit of 99 to be able to be written again, thus overwriting the original files.

Setting the file-number argument to zero (the default) disables the collection of the dump from the DSP. In this case, the memory is not collected from the DSP, and the stack is not displayed on the console. If the keepalive mechanism detects a crashed DSP, the DSP is simply restarted.

Setting the file-number argument to a non-zero number but having a null destination url causes the dump to be collected and the stack to be displayed on the console, but no dump file is written.

If auto-recovery is turned off for the router, no DSP dump functions are enabled, no keepalive checks are done, and no dumps are collected or written.

Note Some types of flash need to be completely erased to free up space from deleted files, and some types of flash cannot have files overwritten with new versions until the entire flash is erased. As a result, you might want to set the file limit so that only one or two dump files are written to flash. This prevents flash from being filled up.

Note It is not recommended to write crash dump files to internal flash or bootflash, because these files are normally used to hold configuration information and Cisco IOS software images. Cisco recommends writing crash dump files to spare flash cards, which can be inserted into slot 0 or slot 1 on many of the routers. These cards usually do not hold critical information and may be erased. Additionally, these cards can be conveniently removed from the router and sent to Cisco, so that the crash dump files can be analyzed.

Examples

The following example enables the crash dump feature and specifies the destination file in slot 0:

```
Router configure terminal
Enter configuration commands, one per line.  End with CNTL/Z.
Router(config)# voice dsp crash-dump destination slot0:banjo-152-s
Router# end
1w0d:%SYS-5-CONFIG_I:Configured from console by console
```

Check your configuration by entering the show voice dsp crash-dump command in privileged EXEC configuration mode:

```
Router# show voice dsp crash-dump
Voice DSP Crash-dump status:
    Destination file url is slot0:banjo-152-s
File limit is 20
    Last DSP dump file written was
```

```
      tftp://112.29.248.12/tester/26-152-t2
Next DSP dump file written will be slot0:banjo-152-s1
```

Related Commands

Command	Description
debug voice dsp crash-dump	Displays crash dump debug information.
show voice dsp crash-dump	Displays voice dsp crash dump information.

voice echo-canceller extended

To enable the extended G.168 echo canceller (EC) on the Cisco 1700 series, Cisco ICS7750, or Cisco AS5300, use the **voice echo-canceller extended**command in global configuration mode. To reset to the default, use the **no** form of this command.

Cisco 1700 series and Cisco ICS 7750

voice echo-canceller extended

no voice echo-canceller extended

Cisco AS5300

voice echo-canceller extended [**codec small** *codec* **large** *codec*]

no voice echo-canceller extended

Syntax Description

codec	(Optional) Defines restricted codecs, both small and large.
small *codec*	Small footprint codec. Valid values for the *codec* argument are: • **g711** • **g726**
large *codec*	Large footprint codec. Valid values for the *codec* argument are: • **fax-relay** • **g723** • **g728** • **g729** • **gsmefr** • **gsmfr**

Command Default

Proprietary Cisco G.165 EC is enabled.

Command Modes

Global configuration

Command History

Release	Modification
12.2(13)T	This command was introduced.
12.3(3)	This command was modified to allow unrestricted codecs on the Cisco AS5300. The **codec** keyword was made optional.

Usage Guidelines

Cisco 1700 series and Cisco ICS7750

You do not have to shut down all the voice ports on the Cisco 1700 series or Cisco ICS7750 to switch the echo canceller, but you should make sure that when you switch the echo canceller, there are no active calls on the router.

Because echo cancellation is an invasive process that can minimally degrade voice quality, you should disable this command if it is not needed.

Cisco AS5300

This command is available only on the Cisco AS5300 with C542 or C549 digital signal processor module (DSPM) high-complexity firmware.

The **voice echo-canceller extended** command enables the extended EC on a Cisco AS5300 using C549 DSP firmware with one channel of voice per DSP and unrestricted codecs. Any codec is supported.

The **voice echo-canceller extended codec** command enables the extended EC on a Cisco AS5300 using C542 or C549 DSP firmware with two channels of voice per DSP and restricted codecs. Only specific codecs can be used with the extended EC.

If fax-relay is not selected as the large codec, the VoIP dial peer requires that you use the fax rate disabled command in dial-peer configuration mode.

After choosing the codecs to be supported by the extended echo canceller, either remove all dial peers with different codecs not supported by your new configuration or modify the dial-peer codec selection by selecting a voice codec or fax-relay. When codecs are restricted, only one selection is allowed. You must have a VoIP dial peer configured with an extended EC-compatible codec to ensure voice quality on the connection.

This command is not accepted if there are active calls. If the EC is already in effect and a codec choice is changed, the system scans the dial peers. Any dial peers that do not conform to the new global command settings are changed, and the user is informed of the changes. Similarly, modem relay is incompatible with the extended EC and must be disabled globally for all dial peers.

Note This command is valid only when the **echo-cancel enable** command and the echo-cancel coveragecommand are enabled.

Examples

The following example sets the extended G.168 EC on the Cisco 1700 series or Cisco ICS7750:

```
Router(config)# voice echo-canceller extended
```

The following example sets the extended G.168 EC on the Cisco AS5300 with restricted codecs:

```
Router(config)# voice echo-canceller extended codec small g711 large g726
```

The following example shows an error message that displays when a restricted codec is not allowed:

```
Cannot configure now, dial-peer 8800 is configured with codec=g728, fax rate=disable,
modem=passthrough system.If necessary set this command to 'no', re-configure dial-peer
codec, fax rate and/or modem. Then re-enter this command.
```

In the above example, dial peer 8800 is misconfigured with a codec type, g728, that was not selected for the large codec type using the **voice echo-canceller extended** command.

Related Commands

Command	Description
echo-cancel coverage	Enables the cancellation of voice that is sent out the interface and is received on the same interface.
echo-cancel enable	Enables the cancellation of voice that is sent and received on the same interface.

voice enum-match-table

To create an ENUM match table for voice calls, use the **voice enum-match-table** in global configuration mode. To delete the ENUM match table, use the **no**form of this command.

voice enum-match-table *table-number*

no voice enum-match-table *table-number*

Syntax Description

table-number	Number of the ENUM match table. Range is from 1 to 15. There is no default value.

Command Default

No default behavior or values

Command Modes

Global configuration

Command History

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

Usage Guidelines

The ENUM match table is a set of rules for matching incoming calls. When a call comes in, its called number is matched against the match pattern of the rule with the highest preference.

If it matches, the replacement pattern is applied to the number. The resulting number and the domain name of the rule are used to make an ENUM query.

If the called number does not match the match pattern, the next rule in order of preference is selected.

Examples

The following example creates ENUM match table 3 for voice calls:

```
Router(config)# voice enum-match-table 3
Router(config-enum)# rule 1 5/(.*)/ /\1/e164.cisco.com
Router(config-enum)# rule 2 4/^9011\(.*\)/ /\1/e164.arpa
```

In this table, rule 1 matches any number. The resulting number is the same as the called number. That number and the domain name "e164.cisco.com" are used to make an ENUM query.

Rule 2 matches any number that starts with 9011. The 9011 is removed from the incoming number. The resulting number and the domain name "e164.arpa" are used for the ENUM query.

Suppose an incoming call has a called number of 4085550112. [Rule 2 is applied] first because it has a higher preference. The first few digits, 4085, do not match the 9011 pattern of rule 2, so [rule 1 is applied] next. The called number matches rule 1, and the resulting number is 4085550112. This number and "e164.cisco.com" form the ENUM query (2.1.2.1.5.5.5.8.0.4.e164.cisco.com).

Related Commands

Command	Description
rule (ENUM configuration)	Defines the matching, replacement, and rejection patterns for an ENUM match table.
show voice enum-match-table	Displays the configuration of voice ENUM match tables.
test enum	Tests the functionality of an ENUM match table.

voice hpi capture

To allocate the Host Port Interface (HPI) capture buffer size (in bytes) and to set up or change the destination URL for captured data, use the **voice hpi capture** command in global configuration mode. To stop all logging and file operations, to disable data transport from the capture buffer, and to automatically set the buffer size to 328, use the **no** form of this command.

voice hpi capture [**buffer** *size*| **destination** *url*]

no voice hpi capture buffer *size*

Syntax Description

buffer *size*	(Optional) Size of HPI capture buffer, in bytes. Range is from 328 to 9000000. The default is 328.
destination *url*	(Optional) Destination URL for storing captured data.

Command Default

328 bytes (no buffer is used if it is not configured explicitly)

Command Modes

Global configuration

Command History

Release	Modification
12.2(10)	This command was introduced.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

If you want to change the size of an existing non-zero buffer, you must first reset it to 0 and then change it from 0 to the new size.

The **destination***url* option sets up or changes the destination URL for captured data. To disable data transport from the capture buffer, use the **no** form of the command. If the buffer is allocated, captured data is sent to the current URL (if it was already configured) until the new URL is specified.

If a new URL differs from the current URL and logging is enabled, the current URL is closed and all further data is sent to the new URL. Entering a blank URL or prefixing the command with **no** disables data transport from the capture buffer, and (if capture is enabled) captured data is stored in the capture buffer until it reaches its capacity.

Once the buffer-queueing program is running, the transport process attempts to connect to a new or existing "capture destination" URL. A version message is written to the URL, and if the message is successfully received, any further messages placed into the message queue are written to that URL. If a new URL is entered using the voice hpi capture destination url command, the open URL is closed, and the system attempts to write to the new URL. If the new URL does not work, the transport process exits. The transport process is restarted when another URL is entered or the system is restarted.

The buffer size option sets the maximum amount of memory (in bytes) that the capture system allocates for its buffers when it is active. The capture buffer is where the captured messages are stored before they are sent to the URL specified by the capture destination. The system is started by choosing the amount of memory (greater than 0 bytes) that the buffer-queueing system can allocate to the free message pool. HPI messages can then be captured until buffer capacity is reached. Entering **0** for the buffer size and prefixing the command with **no** stops all logging and file operations and automatically sets the buffer size to 0.

The **voice hpi capture** command can be saved with the router configuration so that the command is active during router startup. This allows you to capture the HPI messages sent during router bootup before the CLI is enabled. After you have configured the buffer size in the running configuration (valid range is from 328 to 9000000), save it to the startup configuration using the **write** command or to the TFTP server using the **copy run tftp** command.

Caution Using the message logger feature in a production network environment impacts CPU and memory usage on the gateway.

Examples

The following example changes the size (in bytes) of the HPI capture buffer and initializes the buffer-queueing program:

```
Router# configure terminal
Enter configuration commands, one per line.  End with CNTL/Z.
Router(config)# voice hpi capture buffer 40000
Router(config)# end
Router#
03:23:31:caplog:caplog_cli_interface:hpi capture buffer size set to 40000 bytes
03:23:31:caplog:caplog_logger_init:TRUE, Started task HPI Logger (PID 64)
03:23:31:caplog:caplog_cache_init:TRUE, malloc_named(39852), 123 elements (each 324 bytes
big)
03:23:31:caplog:caplog_logger_proc:Attempting to open ftp://172.23.184.233/c:b-38-117
03:23:32:%SYS-5-CONFIG_I:Configured from console by console
Router#
```

The following example sets the capture destination by entering a destination URL using FTP:

```
Router# configure terminal
Enter configuration commands, one per line.  End with CNTL/Z.
Router(config)# voice hpi capture destination ftp://172.23.184.233/c:b-38-117a
Router(config)#
04:05:10:caplog:caplog_cli_interface:hpi capture destination:ftp://172.23.184.233/c:b-38-117a
04:05:10:caplog:caplog_logger_init:TRUE, Started task HPI Logger (PID 19)
04:05:10:caplog:caplog_cache_init:Cache must be at least 324 bytes
04:05:10:caplog:caplog_logger_proc:Terminating...
Router(config)# end
Router#
```

Related Commands

Command	Description
debug hpi	Turns on the debug output for the logger.
show voice hpi capture	Displays the capture status and statistics.

voice hunt

To configure an originating or tandem router so that it continues dial-peer hunting if it receives a specified disconnect cause code from a destination router, use the **voice hunt**command in global configuration mode. To configure the router so that it stops dial-peer hunting if it receives a specified disconnect cause code (the default condition), use the **no** form of this command. To restore the default dial-peer hunt setting, use the **default** form of this command.

voice hunt {*disconnect-cause-code*| **all**}

no voice hunt {*disconnect-cause-code*| **all**}

default voice hunt

Syntax Description

disconnect-cause-code	A code returned from the destination router to indicate why an attempted end-to-end call was unsuccessful. If the specified disconnect cause code is returned from the last destination endpoint, dial peer hunting is enabled or disabled. The table below in the "Usage Guidelines" section describes the possible values. You can enter the keyword, decimal value, or hexadecimal value.
all	Continue dial-peer hunting for all disconnect cause codes returned from the destination endpoint.
default	Restores the default dial-peer hunt setting, that is, the router stops dial-peer hunting if it receives the user-busy or no-answer disconnect cause code.

Command Default

The router stops dial-peer hunting if it receives the user-busy or no-answer disconnect cause code.

Command Modes

Global configuration

Command History

Release	Modification
12.0(5)T	This command was introduced for VoFR on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810. It was also introduced for VoIP on the Cisco 2600 series and Cisco 3600 series.
12.0(7)T	This command was implemented for VoIP on the Cisco AS5300 and Cisco AS5800.
12.0(7)XK	This command was implemented for VoIP on the Cisco MC3810.

Release	Modification
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T and implemented for VoIP on the Cisco MC3810.
12.1(3)XI	The **invalid-number**and **unassigned-number** keywords were added, and the command name was changed to **voice hunt**.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.2(4)T	Keywords were added for more disconnect cause codes.
12.3(8)T	The *disconnect-cause-code* argument was modified to accept nonstandard disconnect cause codes.

Usage Guidelines

This command is used with routers that act as originating or tandem nodes in a VoIP, VoFR, or Voice over ATM environment.

For an outgoing call from an originating VoIP gateway configured for rotary dial-peer hunting, more than one dial peer may match the same destination number. The matching dial peers may have different routes. After the voice call using the first dial peer gets disconnected, it will return a disconnect cause code. To have the router to pick up the next matching dial peer in the rotary group and set up a call, the router must be configure to continue hunting the various routes. Use this command to configure the router's hunting behavior when specified cause codes are received.

You can use this command to enable and disable dial-peer hunting when nonstandard disconnect cause codes are received. Nonstandard disconnect cause codes are those that are not defined in ITU-T Recommendation Q.931, but are used by service providers. When this command is used to disable dial-peer hunting for a specific disconnect cause code, it appears in the running configuration of the router.

The disconnect cause codes are described in the table below. The decimal and hexadecimal value of the disconnect cause code follows the description of each possible keyword.

Table 3: Standard Disconnect Cause Codes

Keyword	Description	Decimal	Hex
access-info-discard	Access information discarded.	43	0x2b
all	Continue dial-peer hunting for all disconnect cause codes received from a destination router.		
b-cap-not-implemented	Bearer capability not implemented.	65	0x41
b-cap-restrict	Restricted digital information bearer capability only.	70	0x46

Keyword	Description	Decimal	Hex
b-cap-unauthorized	Bearer capability not authorized.	57	0x39
b-cap-unavail	Bearer capability not available.	58	0x3a
call-awarded	Call awarded.	7	0x7
call-cid-in-use	Call exists, call ID in use.	83	0x53
call-clear	Call cleared.	86	0x56
call-reject	Call rejected.	21	0x15
cell-rate-unavail	Cell rate not available.	37	0x25
channel-unacceptable	Channel unacceptable.	6	0x6
chantype-not-implement	Channel type not implemented.	66	0x42
cid-in-use	Call ID in use.	84	0x54
codec-incompatible	Codec incompatible.	171	0xab
cug-incalls-bar	Closed user group (CUG) incoming calls barred.	55	0x37
cug-outcalls-bar	CUG outgoing calls barred.	53	0x35
dest-incompatible	Destination incompatible.	88	0x58
dest-out-of-order	Destination out of order.	27	0x1b
dest-unroutable	No route to destination.	3	0x3
dsp-error	Digital signal processor (DSP) error.	172	0xac
dtl-trans-not-node-id	Designated transit list (DTL) transit not my node ID.	160	0xa0
facility-not-implemented	Facility not implemented.	69	0x45
facility-not-subscribed	Facility not subscribed.	50	0x32

Keyword	Description	Decimal	Hex
facility-reject	Facility rejected.	29	0x1d
glare	Glare.	15	0xf
glaring-switch-pri	Glaring switch PRI.	180	0xb4
htspm-oos	Holst Telephony Service Provider Module (HTSPM) out of service.	129	0x81
ie-missing	Mandatory information element missing.	96	0x60
ie-not-implemented	Information element not implemented.	99	0x63
info-class-inconsistent	Inconsistency in information and class.	62	0x3e
interworking	Interworking.	127	0x7f
invalid-call-ref	Invalid call reference value.	81	0x51
invalid-ie	Invalid information element contents.	100	0x64
invalid-msg	Invalid message.	95	0x5f
invalid-number	Invalid number.	28	0x1c
invalid-transit-net	Invalid transit network.	91	0x5b
misdialled-trunk-prefix	Misdialed trunk prefix.	5	0x5
msg-incomp-call-state	Message in incomplete call state.	101	0x65
msg-not-implemented	Message type not implemented.	97	0x61
msgtype-incompatible	Message type not compatible.	98	0x62
net-out-of-order	Network out of order.	38	0x26
next-node-unreachable	Next node unreachable.	128	0x80
no-answer	No user answer.	19	0x13

Keyword	Description	Decimal	Hex
no-call-suspend	No call suspended.	85	0x55
no-channel	Channel does not exist.	82	0x52
no-circuit	No circuit.	34	0x22
no-cug	Nonexistent CUG.	90	0x5a
no-dsp-channel	No DSP channel.	170	0xaa
no-req-circuit	No requested circuit.	44	0x2c
no-resource	No resource.	47	0x2f
no-response	No user response.	18	0x12
no-voice-resources	No voice resources available.	126	0x7e
non-select-user-clear	Nonselected user clearing.	26	0x1a
normal-call-clear	Normal call clearing.	16	0x10
normal-unspecified	Normal, unspecified.	31	0x1f
not-in-cug	User not in CUG.	87	0x57
number-changeed	Number changed.	22	0x16
param-not-implemented	Nonimplemented parameter passed on.	103	0x67
perm-frame-mode-oos	Permanent frame mode out of service.	39	0x27
perm-frame-mode-oper	Permanent frame mode operational.	40	0x28
precedence-call-block	Precedence call blocked.	46	0x2e
preempt	Preemption.	8	0x8
preempt-reserved	Preemption reserved.	9	0x9
protocol-error	Protocol error.	111	0x6f
qos-unavail	QoS unavailable.	49	0x31

Keyword	Description	Decimal	Hex
rec-timer-exp	Recovery on timer expiry.	102	0x66
redirect-to-new-destination	Redirect to new destination.	23	0x17
req-vpci-vci-unavail	Requested VPCI VCI not available.	35	0x23
send-infotone	Send information tone.	4	0x4
serv-not-implemented	Service not implemented.	79	0x4f
serv/opt-unavail-unspecified	Service or option not available, unspecified.	63	0x3f
stat-enquiry-resp	Response to status enquiry.	30	0x1e
subscriber-absent	Subscriber absent.	20	0x14
switch-congestion	Switch congestion.	42	0x2a
temp-fail	Temporary failure.	41	0x29
transit-net-unroutable	No route to transit network.	2	0x2
unassigned-number	Unassigned number.	1	0x1
unknown-param-msg-discard	Unrecognized parameter message discarded.	110	0x6e
unsupported-aal-parms	ATM adaptation layer (AAL) parameters not supported.	93	0x5d
user-busy	User busy.	17	0x11
vpci-vci-assign-fail	Virtual path connection identifier virtual channel identifier (VPCI VCI) assignment failure.	36	0x24
vpci-vci-unavail	No VPCI VCI available.	45	0x2d

Examples

The following example configures the originating or tandem router to continue dial-peer hunting if it receives a user-busy disconnect cause code from a destination router:

```
voice hunt user-busy
```

The following example configures the originating or tandem router to continue dial-peer hunting if it receives an invalid-number disconnect cause code from a destination router:

```
voice hunt 28
```

The following example configures the originating or tandem router to continue dial-peer hunting if it receives a facility-not-subscribed disconnect cause code from a destination router:

```
voice hunt 0x32
```

Related Commands

Command	Description
huntstop	Disables all further dial-peer hunting if a call fails when using hunt groups.
preference	Indicates the preferred order of a dial peer within a rotary hunt group.

voice iec syslog

To enable viewing of Internal Error Codes as they are encountered in real time, use the voice iec syslog command in global configuration mode. To disable IEC syslog messages, use the **no** form of this command.

voice iec syslog

no voice iec syslog

Syntax Description

This command has no arguments or keywords.

Command Default

IEC syslog messages are disabled.

Command Modes

Global configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Examples

The following example enables IEC syslog messages:

```
Router(config)# voice iec syslog
```

Related Commands

Command	Description
clear voice statistics	Clears voice statistics, resetting the statistics collection.
show voice statistics iec	Displays iec statistics
show voice statistics interval-tag	Displays interval options available for IEC statistics
voice statistics type iec	Enables collection of IEC statistics

voice local-bypass

To configure local calls to bypass the digital signal processor (DSP), use the **voice local-bypass command in**global configuration mode. To direct local calls through the DSP, use the **no** form of this command.

voice local-bypass

no voice local-bypass

Syntax Description

This command has no arguments or keywords.

Command Default

Local calls bypass the DSP.

Command Modes

Global configuration

Command History

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

Usage Guidelines

Local calls (calls between voice ports on a router or concentrator) normally bypass the DSP to minimize use of system resources. Use the **no** form of the **voice local-bypass**command if you need to direct local calls through the DSP. Input gain and output attenuation can be configured only if calls are directed through the DSP.

Examples

The following example configures a Cisco router to pass local calls through the DSP:

```
no voice local-bypass
```

Related Commands

Command	Description
input gain	Configures a specific input gain value.
output attenuation	Configures a specific output attenuation value.

voice mlpp

To enter MLPP configuration mode to enable MLPP service, use the voice service command in global configuration mode. To disable MLPP service, use the **no** form of this command.

voice mlpp

no voice mlpp

Syntax Description

This command has no keywords or arguments.

Command Default

No default behavior or values.

Command Modes

G lobal configuration (config)

Command History

Cisco IOS Release	Cisco Products	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Voice-mlpp configuration mode is used for the gateway globally.

Examples

The following example shows how to enter voice-mlpp configuration mode:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# access-digit
```

Related Commands

Command	Description
access-digit	Defines the access digit that phone users dial to request a precedence call.
mlpp preemption	Enables calls on an SCCP phone or analog FXS port to be preempted.
preemption trunkgroup	Enables preemption capabilities on a trunk group.

voicemail (stcapp-fsd)

To designate an SCCP telephony control (STC) application feature speed-dial code to speed dial the voice-mail number, use the **voicemail** command in STC application feature speed-dial configuration mode. To return the code to its default, use the **no** form of this command.

voicemail *keypad-character*

no voicemail

Syntax Description

keypad-character	One or two digits that can be dialed on a telephone keypad. Range is 0 to 9 for one-digit codes; 00 to 99 for two-digit codes. Default is 0 (zero) for one-digit codes; 00 (two zeroes) for two-digit codes. **Note** Number of digits depends on the value set with the **digit** command.

Command Default

The default voice-mail code is 0 (zero) for one-digit codes; 00 (two zeros) for two-digit codes.

Command Modes

STC application feature speed-dial configuration

Command History

Release	Modification
12.4(2)T	This command was introduced.
12.4(6)T	The *keypad-character* argument was modified to allow two-digit codes.

Usage Guidelines

This command is used with the STC application, which enables certain features on analog FXS endpoints that use Skinny Client Control Protocol (SCCP) for call control.

To use the speed-dial to voice-mail feature on a phone, dial the feature speed-dial (FSD) prefix and the code that has been configured with this command (or the default if this command was not used). For example, if the FSD prefix is * (the default), and you want to dial the voice-mail phone number, dial *0.

Note that the number that will be speed-dialed for voice mail must be set on Cisco CallManager or the Cisco CallManager Express system.

This command is reset to its default value if you modify the value of the **digit** command. For example, if you set the **digit** command to 2, then change the **digit** command back to its default of 1, the voice-mail FSD code is reset to 0 (zero).

If you set this code to a value that is already in use for another FSD code, you receive a warning 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 **show stcapp feature codes** command.

The **show running-config** command displays nondefault FSD codes only. The **show stcapp feature codes** command displays all FSD codes.

Examples

The following example sets an FSD prefix of two pound signs (##) and a voice-mail code of 8. After these values have been configured, a phone user presses ##8 to dial the voice-mail number.

```
Router(config)# stcapp feature speed-dial
Router(stcapp-fsd)# prefix ##
Router(stcapp-fsd)# voicemail 8
Router(stcapp-fsd)# exit
```

Related Commands

Command	Description
digit	Designates the number of digits for STC application feature speed-dial codes.
prefix (stcapp-fsd)	Designates a prefix to precede the dialing of an STC application feature speed-dial code.
redial	Designates an STC application feature speed-dial code to dial again the last number that was dialed.
show running-config	Displays current nondefault configuration settings.
show stcapp feature codes	Displays configured and default STC application feature codes.
speed dial	Designates a range of STC application feature speed-dial codes.
stcapp feature speed-dial	Enters STC application feature speed-dial configuration mode to set feature speed-dial codes.

voice pcm capture

To allocate the number of Pulse Code Modulation (PCM) capture buffers, to set up or change the destination URL for captured data, to enable PCM capture on-demand, and to change the PCM capture trigger string by the user, use the **voice pcm capture** command in global configuration mode. To stop all logging and file operations, to disable data transport from the capture buffer, and to automatically set the number of buffers to 0, use the **no** form of this command.

voice pcm capture {**buffer** *number*| **destination** *url*| **on-demand-trigger**| **user-trigger-string** *start-string stop-string* **stream** *bitmap* **duration** *call-duration*}

no voice pcm capture {**buffer** *number*| **destination** *url*| **on-demand-trigger**| **user-trigger-string**}

Syntax Description

buffer *number*	Allocates the number of PCM capture buffers. The range is from 0 to 200000. The default is 0.
destination *url*	Specifies the destination URL for storing captured data.
on-demand-trigger	(Optional) Configures PCM capture user trigger on-demand.
user-trigger-string *start-string stop-string* **stream** *bitmap* **duration** *call-duration*	(Optional) Configures PCM user trigger string. • *start-string*—Start string up to 15 characters. • *stop-string*—Stop string up to 15 characters. • **stream**—Configures the PCM capture stream bitmap. • *bitmap*—PCM stream bitmap in hexadecimal. The range is from 1 to FFFFFFF. The default is 7. • **duration**—Configures the duration for PCM capture. • *call-duration*—Duration of call. The range is from 0 to 255. The default is 0.

Command Default

The default values are as follows:

- Number of buffers: 0
- Start string: 123
- Stop string: 456
- Stream: 7

- Call duration: 0

Command Modes

Global configuration (config)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

If you want to change the number of an existing nonzero buffer, you must first reset it to 0 and then change it from 0 to the new number.

The **destination** *url* option sets up or changes the destination URL for captured data. To disable data transport from the capture buffer, use the **no** form of this command. If the buffer is allocated, captured data is sent to the current URL (if it was already configured) until the new URL is specified.

If a new URL differs from the current URL and logging is enabled, the current URL is closed and all further data is sent to the new URL. Entering a blank URL or prefixing the command with **no** disables data transport from the capture buffer, and (if capture is enabled) captured data is stored in the capture buffer until it reaches its capacity.

Once the buffer-queueing program is running, the transport process attempts to connect to a new or existing "capture destination" URL. A version message is written to the URL, and if the message is successfully received, any further messages placed into the message queue are written to that URL. If a new URL is entered using the **voice pcm capture destination url** command, the open URL is closed, and the system attempts to write to the new URL. If the new URL does not work, the transport process exits. The transport process is restarted when another URL is entered or the system is restarted.

Examples

The following example shows how to configure the number of PCM capture buffers:

```
Router> enable
Router# configure terminal
Router(config)# voice pcm capture buffer 200
```

The following example shows how to configure the destination URL for storing captured data:

```
Router> enable
Router# configure terminal
Router(config)# voice pcm capture destination tftp://10.0.1.10/acphan/
```

The following example shows how to configure user trigger PCM capture:

```
Router> enable
Router# configure terminal
Router(config)# voice pcm capture on-demand-trigger
```

The following example shows how to change the default user trigger PCM capture start and stop string, stream, and call duration:

```
Router> enable
Router# configure terminal
```

```
Router(config)# voice pcm capture #132 #543 stream ff duration 230
```

Related Commands

Command	Description
show voice pcm capture	Displays PCM capture status and statistics.

voiceport

To enable a private line automatic ringdown (PLAR) connection for an analog phone, use the **voiceport** command in SCCP PLAR configuration mode. To remove PLAR from the voice port, use the **no** form of this command.

voiceport *port-number* **dial** *dial-string* [**digit** *dtmf-digits* [**wait-connect** *wait-msecs*] [**interval** *inter-digit-msecs*]]

no voiceport *port-number*

Syntax Description

port-number	Analog foreign exchange station (FXS) voice port number. Range: 2/0 to 2/23.
dial *dial-string*	String of up to 16 characters that can be dialed on a telephone keypad. Valid characters are 0 through 9, A through D, an * (asterisk) and # (pound sign). The voice gateway sends this string to the call-control system when the analog phone goes off hook.
digit *dtmf-digits*	(Optional) String of up to 16 characters that can be dialed on a telephone keypad. Valid characters are 0 through 9, A through D, an * (asterisk), # (pound sign), and comma (,). The voice gateway sends this string to the call-control system after the *wait-msecs* expires. Each comma represents a one second wait.
wait-connect *wait-msecs*	(Optional) Number of milliseconds that the voice gateway waits after voice cut-through before out-pulsing the DTMF digits. Range: 0 to 30000, in multiples of 50. Default: 50. If 0, DTMF digits are sent automatically by voice gateway after call is connected.
interval *inter-digit-msecs*	(Optional) Number of milliseconds between the DTMF digits. Range: 50 to 500, in multiples of 50. Default: 50.

Command Default

Disabled (PLAR is not set for the voice port).

Command Modes

SCCP PLAR configuration

Command History

Release	Modification
12.4(6)T	This command was introduced.

Usage Guidelines

This command enables PLAR on analog FXS ports that use Skinny Client Control Protocol (SCCP) for call control. If the **digit** keyword is not used, DTMF digits are not out-pulsed; the voice port uses a simple PLAR connection and the other keywords are not available.

Voice ports can be configured in any order. For example, you can configure port 2/23 before port 2/0. The **show running-config** command lists the ports in ascending order.

Before a PLAR port can become operational, the STC application must first be enabled in the corresponding dial-peer using the **service stcapp** command. If you configure a port for PLAR before enabling the STC application in the dial peer you receive a warning message.

PLAR phones support most of the same features as normal analog phones. The PLAR phone handles incoming calls and supports hookflash for basic supplementary features such as call transfer, call waiting, and conference. The PLAR phone does not support other features such as call forwarding, redial, speed dial, call park, call pick up from a PLAR phone, AMWI, or caller ID.

Examples

The following example enables the PLAR feature on port 2/0, 2/1, and 2/3. When a phone user picks up the handset on the analog phone connected to port 2/0, the system automatically rings extension 3660 and after waiting 500 milliseconds, dials 1234. The DTMF digits are out-pulsed to the destination port at an interval of 200 milliseconds.

```
Router(config)# sccp plar
Router(config-sccp-plar)# voiceport 2/0 dial 3660 digit 1234 wait-connect 500 interval 200
Router(config-sccp-plar)# voiceport 2/1 dial 3264 digit 678,,,9*0,,#123 interval 100
Router(config-sccp-plar)# voiceport 2/3 dial 3478 digit 34567 wait-connect 500
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and defines a dial peer.
sccp plar	Enters SCCP PLAR configuration mode.

voice-port

To enter voice-port configuration mode, use the **voice-port** command in global configuration mode.

Cisco 1750 and Cisco 1751

voice-port *slot-number*/*port*

Cisco 2600 series, Cisco 3600 Series, and Cisco 7200 Series

voice-port {*slot-number*/*subunit-number*/*port*| *slot*/*port*:*ds0-group-no*}

Cisco 2600 and Cisco 3600 Series with a High-Density Analog Network Module (NM-HDA)

slot-number/*subunit-number*/*port***voice-port**

Cisco AS5300

voice-port *controller-number* **:D**

Syntax Description

slot-number	Number of the slot in the router in which the voice interface card (VIC) is installed. Valid entries are from 0 to 2, depending on the slot in which it has been installed.
port	Voice port number. Valid entries are 0 and 1.

Syntax Description

slot-number	Number of the slot in the router in which the VIC is installed. Valid entries are from 0 to 3, depending on the slot in which it has been installed.
subunit-number	Subunit on the VIC in which the voice port is located. Valid entries are 0 or 1.
port	Voice port number. Valid entries are 0 and 1.
slot	The router location in which the voice port adapter is installed. Valid entries are from 0 to 3.
port:	Indicates the voice interface card location. Valid entries are 0 and 3.

ds0-group-no	Indicates the defined DS0 group number. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.

Syntax Description

controller-number	T1 or E1 controller.
:D	D channel associated with ISDN PRI.

Command Default

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
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.
12.0(7)T	This command was implemented on the Cisco AS5800, Cisco 7200 series, and Cisco 1750. Arguments were added for the Cisco 2600 series and Cisco 3600 series.
12.2(8)T	This command was implemented on Cisco 1751 and Cisco 1760. This command was modified to accommodate the additional ports of the NM-HDA on the Cisco 2600 series, Cisco 3640, and Cisco 3660.
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(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.
12.2(13)T	This command was integrated into Cisco IOS Release 12.2(13)T. This command does not support the extended echo canceller (EC) feature on the Cisco AS5300 or the Cisco AS5800.

Usage Guidelines

Use the **voice-port**global configuration command to switch to voice-port configuration mode from global configuration mode. Use the **exit** command to exit voice-port configuration mode and return to global configuration mode.

Note This command does not support the extended echo canceller (EC) feature on the Cisco AS5300.

Examples

The following example accesses voice-port configuration mode for port 0, located on subunit 0 on a VIC installed in slot 1:

```
voice-port 1/0/0
```

The following example accesses voice-port configuration mode for a Cisco AS5300:

```
voice-port 1:D
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice encapsulation.

voice-port (MGCP profile)

The **voice-port**(MGCP profile)command is replaced by the **port**(MGCP profile) command in Cisco IOS Release 12.2(8)T. See the **port** (MGCP profile) command for more information.

voice-port busyout

To place all voice ports associated with a serial or ATM interface into a busyout state, use the **voice-port busyout**command in interfaceconfiguration mode. To remove the busyout state on the voice ports associated with this interface, use the **no** form of this command.

voice-port busyout

no voice-port busyout

Syntax Description

This command has no arguments or keywords.

Command Default

The voice ports on the interface are not in busyout state.

Command Modes

Interface configuration

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco MC3810.

Usage Guidelines

This command busies out all voice ports associated with the interface, except any voice ports configured to busy out under specific conditions using the **busyout monitor** and **busyout seize** commands.

Examples

The following example places the voice ports associated with serial interface 1 into busyout state:

```
interface serial 1
 voice-port busyout
```

The following example places the voice ports associated with ATM interface 0 into busyout state:

```
interface atm 0
 voice-port busyout
```

Related Commands

Command	Description
busyout forced	Forces a voice port into the busyout state.
busyout monitor	Places a voice port into the busyout monitor state.
busyout seize	Changes the busyout action for an FXO or FXS voice port.
show voice busyout	Displays information about the voice busyout state.

voice rtp send-recv

To establish a two-way voice path when the Real-Time Transport Protocol (RTP) channel is opened, use the **voice rtp send-recv command in**global configuration mode. To reset to the default, use the **no** form of this command.

voice rtp send-recv

no voice rtp send-recv

Syntax Description

This command has no arguments or keywords.

Command Default

The voice path is cut-through in only the backward direction when the RTP channel is opened.

Command Modes

Global configuration

Command History

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

Usage Guidelines

This command should be enabled only when the voice path must be cut-through (established) in both the backward and forward directions before a Connect message is received from the destination switch. This command affects all VoIP calls when it is enabled.

Examples

The following example enables the voice path to cut-through in both directions when the RTP channel is opened:

```
voice rtp send-recv
```

voice-service dsp-reservation

To specify the percentage of DSP resources that are reserved strictly for VOIP on the voice card, use the **voice-service dsp-reservation**command in voice-card configuration. To reset the percentage of DSP resources, use the **no** form of this command.

voice-service-dsp reservation *percentage*

no voice-service-dsp reservation *percentage*

Syntax Description

percentage	Percentage of DSP resources on this voice card that are reserved for voice services. The remaining DSP resources will be available for video services.

Command Default

The default voice reservation is 100%.

Command Modes

voice-card configuration (config-voicecard)

Command History

Release	Modification
15.1(4)M	The command was introduced.

Usage Guidelines

Use this command to reserve a percentage of the voice card for voice services. The remaining DSP resources will be used for video services. A reservation of 100% specified that all DSP resources will be used for voice services.

Note You can configure a percentage less than 100% only when there is a video license and the appropriate PVDM# modules are installed.

Tip DSP can become fragmented when you change the percentage of DSP resources reserved for voice services when there are TDM voice or DSP farm profiles configured. To ensure the best system performance, reload the router when you change the **voice-service-dsp-reservation**.

Examples

The following example enters voice-card configuration mode and sets the percentage of DSP resources for voice to 60%:

```
Router(config)# voice card 0
Router(config-voicecard)# voice-service dsp-reservation 60
```

Related Commands

Command	Description
dspfarm profile	Adds the specified voice card to those participating in a DSP resource pool.

voice service

To enter voice-service configuration mode and to specify a voice-encapsulation type, use the voice service command in global configuration mode..

voice service {**pots**| **voatm**| **vofr**| **voip**}

Syntax Description

pots	Telephony voice service.
voatm	Voice over ATM (VoATM) encapsulation.
vofr	Voice over Frame Relay (VoFR) encapsulation.
voip	Voice over IP (VoIP) encapsulation.

Command Default

No default behavior or values.

Command Modes

Global 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.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T for VoIP on the Cisco 2600 series and the Cisco 3600 series.
12.1(3)XI	This command was implemented on the Cisco AS5300.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.1(5)XM	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(2)T	This command was integrated into Cisco IOS Release 12.2(2)T and implemented on the Cisco 7200 series.
12.2(11)T	This command was implemented on the Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.

Usage Guidelines

Voice-service configuration mode is used for packet telephony service commands that affect the gateway globally.

Examples

The following example enters voice-service configuration mode for VoATM service commands:

```
voice service voatm
```

voice source-group

To define a source IP group for voice calls, use the **voice source-group** command in global configuration mode. To delete the source IP group, use the **no** form of this command.

voice source-group *name*

no voice source-group *name*

Syntax Description

name	Name of the IP group. Maximum length of the source IP group name is 31 alphanumeric characters.

Command Default

No default behavior or values

Command Modes

Global configuration

Command History

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

Usage Guidelines

Use the **voice source-group** command to assign a name to a set of source IP group characteristics. The terminating gateway uses these characteristics to identify and translate the incoming VoIP call.

Carrier IDs and trunk group labels must not have the same names.

Do not mix carrier IDs and trunk group labels within a source IP group.

A terminating gateway can be configured with carrier ID source IP groups and trunk-group-label source IP groups. Th e name of the source IP group must be unique to the gateway.

Examples

The following example initiates source IP group "utah2" for VoIP calls:

```
Router(config)# voice source-group utah2
```

Related Commands

Command	Description
access-list	Defines a list of source groups for identifying incoming calls.
carrier-id (voice source group)	Specifies the carrier handling a VoIP call.

Command	Description
description (voice source group)	Assigns a disconnect cause to a source IP group.
h323zone-id (voice source group)	Assigns a zone ID to an incoming H.323 call.
translation-profile (source group)	Assigns a translation profile to a source IP group.
trunk-group-label (voice source group)	Specifies the trunk handling a VoIP call.

voice statistics accounting method

To enable voice accounting statistics to be collected for a specific accounting method list and to specify the pass criteria for call legs, use the **voice statistics accounting method**command in global configuration mode. To disable the collection of statistics for the accounting method, use the **no** form of this command.

voice statistics accounting method *method-list-name* **pass** {**start-interim-stop| start-stop| stop-only**}

no voice statistics accounting method *method-list-name* **pass** {**start-interim-stop| start-stop| stop-only**}

Syntax Description

method-list-name	Name of the accounting method list. The method-list-name argument is the same as that configured using the **method** command in gateway accounting AAA configuration mode.
pass	The pass criteria for call legs (PSTN or IP) and call directions (inbound or outbound) that is used by the method list. **Note** The definition of pass implies that all start, stop, or interim messages are acknowledged by the designated servers. The definition of failure implies that any start, stop, or interim message is rejected or is timed out by the designated servers.
start-interim-stop	All start, interim, and stop pass criteria records are counted.
start-stop	All start and stop pass criteria records are counted.
stop-only	Only stop pass criteria records are counted.

Command Default

No statistics for the specified accounting method list are collected.

Command Modes

Global configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Examples

The following example shows that h323 is specified as the method list and that the pass criterion is stop-only:

```
Router(config)# voice statistics accounting method h323 pass stop-only
```

Related Commands

Command	Description
method	Specifies the AAA method list name to be used.
show voice statistics csr interval accounting	Displays statistical information by configured intervals for accounting statistics.
show voice statistics csr since-reset accounting	Displays all accounting CSRs since the last reset.
voice statistics display-format separator	Specifies the format for CSR display.
voice statistics field-params	Specifies MCD, lost-packet, packet-latency, and packet-jitter parameters.
voice statistics max-storage-duration	Specifies the maximum time for which CSRs are stored in system memory.
voice statistics push	Specifies an FTP or syslog server for downloading CSRs, the maximum file size, and the maximum message size.
voice statistics time-range	Specifies the time range to collect CSRs.
voice statistics type	Enables the collection of accounting and signaling CSRs.

voice statistics display-format separator

To configure the display format of the statistics on the gateway, use the **voice statistics display-format separator**command in global configuration mode. To return the display format of the statistics to the default value, use the **no** form of this command.

voice statistics display-format separator {**space**| **tab**| **new-line**| **char char**}

no voice statistics display-format separator {**space**| **tab**| **new-line**| **char char**}

Syntax Description

separator	Type of separator used in the displayed format.
space	A space is used for the formatting between each statistic in the displayed output.
tab	A tab is used for the formatting between each statistic in the displayed output.
new-line	A new line is used for the formatting between each statistic in the displayed output.
char *char*	A character is used for the formatting between each statistic in the displayed output. The char argument is a visible ASCII character used for the formatting between each statistic in the displayed output.

Command Default

A comma (,) is the default separator.

Command Modes

Global configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Examples

The following example shows that a space is specified as the display separator:

```
Router(config)# voice statistics display-format separator space
```

Related Commands

Command	Description
voice statistics accounting method	Enables the accounting method and the pass and fail criteria.
voice statistics field-params	Specifies MCD, lost-packet, packet-latency, and packet-jitter parameters.
voice statistics max-storage-duration	Specifies the maximum time for which CSRs are stored in system memory.
voice statistics push	Specifies an FTP or syslog server for downloading CSRs, the maximum file size, and the maximum message size.
voice statistics time-range	Specifies the time range to collect CSRs.
voice statistics type	Enables the collection of accounting and signaling CSRs.

voice statistics field-params

To configure the parameters of call statistics fields on the gateway, use the **voice statistics field-params** command in global configuration mode. To return the call statistics parameters to the default values, use the **no** form of this command.

voice statistics field-params {**mcd** *value*| **lost-packet** *value*| **packet-latency** *value*| **packet-jitter value**}

no voice statistics field-params {**mcd** *value*| **lost-packet** *value*| **packet-latency** *value*| **packet-jitter value**}

Syntax Description

mcd	Minimum call duration. The value argument is an integer that represents the number of milliseconds. Valid values are from 0 to 30. The default is 2.
lost-packet	Lost voice packet threshold. The value argument is an integer that represents milliseconds. Valid values are from 0 to 65535. The default is 1000.
packet-latency	Voice packet latency threshold. The value argument is an integer that represents milliseconds. Valid values are from 0 to 500. The default is 250.
packet-jitter	Voice packet jitter threshold. The value argument is an integer that represents milliseconds. Valid values are from 0 to 1000. The default is 250.

Command Default

MCD is 2 milliseconds. Lost packet threshold is 1000 milliseconds. Packet latency threshold is 250 milliseconds. Packet jitter threshold is 250 milliseconds.

Command Modes

Global configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Examples

The following example configures a minimum call duration of 5 milliseconds:

```
Router(config)# voice statistics field-params mcd 5
```

The following example configures a lost packet threshold of 250 milliseconds:

```
Router(config)# voice statistics field-params lost-packet 250
```

The following example configures a packet-latency threshold of 300 milliseconds:

```
Router(config)# voice statistics field-params packet-latency 300
```

The following example configures a packet-jitter threshold of 245 milliseconds:

```
Router(config)# voice statistics field-params packet-jitter 245
```

Related Commands

Command	Description
voice statistics accounting method	Enables the accounting method and the pass and fail criteria.
voice statistics display-format separator	Specifies the format for CSR display.
voice statistics max-storage-duration	Specifies the maximum time for which CSRs are stored in system memory.
voice statistics push	Specifies an FTP or syslog server for downloading CSRs, the maximum file size, and the maximum message size.
voice statistics time-range	Specifies the time range to collect CSRs.
voice statistics type	Enables the collection of accounting and signaling CSRs.

voice statistics max-storage-duration

To configure the maximum amount of time for which collected statistics are stored in the system memory of the gateway, use the **voice statistics max-storage-duration**command in global configuration mode. To remove the configured maximum storage duration, use the **no** form of this command.

voice statistics max-storage-duration{**day***value* | **hour** *value* | **minute***value*}

no voice statistics max-storage-duration{**day***value* | **hour** *value* | **minute***value*}

Syntax Description

day	Number of days for which call statistics data are to be stored. The value argument has a valid range from 0 to 365.
hour	Number of hours for which call statistics data are to be stored. The value argument has a valid range from 0 to 720.
minute	Number of minutes for which call statistics data are to be stored. The value argument has a valid range from 0 to 1440.

Command Default

If no length of time is configured, no memory is allocated for those call statistic records that have stopped after the end of their collection intervals. If no memory is allocated, only active call statistic record buffers are kept in system memory.

Command Modes

Global configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Usage Guidelines

The maximum storage duration means the time-to-exist duration of the call statistic records on the gateway.

The values entered using this command also apply to the collection of VoIP internal error codes (IECs).

Examples

The following example shows that the maximum storage duration for the collection of voice call statistics has been set for 60 minutes:

```
Router(config)# voice statistics max-storage-duration minute 60
```

Related Commands

Command	Description
voice statistics accounting method	Enables the accounting method and the pass and fail criteria.
voice statistics display-format separator	Specifies the format for CSR display.
voice statistics field-params	Specifies MCD, lost-packet, packet-latency, and packet-jitter parameters.
voice statistics push	Specifies an FTP or syslog server for downloading CSRs, the maximum file size, and the maximum message size.
voice statistics time-range	Specifies the time range to collect CSRs.
voice statistics type	Enables the collection of accounting and signaling CSRs.

voice statistics push

To configure the method for pushing signaling statistics, VoIP AAA accounting statistics, or Cisco internal error codes (IECs) to an FTP or syslog server, use the **voice statistics push** command in global configuration mode. To disable the configured push method, use the **no** form of this command.

{**voice statistics push ftp url** *ftp-url* [**max-file-size** *value*]| **syslog** [**max-msg-size** *value*]}

{**no voice statistics push ftp url** *ftp-url* [**max-file-size** *value*]| **syslog** [**max-msg-size** *value*]}

Syntax Description

ftp url	URL of the FTP server to which voice statistics are to be pushed. The syntax of the ftp-url argument follows: ftp://user:password@host:port//directory1/directory2
max-file-size	(Optional) Maximum size of a voice statistics file to be pushed to an FTP server, in bytes. The valid range of the *value* argument is from 1024 to 4294967296. The default value is 400000000 (4 GB).
syslog	Voice statistics are pushed to a syslog server.
max-msg-size	(Optional) Maximum size of a voice statistics file to be pushed to a syslog server, in bytes. The valid range of the *value* argument is from 1024 to 4294967296. The default value is 400000000 (4 GB).

Command Default

Voice statistics are not pushed to an FTP or syslog server.

Command Modes

Global configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Usage Guidelines

The gateway configuration should be consistent with the configuration on the FTP or syslog servers. This command may also be used to push Cisco VoIP internal error codes (IECs) to either an FTP server or a syslog server.

Examples

The following is a configuration example showing a specified FTP server and maximum file size:

```
Router(config)# voice statistics push ftp url
ftp://john:doe@abc:23//directory1/directory2 max-file-size 10000
```

Related Commands

Command	Description
voice statistics accounting method	Enables the accounting method and the pass and fail criteria.
voice statistics display-format separator	Specifies the format for CSR display.
voice statistics field-params	Specifies MCD, lost-packet, packet-latency, and packet-jitter parameters.
voice statistics max-storage-duration	Specifies the maximum time for which CSRs are stored in system memory.
voice statistics time-range	Specifies the time range to collect CSRs.
voice statistics type	Enables the collection of accounting and signaling CSRs.

voice statistics time-range

To specify a time range to collect statistics from the gateway on a periodic basis, since the last reset, or for a specific time duration , use the **voice statistics time-range**command in global configuration mode. To disable the time-range settings, use the **no** form of this command.

Statistics Collection on a Periodic Basis

voice statistics time-rangeperiodic*interval***start***hh:mm*{**days-of-week**{**Monday| Tuesday| Wednesday| Thursday| Friday| Saturday| Sunday| daily| weekday| weekend**}}[**end***hh:mm*{**days-of-week| Monday| Tuesday| Wednesday| Thursday| Friday| Saturday| Sunday| daily| weekday| weekend**}]

no voice statistics time-rangeperiodic*interval***start***hh:mm*{**days-of-week**{**Monday| Tuesday| Wednesday| Thursday| Friday| Saturday| Sunday| daily| weekday| weekend**}}[**end***hh:mm*{**days-of-week| Monday| Tuesday| Wednesday| Thursday| Friday| Saturday| Sunday| daily| weekday| weekend**}]

Statistics Collection Since the Last Reset or Reboot of the Gateway

voice statistics time-range since-reset

no voice statistics time-range since-reset

Statistics Collection at a Specific Time Duration

voice statistics time-range specific start *hh* **:** *mm day month year end hh* **:** *mm day month year*

no voice statistics time-range specific start *hh* **:** *mm day month year end hh* **:** *mm day month year*

Syntax Description

Statistics Collection on a Periodic Basis:	
periodic	Call statistics are collected for a configured period.
interval	Specifies the periodic interval during which statistics will be collected. Valid entries for this value are **5minutes**, **15minutes**, **30minutes**, **60minutes**, or **1day**.
start/end	Specifies the start and ending periods of the statistics collection. If no end time is entered, then the statistics collection continues nonstop. By default, there is no end of the collection period.
hh:mm	Specifies the start and ending times for the periodic statistics collection in hours and minutes. The times entered must be in 24-hour format.

days-of-week	Specifies the start and ending days of the week that call statistics are collected. You can configure a specific day of the week, or one of the following: • daily--Call statistics are collected daily. • weekdays--Call statistics are collected on weekdays only. • weekend--Call statistics are collected on weekends only. The default value is daily.
Statistics Collection Since the Last Reset or Reboot of the Gateway	
since-reset	Call statistics are collected only since a reset or reboot of the gateway. **Note** Voice statistics collection on the gateway is reset using the **clear voice statistics csr**command.
Statistics Collection at a Specified Time Duration:	
specific	Call statistics are collected for a specific time duration.
start/end	Specifies the start and end times of the statistics collection. The required arguments for both the start and end keywords are as follows: • hh**:**mm--Hour and minute. The times entered must be in 24-hour format. • day--Day of the month. Valid values are from 1 to 31. • month--Month for the statistics collection to start. Enter the month name, for example, January, or February. The default is the current month. • year--Year. Valid values are from 1993 to 2035. The default is the current year.

No statistics are collected by default.

Command Modes

Global configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Usage Guidelines

There should be only one specific or periodic configuration at any one time. If a second specific or periodic configuration is configured, the request is rejected and a warning message displays. If the no form of the command is used during the specific time range, the corresponding collection will stop and FTP or syslog messages will not be sent.

Examples

The following example shows that the time range is periodic and set to collect statistics for a 60-minute period on weekdays only beginning at 12:00 a.m.:

```
Router(config)# voice statistics time-range periodic 60minutes start 12:00 days-of-week
weekdays
```

The following example configures the gateway to collect call statistics since the last reset (specified with the **clear voice statistics csr**command) or since the last time the gateway was rebooted:

```
Router(config)# voice statistics time-range since-reset
```

The following example configures the gateway to collect statistics from 10:00 a.m. on the first day of January to 12:00 a.m. on the second day of January:

```
Router(config)
# voice statistics time-range specific start 10:00 1 January 2004 end 12:00 2 January 2004
```

Related Commands

Command	Description
clear voice statistics	Clears voice statistics, resetting the statistics collection.
voice statistics accounting method	Enables the accounting method and the pass and fail criteria.
voice statistics display-format separator	Specifies the format for CSR display.
voice statistics field-params	Specifies MCD, lost-packet, packet-latency, and packet-jitter parameters.
voice statistics max-storage-duration	Specifies the maximum time for which CSRs are stored in system memory.
voice statistics push	Specifies an FTP or syslog server for downloading CSRs, the maximum file size, and the maximum message size.

Command	Description
voice statistics type	Enables the collection of accounting and signaling CSRs.

voice statistics type csr

To configure a gateway to collect VoIP AAA accounting statistics or voice signaling statistics, independently or at the same time, use the **voice statistics type csr**command in global configuration mode. To disable the counters, use the **no** form of this command.

voice statistics type csr [**accounting**| **signaling**]

no voice statistics type csr [**accounting**| **signaling**]

Syntax Description

accounting	(Optional) VoIP AAA accounting statistics are collected.
signaling	(Optional) Voice signaling statistics are collected.

Command Default

No accounting or signaling call statistics records (CSRs) are collected on the gateway.

Command Modes

Global configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Usage Guidelines

If you do not specify a keyword, both accounting and signaling CSRs are collected. Accounting and signaling CSR collection can be enabled and disabled independently.

Examples

The following example shows that both types of CSRs will be collected:

```
Router(config)# voice statistics type csr
```

The following example enables accounting CSRs to be collected:

```
Router(config)# voice statistics type csr accounting
```

The following example enables signaling CSRs to be collected:

```
Router(config)# voice statistics type csr signaling
```

The following example disables the collection of both signaling and accounting CSRs:

```
Router(config)# no
voice statistics type csr
```

The following example disables the collection of signaling CSRs only:

```
Router(config)# no
voice statistics type csr signaling
```

Related Commands

Command	Description
voice statistics accounting method	Enables the accounting method and the pass and fail criteria.
voice statistics display-format separator	Specifies the format for CSR display.
voice statistics field-params	Specifies MCD, lost-packet, packet-latency, and packet-jitter parameters.
voice statistics max-storage-duration	Specifies the maximum time for which CSRs are stored in system memory.
voice statistics push	Specifies an FTP or syslog server for downloading CSRs, the maximum file size, and the maximum message size.
voice statistics time range	Specifies the time range to collect CSRs.

voice statistics type iec

To enable collection of Internal Error Code (IEC) statistics, use the voice statistics type iec command in global configuration mode. To disable IEC statistics collection, use the **no** form of this command.

voice statistics type iec

no voice statistics type iec

Syntax Description

This command has no arguments or keywords.

Command Default

IEC statistics collection is disabled.

Command Modes

Global configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Examples

The following example enables IEC statistics collection:

```
Router(config)# voice statistics type iec
```

Related Commands

Command	Description
clear voice statistics	Clears voice statistics, resetting the statistics collection.
show voice statistics	Displays voice statistics
show voice statistics interval-tag	Displays interval options available for IEC statistics
voice statistics time-range since-reset	Enables collection of call statistics accumulated since the last resetting of IEC counters

voice translation-profile

To define a translation profile for voice calls, use the **voice translation-profile** command in global configuration mode. To delete the translation profile, use the **no** form of this command.

voice translation-profile *name*

no voice translation-profile *name*

Syntax Description

name	Name of the translation profile. Maximum length of the voice translation profile name is 31 alphanumeric characters.

Command Default

No default behavior or values

Command Modes

Global configuration

Command History

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

Usage Guidelines

After translation rules are defined, they are grouped into profiles. The profiles collect a set of rules that, taken together, translate the called, calling, and redirected numbers in specific ways. Up to 1000 profiles can be defined. Each profile must have a unique name .

These profiles are referenced by trunk groups, dial peers, source IP groups, voice ports, and interfaces for handling call translations.

Examples

The following example initiates translation profile "westcoast" for voice calls. The profile uses translation rules 1, 2, and 3 for various types of calls.

```
Router(config)# voice translation-profile westcoast
Router(cfg-translation-profile)# translate calling 2
Router(cfg-translation-profile)# translate called 1
Router(cfg-translation-profile)# translate redirect-called 3
```

Related Commands

Command	Description
rule (voice translation-rule)	Defines call translation criteria.
show voice translation-profile	Displays one or more translation profiles.

Command	Description
translate (translation profiles)	Associates a translation rule with a voice translation profile.

voice translation-rule

To define a translation rule for voice calls, use the **voice translation-rule** command in global configuration mode. To delete the translation rule, use the **no** form of this command.

voice translation-rule *number*

no voice translation-rule *number*

Syntax Description

number	Number that identifies the translation rule. Range is from 1 to 2147483647.

Command Default

No default behavior or values

Command Modes

Global configuration

Command History

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

Usage Guidelines

Use the **voice translation-rule** command to create the definition of a translation rule. Each definition includes up to 15 rules that include SED-like expressions for processing the call translation. A maximum of 128 translation rules are supported.

These translation rules are grouped into profiles that are referenced by trunk groups, dial peers, source IP groups, voice ports, and interfaces.

Examples

The following example initiates translation rule 150, Which includes two rules:

```
Router(config)# voice translation-rule 150
Router(cfg-translation-rule)# rule 1 reject /^408\(.(\)/
Router(cfg-translation-rule)# rule 2 /\(^...\)853\(...\)/ /\1525\2/
```

Related Commands

Command	Description
rule (voice translation-rule)	Defines the matching, replacement, and rejection patterns for a translation rule.
show voice translation-rule	Displays the configuration of a translation rule.

voice vad-time

To change the minimum silence detection time for voice activity detection (VAD), use the **voice vad-time** command in global configuration mode. To reset to the default, use the **no** form of this command.

voice vad-time *milliseconds*

no voice vad-time

Syntax Description

milliseconds	Waiting period, in milliseconds, before silence detection and suppression of voice-packet transmission. Range is from 250 to 65536. The default is 250.

Command Default

250 milliseconds

Command Modes

Global configuration

Command History

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

Usage Guidelines

Thiscommand affects all voice ports on a router or concentrator, but it does not affect calls already in progress.

You can use this command in transparent common-channel signaling (CCS) applications in which you want VAD to activate when the voice channel is idle, but not during active calls. With a longer silence detection delay, VAD reacts to the silence of an idle voice channel, but not to pauses in conversation.

Thiscommand does not affect voice codecs that have ITU-standardized built-in VAD features--for example, G.729B, G.729AB, G.723.1A. The VAD behavior and parameters of these codecs are defined exclusively by the applicable ITU standard.

Examples

The following example configures a 20-second delay before VAD silence detection is enabled:

```
voice vad-time 20000
```

Related Commands

Command	Description
vad (dial peer)	Enables voice activity detection on a network dial peer.

voice vrf

To configure a voice VRF, use the **voice vrf**command in global configuration mode. To remove the voice VRF configuration, use the **no** form of this command.

voice vrf *vrfname*

no voice vrf *vrfname*

Syntax Description

vrfname	A name assigned to the voice vrf.

Command Default

No voice VRF is configured.

Command Modes

Global configuration

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

You must create a VRF using the **ip vrf** *vrfname* command before you can configure it as a voice VRF.

To ensure there are no active calls on the voice gateway during a VRF change, voices services must be shut down on the voice gateway before you configure or make changes to a voice VRF.

Examples

The following example shows that a VRF called *vrf1* was created and then configured as a voice VRF:

```
ip vrf vrf1
 rd 1:1
 route-target export 1:2
 route-target import 1:2
!
voice vrf vrf1
!
voice service voip
```

Related Commands

Command	Description
ip vrf	Defines a VPN VRF instance and enters VRF configuration mode.

voip-incoming translation-profile

To specify a translation profile for all incoming VoIP calls, use the **voip-incoming translation-profile** command in global configuration mode. To delete the profile, use the **no** form of this command.

voip-incoming translation-profile *name*

no voip-incoming translation-profile *name*

Syntax Description

name	Name of the translation profile.

Command Default

No default behavior or values

Command Modes

Global configuration

Command History

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

Usage Guidelines

Use the **voip-incoming translation-profile** command to globally assign a translation profile for all incoming VoIP calls. The translation profile was previously defined using the **voice translation-profile** command. The **voip-incoming translation-profile**command does not require additional steps to complete its definition.

If an H.323 call comes in and the call is associated with a source IP group that is defined with a translation profile, the source IP group translation profile overrides the global translation profile.

Examples

The following example assigns the translation profile named "global-definition" to all incoming VoIP calls:

```
Router(config)# voip-incoming translation-profile global-definition
```

Related Commands

Command	Description
show voice translation-profile	Displays the configurations for all voice translation profiles.
test voice translation-rule	Tests the voice translation rule definition.
voice translation-profile	Initiates a translation profile definition.

voip-incoming translation-rule

To set the incoming translation rule for calls that originate from H.323-compatible clients, use the **voip-incoming translation-rule** command in global configuration mode. To disable the incoming translation rule, use the **no** form of this command.

voip-incoming translation-rule {**calling|** **called**} **name-tag**

no voip-incoming translation-rule {**calling|** **called**} **name-tag**

Syntax Description

name-tag	Tag number by which the rule set is referenced. This is an arbitrarily chosen number. Range is from 1 to 2147483647. There is no default value.
calling	Automatic number identification (ANI) number or the number of the calling party.
called	Dial Number Information Service (DNIS) number or the number of the called party.

Command Default

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XR1	This command was introduced for VoIP on the Cisco AS5300.
12.0(7)XK	This command was implemented for VoIP on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T and implemented for VoIP on the Cisco 1750, Cisco AS5300, Cisco 7200 series, and Cisco 7500 series platforms.
12.1(2)T	This command was implemented for VoIP on Cisco MC3810.
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

With this command, all IP-based calls are captured and handled, depending on either the calling number or the called number to the specified tag name.

Examples

The following example identifies the rule set for calls that originate from H.323-compatible clients:

```
Router(config)# voip-incoming translation-rule called 5
```

Related Commands

Command	Description
numbering-type	Matches one number type for a dial-peer call leg.
rule	Applies a translation rule to a calling party number or a called party number for both incoming and outgoing calls.
show translation-rule	Displays the contents of all the rules that have been configured for a specific translation name.
test translation-rule	Tests the execution of the translation rules on a specific name-tag.
translate	Applies a translation rule to a calling party number or a called party number for incoming calls.
translate-outgoing	Applies a translation rule to a calling party number or a called party number for outgoing calls.
translation-rule	Creates a translation name and enters translation-rule configuration mode.

voip trunk group

To define or modify a VOIP trunk group and to enter trunk group configuration mode, use the **voip trunk group** command in global configuration mode. To delete the VOIP trunk group, use the **no** form of this command.

voip trunk group *name*

no voip trunk group *name*

Syntax Description

name	Name of the voip trunk group. Valid names contain a maximum of 63 alphanumeric characters.

Command Default

No voip trunk group is defined.

Command Modes

Global configuration

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

Use the **voip trunk group** command to define the VOIP trunk and extend serviceability to the trunk. By default, the session protocol of the IP trunk is h323. Up to 1000 trunk groups can be configured on the gateway provided that the gateway has sufficient memory to store the profiles

Examples

The following example enables creates a VOIP trunk group and enables monitoring.

```
Router(config)# voip trunk group siptrk1
Router(config-voip-trk)# session protocol sipv2
Router(config-voip-trk)# target ipv4: 10.1.1.15
Router(config-voip-trk)# xsvc
```

Related Commands

Command	Description
show voip trunk group	Displays internal list of voip trunk groups.
xsvc	Enables monitoring on the trunk.

volume

To set the receiver volume level for a POTS port on a router, use the **volume** command in dial-peer voice configuration mode. To reset to the default, use the **no** form of this command.

volume *number*

no volume *number*

Syntax Description

number	A number from 1 to 5 representing decibels (dB) of gain. Range is as follows: • 1: -11.99 dB • 2: -9.7dB • 3: -7.7dB • 4: -5.7dB • 5: -3.7dB Default is 3 (-7.7 dB gain).

Command Default

3 (-7.7 dB gain)

Command Modes

Dial-peer voice configuration

Command History

Release	Modification
12.2(8)T	This command was introduced on Cisco 803, Cisco 804, and Cisco 813 routers.

Usage Guidelines

Set the **volume** command for each POTS port separately. Setting the volume level affects only the port for which it has been set.

Note Only the receiver volume is set with this command.

Use the **show pots volume** command to check the volume status and level.

Examples

The following example shows a volume level of 4 for POTS port 1 and a volume level of 2 for POTS port 2.

```
dial-peer voice 1 pots
 destination-pattern 5551111
 port 1
 no call-waiting
 ring 0
 volume 4
dial-peer voice 2 pots
 destination-pattern 5552222
 port 2
 no call-waiting
 ring 0
 volume 2
```

Related Commands

Command	Description
show pots volume	Shows the receiver volume configured for each POTS port on a router.

vxml allow-star-digit

To configure a Voice Extensible Markup Language (VXML) interpreter to allow the star digit for built-in type digits, use the **vxml allow-star-digit** command in global configuration mode. To disable the configuration, use the **no** form of this command.

vxml allow-star-digit

no vxml allow-star-digit

Syntax Description

This command has no arguments or keywords.

Command Default

A VXML interpreter is not configured.

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.

Examples

The following example shows how to configure a VXML interpreter to allow the star digit for built-in type digits:

```
Router# configure terminal
Router(config)# vxml allow-star-digit
```

Related Commands

Command	Description
vxml audioerror	Enables throwing an error event when audio playout fails.
vxml version pre2.0	Enables VoiceXML 2.0 features.

vxml audioerror

To enable throwing an error event when audio playout fails, use the **vxml audioerror** command in global configuration mode. To return to the default, use the **no** form of this command.

vxml audioerror

no vxml audioerror

Syntax Description

This command has no arguments or keywords.

Command Default

An audio error event, error.badfetch, is not thrown when an audio file cannot be played.

Command Modes

Global configuration

Command History

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

Usage Guidelines

Entering this command causes an audio error event, error.badfetch, to be thrown when an audio file cannot be played, for instance, because the file is in an unsupported format, the src attribute references an invalid URI, or the expr attribute evaluates to an invalid URI.

The **vxml audioerror** command overrides the **vxml version 2.0** command, so that if both commands are entered, the audio error event will be thrown when an audio file cannot be played.

Examples

The following example enables the audio error feature:

```
Router(config)# vxml audioerror
```

Related Commands

Command	Description
vxml version pre2.0	Enables features compatible with versions earlier than VoiceXML 2.0.

vxml tree memory

To set the maximum memory size for the VoiceXML parser tree, use the **vxml tree memory** command in global configuration mode. To reset to the default, use the **no** form of this command.

vxml tree memory *size*

no vxml tree memory

Syntax Description

size	Maximum memory size, in kilobytes. Range is 64 to 100000. Default is 1000.

Command Default

1000 KB

Command Modes

Global configuration

Command History

Release	Modification
12.2(15)T	This command was introduced.
12.4(15)T	The default was changed from 64 to 1000.

Usage Guidelines

This command limits the memory resources available for parsing VoiceXML documents, preventing large documents from consuming excessive system memory. Increasing the maximum memory size for the VoiceXML tree enables calls to use larger VoiceXML documents. If a VoiceXML document exceeds the limit, the gateway aborts the document execution and the **debug vxml error** command displays a "vxml malloc fail" error.

Note In Cisco IOS Release 12.3(4)T and later releases, less memory is consumed when parsing a VoiceXML document because the document is not stored by the VoiceXML tree.

Examples

The following example sets the maximum memory size to 128 KB:

```
vxml tree memory 128
```

Related Commands

Command	Description
debug vxml error	Displays VoiceXML application error messages.
ivr prompt memory	Sets the maximum amount of memory that dynamic audio files (prompts) occupy in memory.
ivr record memory system	Sets the maximum amount of memory for storing all voice recordings on the gateway.

vxml version 2.0

To enable VoiceXML 2.0 features, use the **vxml version 2.0** command in global configuration mode. To return to the default, use the **no** form of this command.

vxml version 2.0

no vxml version 2.0

Syntax Description

This command has no arguments or keywords.

Command Default

The default VoiceXML behavior is compatible with versions earlier than W3C VoiceXML 2.0 Specification .

Command Modes

Global configuration

Command History

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

Usage Guidelines

This command enables the following VoiceXML features:

- An audio error event, error.badfetch, is not thrown when an audio file cannot be played, for instance, because the file is in an unsupported format, the src attribute references an invalid URI, or the expr attribute evaluates to an invalid URI.
- Support for the beep attribute of the <record> element.
- Blind transfer compliant with *W3C VoiceXML 2.0* and not the same as consultation transfer.
- Compatibility with W3C VoiceXML 2.0 Specification.
- A semantic error is generated if an undeclared variable is used. You must declare variables before using them.

Examples

The following example enables VoiceXML version 2.0 features:

```
Router(config)# vxml version 2.0
```