show aal2 profile through show call filter match-list

show aal2 profile

To display the ATM adaptation layer 2 (AAL2) profiles configured on the system, use the **show aal2 profile** command in privileged EXEC mode.

show aal2 profile all {**itut** *profile-number*| **atmf** *profile-number*| **custom** *profile-number*}

Syntax Description

all	Displays ITU-T, ATMF, and custom AAL2 profiles configured on the system.
itut	Displays ITU-T profiles configured on the system.
atmf	Displays ATMF profiles configured on the system.
custom	Displays custom profiles configured on the system.
profile -number	AAL2 profile number to display. Choices are as follows: For ITU-T: • 1 = G.711 u-law • 2 = G.711 u-law with silence insertion descriptor (SID) • 7 = G.711 u-law and G.729ar8 For ATMF: None. ATMF is not supported. For custom: • 100 = G.711 u-law and G.726r32 • 110 = G.711 u-law, G.726r32, and G.729ar8

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.1(1)XA	This command was introduced on the Cisco MC3810.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.2(2)T	This command was implemented on the Cisco 7200 series.

Usage Guidelines

This command applies to AAL2 VoATM applications on the Cisco 7200 series routers.

Examples

The following command displays all of the configured profiles in the system:

```
Router# show aal2 profile all
Printing all the Profiles in the system
Profile Type: ITUT Profile Number: 1 SID Support: 0
Red enable: 1 Num entries: 1
Coding type: g711ulaw Packet length: 40 UUI min: 0 UUI max: 15
Profile Type: ITUT Profile Number: 2 SID Support: 1
Red enable: 1 Num entries: 1
Coding type: g711ulaw Packet length: 40 UUI min: 0 UUI max: 15
Profile Type: custom Profile Number: 100 SID Support: 1
Red enable: 1 Num entries: 2
Coding type: g711ulaw Packet length: 40 UUI min: 0 UUI max: 7
Coding type: g726r32 Packet length: 40 UUI min: 8 UUI max: 15
Profile Type: ITUT Profile Number: 7 SID Support: 1
Red enable: 1 Num entries: 2
Coding type: g711ulaw Packet length: 40 UUI min: 0 UUI max: 15
Coding type: g729ar8 Packet length: 10 UUI min: 0 UUI max: 15
Profile Type: custom Profile Number: 110 SID Support: 1
Red enable: 1 Num entries: 3
Coding type: g711ulaw Packet length: 40 UUI min: 0 UUI max: 7
Coding type: g726r32 Packet length: 40 UUI min: 8 UUI max: 15
Coding type: g729ar8 Packet length: 30 UUI min: 8 UUI max: 15
```

The table below describes significant fields shown in this output.

Table 1: show aal2 profile all Field Descriptions

Field	Description
Coding type	Voice compression algorithm.
ITUT Profile Number	Predefined combination of one or more codec types configured for a digital signal processor (DSP).
Num entries	Number of profile elements.
Packet length	Sample size.
Profile Type	Category of codec types configured on DSP. Possible types are ITU-T, ATMF, and custom.
Red enable	Redundancy for type 3 packets.
SID Support	Silence insertion descriptor.
UUI max	Maximum sequence number on the voice packets.
UUI min	Minimum sequence number on the voice packets.

Related Commands

Command	Description
codec aal2-profile	Sets the codec profile for a DSP on a per-call basis.

show atm video-voice address

To display the network service access point (NSAP) address for the ATM interface, enter the **show atm video-voice address**command inprivileged EXEC mode.

show atm video-voice address

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

Use this command to review ATM interface NSAP addresses that have been assigned with the atm video aesa command and to ensure that ATM management is confirmed for those addresses.

Examples

The following example displays ATM interface NSAP addresses:

```
Router# show atm video-voice address
nsap address                                      type          ilmi status
47.0091810000000002F26D4901.00107B4832E1.FE   VOICE_AAL5    Confirmed
47.0091810000000002F26D4901.00107B4832E1.C8   VIDEO_AAL1    Confirmed
```

The table below describes the significant fields shown in the output.

Table 2: show atm video-voice address Field Descriptions

Field	Description
NSAP address	NSAP address for the ATM interface.
Type	Type of ATM interface.
ILMI status	Integrated Local management Interface (ILMI) protocol status for the ATM interface.

Related Commands

Command	Description
codec aal2-profile	Sets the codec profile for a DSP on a per-call basis.

show auto-config

To display the current status of auto-configuration applications, use the **show auto-config**command in privileged EXEC mode.

show auto-config [**application sccp**]

Syntax Description

application sccp	Displays the current status of only the Skinny Client Control Protocol (SCCP) application.

Command Modes

Privileged EXEC (#)

Command History

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

Examples

The following is sample output from **show auto-config** command:

```
Router# show auto-config application sccp
 auto-config application: sccp
 auto-config admin state: ENABLED & ACTIVE
 download retries: (3)
 download timeout: no timeout, continuous retry
 server(s):  172.19.240.41 172.19.240.40 172.19.240.42
Configuration Download statistics:
        Download Attempted            : 2
          Download Successful         : 2
          Download Failed             : 0
        Configuration Attempted       : 2
          Configuration Successful    : 2
          Configuration Failed(parsing): 0
          Configuration Failed(config) : 0
Configuration Error History:
```

The table below describes the significant fields shown in the display.

Table 3: show auto-config Field Descriptions

Field	Description
ENABLED	Shows auto-config application: SCCP is enabled.
ACTIVE	Shows the SCCP application has registered to use auto-configuration.

Field	Description
timeout	Shows timeout is set to 0, continuous retry without timeout.

Related Commands

Command	Description
auto-config	Enables auto-configuration or enters auto-config application configuration mode for the SCCP application.
debug auto-config	Enables debugging for auto-configuration applications.
debug sccp config	Enables SCCP event debugging.

show backhaul-session-manager group

To display the status, statistics, or configuration for a particular session group or all available session groups, use the **show backhaul-session-manager group**command in privileged EXEC mode.

show backhaul-session-manager group {**status**| **stats**| **cfg**} {**all**| **name** *group-name*}

Syntax Description

status	Status for available session groups.
stats	Statistics for available session groups.
cfg	Configuration for available session groups.
all	Specified parameters for all session groups.
name *group -name*	A particular session group.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.1(1)T	This command was introduced on the Cisco AS5300.
12.2(2)T	This command was implemented on the Cisco 7200 series.
12.2(4)T	This command was implemented on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.2(2)XB	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and was implemented on the Cisco IAD2420 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release.

Examples

The following example displays statistics for all session groups:

```
Router# show backhaul-session-manager group stats all
Session-Group grp1 statistics
```

```
   Successful Fail-Overs      :0
   Un-Successful Fail-Over attempts:0
   Active Pkts receive count  :0
   Standby Pkts receive count :0
   Total PDUs dispatch err    :0
```

The following example displays the current configuration for all session groups:

```
Router# show backhaul-session-manager group cfg all
Session-Group
   Group Name :grp1
   Set Name   :set1
   Sessions   :3
    Dest:10.5.0.3 8304  Local:10.1.2.15 8304  Priority:0
    Dest:10.5.0.3 8300  Local:10.1.2.15 8300  Priority:2
    Dest:10.5.0.3 8303  Local:10.1.2.15 8303  Priority:2
    RUDP Options
      timer cumulative ack :100
      timer keepalive      :1000
      timer retransmit     :300
      timer transfer state :2000
      receive max          :32
      cumulative ack max   :3
      retrans max          :2
      out-of-sequence max  :3
      auto-reset max       :5
```

The following example displays the current state of all session groups. The group named "grp1" belongs to the set named "set1".

```
Router# show backhaul-session-manager group status all
Session-Group
Group Name   :grp1
   Set Name     :set1
   Status       :Group-OutOfService
   Status (use) :Group-None
```

The table below describes the significant fields shown in the output.

Table 4: show backhaul-session-manager group Field Descriptions

Field	Descrption
RUDP Options	Reliable User datagram Protocol (RUDP) options.
Status	One of the following: • Group-OutOfService--No session in the group has been established. • Group-Inservice--At least one session in the group has been established.
Status (use)	One of the following: • Group-Standby--The virtual switch controller (VSC) connected to the other end of this group goes into standby mode. • Group-Active--The VSC connected to the other end of this group is the active VSC. • Group-None--The VSC has not yet declared its intent.

Related Commands

Command	Description
show backhaul-session-manager session	Displays status, statistics, or configuration of sessions.
show backhaul-session-manager set	Displays session groups associated with a specific session set or all session sets.

show backhaul-session-manager session

To display various information about a session or sessions, use the **show backhaul-session-manager session** command in privileged EXEC mode.

show backhaul-session-manager session {**all**| **ip** *ip-address*}

Syntax Description

all	Information is displayed about all available sessions.
ip	Information is displayed about the session associated with this IP address only.
ip -address	IP address of the local or remote session.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.1(1)T	This command was introduced on the Cisco AS5300.
12.2(2)T	This command was implemented on the Cisco 7200 series.
12.2(4)T	This command was implemented on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.2(2)XB	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and was implemented on the Cisco IAD2420 series. Support for the Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command was implemented on the Cisco AS5350, Cisco AS5400, and Cisco AS5850.

Examples

The following command displays information for all available sessions:

```
Router# show backhaul-session-manager session all
Session information --
Session-id:35
  Group:grp1  /*this session belongs to the group named 'grp1' */
Configuration:
     Local:10.1.2.15      , port:8303
```

```
    Remote:10.5.0.3       , port:8303
  Priority:2
  RUDP Option:Client, Conn Id:0x2
State:
  Status:OPEN_WAIT, Use-status:OOS,  /*see explanation below */
Statistics:
  # of resets:0
  # of auto_resets 0
  # of unexpected RUDP transitions (total) 0
  # of unexpected RUDP transitions (since last reset) 0
  Receive pkts -  Total:0 , Since Last Reset:0
  Recieve failures -  Total:0 ,Since Last Reset:0
  Transmit pkts - Total:0, Since Last Reset:0
  Transmit Failures (PDU Only)
         Due to Blocking (Not an Error) - Total:0, Since Last Reset:0
         Due to causes other than Blocking - Total:0, Since Last
Reset:0
  Transmit Failures (NON-PDU Only)
         Due to Blocking(Not an Error) - Total:0, Since Last Reset:0
         Due to causes other than Blocking - Total:0, Since Last
Reset:0
  RUDP statistics
         Open failures:0
         Not ready failures:0
         Conn Not Open failures:0
         Send window full  failures:0
         Resource unavailble failures:0
         Enqueue failures:0
```

The table below describes significant fields shown in this output.

Table 5: show backhaul-session-manager session Field Descriptions

Field	Description
State	Can be any of the following: • OPEN--The connection is established. • OPEN_WAIT--The connection is awaiting establishment. • OPEN_XFER--Session failover is in progress for this session, which is a transient state. • CLOSE--The session is down, also a transient state. The session waits a fixed amount of time and then moves to OPEN_WAIT.
Use-status	Indicates whether PRI signaling traffic is currently being transported over this session. Can be either of the following: • OOS--The session is not being used to transport signaling traffic. Out of service (OOS) does not indicate if the connection is established. • IS--The session is being used currently to transport all PRI signaling traffic. In service (IS) indicates that the connection is established.

Related Commands

Command	Description
show backhaul-session-manager group	Displays status, statistics, or configuration of a specific session group or all session groups.
show backhaul-session-manager set	Displays session groups associated with a specific session set or all session sets.

show backhaul-session-manager set

To display session groups associated with a specified session set or all session sets, use the **show backhaul-session-manager set**command in privileged EXEC mode.

show backhaul-session-manager set {**all**| **name** *session-set-name*}

Syntax Description

all	All available session sets.
name *session -set -name*	A specified session set.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.1(1)T	This command was introduced on the Cisco AS5300.
12.2(2)T	This command was implemented on the Cisco 7200 series.
12.2(4)T	This command was implemented on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.2(2)XB	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and was implemented on the Cisco IAD2420 series. Support for the Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command is supported on the Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release.

Examples

The following command displays session groups associated with all session sets:

```
Router# show backhaul-session-manager set all
```

Related Commands

Command	Description
show backhaul -session-manager group	Displays status, statistics, or configuration of a specific session group or all session groups.

Command	Description
show backhaul -session-manager session	Displays status, statistics, or configuration of a session or all sessions.

show call accounting-template voice

To display accounting template activity, use the **show call accounting-template voice** command in privileged EXEC mode.

show call accounting-template voice [*acctTempName*| **master**| **qdump**| **summary**]

Syntax Description

acctTempName	(Optional) Name of the accounting template.
master	(Optional) Displays all vendor-specific attributes (VSAs) that are filtered by accounting templates.
qdump	(Optional) Displays template activity in the service and free queues.
summary	(Optional) Lists names of all the accounting templates and the number of attributes in each template currently being used.

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

- The **show call accounting-template voice** command displays the status and attributes defined in each template after it is configured.
- The **show call accounting-template voice** *acctTempName* command displays the status of a specific template and the attributes (VSAs) that are defined for that template.
- The **show call accounting-template voice master** command displays all VSAs that can be filtered by accounting templates.
- The **show call accounting-template voice qdump** command displays template activity in the service (svc) and free queues. It displays the template URL, the number of legs on which a template is active, and the state of a template.
    - After an accounting template is defined, it is put in the svc queue to serve new incoming calls. When a running accounting template is undefined or reloaded during an active call, the template is moved from the svc queue to the free queue and can be reused after all the active calls stop

referencing it. Templates that are reloaded or undefined and that are referenced during an active call are considered to be in a "dirty" state and are called dirty templates.

- To ensure that start and stop records correspond on an active call that is referencing a dirty template, all dirty templates must be kept alive until all active calls referencing that dirty template are released. After all active calls are released, the reloaded templates are applied to the next call.

- The **show call accounting-template voice summary** command displays the current status of all the accounting templates that are configured. It shows if the template was loaded and if it is running successfully.

Examples

The following example displays details about two templates named "cdr1" and "cdr2".

```
Router# show call accounting-template voice
CDR template cdr1 is running
url: tftp://sanjoe/santa/abc/Templates/cdr1.cdr
The last load was successful.
attr:  h323-call-origin (56)
attr:  h323-call-type (57)
attr:  h323-gw-id (65)
attr:  subscriber (79)
attr:  in-portgrp-id (80)
attr:  out-portgrp-id (81)
Totally 6 attrs defined.
CDR template cdr2 is running
url: tftp://sanjoe/santa/abc/Templates/cdr2.cdr
The last load was successful.
attr:  h323-call-origin (56)
attr:  h323-call-type (57)
attr:  h323-connect-time (59)
attr:  h323-disconnect-time (64)
attr:  h323-gw-id (65)
attr:  h323-setup-time (76)
attr:  h323-voice-quality (78)
Totally 7 attrs defined.
```

The following example displays details about the template named "cdr1" only.

```
Router# show call accounting-template voice cdr1
CDR template cdr1 is running
url: tftp://sanjoe/santa/abc/Templates/cdr1.cdr
The last load was successful.
attr:  h323-call-origin (56)
attr:  h323-call-type (57)
attr:  h323-gw-id (65)
attr:  subscriber (79)
attr:  in-portgrp-id (80)
attr:  out-portgrp-id (81)
Totally 6 attrs defined.
```

The following example displays all 64 attributes that can be filtered by a template.

```
Router# show call accounting-template voice master
h323-call-origin
h323-call-type
h323-gw-id
h323-setup-time
h323-connect-time
h323-disconnect-time
h323-disconnect-cause
.
.
.
calling-party-category
originating-line-info
charge-number
```

```
transmission-medium-req
redirecting-number
backward-call-indicators
Totally 64 attributes are filterable.
```

The following example displays template activity in the service queue. Initially, no templates are in the dirty state.

```
Router# show call accounting-template voice qdump
name         url                              is_dirty   no_of_legs
=========================================================================
cdr1         tftp://sanjoe/santa/abc                         0
cdr2         tftp://sanjoe/santa/abc                         0
cdr3         tftp://sanjoe/santa/abc                         0
```

After the templates are reloaded during active calls, the display below shows the templates named "cdr1" and "cdr2" to be in a dirty state.

```
.
.
.
Templates in freeq
cdr1         tftp://sanjoe/santa/abc            dirty       1
cdr2         tftp://sanjoe/santa/abc            dirty       1
```

The following example displays a summary of all configured accounting templates. The template named "cdr3" is not in running mode, either because it has been rejected or because it does not exist at the given URL.

```
Router# show call accounting-template voice summary
name         url                                    last_load  is_running
=========================================================================
cdr1         tftp://sanjoe/santa/abc                success    is running
cdr2         tftp://sanjoe/santa/abc                success    is running
cdr3         tftp://sanjoe/santa/abc                fail       is not running
```

The table below describes the fields shown in the **show call accounting-template voice** display.

Table 6: show call accounting-template voice Field Descriptions

Field	Description
name	Name of the accounting template.
url	Location of the accounting template.
last_load	Describes if the accounting template was successfully or unsuccessfully loaded from its location.
is_running	Describes if the accounting template was activated after it was successfully loaded from its location.
is_dirty	Shows that the accounting template was reloaded during an active call.
no_of_legs	Number of call legs.
attr	Vendor-specific attributes (VSAs) defined in an accounting template.

Related Commands

Command	Description
gw-accounting aaa	Configures a new accounting template.

show call active fax

To display call information for T.37 store-and-forward fax transmissions in progress, use the **show call active fax** command in user EXEC or privileged EXEC mode.

show call active fax [**brief** [**id** *identifier*]| **compact** [**duration** {**less** *seconds*| **more** *seconds*}]| **id** *identifier*]

Syntax Description

brief	(Optional) Displays a truncated version of fax call information.
id *identifier*	(Optional) Displays only the call with the specified *identifier* . Range is a hex value from 1 to FFFF.
compact	(Optional) Displays a compact version of the fax call information.
duration	(Optional) Displays active calls that are longer or shorter than a specified *seconds* value. The arguments and keywords are as follows: • **less** --Displays calls shorter than the *seconds* value. • **more** --Displays calls longer than the *seconds* value. • *seconds* --Elapsed time, in seconds. Range is from 1 to 2147483647. There is no default value.

Command Modes

User EXEC (>) Privileged EXEC (#)

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 2600 series and Cisco 3600 series.
12.0(3)XG	This command was modified. Support for Voice over Frame Relay (VoFR) was added.
12.0(4)XJ	This command was implemented for store-and-forward fax on the Cisco AS5300.
12.0(4)T	This command was implemented on the Cisco 7200 series.
12.0(7)XK	This command was implemented on the Cisco MC3810.

Release	Modification
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.1(3)T	This command was modified. This command was implemented for modem pass-through over VoIP on the Cisco AS5300.
12.1(5)XM	This command was implemented on the Cisco AS5800.
12.1(5)XM2	The command was implemented on the Cisco AS5350 and Cisco AS5400.
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 was not included for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850.
12.2(11)T	Support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850.
12.3(14)T	This command was modified. T.38 fax relay call statistics were made available to Call Detail Records (CDRs) through vendor-specific attributes (VSAs) and added to the call log.
12.4(2)T	This command was modified. The LocalHostname display field was added to the VoIP call leg record.
12.4(15)T	This command was modified. The Port and BearerChannel display fields were added to the TELE call leg record of the command output.
12.4(16)	This command was modified. The Port and BearerChannel display fields were added to the TELE call leg record of the command output.
12.4(22)T	This command was modified. Command output was updated to show IPv6 information.

Usage Guidelines

Use this command to display the contents of the active call table. This command displays information about call times, dial peers, connections, quality of service, and other status and statistical information for T.37 store-and-forward fax calls currently connected through the router. This command works with both on-ramp and off-ramp store-and-forward fax functions.

To display information about fax relay calls in progress, use the **show call active voice** command.

Examples

The following is sample output from the **show call active fax** command:

```
Router# show call active fax
GENERIC:
SetupTime=22021 ms
Index=1
PeerAddress=peer one
PeerSubAddress=
```

```
PeerId=0
PeerIfIndex=0
LogicalIfIndex=0
ConnectTime=24284
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=10
TransmitPackets=0
TransmitBytes=0
ReceivePackets=0
ReceiveBytes=41190
MMOIP:
ConnectionId[0x37EC7F41 0xB0110001 0x0 0x35C34]
CallID=1
RemoteIPAddress=10.0.0.0
SessionProtocol=SMTP
SessionTarget=
MessageId=
AccountId=
ImgEncodingType=MH
ImgResolution=fine
AcceptedMimeTypes=2
DiscardedMimeTypes=1
Notification=None
GENERIC:
SetupTime=23193 ms
Index=1
PeerAddress=527....
PeerSubAddress=
PeerId=3469
PeerIfIndex=157
LogicalIfIndex=30
ConnectTime=24284
CallState=4
CallOrigin=1
ChargedUnits=0
InfoType=10
TransmitPackets=5
TransmitBytes=6513
ReceivePackets=0
ReceiveBytes=0
TELE:
ConnectionId=[0x37EC7F41 0xB0110001 0x0 0x35C34]
CallID=2
Port=3/0/0 (2)
BearerChannel=3/0/0.1
TxDuration=24010 ms
FaxTxDuration=10910 ms
FaxRate=14400
NoiseLevel=-1
ACOMLevel=-1
OutSignalLevel=0
InSignalLevel=0
InfoActivity=0
ERLLevel=-1
SessionTarget=
ImgPages=0
```

The table below provides an alphabetical listing of the fields displayed in the output of the **show call active fax**command and a description of each field.

Table 7: show call active fax Field Descriptions

Field	Description
ACOM Level	Current ACOM level for this call. ACOM is the combined loss achieved by the echo canceler, which is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.
BearerChannel	Identification of the bearer channel carrying the call.
Buffer Drain Events	Total number of jitter buffer drain events.
Buffer Fill Events	Total number of jitter buffer fill events.
CallDuration	Length of the call, in hours, minutes, and seconds, hh:mm:ss.
CallOrigin	Call origin: answer or originate.
CallState	Current state of the call.
ChargedUnits	Total number of charging units that apply to this peer since system startup. The unit of measure for this field is hundredths of second.
CodecBytes	Payload size, in bytes, for the codec used.
CoderTypeRate	Negotiated coder rate. This value specifies the send rate of voice or fax compression to its associated call leg for this call.
ConnectionId	Global call identifier for this gateway call.
ConnectTime	Time, in milliseconds, at which the call was connected.
Consecutive-packets-lost Events	Total number of consecutive (two or more) packet-loss events.
Corrected packet-loss Events	Total number of packet-loss events that were corrected using the RFC 2198 method.
Dial-Peer	Tag of the dial peer sending this call.
EchoCancellerMaxReflector=64	The location of the largest reflector, in milliseconds (ms). The reflector size does not exceed the configured echo path capacity. For example, if 32 ms is configured, the reflector does not report beyond 32 ms.

Field	Description
ERLLevel	Current echo return loss (ERL) level for this call.
FaxTxDuration	Duration of fax transmission from this peer to the voice gateway for this call. You can derive the Fax Utilization Rate by dividing the FaxTxDuration value by the TxDuration value.
GapFillWithInterpolation	Duration of a voice signal played out with a signal synthesized from parameters, or samples of data preceding and following in time because voice data was lost or not received in time from the voice gateway for this call.
GapFillWithPrediction	Duration of the voice signal played out with signal synthesized from parameters, or samples of data preceding in time, because voice data was lost or not received in time from the voice gateway for this call. Examples of such pullout are frame-eraser and frame-concealment strategies in G.729 and G.723.1 compression algorithms.
GapFillWithRedundancy	Duration of a voice signal played out with a signal synthesized from available redundancy parameters because voice data was lost or not received in time from the voice gateway for this call.
GapFillWithSilence	Duration of a voice signal replaced with silence because voice data was lost or not received in time for this call.
GENERIC	Generic or common parameters, that is, parameters that are common for VoIP and telephony call legs.
H323 call-legs	Total H.323 call legs for which call records are available.
HiWaterPlayoutDelay	High-water-mark Voice Playout FIFO Delay during this call, in ms.
Index	Dial peer identification number.
InfoActivity	Active information transfer activity state for this call.
InfoType	Information type for this call; for example, voice or fax.
InSignalLevel	Active input signal level from the telephony interface used by this call.

Field	Description
Last Buffer Drain/Fill Event	Elapsed time since the last jitter buffer drain or fill event, in seconds.
LocalHostname	Local hostnames used for locally generated gateway URLs.
LogicalIfIndex	Index number of the logical interface for this call.
LoWaterPlayoutDelay	Low-water-mark Voice Playout FIFO Delay during this call, in ms.
LowerIFName	Physical lower interface information. Appears only if the medium is ATM, Frame Relay (FR), or High-Level Data Link Control (HDLC).
Media	Medium over which the call is carried. If the call is carried over the (telephone) access side, the entry is TELE. If the call is carried over the voice network side, the entry is either ATM, FR, or HDLC.
Modem passthrough signaling method in use	Indicates that this is a modem pass-through call and that named signaling events (NSEs)--a Cisco-proprietary version of named telephone events in RFC 2833--are used for signaling codec upspeed. The upspeed method is the method used to dynamically change the codec type and speed to meet network conditions. This means that you might move to a faster codec when you have both voice and data calls and then slow down when there is only voice traffic.
NoiseLevel	Active noise level for this call.
OnTimeRvPlayout	Duration of voice playout from data received on time for this call. Derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values.
OutSignalLevel	Active output signal level to the telephony interface used by this call.
PeerAddress	Destination pattern or number associated with this peer.
PeerId	ID value of the peer table entry to which this call was made.

Field	Description
PeerIfIndex	Voice port index number for this peer. For ISDN media, this would be the index number of the B channel used for this call.
PeerSubAddress	Subaddress when this call is connected.
Percent Packet Loss	Total percent packet loss.
Port	Identification of the time-division multiplexing (TDM) voice port carrying the call.
ReceiveBytes	Number of bytes received by the peer during this call.
ReceiveDelay	Average Playout FIFO Delay plus the Decoder Delay during this voice call, in ms.
ReceivePackets	Number of packets received by this peer during this call.
ReleaseSource	Number value of the release source.
RemoteIPAddress	Remote system IP address for the VoIP call.
RemoteUDPPort	Remote system User Datagram Protocol (UDP) listener port to which voice packets are sent.
RoundTripDelay	Voice packet round-trip delay between the local and remote systems on the IP backbone for this call.
SelectedQoS	Selected Resourse Reservation Protocol (RSVP) quality of service (QoS) for this call.
SessionProtocol	Session protocol used for an Internet call between the local and remote routers through the IP backbone.
SessionTarget	Session target of the peer used for this call.
SetupTime	Value of the system UpTime, in milliseconds, when the call associated with this entry was started.
SignalingType	Signaling type for this call; for example, channel-associated signaling (CAS) or common channel signaling (CCS).
SIP call-legs	Total Session Initiation Protocol (SIP) call legs for which call records are available.
Telephony call-legs	Total telephony call legs for which call records are available.

Field	Description
Time between Buffer Drain/Fills	Minimum and maximum durations between jitter buffer drain or fill events, in seconds.
TransmitBytes	Number of bytes sent by this peer during this call.
TransmitPackets	Number of packets sent by this peer during this call.
TxDuration	The length of the call. Appears only if the medium is TELE.
VAD	Whether voice activation detection (VAD) was enabled for this call.
VoiceTxDuration	Duration of voice transmission from this peer to the voice gateway for this call, in ms. Derive the Voice Utilization Rate by dividing the VoiceTxDuration value by the TxDuration value.

The following is sample output from the **show call active fax brief** command:

```
Router# show call active fax brief
<ID>: <start>hs.<index> +<connect> pid:<peer_id> <dir> <addr> <state> \
  tx:<packets>/<bytes> rx:<packets>/<bytes> <state>
IP <ip>:<udp> rtt:<time>ms pl:<play>/<gap>ms lost:<lost>/<early>/<late>
  delay:<last>/<min>/<max>ms <codec>
FR <protocol> [int dlci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
  sig:<on/off> <codec> (payload size)
Tele <int>: tx:<tot>/<v>/<fax>ms <codec> noise:<l> acom:<l> i/o:<l>/<l> dBm

1    : 22021hs.1 +2263 pid:0 Answer wook song active
tx:0/0 rx:0/41190
IP 0.0.0.0 AcceptedMime:2 DiscardedMime:1

1    : 23193hs.1 +1091 pid:3469 Originate 527.... active
tx:10/13838 rx:0/0
Tele : tx:31200/10910/20290ms  noise:-1 acom:-1  i/0:0/0 dBm
```

The following is sample output from the **show call active fax** command displaying T.38 fax relay statistics:

```
Router# show call active fax
Telephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 0
MGCP call-legs: 0
Multicast call-legs: 0
Total call-legs: 1
 GENERIC:
SetupTime=1874690 ms
Index=1
PeerAddress=5551234
PeerSubAddress=
PeerId=3
PeerIfIndex=244
LogicalIfIndex=118
ConnectTime=187875
CallDuration=00:00:44 sec
CallState=4
CallOrigin=2
ChargedUnits=0
```

```
InfoType=fax
TransmitPackets=309
TransmitBytes=5661
ReceivePackets=1124
ReceiveBytes=49189
TELE:
ConnectionId=[0x6B241E98 0xA78111D8 0x8002000A 0xF4107CA0]
IncomingConnectionId=[0x6B241E98 0xA78111D8 0x8002000A 0xF4107CA0]
CallID=1
Port=3/0/0 (1)
BearerChannel=3/0/0.1
TxDuration=2840 ms
VoiceTxDuration=0 ms
FaxTxDuration=0 ms
FaxRate=disable bps
FaxRelayMaxJitBufDepth 346
FaxRelayJitterBufOverflow 0
Initial HS Modulation is V.17/long/14400
Recent HS modulation is V.17/short/14400
Number of pages 1
Direction of transmission is Transmit
Num of Packets TX'ed/RX'ed 932/52
Packet loss conceal is 0
Encapsulation protocol is T.38 (UDPTL)
ECM is DISABLED
NoiseLevel=0
ACOMLevel=0
OutSignalLevel=0
InSignalLevel=0
InfoActivity=0
ERLLevel=0
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=5551234
OriginalCallingOctet=0x80
OriginalCalledNumber=5555678
OriginalCalledOctet=0x80
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0xFF
TranslatedCallingNumber=5551234
TranslatedCallingOctet=0x80
TranslatedCalledNumber=5555678
TranslatedCalledOctet=0x80
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwReceivedCalledNumber=5555678
GwReceivedCalledOctet3=0x80
GwReceivedCallingNumber=5551234
GwReceivedCallingOctet3=0x80
GwReceivedCallingOctet3a=0x0
DSPIdentifier=1/0:0
Telephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 0
MGCP call-legs: 0
Multicast call-legs: 0
Total call-legs: 1
```

The table below provides an alphabetical listing of the fields displayed in the output of the **show call active fax**command for T.38 fax relay statistics and a description of each field.

Table 8: show call active fax Field Descriptions for Significant T.38 Fax Relay Statistics

Field	Description
ACOMLevel	Current ACOM level estimate in 0.1 dB increments. The term ACOM is used in G.165, *General Characteristics of International Telephone Connections and International Telephone Circuits: Echo Cancellers* . ACOM is the combined loss achieved by the echo canceller, which is the sum of the ERL, ERL enhancement, and nonlinear processing loss for the call.
BearerChannel	Identification of the bearer channel carrying the call.
ERLLevel	Current ERL level estimate in 0.1 dB increments.
FaxRate	Fax transmission rate from this peer to the specified dial peer, in bits per second (bps).
FaxRelayJitterBufOverflow	Fax relay jitter buffer overflow, in ms.
FaxRelayMaxJitBufDepth	Fax relay maximum jitter buffer depth, in ms.
FaxTxDuration	Duration of fax transmission from this peer to the voice gateway for this call, in ms.
GwReceivedCalledNumber, GwReceivedCalledOctet3	Call information received at the gateway.
H323 call-legs	Type of call: H.323.
Initial HS Modulation	Initial high speed modulation used.
LogicalIfIndex	Index number of the logical interface for this call.
MGCP call-legs	Type of call: Media Gateway Control Protocol (MGCP).
Multicast call-legs	Type of call: Multicast.
OriginalCallingNumber, OriginalCalling Octet, OriginalCalledNumber, OriginalCalledOctet, OriginalRedirectCalledNumber, OriginalRedirectCalledOctet	Original call information regarding calling, called, and redirect numbers, and octet-3s. Octet-3s are information elements (IEs) of Q.931 that include type of number, numbering plan indicator, presentation indicator, and redirect reason information.
PeerIfIndex	Voice port index number for this peer. For ISDN media, this would be the index number of the B channel used for this call.

Field	Description
Port	Identification of the TDM voice port carrying the call.
Recent HS Modulation	Most recent high-speed modulation used.
SIP call-legs	Type of call: SIP.
Telephony call-legs	Type of call: Telephony.
Total call-legs	Total calls.
TranslatedCallingNumber, TranslatedCallingOctet, TranslatedCalledNumber, TranslatedCalledOctet, TranslatedRedirectCalledNumber, TranslatedRedirectCalledOctet	Translated call information.
TxDuration	Duration of transmit path open from this peer to the voice gateway for this call, in ms.
VoiceTxDuration	Duration of voice transmission from this peer to the voice gateway for this call, in ms.

Related Commands

Command	Description
show call active voice	Displays call information for voice calls that are in progress.
show call history	Displays the call history table.
show call-router routes	Displays the dynamic routes in the cache of the BE.
show call-router status	Displays the Annex G BE status.
show voice port	Displays configuration information about a specific voice port.

show call active media

To display call information for media calls in progress, use the **show call active media** command in user EXEC or privileged EXEC mode.

show call active media [[**brief**] [**id** *identifier*]| **compact** [**duration** {**less** *seconds*| **more** *seconds*}]]

Syntax Description

brief	(Optional) Displays a truncated version of call information.
id *identifier*	(Optional) Displays only the call with the specified *identifier*. The range is a hexadecimal value from 1 to FFFF.
compact	(Optional) Displays a compact version of call information.
duration	(Optional) Displays the call history for the specified time duration.
less *seconds*	(Optional) Displays the call history for shorter duration calls, in seconds. The range is from 1 to 2147483647.
more *seconds*	(Optional) Displays the call history for longer duration calls, in seconds. The range is from 1 to 2147483647.

Command Modes

User EXEC (>) Privileged EXEC (#)

Command History

Release	Modification
12.4(15)T	This command was introduced.
12.4(18)M	This command was modified. The **less**keyword, **more**keyword, and *seconds* argument were added.

Usage Guidelines

Use this command to display the contents of the active call table. This command displays information about call times, dial peers, connections, quality of service, and other status and statistical information for media calls currently connected through the router.

When a media call is no longer active, its record is stored. You can display the record by using the **show call history media**command.

Examples

The following is sample output from the **show call active media**command:

```
Router# show call active media
Telephony call-legs: 0
SIP call-legs: 0
H323 call-legs: 0
Call agent controlled call-legs: 0
SCCP call-legs: 0
Multicast call-legs: 0
Media call-legs: 2
Total call-legs: 2
 GENERIC:
SetupTime=408040 ms
Index=1
PeerAddress=sip:mrcpv2TTSServer@10.5.18.224:5060
PeerSubAddress=
PeerId=2235
PeerIfIndex=185
LogicalIfIndex=0
ConnectTime=408130 ms
CallDuration=00:00:01 sec
CallState=4
CallOrigin=1
ChargedUnits=0
InfoType=speech
TransmitPackets=0
TransmitBytes=0
ReceivePackets=57
ReceiveBytes=9120
VOIP-MEDIA:
ConnectionId[0x6B02FC0C 0xC3511DB 0x8006000B 0x5FDA0EF4]
IncomingConnectionId[0x6B02FC0C 0xC3511DB 0x8006000B 0x5FDA0EF4]
CallID=18
RemoteIPAddress=10.5.18.224
RemoteUDPPort=10000
RemoteSignallingIPAddress=10.5.18.224
RemoteSignallingPort=5060
RemoteMediaIPAddress=10.5.18.224
RemoteMediaPort=10000
RoundTripDelay=0 ms
SelectedQoS=best-effort
tx_DtmfRelay=rtp-nte
FastConnect=FALSE
AnnexE=FALSE
Separate H245 Connection=FALSE
H245 Tunneling=FALSE
SessionProtocol=sipv2
ProtocolCallId=6B0CC055-C3511DB-801BC48C-6A894889@10.5.14.2
SessionTarget=10.5.18.224
OnTimeRvPlayout=0
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=0 ms
LoWaterPlayoutDelay=0 ms
TxPakNumber=0
TxSignalPak=0
TxComfortNoisePak=0
TxDuration=0
TxVoiceDuration=0
RxPakNumber=0
RxSignalPak=0
RxComfortNoisePak=0
RxDuration=0
RxVoiceDuration=0
RxOutOfSeq=0
RxLatePak=0
RxEarlyPak=0
```

```
RxBadProtocol=0
PlayDelayCurrent=0
PlayDelayMin=0
PlayDelayMax=0
PlayDelayClockOffset=0
PlayDelayJitter=0
PlayErrPredictive=0
PlayErrInterpolative=0
PlayErrSilence=0
PlayErrBufferOverFlow=0
PlayErrRetroactive=0
PlayErrTalkspurt=0
OutSignalLevel=0
InSignalLevel=0
LevelTxPowerMean=0
LevelRxPowerMean=0
LevelBgNoise=0
ERLLevel=0
ACOMLevel=0
ErrRxDrop=0
ErrTxDrop=0
ErrTxControl=0
ErrRxControl=0
Source tg label=test5
ReceiveDelay=0 ms
LostPackets=0
EarlyPackets=0
LatePackets=0
SRTP = off
TextRelay = off
VAD = disabled
CoderTypeRate=g711ulaw
CodecBytes=160
Media Setting=flow-through
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=
OriginalCalledOctet=0x0
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=4085254655
TranslatedCallingOctet=0x21
TranslatedCalledNumber=
TranslatedCalledOctet=0xC1
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwOutpulsedCallingNumber=4085254655
GwOutpulsedCallingOctet3=0x21
GwOutpulsedCallingOctet3a=0x81
MediaInactiveDetected=no
MediaInactiveTimestamp=
MediaControlReceived=
LongDurationCallDetected=no
LongDurCallTimestamp=
LongDurcallDuration=
Username=
 GENERIC:
SetupTime=408050 ms
Index=1
PeerAddress=sip:mrcpv2ASRServer@10.5.18.224:5060
PeerSubAddress=
PeerId=2234
PeerIfIndex=184
LogicalIfIndex=0
ConnectTime=408160 ms
CallDuration=00:00:03 sec
CallState=4
CallOrigin=1
ChargedUnits=0
InfoType=speech
TransmitPackets=188
```

```
TransmitBytes=30080
ReceivePackets=0
ReceiveBytes=0
VOIP-MEDIA:
ConnectionId[0x6B02FC0C 0xC3511DB 0x8006000B 0x5FDA0EF4]
IncomingConnectionId[0x6B02FC0C 0xC3511DB 0x8006000B 0x5FDA0EF4]
CallID=19
RemoteIPAddress=10.5.18.224
RemoteUDPPort=10002
RemoteSignallingIPAddress=10.5.18.224
RemoteSignallingPort=5060
RemoteMediaIPAddress=10.5.18.224
RemoteMediaPort=10002
RoundTripDelay=0 ms
SelectedQoS=best-effort
tx_DtmfRelay=rtp-nte
FastConnect=FALSE
AnnexE=FALSE
Separate H245 Connection=FALSE
H245 Tunneling=FALSE
SessionProtocol=sipv2
ProtocolCallId=6B0E94CD-C3511DB-801DC48C-6A894889@10.5.14.2
SessionTarget=10.5.18.224
OnTimeRvPlayout=1000
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=1495 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=100 ms
LoWaterPlayoutDelay=95 ms
TxPakNumber=0
TxSignalPak=0
TxComfortNoisePak=0
TxDuration=0
TxVoiceDuration=0
RxPakNumber=0
RxSignalPak=0
RxComfortNoisePak=0
RxDuration=0
RxVoiceDuration=0
RxOutOfSeq=0
RxLatePak=0
RxEarlyPak=0
RxBadProtocol=0
PlayDelayCurrent=0
PlayDelayMin=0
PlayDelayMax=0
PlayDelayClockOffset=0
PlayDelayJitter=0
PlayErrPredictive=0
PlayErrInterpolative=0
PlayErrSilence=0
PlayErrBufferOverFlow=0
PlayErrRetroactive=0
PlayErrTalkspurt=0
OutSignalLevel=0
InSignalLevel=0
LevelTxPowerMean=0
LevelRxPowerMean=0
LevelBgNoise=0
ERLLevel=0
ACOMLevel=0
ErrRxDrop=0
ErrTxDrop=0
ErrTxControl=0
ErrRxControl=0
Source tg label=test5
ReceiveDelay=100 ms
LostPackets=0
EarlyPackets=0
LatePackets=0
SRTP = off
TextRelay = off
```

```
VAD = disabled
CoderTypeRate=g711ulaw
CodecBytes=160
Media Setting=flow-through
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=
OriginalCalledOctet=0x0
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=4085254655
TranslatedCallingOctet=0x21
TranslatedCalledNumber=
TranslatedCalledOctet=0xC1
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwOutpulsedCallingNumber=4085254655
GwOutpulsedCallingOctet3=0x21
GwOutpulsedCallingOctet3a=0x81
MediaInactiveDetected=no
MediaInactiveTimestamp=
MediaControlReceived=
LongDurationCallDetected=no
LongDurCallTimestamp=
LongDurcallDuration=
Username=
Telephony call-legs: 0
SIP call-legs: 0
H323 call-legs: 0
Call agent controlled call-legs: 0
SCCP call-legs: 0
Multicast call-legs: 0
Media call-legs: 2
Total call-legs: 2
```

The table below describes the significant fields shown in the display.

Table 9: show call active media Field Descriptions

Field	Description
Telephony call-legs	Total telephony call legs for which call records are available.
SIP call-legs	Total session initiation protocol (SIP) call legs for which call records are available.
H323 call-legs	Total H.323 call legs for which call records are available.
Media	Medium over which the call is carried. If the call is carried over the (telephone) access side, the entry is TELE. If the call is carried over the voice network side, the entry is either ATM, FR (for Frame Relay), or HDLC (for High-Level Data Link Control).
GENERIC	Generic or common parameters, that is, parameters that are common for VoIP and telephony call legs.
SetupTime	Value of the system UpTime, in milliseconds, when the call associated with this entry was started.

Field	Description
Index	Dial peer identification number.
PeerAddress	Destination pattern or number associated with this peer.
PeerId	ID value of the peer table entry to which this call was made.
PeerIfIndex	Voice port index number for this peer. For ISDN media, this would be the index number of the B channel used for this call.
LogicalIfIndex	Index number of the logical interface for this call.
ConnectTime	Time, in milliseconds, at which the call was connected.
CallDuration	Length of the call, in hours, minutes, and seconds, hh:mm:ss.
CallOrigin	Call origin: answer or originate.
CallState	Current state of the call.
ChargedUnits	Total number of charging units that apply to this peer since system startup. The unit of measure for this field is hundredths of second.
InfoType	Information type for this call; for example, voice or fax.
TransmitBytes	Number of bytes sent by this peer during this call.
TransmitPackets	Number of packets sent by this peer during this call.
ReceivePackets	Number of packets received by this peer during this call.
ReceiveBytes	Number of bytes received by the peer during this call.
ReceiveDelay	Average Playout FIFO Delay plus the Decoder Delay during this voice call, in ms.
ConnectionId	Global call identifier for this gateway call.
RemoteIPAddress	Remote system IP address for the VoIP call.
RemoteUDPPort	Remote system User Datagram Protocol (UDP) listener port to which voice packets are sent.

Field	Description
SelectedQoS	Selected Resource Reservation Protocol (RSVP) quality of service (QoS) for this call.
SessionTarget	Session target of the peer used for this call.
OnTimeRvPlayout	Duration of voice playout from data received on time for this call. Derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values.
GapFillWithInterpolation	Duration of a voice signal played out with a signal synthesized from parameters, or samples of data preceding and following in time because voice data was lost or not received in time from the voice gateway for this call.
GapFillWithRedundancy	Duration of a voice signal played out with a signal synthesized from available redundancy parameters because voice data was lost or not received in time from the voice gateway for this call.
GapFillWithPrediction	Duration of the voice signal played out with signal synthesized from parameters, or samples of data preceding in time, because voice data was lost or not received in time from the voice gateway for this call. Examples of such pullout are frame-eraser and frame-concealment strategies in G.729 and G.723.1 compression algorithms.
GapFillWithSilence	Duration of a voice signal replaced with silence because voice data was lost or not received in time for this call.
HiWaterPlayoutDelay	High-water-mark Voice Playout FIFO Delay during this call, in ms.
LoWaterPlayoutDelay	Low-water-mark Voice Playout FIFO Delay during this call, in ms.
CodecBytes	Payload size, in bytes, for the codec used.
CoderTypeRate	Negotiated coder rate. This value specifies the send rate of voice or fax compression to its associated call leg for this call.
InSignalLevel	Active input signal level from the telephony interface used by this call.

Field	Description
OutSignalLevel	Active output signal level to the telephony interface used by this call.
ERLLevel	Current echo return loss (ERL) level for this call.
ACOMLevel	Current ACOM level for this call. ACOM is the combined loss achieved by the echo canceler, which is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.
PeerSubAddress	Subaddress when this call is connected.
RoundTripDelay	Voice packet round-trip delay between the local and remote systems on the IP backbone for this call.
SessionProtocol	Session protocol used for an Internet call between the local and remote routers through the IP backbone.
TxDuration	The length of the call. Appears only if the medium is TELE.
VAD	Whether voice activation detection (VAD) was enabled for this call.

Related Commands

Command	Description
show call history media	Displays the call history table.

show call active video

To display call information for Signaling Connection Control Protocol (SCCP), Session Initiation Protocol (SIP), and H.323 video calls in progress, use the **show call active video** command in user EXEC or privileged EXEC mode.

show call active video [[**brief**] [**id** *call-identifier*]| **compact** [**duration** {**less**| **more**} *seconds*]| **echo-canceller** *call-id*| **stats**]

Syntax Description

brief	(Optional) Displays a truncated version of active video call information.
id *call-identifier*	(Optional) Displays only the video calls with the specified identifier. The range is from 1 to FFFF.
compact	(Optional) Displays a compact version of active video call information.
duration	(Optional) Displays call history for the specified time duration.
less	Displays call history for shorter duration calls.
more	Displays call history for longer duration calls.
seconds	Time, in seconds. The range is from 1 to 2147483647.
echo-canceller *call-id*	(Optional) Displays information about the state of the extended echo canceller (EC). The range is from 0 to FFFFFFFF.
stats	(Optional) Displays information about DSP statistics and video quality metrics.

Command Modes

User EXEC (>) Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Cisco IOS Release	Cisco Product	Modification
12.4(11)T	--	This command was modified. Support was added for SIP and H.323 calls.
12.4(16); 12.4(15)T	--	This command was modified. The Port and BearerChannel display fields were added to the TELE call leg record of the command output.
15.1(4)M	Cisco Unified CME 8.6	This command was modified. The **stats** keyword was added.

Usage Guidelines

Use this command to display the contents of the active video call table.

Before you can query the echo state, you need to know the hexadecimal ID. Use the **show call active video brief** command to find the hexadecimal ID.

Examples

The following is sample output from the **show call active video brief**command:

```
Router # show call active video brief
<ID>: <CallID> <start>hs.<index> +<connect> pid:<peer_id> <dir> <addr> <state>
  dur hh:mm:ss tx:<packets>/<bytes> rx:<packets>/<bytes>
 IP <ip>:<udp> rtt:<time>ms pl:<play>/<gap>ms lost:<lost>/<early>/<late>
  delay:<last>/<min>/<max>ms <codec>
 media inactive detected:<y/n> media cntrl rcvd:<y/n> timestamp:<time>
 long duration call detected:<y/n> long duration call duration :<sec> timestamp:<time>
  MODEMPASS <method> buf:<fills>/<drains> loss <overall%> <multipkt>/<corrected>
   last <buf event time>s dur:<Min>/<Max>s
 FR <protocol> [int dlci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
  <codec> (payload size)
 ATM <protocol> [int vpi/vci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
  <codec> (payload size)
 Tele <int> (callID) [channel_id] tx:<tot>/<v>/<fax>ms <codec> noise:<l> acom:<l> i/o:<l>/<l>
 dBm
   video: h320:<type> tx:<video codec> <video pkts>/<video bytes> rx:<video codec> <video
pkts>/<video bytes>
  MODEMRELAY info:<rcvd>/<sent>/<resent> xid:<rcvd>/<sent> total:<rcvd>/<sent>/<drops>
         speeds(bps): local <rx>/<tx> remote <rx>/<tx>
 Proxy <ip>:<audio udp>,<video udp>,<tcp0>,<tcp1>,<tcp2>,<tcp3> endpt: <type>/<manf>
 bw: <req>/<act> codec: <audio>/<video>
  tx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>
 rx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>
Telephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 1
Call agent controlled call-legs: 0
SCCP call-legs: 0
Multicast call-legs: 0
Media call-legs: 0
Total call-legs: 2
141D : 83 165385200ms.1 +3180 pid:6 Answer 2004 active
 dur 00:00:36 tx:1602/1232038768 rx:3237/1192797
 IP 192.0.2.0:5445 SRTP: off rtt:0ms pl:27980/0ms lost:0/0/0 delay:0/0/0ms g711ulaw TextRelay:
 off
 media inactive detected:n media contrl rcvd:n/a timestamp:n/a
 long duration call detected:n long duration call duration:n/a timestamp:n/a
141D : 84 165385200ms.2 +3170 pid:20008 Originate 1008 active
 dur 00:00:36 tx:1698/271680 rx:1796/287360
 Tele 50/0/8 (84) [50/0/8.0] tx:33960/33960/0ms g711ulaw noise:0 acom:0  i/0:0/0 dBm
Telephony call-legs: 1
SIP call-legs: 0
```

```
H323 call-legs: 1
Call agent controlled call-legs: 0
SCCP call-legs: 0
Multicast call-legs: 0
Media call-legs: 0
Total call-legs: 2
```

The following is sample output from the **show call active video**command:

```
Router# show call active video
Telephony call-legs: 4
SIP call-legs: 0
H323 call-legs: 0
Call agent controlled call-legs: 0
SCCP call-legs: 2
Multicast call-legs: 0
Total call-legs: 6
 GENERIC:
SetupTime=169281770 ms
Index=2
PeerAddress=
PeerSubAddress=
PeerId=0
PeerIfIndex=0
LogicalIfIndex=0
ConnectTime=169281770 ms
CallDuration=01:20:44 sec
CallState=2
CallOrigin=1
ChargedUnits=0
InfoType=speech
TransmitPackets=819728
TransmitBytes=571031017
ReceivePackets=796308
ReceiveBytes=566120602
VOIP:
ConnectionId[0x0 0x0 0x0 0x0]
IncomingConnectionId[0x0 0x0 0x0 0x0]
CallID=85
GlobalCallId=[0x0 0x0 0x0 0x0]
CallReferenceId=25666520
CallServiceType=Video Conference
RTP Loopback Call=FALSE RemoteIPAddress=0.0.0.0
RemoteUDPPort=2000
RemoteSignallingIPAddress=0.0.0.0
RemoteSignallingPort=0
RemoteMediaIPAddress=1.4.211.39
RemoteMediaPort=2000
RoundTripDelay=0 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
FastConnect=FALSE
AnnexE=FALSE
Separate H245 Connection=FALSE
H245 Tunneling=FALSE
SessionProtocol=other
ProtocolCallId=
SessionTarget=
SafEnabled=FALSE
OnTimeRvPlayout=0
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=0 ms
LoWaterPlayoutDelay=0 ms
Video Conferee Statistics
ConfereeActualFrameRate=0
ConfereeActualBitrate=934600
ConfereeTotalRxPackets=129853
ConfereeTotalRxBytes=125825024
ConfereeTotalTxPackets=129853
```

```
ConfereeTotalTxBytes=125825085
ConfereeTotalPacketsDropped=313
ConfereeCurrentPacketsDropped=0
ConfereeTotalPacketsOutOfOrder=296
ConfereeCurrentPacketsOutOfOrder=0
ConfereeMaxJitter=0
ConfereeCurJitter=0
ConfereeMaxDelay=0
ConfereeCurDelay=0
ConfereeMaxOutOfSyncDelay=0
ConfereeCurrentOutOfSyncDelay=0
ConfereeFastVideoUpdateRate=0
ConfereeVideoDuration=1076
Video Quality Scores
RxVideoMOSInstant=78/100 (Good)
RxVideoMOSAverage=70/100 (Good)
VIDEO:
VideoTransmitCodec=H264
VideoTransmitPictureWidth=640
VideoTransmitPictureHeight=480
VideoTransmitFrameRate=30
VideoTransmitBitrate=934600 bps
VideoTransmitLevel=2
VideoTransmitProfile=Baseline
VideoTransmitPayloadFormat=RFC3984
VideoTransmitPackets=129853
VideoTransmitBytes=125825085
VideoTransmitDuration=1076 seconds
VideoReceiveCodec=H264
VideoReceivePictureWidth=640
VideoReceivePictureHeight=480
VideoReceiveFrameRate=30
VideoReceiveBitrate=934600 bps
VideoReceiveLevel=2
VideoReceiveProfile=Baseline
VideoReceivePayloadFormat=RFC3984
VideoReceivePackets=129853
VideoReceiveBytes=125825024
VideoReceiveDuration=1076 seconds
VideoCap_Codec=H264
VideoCap_Format=CUSTOM
VideoCap_PictureWidth=640
VideoCap_PictureHeight=480
VideoCap_FrameRate=30
VideoCap_Bitrate=960000 bps
VideoCap_Level=2
VideoCap_Profile=Baseline
VideoCap_PayloadFormat=RFC3984
VideoLostPackets=0
VideoEarlyPackets=0
VideoLatePackets=0
VideoUsedBandwidth=934600
VideoNumberOfChannels=0

PlayoutMode = undefined
PlayoutInitialDelay=0 ms
ReceiveDelay=0 ms
LostPackets=0
EarlyPackets=0
LatePackets=0
SRTP = off
TextRelay = off
VAD = disabled
CoderTypeRate=h264
CodecBytes=0
Media Setting=flow-around
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=
OriginalCalledOctet=0x0
OriginalRedirectCalledNumber=
```

```
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=
TranslatedCallingOctet=0x0
TranslatedCalledNumber=
TranslatedCalledOctet=0x0
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0x0
MediaInactiveDetected=no
MediaInactiveTimestamp=
MediaControlReceived=
LongDurationCallDetected=no
LongDurCallTimestamp=
LongDurcallDuration=
Username=
MlppServiceDomainNW=0 (none)
MlppServiceDomainID=
PrecedenceLevel=0 (PRECEDENCE_LEVEL_NONE)
```

The following shows sample output from the **show call active video stats**command:

```
Router# show call active video stats
<ID>: <CallID> <start>ms.<index> +<connect> +<disc> pid:<peer_id> <direction> <addr>
  dur hh:mm:ss tx:<packets>/<bytes> rx:<packets>/<bytes>
Telephony call-legs: 0
SIP call-legs: 0
H323 call-legs: 0
Call agent controlled call-legs: 0
SCCP call-legs: 1
Multicast call-legs: 0
Total call-legs: 1
0    : 5 *10:54:50.661 PDT Tue Jan 11 2011.2 +0 pid:0 Originate  connecting
 dur 00:17:27 tx:126342/122451295 rx:126640/122453063
Video Conferee Statistics
ConfereeActualFrameRate=0  ConfereeActualBitrate=934300
ConfereeTotalRxPackets=126166  ConfereeTotalRxBytes=122282402
ConfereeTotalTxPackets=126166  ConfereeTotalTxBytes=122282463
ConfereeTotalPacketsDropped=295  ConfereeCurrentPacketsDropped=0
ConfereeTotalPacketsOutOfOrder=278  ConfereeCurrentPacketsOutOfOrder=0
ConfereeMaxJitter=0  ConfereeCurJitter=0
ConfereeMaxDelay=0  ConfereeCurDelay=0
ConfereeMaxOutOfSyncDelay=0  ConfereeCurrentOutOfSyncDelay=0
ConfereeFastVideoUpdateRate=0  ConfereeVideoDuration=1046
Video Quality Scores
RxVideoMOSInstant=78/100 (Good)
  (Compression Degradation: 86%, Network Degradation: 13%, Transcoding Degradation: 0%)
RxVideoMOSAverage=70/100 (Good)
  (Compression Degradation: 93%, Network Degradation: 6%, Transcoding Degradation: 0%)
```

The table below describes the significant fields shown in the display, in alphabetical order.

Table 10: show call active video Field Descriptions

Field	Description
CallDuration	Length of the call, in hours, minutes, and seconds, hh:mm:ss.
CallState	Current state of the call.
Call agent controlled call-legs	Displays call legs for devices that are not telephony endpoints; for example, transcoding and conferencing
ChargedUnits	Total number of charging units that apply to this peer since system startup. The unit of measure for this field is hundredths of a second.

Field	Description
CodecBytes	Payload size, in bytes, for the codec used.
CoderTypeRate	Negotiated coder rate. This value specifies the send rate of voice or fax compression to its associated call leg for this call.
ConnectionId	Global call identifier for this gateway call.
ConnectTime	Time, in milliseconds (ms), during which the call was connected.
EchoCancellerMaxReflector	Size of the largest reflector, in ms. The reflector size cannot exceed the configured echo path capacity. For example, if 32 ms is configured, the reflector does not report capacity beyond 32 ms.
ERLLevel	Current echo return loss (ERL) level for this call.
FaxTxDuration	Duration, in ms, of fax transmission from this peer to the voice gateway for this call. You can derive the Fax Utilization Rate by dividing the FaxTxDuration value by the TxDuration value.
GapFillWithInterpolation	Duration, in ms, of a voice signal played out with a signal synthesized from parameters, or samples of data preceding and following in time because voice data was lost or not received in time from the voice gateway for this call.
GapFillWithRedundancy	Duration, in ms, of a voice signal played out with a signal synthesized from available redundancy parameters because voice data was lost or not received in time from the voice gateway for this call.
GapFillWithPrediction	Duration, in ms, of the voice signal played out with a signal synthesized from parameters, or samples of data preceding in time, because voice data was lost or not received in time from the voice gateway for this call. Examples of such pullout are frame-eraser and frame-concealment strategies in G.729 and G.723.1 compression algorithms.
GapFillWithSilence	Duration, in ms, of a voice signal replaced with silence because voice data was lost or not received in time for this call.
GENERIC	Generic or common parameters, that is, parameters that are common for VoIP and telephony call legs.

Field	Description
H320CallType	Total H320 call types available.
H323 call-legs	Total H.323 call legs for which call records are available.
HiWaterPlayoutDelay	High-water-mark voice playout first in first out (FIFO) delay during this call, in ms.
Index	Dial peer identification number.
InfoActivity	Active information transfer activity state for this call.
InfoType	Information type for this call; for example, voice, speech, or fax.
InSignalLevel	Active input signal level from the telephony interface used by this call.
Last Buffer Drain/Fill Event	Elapsed time since the last jitter buffer drain or fill event, in seconds.
LocalHostname	Local hostnames used for locally generated gateway URLs.
LogicalIfIndex	Index number of the logical interface for this call.
LoWaterPlayoutDelay	Low-water-mark voice playout FIFO delay during this call, in ms.
LowerIFName	Physical lower interface information. Appears only if the medium is ATM, Frame Relay (FR), or High-Level Data Link Control (HDLC).
Media	Medium over which the call is carried. If the call is carried over the (telephone) access side, the entry is TELE. If the call is carried over the voice network side, the entry is either ATM, FR, or HDLC.
Multicast call-legs	Total multicast call legs for which call records are available.
NoiseLevel	Active noise level for this call.
OnTimeRvPlayout	Duration of voice playout from data received on time for this call. Derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values.

Field	Description
OutSignalLevel	Active output signal level to the telephony interface used by this call.
PeerAddress	Destination pattern or number associated with this peer.
PeerId	ID value of the peer table entry to which this call was made.
PeerIfIndex	Voice port index number for this peer. For ISDN media, this would be the index number of the B channel used for this call.
PeerSubAddress	Subaddress when this call is connected.
ReceiveBytes	Number of bytes received by the peer during this call.
ReceiveDelay	Average playout FIFO delay plus the decoder delay during this voice call, in ms.
ReceivePackets	Number of packets received by this peer during this call.
RemoteIPAddress	Remote system IP address for the VoIP call.
RemoteUDPPort	Remote system User Datagram Protocol (UDP) listener port to which voice packets are sent.
RoundTripDelay	Voice packet round-trip delay, in ms, between the local and remote systems on the IP backbone for this call.
SCCP call-legs	Call legs for SCCP telephony endpoints.
SelectedQoS	Selected Resource Reservation Protocol (RSVP) quality of service (QoS) for this call.
SessionProtocol	Session protocol used for an Internet call between the local and remote routers through the IP backbone.
SessionTarget	Session target of the peer used for this call.
SetupTime	Value of the system UpTime, in milliseconds, when the call associated with this entry was started.
SIP call-legs	Total SIP call legs for which call records are available.
Telephony call-legs	Total telephony call legs for which call records are available.

Field	Description
Total call-legs	Total number of call legs for the call.
TransmitBytes	Number of bytes sent by this peer during this call.
TransmitPackets	Number of packets sent by this peer during this call.
TxDuration	The length of the call. Appears only if the medium is TELE.
VAD	Whether voice activation detection (VAD) was enabled for this call.
VideoCap_Annex	Extension of the video stream; for example, annex D1 and E.
VideoCap_Bitrate	Negotiated bitrate of the video stream; for example, 128000 b/s.
VideoCap_Codec	Codec for the active video call.
VideoCap_Format	Video format for the active video call.
VideoCap_FrameRate	Negotiated frame rate of the video stream; for example, 15 or 30 f/s.
VideoCap_PictureHeight	Height of the video resolution.
VideoCap_PictureWidth	Width of the video resolution.
VideoEarlyPackets	Number of early packets for a video call.
VideoLatePackets	Number of late packets in a video call.
VideoLostPackets	Number of lost packets in a video call.
VideoNumberOfChannels	Number of channels used for a video call.

Field	Description
Video Quality Score	Instantaneous and average Mean Opinion Score (MOS) for each active call leg. The MOS score is based on the amount of video quality degradation caused by compression distortion and the amount of video quality degradation caused by packet loss. The scale for the MOS score is as follows: • Excellent--(80--100) • Good--(60--80) • Fair--(40--60) • Poor--(20--40) • Bad--(0--20)
VideoReceiveBytes	Number of bytes received in the video call.
VideoReceiveCodec	Type of video codec used in the receiving stream.
VideoReceivePackets	Number of packets received in the video call.
VideoTransmitBytes	Number of bytes transmitted in the video call.
VideoTransmitCodec	Type of video codec used in the transmission stream.
VideoTransmitPackets	Number of packets transmitted in the video call.
VideoUsedBandwidth	Bandwidth, in kbps, used for a video call.
VoiceTxDuration	Duration of voice transmission from this peer to the voice gateway for this call, in milliseconds. Derive the Voice Utilization Rate by dividing the VoiceTxDuration value by the TxDuration value.

Related Commands

Command	Description
show call history video	Displays call history information for SCCP video calls.

show call active voice

To display call information for voice calls in progress, use the **show call active voice** command in user EXEC or privileged EXEC mode.

show call active voice [[**brief**] [**long-dur-call-inactive**| **media-inactive**] [**called-number** *number*| **calling-number** *number*] [**id** *call-identifier*]| **compact** [**duration** {**less**| **more**} *seconds*]| **dest-route-string** *tag*| **echo-canceller** {*hexadecimal-id*| **port** *slot-number*| **summary**}| **long-dur-call** [**called-number** *number*| **calling-number** *number*]| **redirect tbct**| **stats**]

Syntax in Cisco IOS Release 12.2(33)SXH and Subsequent 12.2SX Releases

show call active [brief]

Syntax Description

brief	(Optional) Displays a truncated version of call information.
long-dur-call-inactive	(Optional) Displays long duration calls that are detected and notified.
media-inactive	(Optional) Displays information about inactive media that have been detected.
called-number *number*	(Optional) Displays a specific called number pattern.
calling-number *number*	(Optional) Displays a specific calling number pattern.
id *call-identifier*	(Optional) Displays only the call with the specified *call-identifier* value. The range is from 1 to FFFF.
compact	(Optional) Displays a compact version of call information.
duration	(Optional) Displays the call history for the specified time duration.
less *seconds*	Displays the call history for shorter duration calls, in seconds. The range is from 1 to 2147483647.
more *seconds*	Displays the call history for longer duration calls, in seconds. The range is from 1 to 2147483647.
dest-route-string *tag*	(Optional) Displays only the call with the specified destination route *tag* value. The range is from 1 to 10000.
echo-canceller	(Optional) Displays information about the state of the extended echo canceller (EC).

hexadecimal-id	The hexadecimal ID of an active voice call. The range is from 0x0 to 0xFFFFFFFF.
port *slot-number*	Displays EC details for a specified active voice port. The range varies depending on the voice ports available on the router.
summary	Displays an EC summary for all active voice calls.
long-dur-call	(Optional) Displays long duration calls that are detected and notified.
redirect	(Optional) Displays information about active calls that are being redirected using Release-to-Pivot (RTPvt) or Two B-Channel Transfer (TBCT).
tbct	Displays information about TBCT calls.
stats	(Optional) Displays information about digital signal processing (DSP) voice quality metrics.

Command Modes

User EXEC (>) Privileged EXEC (#)

Command History

Release	Modification
11.3(1)T	This command was introduced.
12.0(3)XG	This command was modified. Support for Voice over Frame Relay (VoFR) was added.
12.0(4)XJ	This command was implemented for store-and-forward fax on the Cisco AS5300.
12.0(4)T	This command was implemented on the Cisco 7200 series.
12.0(7)XK	This command was implemented on the Cisco MC3810.
12.1(3)T	This command was implemented for modem pass-through over VoIP on the Cisco AS5300.
12.1(5)XM	This command was implemented on the Cisco AS5800.
12.1(5)XM2	The command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.

Release	Modification
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T. Support was not included for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850.
12.2(11)T	Support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850.
12.2(13)T	This command was modified. The **echo-canceller**keyword was added. The command output was modified with an extra reflector location when the extended EC is present; the largest reflector location is shown.
12.3(1)	This command was modified. The **redirect** keyword was added.
12.3(4)T	This command was modified. The **called-number**, **calling-number**, and **media-inactive**keywords were added.
12.3(14)T	This command was modified. New output relating to Skinny Client Control Protocol (SCCP), SCCP Telephony Control Application (STCAPP), and modem pass-through traffic was added.
12.4(2)T	This command was modified. The LocalHostname display field was added to the VoIP call leg record and command output was enhanced to display modem relay physical layer and error correction protocols.
12.4(4)T	This command was modified. The **long-dur-call** keyword was added.
12.4(11)XW	This command was modified. The **stats** keyword was added.
12.4(15)T	This command was modified. The Port and BearerChannel display fields were added to the TELE call leg record of the command output.
12.2(33)SXH	This command was integrated into Cisco IOS Release 12.2(33)SXH.
12.4(16)	This command was modified. The Port and BearerChannel display fields were added to the TELE call leg record of the command output.
12.4(22)T	This command was modified. Command output was updated to show IPv6 information.
15.3(3)M	This command was modified. The **dest-route-string** keyword was added.
Cisco IOS XE Release 3.10S	This command was integrated into Cisco IOS XE Release 3.10S.

Usage Guidelines

Use this command to display the contents of the active voice call table. This command displays information about call times, dial peers, connections, and quality of service, and other status and statistical information for voice calls currently connected through the router.

Before you can query the echo state, you need to know the hexadecimal ID. To find the hexadecimal ID, enter the **show call active voice brief** command or use the **show voice call status**command.

When the extended EC is present, the **show call active voice** command displays the contents of the Ditech EC_CHAN_CTRL structure. The table below contains names and descriptions of the fields in the EC_CHAN_CTRL structure. The table also provides a listing of the information types associated with this command.

Use the **show call active voice dest-route-string** command to display only the active voice calls with call routing configured using specified destination-route-string globally and in dial-peer level.

Table 11: EC_CHAN_CTRL Field Descriptions

Symbol	Field	Description
BYP0	Channel bypass	• 1 = Transparent bypass; EC is disabled. • 0 = Cancel; EC is enabled.
TAIL3	Max tail	• 0 = 24 milliseconds. • 1 = 32 milliseconds. • 2 = 48 milliseconds. • 3 = 64 milliseconds. **Note** This field should be set just greater than the anticipated worst round-trip tail delay.
REC3	Residual echo control	• 0 = Cancel only; echo is the result of linear processing; no nonlinear processing is applied. • 1 = Suppress residual; residual echo is zeroed; simple nonlinear processing is applied (you might experience "dead air" when talking). • 2 = Reserved. • 3 = Generate comfort noise (default).
FRZ0	h-register hold	1 = Freezes h-register; used for testing.

Symbol	Field	Description
HZ0	h-register clear	Sending the channel command with this bit set clears the h-register.
TD3	Modem tone disable	• 0 = Ignore 2100 Hz modem answer tone. • 1 = G.164 mode (bypass canceller if 2100 Hz tone). • 2 = R. • 3 = G.165 mode (bypass canceller for phase reversing tone only).
ERL0	Echo return loss	• 0 = 6 decibel (dB). • 1 = 3 dB. • 2 = 0 dB. • 3 = R. Worst echo return loss (ERL) situation in which canceller still works.
HLC1	High level compensation	• 0 = No attenuation. • 1 = 6 dB if clipped. On loud circuits, the received direction can be attenuated 6 dB if clipping is observed.
R0	Reserved	Must be set to 0 to ensure compatibility with future releases.

Use the **show call active voice redirect tbct**command to monitor any active calls that implement RTPvt or TBCT.

When a call is no longer active, its record is stored. You can display the record by using the **show call history voice**command.

Examples

The following is sample output from the **show call active voice** command for modem relay traffic:

```
Router# show call active voice
Modem Relay Local Rx Speed=0 bps
Modem Relay Local Tx Speed=0 bps
Modem Relay Remote Rx Speed=0 bps
Modem Relay Remote Tx Speed=0 bps
Modem Relay Phy Layer Protocol=v34
```

```
Modem Relay Ec Layer Protocol=v14
SPRTInfoFramesReceived=0
SPRTInfoTFramesSent=0
SPRTInfoTFramesResent=0
SPRTXidFramesReceived=0
SPRTXidFramesSent=0
SPRTTotalInfoBytesReceived=0
SPRTTotalInfoBytesSent=0
SPRTPacketDrops=0
```

The table below describes the significant fields shown in the display.

Table 12: show show call active voice Field Descriptions

Field	Description
Modem Relay Local Rx Speed	Download speed, in bits per second, of the local modem relay.
Modem Relay Local Tx Speed	Upload speed of the local modem relay.
Modem Relay Remote Rx Speed	Download speed of the remote modem relay.
Modem Relay Remote Tx Speed	Upload speed of the remote modem relay.
Modem Relay Phy Layer Protocol	Physical protocol of the modem relay.
Modem Relay Ec Layer Protocol	EC layer protocol of the modem relay.
SPRTInfoFramesReceived	Total number of simple packet relay transport (SPRT) protocol frames received.
SPRTInfoTFramesSent	Total number of SPRT frames sent.
SPRTInfoTFramesResent	Total number of SPRT frames sent again.
SPRTXidFramesReceived	Total number of SPRTS ID frames received.
SPRTXidFramesSent	Total number of SPRTS ID frames sent.
SPRTTotalInfoBytesReceived	Total number of SPRT bytes received.
SPRTTotalInfoBytesSent	Total number of SPRT bytes sent.
SPRTPacketDrops	Total number of SPRT packets dropped.

The following is sample output from the **show call active voice**command:

```
Router# show call active voice
Telephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 1
Call agent controlled call-legs: 0
SCCP call-legs: 0
Multicast call-legs: 0
Total call-legs: 2
```

```
 GENERIC:
SetupTime=1072620 ms
Index=1
PeerAddress=9193927582
PeerSubAddress=
PeerId=8
PeerIfIndex=19
LogicalIfIndex=0
ConnectTime=1078940 ms
CallDuration=00:00:51 sec
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=speech
TransmitPackets=1490
TransmitBytes=0
ReceivePackets=2839
ReceiveBytes=56780
VOIP:
ConnectionId[0xE28B6D1D 0x3D9011D6 0x800400D0 0xBA0D97A1]
IncomingConnectionId[0xE28B6D1D 0x3D9011D6 0x800400D0 0xBA0D97A1]
CallID=1
RemoteIPAddress=10.44.44.44
RemoteUDPPort=17096
RemoteSignallingIPAddress=10.44.44.44
RemoteSignallingPort=56434
RemoteMediaIPAddress=10.44.44.44
RemoteMediaPort=17096
RoundTripDelay=6 ms
SelectedQoS=best-effort
tx_DtmfRelay=h245-signal
FastConnect=TRUE
AnnexE=FALSE
Separate H245 Connection=FALSE
H245 Tunneling=TRUE
SessionProtocol=cisco
ProtocolCallId=
SessionTarget=
OnTimeRvPlayout=54160
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=70 ms
LoWaterPlayoutDelay=60 ms
TxPakNumber=1490
TxSignalPak=0
TxComfortNoisePak=1
TxDuration=54240
TxVoiceDuration=29790
RxPakNumber=2711
RxSignalPak=0
RxDuration=0
TxVoiceDuration=54210
VoiceRxDuration=54160
RxOutOfSeq=0
RxLatePak=0
RxEarlyPak=0
PlayDelayCurrent=60
PlayDelayMin=60
PlayDelayMax=70
PlayDelayClockOffset=212491899
PlayDelayJitter=0 ms
PlayErrPredictive=0
PlayErrInterpolative=0
PlayErrSilence=0
PlayErrBufferOverFlow=10
PlayErrRetroactive=0
PlayErrTalkspurt=0
OutSignalLevel=-57
InSignalLevel=-51
LevelTxPowerMean=0
LevelRxPowerMean=-510
```

```
LevelBgNoise=0
ERLLevel=16
ACOMLevel=16
ErrRxDrop=0
ErrTxDrop=0
ErrTxControl=0
ErrRxControl=0
ReceiveDelay=60 ms
LostPackets=0
EarlyPackets=0
LatePackets=0
SRTP = off
VAD = enabled
CoderTypeRate=g729r8
CodecBytes=20
Media Setting=flow-through
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=9193927582
OriginalCallingOctet=0x21
OriginalCalledNumber=93615494
OriginalCalledOctet=0xC1
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0xFF
TranslatedCallingNumber=9193927582
TranslatedCallingOctet=0x21
TranslatedCalledNumber=93615494
TranslatedCalledOctet=0xC1
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwReceivedCalledNumber=93615494
GwReceivedCalledOctet3=0xC1
GwReceivedCallingNumber=9193927582
GwReceivedCallingOctet3=0x21
GwReceivedCallingOctet3a=0x81
MediaInactiveDetected=no
MediaInactiveTimestamp=
MediaControlReceived=
Username=
 GENERIC:
SetupTime=1072760 ms
Index=1
PeerAddress=93615494
PeerSubAddress=
PeerId=9
PeerIfIndex=18
LogicalIfIndex=4
ConnectTime=1078940 ms
CallDuration=00:00:53 sec
CallState=4
CallOrigin=1
ChargedUnits=0
InfoType=speech
TransmitPackets=2953
TransmitBytes=82684
ReceivePackets=1490
ReceiveBytes=29781
TELE:
ConnectionId=[0xE28B6D1D 0x3D9011D6 0x800400D0 0xBA0D97A1]
IncomingConnectionId=[0xE28B6D1D 0x3D9011D6 0x800400D0 0xBA0D97A1]
CallID=2
Port=3/0/0 (1)
BearerChannel=3/0/0.2
TxDuration=59080 ms
VoiceTxDuration=29790 ms
FaxTxDuration=0 ms
CoderTypeRate=g729r8
NoiseLevel=-54
ACOMLevel=16
OutSignalLevel=-57
InSignalLevel=-51
InfoActivity=1
ERLLevel=16
```

```
EchoCancellerMaxReflector=8
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
AlertTimepoint=1073340 ms
OriginalCallingNumber=9193927582
OriginalCallingOctet=0x21
OriginalCalledNumber=93615494
OriginalCalledOctet=0xC1
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0xFF
TranslatedCallingNumber=9193927582
TranslatedCallingOctet=0x21
TranslatedCalledNumber=93615494
TranslatedCalledOctet=0xC1
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwReceivedCalledNumber=93615494
GwReceivedCalledOctet3=0xC1
GwOutpulsedCalledNumber=93615494
GwOutpulsedCalledOctet3=0xC1
GwReceivedCallingNumber=9193927582
GwReceivedCallingOctet3=0x21
GwReceivedCallingOctet3a=0x81
GwOutpulsedCallingNumber=9193927582
GwOutpulsedCallingOctet3=0x21
GwOutpulsedCallingOctet3a=0x81
DSPIdentifier=3/1:1
Telephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 1
Call agent controlled call-legs: 0
SCCP call-legs: 0
Multicast call-legs: 0
Total call-legs: 2
```

The first table above and the table below describe the significant fields shown in the display, in alphabetical order.

Table 13: show call active voice Field Descriptions

Field	Description
CallDuration	Length of the call, in hours, minutes, and seconds, hh:mm:ss.
CallState	Current state of the call.
Call agent controlled call-legs	Displays call legs for devices that are not telephony endpoints; for example, transcoding and conferencing
ChargedUnits	Total number of charging units that apply to this peer since system startup. The unit of measure for this field is hundredths of second.
CodecBytes	Payload size, in bytes, for the codec used.
CoderTypeRate	Negotiated coder rate. This value specifies the send rate of voice or fax compression to its associated call leg for this call.
ConnectionId	Global call identifier for this gateway call.

Field	Description
ConnectTime	Time, in ms, during which the call was connected.
EchoCancellerMaxReflector	Size of the largest reflector, in ms. The reflector size cannot exceed the configured echo path capacity. For example, if 32 ms is configured, the reflector does not report capacity beyond 32 ms.
ERLLevel	Current echo return loss (ERL) level for this call.
FaxTxDuration	Duration, in ms, of fax transmission from this peer to the voice gateway for this call. You can derive the Fax Utilization Rate by dividing the FaxTxDuration value by the TxDuration value.
GapFillWithInterpolation	Duration, in ms, of a voice signal played out with a signal synthesized from parameters, or samples of data preceding and following in time because voice data was lost or not received in time from the voice gateway for this call.
GapFillWithRedundancy	Duration, in ms, of a voice signal played out with a signal synthesized from available redundancy parameters because voice data was lost or not received in time from the voice gateway for this call.
GapFillWithPrediction	Duration, in ms, of the voice signal played out with a signal synthesized from parameters, or samples of data preceding in time, because voice data was lost or not received in time from the voice gateway for this call. Examples of such pullout are frame-eraser and frame-concealment strategies in G.729 and G.723.1 compression algorithms.
GapFillWithSilence	Duration, in ms, of a voice signal replaced with silence because voice data was lost or not received in time for this call.
GENERIC	Generic or common parameters; that is, parameters that are common for VoIP and telephony call legs.
H320CallType	Total H320 call types available.
H323 call-legs	Total H.323 call legs for which call records are available.
HiWaterPlayoutDelay	High-water-mark voice playout first in first out (FIFO) delay during this call, in ms.
Index	Dial peer identification number.

Field	Description
InfoActivity	Active information transfer activity state for this call.
InfoType	Information type for this call; for example, voice, speech, or fax.
InSignalLevel	Active input signal level from the telephony interface used by this call.
LogicalIfIndex	Index number of the logical interface for this call.
LoWaterPlayoutDelay	Low-water-mark voice playout FIFO delay during this call, in ms.
Media	Medium over which the call is carried. If the call is carried over the (telephone) access side, the entry is TELE. If the call is carried over the voice network side, the entry is either ATM, Frame Relay (FR), or High-Level Data Link Control (HDLC).
Multicast call-legs	Total multicast call legs for which call records are available.
NoiseLevel	Active noise level for this call.
OnTimeRvPlayout	Duration of voice playout from data received on time for this call. Derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values.
OutSignalLevel	Active output signal level to the telephony interface used by this call.
PeerAddress	Destination pattern or number associated with this peer.
PeerId	ID value of the peer table entry to which this call was made.
PeerIfIndex	Voice port index number for this peer. For ISDN media, this would be the index number of the B channel used for this call.
PeerSubAddress	Subaddress when this call is connected.
ReceiveBytes	Number of bytes received by the peer during this call.
ReceiveDelay	Average playout FIFO delay plus the decoder delay during this voice call, in ms.

Field	Description
ReceivePackets	Number of packets received by this peer during this call.
RemoteIPAddress	Remote system IP address for the VoIP call.
RemoteUDPPort	Remote system User Datagram Protocol (UDP) listener port to which voice packets are sent.
RoundTripDelay	Voice packet round-trip delay, in ms, between the local and remote systems on the IP backbone for this call.
SCCP call-legs	Call legs for SCCP telephony endpoints.
SelectedQoS	Selected Resource Reservation Protocol (RSVP) quality of service (QoS) for this call.
SessionProtocol	Session protocol used for an Internet call between the local and remote routers through the IP backbone.
SessionTarget	Session target of the peer used for this call.
SetupTime	Value of the system UpTime, in ms, when the call associated with this entry was started.
SIP call-legs	Total SIP call legs for which call records are available.
Telephony call-legs	Total telephony call legs for which call records are available.
Total call-legs	Total number of call legs for the call.
TransmitBytes	Number of bytes sent by this peer during this call.
TransmitPackets	Number of packets sent by this peer during this call.
TxDuration	The length of the call. Appears only if the medium is TELE.
VAD	Whether voice activation detection (VAD) was enabled for this call.
VoiceTxDuration	Duration of voice transmission from this peer to the voice gateway for this call, in ms. Derive the Voice Utilization Rate by dividing the VoiceTxDuration value by the TxDuration value.

The following is sample output from the **show call active voice** command for voice traffic over call-agent controlled call legs. Note that call legs for SCCP telephony endpoints, that is, phones controlled by STCAPP, are displayed under the "Call agent controlled call-legs" field ("SCCP call-legs" displays call legs for devices that are not telephony endpoints; for example, transcoding and conferencing).

```
Router# show call active voice
Telephony call-legs: 2
SIP call-legs: 0
H323 call-legs: 0
Call agent controlled call-legs: 2
SCCP call-legs: 0
Multicast call-legs: 0
Total call-legs: 4
 GENERIC:
SetupTime=1557650 ms
Index=1
PeerAddress=
PeerSubAddress=
PeerId=999100
PeerIfIndex=14
LogicalIfIndex=10
ConnectTime=1562040 ms
CallDuration=00:01:01 sec
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=speech
TransmitPackets=3101
TransmitBytes=519564
ReceivePackets=3094
ReceiveBytes=494572
TELE:
ConnectionId=[0x11B1860C 0x22D711D7 0x8014E4D4 0x8FD15327]
IncomingConnectionId=[0x11B1860C 0x22D711D7 0x8014E4D4 0x8FD15327]
CallID=25
Port=3/0/0 (25)
BearerChannel=3/0/0.1
TxDuration=59670 ms
VoiceTxDuration=59670 ms
FaxTxDuration=0 ms
CoderTypeRate=g711ulaw
NoiseLevel=-12
ACOMLevel=22
OutSignalLevel=-12
InSignalLevel=-11
InfoActivity=1
ERLLevel=22
EchoCancellerMaxReflector=2
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=
OriginalCalledOctet=0x80
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=
TranslatedCallingOctet=0x0
TranslatedCalledNumber=
TranslatedCalledOctet=0x80
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0x0
DSPIdentifier=1/1:1
 GENERIC:
SetupTime=1559430 ms
Index=1
PeerAddress=7702
PeerSubAddress=
PeerId=999100
```

```
PeerIfIndex=14
LogicalIfIndex=11
ConnectTime=1562020 ms
CallDuration=00:01:03 sec
CallState=4
CallOrigin=1
ChargedUnits=0
InfoType=speech
TransmitPackets=3151
TransmitBytes=528900
ReceivePackets=3158
ReceiveBytes=503876
TELE:
ConnectionId=[0x0 0x0 0x0 0x0]
IncomingConnectionId=[0x0 0x0 0x0 0x0]
CallID=26
Port=3/0/0 (26)
BearerChannel=3/0/0.2
TxDuration=60815 ms
VoiceTxDuration=60815 ms
FaxTxDuration=0 ms
CoderTypeRate=g711ulaw
NoiseLevel=-12
ACOMLevel=28
OutSignalLevel=-12
InSignalLevel=-11
InfoActivity=1
ERLLevel=28
EchoCancellerMaxReflector=2
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
AlertTimepoint=1559430 ms
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=
OriginalCalledOctet=0x0
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=7701
TranslatedCallingOctet=0x0
TranslatedCalledNumber=7702
TranslatedCalledOctet=0x0
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0x0
GwOutpulsedCalledNumber=7702
GwOutpulsedCalledOctet3=0x0
GwOutpulsedCallingNumber=7701
GwOutpulsedCallingOctet3=0x0
GwOutpulsedCallingOctet3a=0x0
DSPIdentifier=1/1:2
 GENERIC:
SetupTime=1562040 ms
Index=1
PeerAddress=
PeerSubAddress=
PeerId=0
PeerIfIndex=0
LogicalIfIndex=0
ConnectTime=0 ms
CallDuration=00:00:00 sec
CallState=2
CallOrigin=1
ChargedUnits=0
InfoType=speech
TransmitPackets=3215
TransmitBytes=512996
ReceivePackets=3208
ReceiveBytes=512812
VOIP:
ConnectionId[0x0 0x0 0x0 0x0]
IncomingConnectionId[0x0 0x0 0x0 0x0]
```

```
CallID=27
RemoteIPAddress=10.10.0.0
RemoteUDPPort=17718
RemoteSignallingIPAddress=10.10.0.0
RemoteSignallingPort=0
RemoteMediaIPAddress=10.2.6.10
RemoteMediaPort=17718
RoundTripDelay=0 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
FastConnect=FALSE
AnnexE=FALSE
Separate H245 Connection=FALSE
H245 Tunneling=FALSE
SessionProtocol=other
ProtocolCallId=
SessionTarget=
OnTimeRvPlayout=60640
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=105 ms
LoWaterPlayoutDelay=105 ms
TxPakNumber=3040
TxSignalPak=0
TxComfortNoisePak=0
TxDuration=60815
TxVoiceDuration=60815
RxPakNumber=3035
RxSignalPak=0
RxDuration=0
TxVoiceDuration=60690
VoiceRxDuration=60640
RxOutOfSeq=0
RxLatePak=0
RxEarlyPak=0
PlayDelayCurrent=105
PlayDelayMin=105
PlayDelayMax=105
PlayDelayClockOffset=-1662143961
PlayDelayJitter=0
PlayErrPredictive=0
PlayErrInterpolative=0
PlayErrSilence=0
PlayErrBufferOverFlow=0
PlayErrRetroactive=0
PlayErrTalkspurt=0
OutSignalLevel=-12
InSignalLevel=-11
LevelTxPowerMean=0
LevelRxPowerMean=-115
LevelBgNoise=0
ERLLevel=28
ACOMLevel=28
ErrRxDrop=0
ErrTxDrop=0
ErrTxControl=0
ErrRxControl=0
PlayoutMode = undefined
PlayoutInitialDelay=0 ms
ReceiveDelay=105 ms
LostPackets=0
EarlyPackets=0
LatePackets=0
SRTP = off
VAD = disabled
CoderTypeRate=g711ulaw
CodecBytes=160
Media Setting=flow-around
Modem passthrough signaling method is nse:
Buffer Fill Events = 0
Buffer Drain Events = 0
```

```
Percent Packet Loss = 0
Consecutive-packets-lost Events = 0
Corrected packet-loss Events = 0
Last Buffer Drain/Fill Event = 0sec
Time between Buffer Drain/Fills = Min 0sec Max 0sec
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=
OriginalCalledOctet=0x0
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=
TranslatedCallingOctet=0x0
TranslatedCalledNumber=
TranslatedCalledOctet=0x0
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0x0
MediaInactiveDetected=no
MediaInactiveTimestamp=
MediaControlReceived=
Username=
 GENERIC:
SetupTime=1562040 ms
Index=2
PeerAddress=
PeerSubAddress=
PeerId=0
PeerIfIndex=0
LogicalIfIndex=0
ConnectTime=0 ms
CallDuration=00:00:00 sec
CallState=2
CallOrigin=1
ChargedUnits=0
InfoType=speech
TransmitPackets=3380
TransmitBytes=540332
ReceivePackets=3386
ReceiveBytes=540356
VOIP:
ConnectionId[0x0 0x0 0x0 0x0]
IncomingConnectionId[0x0 0x0 0x0 0x0]
CallID=28
RemoteIPAddress=10.0.0.0
RemoteUDPPort=18630
RemoteSignallingIPAddress=10.10.0.0
RemoteSignallingPort=0
RemoteMediaIPAddress=10.2.6.10
RemoteMediaPort=18630
RoundTripDelay=0 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
FastConnect=FALSE
AnnexE=FALSE
Separate H245 Connection=FALSE
H245 Tunneling=FALSE
SessionProtocol=other
ProtocolCallId=
SessionTarget=
OnTimeRvPlayout=63120
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=105 ms
LoWaterPlayoutDelay=105 ms
TxPakNumber=3158
TxSignalPak=0
TxComfortNoisePak=0
TxDuration=63165
TxVoiceDuration=63165
```

```
RxPakNumber=3164
RxSignalPak=0
RxDuration=0
TxVoiceDuration=63165
VoiceRxDuration=63120
RxOutOfSeq=0
RxLatePak=0
RxEarlyPak=0
PlayDelayCurrent=105
PlayDelayMin=105
PlayDelayMax=105
PlayDelayClockOffset=957554296
PlayDelayJitter=0
PlayErrPredictive=0
PlayErrInterpolative=0
PlayErrSilence=0
PlayErrBufferOverFlow=0
PlayErrRetroactive=0
PlayErrTalkspurt=0
OutSignalLevel=-12
InSignalLevel=-11
LevelTxPowerMean=0
LevelRxPowerMean=-114
LevelBgNoise=0
ERLLevel=22
ACOMLevel=22
ErrRxDrop=0
ErrTxDrop=0
ErrTxControl=0
ErrRxControl=0
PlayoutMode = undefined
PlayoutInitialDelay=0 ms
ReceiveDelay=105 ms
LostPackets=0
EarlyPackets=0
LatePackets=0
SRTP = off
VAD = disabled
CoderTypeRate=g711ulaw
CodecBytes=160
Media Setting=flow-around
Modem passthrough signaling method is nse:
Buffer Fill Events = 0
Buffer Drain Events = 0
Percent Packet Loss = 0
Consecutive-packets-lost Events = 0
Corrected packet-loss Events = 0
Last Buffer Drain/Fill Event = 0sec
Time between Buffer Drain/Fills = Min 0sec Max 0sec
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=
OriginalCalledOctet=0x0
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=
TranslatedCallingOctet=0x0
TranslatedCalledNumber=
TranslatedCalledOctet=0x0
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0x0
MediaInactiveDetected=no
MediaInactiveTimestamp=
MediaControlReceived=
Username=
Telephony call-legs: 2
SIP call-legs: 0
H323 call-legs: 0
Call agent controlled call-legs: 2
SCCP call-legs: 0
```

```
Multicast call-legs: 0
Total call-legs: 4
```

The tables above describe the significant fields shown in the display, in alphabetical order.

The following is sample output from the **show call active voice** command to indicate if Service Advertisement Framework (SAF) is being used:

```
Router# show call active voice
Total call-legs: 2
GENERIC:
SetupTime=1971780 ms
Index=1
PeerAddress=6046692010
PeerSubAddress=
PeerId=20003
PeerIfIndex=17
.
.
.
VOIP:
SessionProtocol=sipv2
ProtocolCallId=7A9E7D9A-EAD311DC-8036BCC4-6EEE85D6@1.5.6.12
SessionTarget=1.5.6.10
SafEnabled=TRUE
SafTrunkRouteId=1
SafPluginDialpeerTag=8
```

The tables above describe the significant fields shown in the display.

The following is sample output from the **show call active voice** command for fax-relay traffic:

```
Router# show call active voice
Telephony call-legs: 0
SIP call-legs: 0
H323 call-legs: 1
MGCP call-legs: 0
Multicast call-legs: 0
Total call-legs: 1
 GENERIC:
SetupTime=1049400 ms
Index=2
PeerAddress=52930
PeerSubAddress=
PeerId=82
PeerIfIndex=222
LogicalIfIndex=0
ConnectTime=105105
CallDuration=00:00:59
CallState=4
CallOrigin=1
ChargedUnits=0
InfoType=10
TransmitPackets=1837
TransmitBytes=29764
ReceivePackets=261
ReceiveBytes=4079
VOIP:
ConnectionId[0xEB630F4B 0x9F5E11D7 0x8008CF18 0xB9C3632]
IncomingConnectionId[0xEB630F4B 0x9F5E11D7 0x8008CF18 0xB9C3632]
RemoteIPAddress=10.7.95.3
RemoteUDPPort=16610
RemoteSignallingIPAddress=10.7.95.3
RemoteSignallingPort=1720
RemoteMediaIPAddress=10.7.95.3
RemoteMediaPort=16610
RoundTripDelay=13 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
FastConnect=TRUE
AnnexE=FALSE
Separate H245 Connection=FALSE
```

```
H245 Tunneling=TRUE
SessionProtocol=cisco
ProtocolCallId=
SessionTarget=ipv4:10.7.95.3
OnTimeRvPlayout=1000
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=110 ms
LoWaterPlayoutDelay=70 ms
ReceiveDelay=70 ms
LostPackets=0
EarlyPackets=1
LatePackets=0
VAD = enabled
CoderTypeRate=t38
CodecBytes=40
Media Setting=flow-through
AlertTimepoint=104972
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=4085550130
OriginalCallingOctet=0x0
OriginalCalledNumber=52930
OriginalCalledOctet=0xE9
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x7F
TranslatedCallingNumber=4085550130
TranslatedCallingOctet=0x0
TranslatedCalledNumber=52930
TranslatedCalledOctet=0xE9
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwReceivedCalledNumber=52930
GwReceivedCalledOctet3=0xE9
GwOutpulsedCalledNumber=52930
GwOutpulsedCalledOctet3=0xE9
GwReceivedCallingNumber=555-0100
GwReceivedCallingOctet3=0x0
GwReceivedCallingOctet3a=0x80
GwOutpulsedCallingNumber=555-0101
GwOutpulsedCallingOctet3=0x0
GwOutpulsedCallingOctet3a=0x80
Username=
FaxRelayMaxJitterBufDepth  = 0 ms
FaxRelayJitterBufOverFlow = 0
FaxRelayHSmodulation = 0
FaxRelayNumberOfPages = 0
Telephony call-legs: 0
SIP call-legs: 0
H323 call-legs: 1
MGCP call-legs: 0
Multicast call-legs: 0
Total call-legs: 1
```

The tables above describe the significant fields shown in the display.

The following is sample output from the **show call active voice brief**command:

```
Router# show call active voice brief

<ID>: <CallID> <start>hs.<index> +<connect> pid:<peer_id> <dir> <addr> <state>
dur hh:mm:ss tx:<packets>/<bytes> rx:<packets>/<bytes>
IP <ip>:<udp> rtt:<time>ms pl:<play>/<gap>ms lost:<lost>/<early>/<late>
delay:<last>/<min>/<max>ms <codec>
media inactive detected:<y/n> media cntrl rcvd:<y/n> timestamp:<time>
long_duration_call_detected:<y/n> long duration call duration:n/a timestamp:n/a
MODEMPASS <method> buf:<fills>/<drains> loss <overall%> <multipkt>/<corrected>
last <buf event time>s dur:<Min>/<Max>s
FR <protocol> [int dlci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
<codec> (payload size)
ATM <protocol> [int vpi/vci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
```

```
<codec> (payload size)
Tele <int> (callID) [channel_id] tx:<tot>/<v>/<fax>ms <codec> noise:<l> acom:<l> i/o:<l>/<l>
 dBm
MODEMRELAY info:<rcvd>/<sent>/<resent> xid:<rcvd>/<sent> total:<rcvd>/<sent>/<drops>
speeds(bps): local <rx>/<tx> remote <rx>/<tx>
Proxy <ip>:<audio udp>,<video udp>,<tcp0>,<tcp1>,<tcp2>,<tcp3> endpt: <type>/<manf>
bw: <req>/<act> codec: <audio>/<video>
tx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>
rx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>
Total call-legs:2
1269 :7587246hs.1 +260 pid:0 Answer  active
 dur 00:07:14 tx:590/11550 rx:21721/434420
IP 172.29.248.111:17394 rtt:3ms pl:431850/0ms lost:0/0/0 dela
y:69/69/70ms g729r8

1269 :7587246hs.2 +259 pid:133001 Originate 133001 active
 dur 00:07:14 tx:21717/434340 rx:590/11550
 Tele 1/0:1 (2):tx:434350/11640/0ms g729r8 noise:-44 acom:-19
i/0:-45/-45 dBm
```

The following is an example of the **show call active voice**command using the **echo-canceller** keyword. The number 9 represents the hexadecimal ID of an active voice call.

```
Router# show call active voice echo-canceller 9
ACOM=-65  ERL=45
Echo canceller control words=6C 0
Bypass=OFF  Tail=64  Residual ecan=Comfort noise
Freeze=OFF  Modem tone disable=Ignore 2100Hz tone
Worst ERL=6  High level compensation=OFF
Max amplitude reflector (in msec)=5
Ecan version = 8180
```

The following is sample output from the **show call active voice echo-canceller**command for a call with a hexadecimal ID of 10:

```
Router# show call active voice echo-canceller 10

ACOM=-15  ERL=7
Echo canceller control words=6C 0
Bypass=OFF  Tail=64  Residual ecan=Comfort noise
Freeze=OFF  Modem tone disable=Ignore 2100Hz tone
Worst ERL=6  High level compensation=OFF
Max amplitude reflector (in msec)=64
```

The call ID number (which is 10 in the preceding example) changes with every new active call. When an active call is up, you must enter the **show call active voice brief**command to obtain the call ID number. The call ID must be converted to hexadecimal value if you want to use the **show call active voice echo-canceller** *x*command (*x* = call ID converted to hexadecimal value).

The table below shows call ID examples converted to hexadecimal values (generally incremented by 2):

Table 14: Call IDs Converted to Hex

Decimal	Hex
2	2
4	4
6	6
8	8
10	A

Decimal	Hex
12	C

Alternatively, you can use the **show voice call status**command to obtain the call ID. The call ID output is already in hexadecimal values form when you use this command:

```
Router# show voice call status


CallID     CID  ccVdb      Port      DSP/Ch  Called #   Codec    Dial-peers
0x1        11CE 0x02407B20 1:0.1     1/1     1000       g711ulaw 2000/1000
```

The following is sample output from the **show call active voice** command using the **compact** keyword:

```
Router# show call active voice compact
<callID>   A/O FAX T<sec> Codec    type    Peer Address    IP R<ip>:<udp>
Total call-legs: 2
58 ANS     T11          g711ulaw   VOIP     Psipp 2001:......:230A:6080
59 ORG     T11          g711ulaw   VOIP     P5000110011       10.13.37.150:6090
```

The following is sample output from the **show call active voice redirect**command using the **tbct** keyword:

```
Router# show call active voice redirect tbct
TBCT:
        Maximum no. of TBCT calls allowed:No limit
        Maximum TBCT call duration:No limit
Total number TBCT calls currently being monitored = 1
ctrl name=T1-2/0, tag=13, call-ids=(7, 8), start_time=*00:12:25.985 UTC Mon Mar 1 1993
```

The table below describes the significant fields shown in the display.

Table 15: show call active voice redirect Field Descriptions

Field	Description
Maximum no. of TBCT calls allowed	Maximum number of calls that can use TBCT as defined by the **tbct max calls** command.
Maximum TBCT call duration	Maximum length allowed for a TBCT call as defined by the **tbct max call-duration** command.
Total number TBCT calls currently being monitored	Total number of active TBCT calls.
ctrl name	Name of the T1 controller where the call originated.
tag	Call tag number that identifies the call.
call-ids	Numbers that uniquely identify the call legs.
start_time	Time, in hours, minutes, and seconds, when the redirected call began.

Related Commands

Command	Description
show call active fax	Displays call information for fax transmissions that are in progress.
show call history	Displays the call history table.
show call-router routes	Displays the dynamic routes in the cache of the BE.
show call-router status	Displays the Annex G BE status.
show dial-peer voice	Displays configuration information for dial peers.
show num-exp	Displays how the number expansions are configured in VoIP.
show voice call status	Displays the call status for voice ports on the Cisco router or concentrator.
show voice port	Displays configuration information about a specific voice port.

show call application app-level

To display application-level statistics for voice applications, use the **show call application app-level**command in privileged EXEC mode.

show call application {**active**| **history**} **app-level** [**app-tag** *application-name*| **summary**]

Syntax Description

active	Displays statistics for active application instances.
history	Displays statistics for terminated application instances.
app-tag *application-name*	Name of a specific voice application. Output displays statistics for that voice application.
summary	Displays a summary for each application.

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

- To display statistics with this command, you must enable statistics collection with the **call application stats** command.
- This command displays gauges and counters that are aggregated per application. The values represent all instances of a particular voice application running on the gateway while statistics collection is enabled.
- To reset application-level counters to zero and subtract the counters from the gateway-level statistics in history, use the **clear call application stats** command. Statistic counters continue accumulating in history until you use the **clear call application stats** command or the gateway reloads.

Note Statistics for an application are automatically cleared if the application is deleted with the **no call application voice** command or its script is reloaded with the **call application voice load** command.

Examples

The following is sample output from the **show call application app-level** command using different keywords:

```
Router# show call application active app-level summary
```

```
Application level active Info:
                             Sessions
App Name                 w/ Stats   Total
session                  0          0
fax_hop_on               0          0
clid_authen              0          0
clid_authen_collect      0          0
clid_authen_npw          0          0
clid_authen_col_npw      0          0
clid_col_npw_3           0          0
clid_col_npw_npw         0          0
Default                  0          0
lib_off_app              0          0
fax_on_vfc_onramp_app    0          0
asr                      0          0
offramp                  0          0
generic                  1          1
smtp_record              0          0
authen                   0          0
authorize                0          0
ram_record_replay        0          0
Router# show call application active app-level app-tag generic

Application level active Info:
Application Name:              generic
url:                           tftp://10.10.10.113/tftplocal/generic.vxml
Total sessions:                                 1
Sessions w/ stats:                              1
Currently connected incoming PSTN legs:         1
Currently connected outgoing PSTN legs:         0
Currently connected incoming VoIP legs:         0
Currently connected outgoing VoIP legs:         0
Placecalls in transit:                          0
Handouts in transit:                            0
Pending ASNL subscriptions:                     0
Pending ASNL unsubscriptions:                   0
Prompts playing (non-TTS):                      0
Recordings:                                     0
TTS prompts playing:                            0
```

For a description of the fields shown in the display above, see Table 38 on page 1363.

```
Router# show call application history app-level summary

Application level history Info:
                                   Sessions                     Last Reset
App Name                 Stats w/ Stats   Total      Errors     Time
session                  N     0          0          0
fax_hop_on               N     0          0          0
clid_authen              N     0          0          0
clid_authen_collect      N     0          0          0
clid_authen_npw          N     0          0          0
clid_authen_col_npw      N     0          0          0
clid_col_npw_3           N     0          0          0
clid_col_npw_npw         N     0          0          0
Default                  N     0          0          0
lib_off_app              N     0          0          0
fax_on_vfc_onramp_app    N     0          0          0
ram_record_replay        N     0          0          0
authorize                N     0          0          0
authen                   N     0          0          0
smtp_record              N     0          0          0
generic                  Y     2          2          4          *Jul  3 15:49:28
offramp                  N     0          0          0
asr                      N     0          0          0
```

The table below describes the fields shown in the display.

Table 16: show call application history app-level Field Descriptions

Field	Description
App Name	Name of the voice application.
Stats	Whether statistics is enabled for this application. **Note** If statistics is enabled, this field displays N until there is at least one active instance of the application.
Sessions w/ stats	Number of terminated application instances that the gauges represent.
Total	Total number of instances of the application.
Errors	Total number of errors for all instances of the application.
Last Reset Time	Time at which the statistics were last cleared with the **clear call application stats** command, or the gateway was restarted.

```
Router# show call application history app-level app-tag generic

Application level history Info:
Application name:              generic
URL:                           tftp://10.10.10.113/tftplocal/generic.vxml
Total sessions:                2
Sessions w/ stats:             2
Last reset time:               *Jul  3 15:49:28 PST
Statistics:
 Subscriber Service - Call
                                             PSTN               VOIP
                                   Incoming Outgoing  Incoming Outgoing
Legs setup:                        2        0         0        0
Total legs connected:              2        0         0        0
Legs handed in:                    0        0         0        0
Legs handed in returned back:      0        0         0        0
Legs handed out:                   0        0         0        0
Legs handed out came back:         0        0         0        0
Legs disconnected normally:        2        0         0        0
Legs disconnected for user error:  0        0         0        0
Legs disconnected for system error:0        0         0        0
 Subscriber Service - Media
                                   Play       Record     TTS
Media attempts:                    3          0          0
Media successes:                   0          0          0
Media aborts:                      0          0          0
Media failures:                    3          0          0
Total media duration (in seconds): 3          0          0
 Application Internal Service - Handoff
                                   Incoming   Outgoing
Bridged handoffs:                  0          0
Bridged handoffs returned:         0          0
Blind handoffs:                    0          0
Handoffs failed:                   x          0
 Application Internal Service - Placecall/transfer
Placecall requests:                0
Placecall successes:               0
```

```
Placecall failures:                         0
 Application Internal Service - Document Read-Write
                                            Read       Write
Doc requests:                               0          0
Doc successes:                              0          0
Doc failures:                               0          0
 Application Internal Service - Downloaded Script
Script parse errors:                        0
 Application Internal Service - ASNL
ASNL notifications:                         0
                                            Subscription   Unsubscription
ASNL requests:                              0              0
ASNL successes:                             0              0
ASNL failures:                              0              0
 Subscriber Interaction - DTMF
DTMFs not matched:                          0
DTMFs matched:                              0
DTMFs no input:                             1
DTMFs long pound:                           0
Subscriber Interaction - ASR
ASRs not matched:                           0
ASRs matched:                               0
ASRs no input:                              0
 Subscriber Interaction - AAA
                                            Authentication Authorization
AAA successes:                              0              1
AAA failures:                               0              0
```

For a description of the fields shown in this display, see Table 41 on page 1379.

Related Commands

Command	Description
call application event-log	Enables event logging for voice application instances.
call application stats	Enables statistics collection for voice applications.
call application voice event-log	Enables event logging for a specific voice application.
clear call application stats	Clears application-level statistics in history and subtracts the statistics from the gateway-level statistics.
show call application gateway-level	Displays gateway-level statistics for voice application instances.
show call application session-level	Displays event logs and statistics for voice application instances.

show call application gateway-level

To display gateway-level statistics for voice application instances, use the **show call application gateway-level** command in privileged EXEC mode.

show call application {**active**| **history**} **gateway-level**

Syntax Description

active	Displays statistics for active application instances.
history	Displays statistics for terminated application instances.

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

- To display statistics with this command, you must enable statistics collection with the **call application stats** command.
- This command displays gauges and counters that are aggregated per gateway. The values represent all instances of all voice applications running on the gateway while statistics collection is enabled.
- To reset application-level counters to zero and subtract the counters from the gateway-level statistics in history, use the **clear call application stats** command. Statistic counters continue accumulating in history until you use the **clear call application stats** command or the gateway reloads.

Note Statistics for an application are automatically cleared if the application is deleted with the **no call application voice** command or its script is reloaded with the **call application voice load** command.

Examples

The following is sample output from the **show call application gateway-level** command using different keywords:

```
Router# show call application active gateway-level
Gateway level statistics for active application sessions:
Sessions w/ stats:                          1
Currently connected incoming PSTN legs:     1
Currently connected outgoing PSTN legs:     0
Currently connected incoming VoIP legs:     0
```

```
Currently connected outgoing VoIP legs:        0
Placecalls in transit:                         0
Handouts in transit:                           0
Pending ASNL subscriptions:                    0
Pending ASNL unsubscriptions:                  0
Prompts playing (non-TTS):                     0
Recordings:                                    0
TTS prompts playing:                           0
```

The table below describes the fields shown in the display.

Table 17: show call application active gateway-level Field Descriptions

Field	Description
Sessions w/ stats	Number of active application instances that the gauges represent.
Currently connected incoming PSTN legs	Number of active call legs that are incoming from the PSTN.
Currently connected outgoing PSTN legs	Number of active call legs that are outgoing to the PSTN.
Currently connected incoming VoIP legs	Number of active call legs that are incoming from the IP network.
Currently connected outgoing VoIP legs	Number of active call legs that are outgoing to the IP network.
Placecalls in transit	Number of outgoing calls in progress for all active application instances. The value is decremented by one after the call is either set up or the setup fails.
Handouts in transit	Number of handoffs in progress for all active application instances. The value is decremented by one after the receiving application either hands back the application or rejects the handoff.
Pending ASNL subscriptions	Number of Application Subscribe Notify Layer (ASNL) subscription requests that are in progress for all active application instances.
Pending ASNL unsubscriptions	Number of ASNL unsubscription requests that are in progress for all active application instances.
Prompts playing (non-TTS)	Number of recorded prompts being played in all active application instances.
Recordings	Number of recordings being made in all active application instances.
TTS prompts playing	Number of text-to-speech (TTS) prompts playing in all active application instances.

```
Router# show call application history gateway-level

Gateway level statistics for history application sessions:
Sessions w/ stats:             2
Last reset time:               *Jul  3 15:49:28 PST
Statistics:
 Subscriber Service - Call
                                              PSTN              VOIP
                                       Incoming Outgoing  Incoming Outgoing
Legs setup:                            2        0         0        0
Total legs connected:                  2        0         0        0
Legs handed in:                        0        0         0        0
Legs handed in returned back:          0        0         0        0
Legs handed out:                       0        0         0        0
Legs handed out came back:             0        0         0        0
Legs disconnected normally:            2        0         0        0
Legs disconnected for user error:      0        0         0        0
Legs disconnected for system error:    0        0         0        0
 Subscriber Service - Media
                                       Play       Record     TTS
Media attempts:                        3          0          0
Media successes:                       0          0          0
Media aborts:                          0          0          0
Media failures:                        3          0          0
Total media duration (in seconds):     3          0          0
 Subscriber Interaction - DTMF
DTMFs not matched:                     0
DTMFs matched:                         0
DTMFs no input:                        1
DTMFs long pound:                      0
```

For a description of the fields shown with the **history** keyword, see the table above.

Related Commands

Command	Description
call application stats	Enables statistics collection for voice applications.
clear call application stats	Clears application-level statistics in history and subtracts the statistics from the gateway-level statistics.
show call application app-level	Displays application-level statistics for voice applications.
show call application session-level	Displays event logs and statistics for voice application instances.

show call application interface

To display event logs and statistics for application interfaces, use the **show call application interface**command in privileged EXEC mode.

show call application interface [**summary**| {**aaa**| **asr**| **flash**| **http**| **ram**| **rtsp**| **smtp**| **tftp**| **tts**} [**server** *server*] [**event-log**| **info**| **summary**]]

Syntax Description

summary	(Optional) Displays a short summary of all interface types or the selected interface.
aaa	Authentication, authorization, and accounting (AAA) interface type.
asr	Automatic speech recognition (ASR) interface type.
flash	Flash memory of the Cisco gateway.
http	Hypertext Transfer Protocol (HTTP) interface type.
ram	Memory of the Cisco gateway.
rtsp	Real Time Streaming Protocol (RTSP) interface type.
smtp	Simple Mail Transfer Protocol (SMTP) interface type.
tftp	Trivial File Transfer Protocol (TFTP) interface type.
tts	Text-to-speech (TTS) interface type.
server *server*	(Optional) Displays event logs or statistics for the specified server.
event-log	(Optional) Displays event logs for the selected interface type or server.
info	(Optional) Displays statistics for the selected interface type or server.

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

If you use the **server** keyword, only statistics or event logs for that server display. To display event logs or statistics with this command, you must enable statistics and event logging with the **call application interface event-log** and **call application interface stats** command, respectively. To reset statistic counters to zero and clear the event logs in history, use the **clear call application interface** command.

Examples

The following is sample output from the **show call application interface** command using different keywords:

```
Router# show call application interface summary
Aggregated statistics for http service:
Stats last reset time *Jul  3 15:24:48 PST
Read requests:                  3
Read successes:                 0
Read failures:                  3
Read aborts:                    0
Total bytes read:               0
Write requests:                 0
Write successes:                0
Write failures:                 0
Write aborts:                   0
Total bytes written:            0
Aggregated statistics for tts service:
Stats last reset time *Jul  3 15:24:48 PST
Read requests:                  0
Read successes:                 0
Read failures:                  0
Read aborts:                    0
Aggregated statistics for asr service:
Stats last reset time *Jul  3 15:24:48 PST
Read requests:                  0
Read successes:                 0
Read failures:                  0
Read aborts:                    0
Aggregated statistics for tftp service:
Stats last reset time *Jul  3 15:24:48 PST
Read requests:                  3
Read successes:                 2
Read failures:                  0
Read aborts:                    1
Total bytes read:               145888
Router# show call application interface tftp summary

Aggregated statistics for tftp service:
Stats last reset time *Jul  3 15:24:48 PST
Read requests:                  3
Read successes:                 2
Read failures:                  0
Read aborts:                    1
Total bytes read:               145888
Server Name          Stats Error Count            Event Log
172.19.139.145       Y     0                      Y
speech-serv          Y     0                      N
Router# show call application interface tftp

Server name:         172.19.139.145
Statistics:
Last reset time *Jul  3 16:08:13 PST
Read requests:                  1
Read successes:                 2
Read failures:                  0
Read aborts:                    1
Total bytes read:               145888
Event log:
Last reset time *Jul  3 16:08:13 PST
buf_size=50K, log_lvl=INFO
```

```
<ctx_id>:<timestamp>:<seq_no>:<severity>:<msg_body>
172.19.139.145:1057277293:53:INFO: ID = 6549D9E0: Read requested for URL =
tftp://172.19.139.145/audio/ch_welcome.au
172.19.139.145:1057277295:54:INFO: ID = 6549D9E0: Streamed read transaction Successful URL
 = tftp://172.19.139.145/audio/ch_welcome.au
172.19.139.145:1057277306:59:INFO: ID = 649A0320: Streamed read transaction Successful URL
 = tftp://172.19.139.145/audio/ch_welcome.au
172.19.139.145:1057277317:65:INFO: ID = 650922A8: Read request aborted for URL =
tftp://172.19.139.145/audio/ch_welcome.au
--------------------------------------------------------------------------------
Router# show call application interface tftp event-log

Server name:         172.19.139.145
Event log:
Last reset time *Jul  3 16:08:13 PST
buf_size=50K, log_lvl=INFO
<ctx_id>:<timestamp>:<seq_no>:<severity>:<msg_body>
172.19.139.145:1057277293:53:INFO: ID = 6549D9E0: Read requested for URL =
tftp://172.19.139.145/audio/ch_welcome.au
172.19.139.145:1057277295:54:INFO: ID = 6549D9E0: Streamed read transaction Successful URL
 = tftp://172.19.139.145/audio/ch_welcome.au
172.19.139.145:1057277306:59:INFO: ID = 649A0320: Streamed read transaction Successful URL
 = tftp://172.19.139.145/audio/ch_welcome.au
172.19.139.145:1057277317:65:INFO: ID = 650922A8: Read request aborted for URL =
tftp://172.19.139.145/audio/ch_welcome.au
--------------------------------------------------------------------------------
Router# show call application interface tftp info

Server name:         172.19.139.145
Statistics:
Last reset time *Jul  3 16:08:13 PST
Read requests:             3
Read successes:            2
Read failures:             0
Read aborts:               1
Total bytes read:                145888
--------------------------------------------------------------------------------
```

The table below describes the significant fields shown in the display.

Table 18: show call application interface Field Descriptions

Field	Description
Last reset time	Time at which the statistics were last cleared with the **clear call application interface** command, or the gateway was restarted.
Read requests	Total number of read requests from applications to this interface type.
Read successes	Number of successful read requests from applications to this interface type.
Read failures	Number of failed read requests from applications to this interface type.
Read aborts	Number of aborted read requests from applications to this interface type.
Total bytes read	Total number of bytes that the application read from this interface type.

Field	Description
Server name	Name of the specific server.
Stats	Whether statistics are enabled for this server.
Error Count	Total number of errors for this server.
Event Log	Whether event logging is enabled for this server.

Related Commands

Command	Description
call application interface event-log	Enables event logging for external interfaces used by voice applications.
call application interface stats	Enables statistics collection for application interfaces.
clear call application interface	Clears application interface statistics and event logs.

show call application services registry

To display a one-line summary of all TCL IVR 2.0 application sessions that have registered as a service, use the **show call application services registry**command in user EXEC or privileged EXEC mode.

show call application services registry

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

User EXEC (>) Privileged EXEC (#)

Command History

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

Usage Guidelines

- The services registry is a database that keeps track of every TCL IVR 2.0 application instance that registers as a service. Other TCL applications can then find and communicate with any registered application.
- A TCL session is not registered as a service through a Cisco IOS command. A running instance of a TCL IVR 2.0 application registers itself as a service with the TCL service register command. For information about the service register command, refer to the TCL IVR API Version 2.0 Programmer's Guide.

Examples

The following is sample output for this command:

```
Router# show call application services registry
There are 1 Registered Services
  Service Name         Session ID  Session Name
  data_service                  4  s1
```

The table below describes significant fields in the display.

Table 19: show call application services registry Field Descriptions

Field	Description
Service Name	Name specified by the TCL service register command.
Session ID	ID of the session that registered as this service. You can use this ID in the **show call application sessions id** command to view details about this session.

Field	Description
Session Name	Name configured by the **call application session start** command, if the session was started on the gateway rather than by an incoming call.

Related Commands

Command	Description
call application session start (global configuration)	Starts a new instance (session) of a TCL application from global configuration mode.
call application session start (privileged EXEC)	Starts a new instance (session) of a TCL application from privileged EXEC mode.
call application session stop	Stops a voice application session that is running.
show call application sessions	Displays summary or detailed information about voice application sessions.

show call application session-level

To display event logs and statistics for individual voice application instances, use the **show call application session-level**command in privileged EXEC mode.

show call application {**active**| **history**} **session-level** [**summary**| [**app-tag** *application-name*| **last** [*number*]| **session-id** *session-id*] [**event-log**| **info**]]

Syntax Description

active	Displays event logs or statistics for active application instances.
history	Displays event logs or statistics for inactive application instances in the history table.
summary	Displays a summary of each application instance.
app-tag *application-name*	Name of a specific voice application. Output displays event logs or statistics for that voice application.
last	(Optional) Displays event logs or statistics for the most recent instance.
number	(Optional) Displays event logs or statistics for this number of most recent previous instances.
session-id *session-id*	Identifies a specific application instance. Output displays event logs or statistics for that instance.
event-log	(Optional) Displays event logs for application instances.
info	(Optional) Displays statistics for application instances.

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

- To display event logs or statistics with this command, you must enable event logging and statistics with the **call application event-log** and **call application stats** command, respectively.

- This command displays gauges and counters that are aggregated per application instance. The values represent an individual instance running on the gateway while statistics collection is enabled.
- The number of records that are included when using the **history** keyword depends on the settings of the **call application history session max-records** and **call application history session retain-timer** commands.

Examples

The following is sample output from the **show call application session-level** command using different keywords and arguments:

```
Router# show call application active session-level summary

SID  Application Name       Stat Err Cnt    Log Start Time
5    generic                Y    6          Y   *Jul  3 15:19:4
6    generic                Y    3          Y   *Jul  3 15:19:5
Router# show call application active session-level last
Session Info:
Session id:          6
Session name:
Application name:    generic
Application URL:     tftp://demo/scripts/master/generic.vxml
Start time:          *Jul  3 15:19:53 PST
Statistics:
 Subscriber Service - Call
                                                 PSTN              VOIP
                                          Incoming Outgoing  Incoming Outgoing
Legs setup:                               1        0         0        0
Total legs connected:                     1        0         0        0
Legs currently connected:                 1        0         0        0
Legs handed in:                           0        0         0        0
Legs handed in returned back:             0        0         0        0
Legs handed out:                          0        0         0        0
Legs handed out came back:                0        0         0        0
Legs disconnected normally:               0        0         0        0
Legs disconnected for user error:         0        0         0        0
Legs disconnected for system error:       0        0         0        0
 Subscriber Service - Media
                                          Play       Record     TTS
Media attempts:                           4          0          0
Media actives:                            0          0          0
Media successes:                          0          0          0
Media aborts:                             0          0          0
Media failures:                           4          0          0
Total media duration (in seconds):        0          0          0
 Subscriber Interaction - DTMF
DTMFs not matched:                        0
DTMFs matched:                            0
DTMFs no input:                           3
DTMFs long pound:                         0
Event log:
buf_size=25K, log_lvl=INFO
<ctx_id>:<timestamp>:<seq_no>:<severity>:<msg_body>
6:1057274393:472:INFO: Session started for App-type = generic, URL =
tftp://demo/scripts/master/generic.vxml
6:1057274393:473:INFO: Incoming Telephony call received, LegID = 10
6:1057274393:474:INFO: LegID = 10: Calling = 4084644753, called = 52927, dial peer = 1
6:1057274393:475:INFO: LegID = 10: Leg State = LEG_INCCONNECTED
6:1057274393:478:INFO: Playing prompt #1: http://172.19.139.145/audio/ch_welcome.au
6:1057274408:517:INFO: Script received event = "error.badfetch"
Router# show call application active session-level info

Session Info:
Session id:          5
Session name:
Application name:    generic
Application URL:     tftp://demo/scripts/master/generic.vxml
Start time:          *Jul  3 15:19:44 PST
```

```
Statistics:
 Subscriber Service - Call
                                           PSTN               VOIP
                                    Incoming Outgoing  Incoming Outgoing
Legs setup:                         1        0         0        0
Total legs connected:               1        0         0        0
Legs currently connected:           1        0         0        0
Legs handed in:                     0        0         0        0
Legs handed in returned back:       0        0         0        0
Legs handed out:                    0        0         0        0
Legs handed out came back:          0        0         0        0
Legs disconnected normally:         0        0         0        0
Legs disconnected for user error:   0        0         0        0
Legs disconnected for system error: 0        0         0        0
 Subscriber Service - Media
                                    Play       Record     TTS
Media attempts:                     9          0          0
Media actives:                      0          0          0
Media successes:                    0          0          0
Media aborts:                       0          0          0
Media failures:                     9          0          0
Total media duration (in seconds):  0          0          0
 Subscriber Interaction - DTMF
DTMFs not matched:                  0
DTMFs matched:                      0
DTMFs no input:                     8
DTMFs long pound:                   0
Session Info:
Session id:          6
Session name:
Application name:    generic
Application URL:     tftp://demo/scripts/master/generic.vxml
Start time:          *Jul  3 15:19:53 PST
Statistics:
 Subscriber Service - Call
                                           PSTN               VOIP
                                    Incoming Outgoing  Incoming Outgoing
Legs setup:                         3        0         0        0
Total legs connected:               3        0         0        0
Legs currently connected:           1        0         0        0
Legs handed in:                     0        0         0        0
Legs handed in returned back:       0        0         0        0
Legs handed out:                    0        0         0        0
Legs handed out came back:          0        0         0        0
Legs disconnected normally:         0        0         0        0
Legs disconnected for user error:   0        0         0        0
Legs disconnected for system error: 0        0         0        0
 Subscriber Service - Media
                                    Play       Record     TTS
Media attempts:                     7          0          0
Media actives:                      0          0          0
Media successes:                    0          0          0
Media aborts:                       0          0          0
Media failures:                     7          0          0
Media duration (in seconds):        0          0          0
 Application Internal Service - Handoff
                                    Incoming   Outgoing
Bridged handoffs:                   0          0
Bridged handoffs returned:          0          0
Blind handoffs:                     0          0
Handoffs in transit:                x          0
Handoffs failed:                    x          0
 Application Internal Service - Placecall/transfer
Placecall requests:                 0
Placecall successes:                0
Placecall failures:                 0
Placecalls in transit:              0
 Application Internal Service - Document Read-Write
                                    Read       Write
Doc requests:                       0          0
Doc successes:                      0          0
Doc failures:                       0          0
 Application Internal Service - Downloaded Script
```

```
Script parse errors:                        0
 Application Internal Service - ASNL
ASNL notifications:                         0
                                            Subscription   Unsubscription
ASNL requests:                              0              0
ASNL successes:                             0              0
ASNL pendings:                              0              0
ASNL failures:                              0              0
 Subscriber Interaction - DTMF
DTMFs not matched:                          0
DTMFs matched:                              0
DTMFs no input:                             6
DTMFs long pound:                           0
 Subscriber Interaction - ASR
ASRs not matched:                           0
ASRs matched:                               0
ASRs no input:                              0
 Subscriber Interaction - AAA
                                            Authentication Authorization
AAA successes:                              0              0
AAA failures:                               0              0
Router# show call application active session-level event-log

Event log:
buf_size=25K, log_lvl=INFO
<ctx_id>:<timestamp>:<seq_no>:<severity>:<msg_body>
5:1057274384:454:INFO: Session started for App-type = generic, URL =
tftp://demo/scripts/master/generic.vxml
5:1057274384:455:INFO: Incoming Telephony call received, LegID = D
5:1057274384:456:INFO: LegID = D: Calling = 4085550198, called = 52927, dial peer = 1
5:1057274384:457:INFO: LegID = D: Leg State = LEG_INCCONNECTED
5:1057274384:460:INFO: Playing prompt #1: http://172.19.139.145/audio/ch_welcome.au
5:1057274384:462:ERR : Prompt play setup failure.
5:1057274384:463:INFO: Script received event = "error.badfetch"
5:1057274389:464:INFO: Timed out waiting for user DTMF digits, no user input.
5:1057274389:465:INFO: Script received event = "noinput"
Event log:
buf_size=25K, log_lvl=INFO
<ctx_id>:<timestamp>:<seq_no>:<severity>:<msg_body>
6:1057274393:472:INFO: Session started for App-type = generic, URL =
tftp://demo/scripts/master/generic.vxml
6:1057274393:473:INFO: Incoming Telephony call received, LegID = 10
6:1057274393:474:INFO: LegID = 10: Calling = 4084644753, called = 52927, dial peer = 1
6:1057274393:475:INFO: LegID = 10: Leg State = LEG_INCCONNECTED
6:1057274393:478:INFO: Playing prompt #1: http://172.19.139.145/audio/ch_welcome.au
6:1057274393:480:ERR : Prompt play setup failure.
6:1057274393:481:INFO: Script received event = "error.badfetch"
6:1057274398:488:INFO: Timed out waiting for user DTMF digits, no user input.
6:1057274398:489:INFO: Script received event = "noinput"
6:1057274398:490:INFO: Playing prompt #1: http://172.19.139.145/audio/ch_welcome.au
Router# show call application active session-level app-tag generic

Session Info:
Session id:          5
Session name:
Application name:    generic
Application URL:     tftp://demo/scripts/master/generic.vxml
Start time:          *Jul  3 15:19:44 PST
Statistics:
 Subscriber Service - Call
                                                   PSTN              VOIP
                                            Incoming Outgoing  Incoming Outgoing
Legs setup:                                 1        0         0        0
Total legs connected:                       1        0         0        0
Legs currently connected:                   1        0         0        0
Legs handed in:                             0        0         0        0
Legs handed in returned back:               0        0         0        0
Legs handed out:                            0        0         0        0
Legs handed out came back:                  0        0         0        0
Legs disconnected normally:                 0        0         0        0
Legs disconnected for user error:           0        0         0        0
Legs disconnected for system error:         0        0         0        0
 Subscriber Service - Media
```

```
                                           Play       Record     TTS
Media attempts:                            16         0          0
Media actives:                             0          0          0
Media successes:                           0          0          0
Media aborts:                              0          0          0
Media failures:                            17         0          0
Total media duration (in seconds):         0          0          0
 Subscriber Interaction - DTMF
DTMFs not matched:                         0
DTMFs matched:                             0
DTMFs no input:                            16
DTMFs long pound:                          0
Event log:
buf_size=25K, log_lvl=INFO
<ctx_id>:<timestamp>:<seq_no>:<severity>:<msg_body>
5:1057274384:454:INFO: Session started for App-type = generic, URL =
tftp://demo/scripts/master/generic.vxml
5:1057274384:455:INFO: Incoming Telephony call received, LegID = D
5:1057274384:456:INFO: LegID = D: Calling = 4085550198, called = 52927, dial peer = 1
5:1057274384:457:INFO: LegID = D: Leg State = LEG_INCCONNECTED
5:1057274384:460:INFO: Playing prompt #1: http://172.19.139.145/audio/ch_welcome.au
5:1057274384:462:ERR : Prompt play setup failure.
5:1057274384:463:INFO: Script received event = "error.badfetch"
5:1057274389:464:INFO: Timed out waiting for user DTMF digits, no user input.
5:1057274389:465:INFO: Script received event = "noinput"
5:1057274389:466:INFO: Playing prompt #1: http://172.19.139.145/audio/ch_welcome.au
Router# show call application active session-level session-id 7
Session Info:
Session id:          7
Session name:
Application name:    generic
Application URL:     tftp://demo/scripts/master/generic.vxml
Start time:          *Jul  3 15:21:26 PST
Statistics:
 Subscriber Service - Call
                                                  PSTN               VOIP
                                           Incoming Outgoing  Incoming Outgoing
Legs setup:                                1        0         0        0
Total legs connected:                      1        0         0        0
Legs currently connected:                  1        0         0        0
Legs handed in:                            0        0         0        0
Legs handed in returned back:              0        0         0        0
Legs handed out:                           0        0         0        0
Legs handed out came back:                 0        0         0        0
Legs disconnected normally:                0        0         0        0
Legs disconnected for user error:          0        0         0        0
Legs disconnected for system error:        0        0         0        0
 Subscriber Service - Media
                                           Play       Record     TTS
Media attempts:                            3          0          0
Media actives:                             0          0          0
Media successes:                           0          0          0
Media aborts:                              0          0          0
Media failures:                            3          0          0
Total media duration (in seconds):         0          0          0
 Subscriber Interaction - DTMF
DTMFs not matched:                         0
DTMFs matched:                             0
DTMFs no input:                            2
DTMFs long pound:                          0
Event log:
buf_size=25K, log_lvl=INFO
<ctx_id>:<timestamp>:<seq_no>:<severity>:<msg_body>
7:1057274486:662:INFO: Session started for App-type = generic, URL =
tftp://demo/scripts/master/generic.vxml
7:1057274486:663:INFO: Incoming Telephony call received, LegID = 13
7:1057274486:664:INFO: LegID = 13: Calling = 4085550198, called = 52927, dial peer = 1
7:1057274486:665:INFO: LegID = 13: Leg State = LEG_INCCONNECTED
7:1057274486:668:INFO: Playing prompt #1: http://172.19.139.145/audio/ch_welcome.au
Router# show call application history session-level summary

SID  Application Name       Stat Err Cnt    Log Stop Time       Duration
1    generic                Y    3          Y   *Jul  3 15:49:2 00:00:11
```

```
2    generic                Y    1          Y   *Jul  3 15:49:3 00:00:03
Router# show call application history session-level last

Session Info:
Session id:          2
Session name:
Application name:    generic
Application URL:     tftp://demo/scripts/master/generic.vxml
Start time:          *Jul  3 15:49:29 PST
Stop time:           *Jul  3 15:49:33 PST
Statistics:
 Subscriber Service - Call
                                                   PSTN              VOIP
                                           Incoming Outgoing  Incoming Outgoing
Legs setup:                                1        0         0        0
Total legs connected:                      1        0         0        0
Legs handed in:                            0        0         0        0
Legs handed in returned back:              0        0         0        0
Legs handed out:                           0        0         0        0
Legs handed out came back:                 0        0         0        0
Legs disconnected normally:                1        0         0        0
Legs disconnected for user error:          0        0         0        0
Legs disconnected for system error:        0        0         0        0
 Subscriber Service - Media
                                           Play       Record     TTS
Media attempts:                            1          0          0
Media successes:                           0          0          0
Media aborts:                              0          0          0
Media failures:                            1          0          0
Total media duration (in seconds):         0          0          0
Event log:
buf_size=25K, log_lvl=INFO
<ctx_id>:<timestamp>:<seq_no>:<severity>:<msg_body>
2:1057276169:28:INFO: Session started for App-type = generic, URL =
tftp://demo/scripts/master/generic.vxml
2:1057276169:29:INFO: Incoming Telephony call received, LegID = 4
2:1057276169:30:INFO: LegID = 4: Calling = 4085550198, called = 52927, dial peer = 1
2:1057276169:31:INFO: LegID = 4: Leg State = LEG_INCCONNECTED
2:1057276169:34:INFO: Playing prompt #1: http://172.19.139.145/audio/ch_welcome.au
2:1057276169:36:ERR : Prompt play setup failure.
2:1057276169:37:INFO: Script received event = "error.badfetch"
2:1057276173:39:INFO: Script received event = "telephone.disconnect.hangup"
2:1057276173:40:INFO: LegID = 4: Call disconnected, cause = normal call clearing (16)
2:1057276173:43:INFO: Session done, terminating cause =
Router# show call application history session-level event-log

Event log:
buf_size=25K, log_lvl=INFO
<ctx_id>:<timestamp>:<seq_no>:<severity>:<msg_body>
1:1057276157:3:INFO: Session started for App-type = generic, URL =
tftp://demo/scripts/master/generic.vxml
1:1057276157:4:INFO: Incoming Telephony call received, LegID = 1
1:1057276157:5:INFO: LegID = 1: Calling = 4085550198, called = 52927, dial peer = 1
1:1057276157:6:INFO: LegID = 1: Leg State = LEG_INCCONNECTED
1:1057276157:9:INFO: Playing prompt #1: http://172.19.139.145/audio/ch_welcome.au
1:1057276160:12:ERR : Prompt play setup failure.
1:1057276160:13:INFO: Script received event = "error.badfetch"
1:1057276165:14:INFO: Timed out waiting for user DTMF digits, no user input.
1:1057276165:15:INFO: Script received event = "noinput"
1:1057276165:16:INFO: Playing prompt #1: http://172.19.139.145/audio/ch_welcome.au
1:1057276165:18:ERR : Prompt play setup failure.
1:1057276165:19:INFO: Script received event = "error.badfetch"
1:1057276168:21:INFO: Script received event = "telephone.disconnect.hangup"
1:1057276168:22:INFO: LegID = 1: Call disconnected, cause = normal call clearing (16)
1:1057276168:25:INFO: Session done, terminating cause =
Event log:
buf_size=25K, log_lvl=INFO
<ctx_id>:<timestamp>:<seq_no>:<severity>:<msg_body>
2:1057276169:28:INFO: Session started for App-type = generic, URL =
tftp://demo/scripts/master/generic.vxml
2:1057276169:29:INFO: Incoming Telephony call received, LegID = 4
2:1057276169:30:INFO: LegID = 4: Calling = 4085550198, called = 52927, dial peer = 1
2:1057276169:31:INFO: LegID = 4: Leg State = LEG_INCCONNECTED
```

```
2:1057276169:34:INFO: Playing prompt #1: http://172.19.139.145/audio/ch_welcome.au
2:1057276169:36:ERR : Prompt play setup failure.
2:1057276169:37:INFO: Script received event = "error.badfetch"
2:1057276173:39:INFO: Script received event = "telephone.disconnect.hangup"
2:1057276173:40:INFO: LegID = 4: Call disconnected, cause = normal call clearing (16)
2:1057276173:43:INFO: Session done, terminating cause =
Router# show call application history session-level info

Session Info:
Session id:          1
Session name:
Application name:    generic
Application URL:     tftp://demo/scripts/master/generic.vxml
Start time:          *Jul  3 15:49:17 PST
Stop time:           *Jul  3 15:49:28 PST
Statistics:
 Subscriber Service - Call
                                                  PSTN              VOIP
                                           Incoming Outgoing  Incoming Outgoing
Legs setup:                                1        0         0        0
Total legs connected:                      1        0         0        0
Legs handed in:                            0        0         0        0
Legs handed in returned back:              0        0         0        0
Legs handed out:                           0        0         0        0
Legs handed out came back:                 0        0         0        0
Legs disconnected normally:                1        0         0        0
Legs disconnected for user error:          0        0         0        0
Legs disconnected for system error:        0        0         0        0
 Subscriber Service - Media
                                           Play       Record     TTS
Media attempts:                            2          0          0
Media successes:                           0          0          0
Media aborts:                              0          0          0
Media failures:                            2          0          0
Total media duration (in seconds):         3          0          0
 Subscriber Interaction - DTMF
DTMFs not matched:                         0
DTMFs matched:                             0
DTMFs no input:                            1
DTMFs long pound:                          0
Session Info:
Session id:          2
Session name:
Application name:    generic
Application URL:     tftp://demo/scripts/master/generic.vxml
Start time:          *Jul  3 15:49:29 PST
Stop time:           *Jul  3 15:49:33 PST
Statistics:
 Subscriber Service - Call
                                                  PSTN              VOIP
                                           Incoming Outgoing  Incoming Outgoing
Legs setup:                                1        0         0        0
Total legs connected:                      1        0         0        0
Legs handed in:                            0        0         0        0
Legs handed in returned back:              0        0         0        0
Legs handed out:                           0        0         0        0
Legs handed out came back:                 0        0         0        0
Legs disconnected normally:                1        0         0        0
Legs disconnected for user error:          0        0         0        0
Legs disconnected for system error:        0        0         0        0
 Subscriber Service - Media
                                           Play       Record     TTS
Media attempts:                            1          0          0
Media successes:                           0          0          0
Media aborts:                              0          0          0
Media failures:                            1          0          0
Total media duration (in seconds):         0          0          0
```

The table below describes significant fields in the displays.

Note These fields display for the **show call application session-level**, **show call application app-level**, and **show call application gateway-level** commands. At the session level, the fields apply to a single application instance. At the application level, the fields apply to all instances of an application. At the gateway level, the fields apply to all instances of all applications.

Table 20: show call application active session-level info Field Descriptions

Field	Description
Session id	Session ID assigned to the instance when it became active.
Session name	Name of the session defined with the **call application session start** command.
Application name	Name of the application defined with the **call application voice** command.
Application URL	Location of the application script defined with the **call application voice** command.
Start time	Time at which the session started.
Subscriber Service -- Call	
Legs setup	Number of calls setup (indications and requests) by an application instance.
Total legs connected	Number of calls connected by an application instance.
Legs currently connected	Number of calls currently connected by an application instance at any moment.
Legs handed in	Number of call legs received as an incoming handoff from another application.
Legs handed in returned back	Number of call legs received as an incoming handoff from another application that were returned to the other application.
Legs handed out	Number of call legs handed off to another application.
Legs handed out came back	Number of call legs handed off to another application that were returned by the other application.
Legs disconnected normally	Number of incoming and outgoing calls disconnected for normal causes.

Field	Description
Legs disconnected for user error	Number of incoming calls disconnected for call failure reasons, such as no answer or busy.
Legs disconnected for system error	Number of incoming calls disconnected for system failure reasons, such as no resources.
Subscriber Service -- Media	
Media attempts	Number of prompt playouts, recordings, and text-to-speech (TTS) attempts on call legs in this application instance.
Media actives,	Number of prompt playouts, recordings, and TTS prompts currently active on call legs in an application instance.
Media successes	Number of prompt playouts, recordings, and TTS prompts that were successful on call legs in an application instance.
Media aborts	Number of prompt playouts, recordings, and TTS prompts that were aborted by the caller on call legs in an application instance.
Media failures	Number of prompt playouts, recording, and TTS attempts that failed on call legs in an application instance.
Total media duration	Total duration, in seconds, of prompt playing, recording, or TTS.
Application Internal Service -- Handoff	
Bridged handoffs, incoming	Number of handoffs received with callback (bridged transfers) in an application instance.
Bridged handoffs, outgoing	Number of handoffs placed with callback (bridged transfers) by an application instance.
Bridged handoffs returned, incoming	Number of incoming bridged handoffs that were returned by an application instance.
Bridged handoffs returned, outgoing	Number of outgoing bridged handoffs that were returned to an application instance.
Blind handoffs, incoming	Number of handoffs received with no callback (blind transfers) in an application instance.

Field	Description
Blind handoffs, outgoing	Number of handoffs placed with no callback (blind transfers) by an application instance.
Handoffs in transit[1]	Number of handoffs in progress for an application instance. The value is decremented by one after the receiving application either hands back the application or rejects the handoff.
Handoffs failed	Number of handoffs that failed (bridged and blind) in an application instance.
Application Internal Service -- Placecall/transfer	
Placecall requests	Number of outgoing call setup requests made by an application instance.
Placecall successes	Number of outgoing calls placed by an application instance.
Placecall failures	Number of outgoing call setup requests that failed for an application instance.
Placecalls in transitshow call application session-level, on page 85	Number of outgoing calls in progress for an application. The value is decremented by one after the call is either set up or the setup fails.
Application Internal Service -- Document Read-Write	
Doc requests	Number of document fetch and submit requests.
Doc successes	Number of successful document fetches and submits.
Doc failures	Number of document fetch and submit failures.
Application Internal Service -- Downloaded Script	
Script parse errors	Number of semantic errors seen by an application instance.
Application Internal Service -- ASNL	
ASNL notifications	Number of Application Subscribe Notify Layer (ASNL) notifications received from servers.
ASNL requests	Number of subscribe or unsubscribe requests made by an application instance.
ASNL successes	Number of subscribe or unsubscribe requests that succeeded for an application instance.

Field	Description
ANSL failures	Number of subscribe or unsubscribe requests that failed for an application instance.
Subscriber Interaction --- DTMF	
DTMFs not matched	Number of DTMF patterns input by a caller that were not matched in an application instance.
DTMFs matched	Number of DTMF patterns input by a caller that were matched in an application instance.
DTMFs no input	Number of "no input" notifications received (includes DTMF timeouts).
DTMFs long pound	Number of long-pound interrupts from a caller seen by an application instance.
Subscriber Interaction -- ASR	
ASR not matched	Number of automatic speech recognition (ASR) phrases from a caller that were not matched in an application instance.
ASR matched	Number of automatic speech recognition (ASR) phrases from a caller that were matched in an application instance.
ASR no inputs	Number of "no input" notifications received from ASR servers.
Subscriber Interaction -- AAA Authentication	
AAA authentication successes	Number of AAA authentication successes.
AAA authentication failures	Number of AAA authentication failures because of invalid passwords.
Subscriber Interaction -- AAA Authorizations	
AAA authorization successes	Number of AAA authorization successes.
AAA authorization failures	Number of AAA authorization failures.

[1] When this gauge is greater than zero, the application instance might stop processing the script and the counters and gauges may appear to freeze. When the handoff or the placecall operation is finished and control is returned to the application instance, the counters and gauges are updated.

Related Commands

Command	Description
call application event-log	Enables event logging for voice application instances.
call application history session max-records	Sets the maximum number of application instance records saved in history.
call application history session retain-timer	Sets the maximum number of minutes for which application instance records are saved in history.
call application stats	Enables statistics collection for voice applications.
call application voice event-log	Enables event logging for a specific voice application.
show call application app-level	Displays application-level statistics for voice applications.
show call application gateway-level	Displays gateway-level statistics for voice application instances.

show call application sessions

To display summary or detailed information about all running or stopped voice application sessions, use the **show call application sessions** command in user EXEC or privileged EXEC mode.

show call application sessions [**callid** *call-id*| **id** *session-id*| **name** *instance-name*]

Syntax Description

callid *call-id*	(Optional) Call-leg ID of an active call that is being controlled by the session.
id *session-id*	(Optional) Session ID for the specific application instance.
name *instance-name*	(Optional) Name assigned to the instance with the **call application session start** command.

Command Default

No default behavior or values

Command Modes

User EXEC (>) Privileged EXEC (#)

Command History

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

Usage Guidelines

- A specific application session is identified by one of three different methods: call ID, session ID, or instance name.
- If a specific session is identified by a **callid**, **id**, or **name** keyword, this command displays information about that specific session only. If you do not use a keyword, this command displays a one-line summary of all sessions, not just those sessions that are started by the **call application session start** command.
- This command lists all running TCL IVR 2.0 and VoiceXML application sessions and TCL sessions that are stopped. A session displays a state of "stopped" if you intentionally stop it with the **call application session stop** or **no call application session start** command, or because there is a syntax error that prevents the script from running. This is the case only if the session is started with the **call application session start** command through global configuration mode.

Note If a session is started with the **call application session start** command in privileged EXEC mode, it is not tracked by the system and is therefore not shown as stopped in the output of the **show call application sessions** command.

Examples

The following is sample output from this command:

```
Router# show call application sessions
TCL Sessions
    There are 1 active TCL sessions
     SID  Name        Called     Calling                 App Name              Legs
       5  serv1                                          sample_service
VXML Sessions
    No running VXML sessions
Stopped Sessions
  Instance Name        App Name             State
  my_instance1         sample               stopped
```

The table below describes significant fields in the display.

Table 21: show call application sessions Field Descriptions

Field	Description
SID	Session identifier for active sessions.
Name	Session name that was configured with the **call application session start** command.
Called	Called number for active calls that are using the session.
Calling	Calling number for active calls that are using the session.
App Name	Name of the application for which the instance was created.
Legs	Any active call legs that are controlled by the session.
State	Shows "stopped" for any session that is no longer running, provided that the session is started with the **call application session start** command in global configuration mode.

The following is sample output for a session named serv1:

```
Router# show call aplication sessions name serv1
Session named serv1 is in the start list in state running
  It is configured to start on GW reboot
  The application it runs is sample_service
  Handle is TCL_HAND*1653710732*0*3193204
```

```
TCL Session ID B
                 App: sample_service
                 URL: tftp://dev/demo/scripts/sample_service.tcl
        Session name: serv1
      Session handle: TCL_HAND*1653710732*0*3193204
           FSM State: start_state
    ID for 'show call active voice id' display: 0
                Legs:
            Services: data_service
```

The table below describes significant fields in the display.

Table 22: show call application sessions name Field Descriptions

Field	Description
App	Name of the application for which the instance was created.
URL	Location of the script used for the application as specified with the **call application voice** command.
Session name	Session name that was configured with the **call application session start** command.
Session handle	Handle that is returned from the TCL mod_handle infotag. A session handle is used in a TCL script on a Cisco gateway to send messages to other sessions.
FSM State	Current state in the TCL IVR 2.0 finite-state machine, as specified with the TCL fsm setstate command in the script.
ID for ‘show call active voice id’ display:	Call identifier.
Legs	Any active call legs that are controlled by this session.
Services	Service name for the session if it registered as a service with the TCL service register command in the script. You can display a list of all registered services with the **show call application services registry** command.

Related Commands

Command	Description
call application session start (global configuration)	Starts a new instance (session) of a TCL application from global configuration mode.
call application session start (privileged EXEC)	Starts a new instance of a TCL application from privileged EXEC mode.
call application session stop	Stops a voice application session that is running.

Command	Description
show call application services registry	Displays a one-line summary of all registered services.

show call application voice

To display information about voice applications, use the **show call application voice** command in EXEC mode.

show call application voice [*name*| **summary**]

Syntax Description

name	(Optional) Name of the desired voice application. Output displays information about that application.
summary	(Optional) Output displays a one-line summary of each voice application.

Command Default

If both the *name* argument and **summary** keyword are omitted, command output displays detailed information about all interactive voice response (IVR) applications.

Command Modes

EXEC (#)

Command History

Release	Modification
11.3(6)NA2	This command was introduced.
12.0(3)T	This command was integrated into Cisco IOS Release 12.0(3)T.
12.1(5)T	This command was implemented on the Cisco AS5800.
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB	This command was modified to support VoiceXML applications.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on the Cisco 1750 and Cisco 1751. This command was not supported on any other platforms in this release.
12.2(8)T	This command was implemented on the Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, Cisco 3745, and Cisco 7200.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T for VoiceXML applications. This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release.
12.3(14)T	New output was added relating to the SCCP Telephony Control Application (STCAPP).

Usage Guidelines

The **show call application voice**command displays a detailed description of each configured application.

If the name of a specific application is entered, the command displays detailed information about only that application.

If the **summary** keyword is entered, the command displays a one-line summary about each application.

If STCAPP is enabled, the **summary** command displays STCAPP as an available call application.

If an asterisk is displayed next to the application name when the **summary** keyword is used, the application is configured, but not running. Normally this is because the application was not successfully loaded, for example:

```
name                 description
*vapptest2           flash:helloworld.vxml
```

TCL scripts and VoiceXML documents can be stored in any of the following locations: TFTP, FTP, or HTTP servers; Flash memory of the gateway; or the removable disks of the Cisco 3600 series. The audio files that they use can be stored in any of these locations and on RTSP servers.

Examples

The following example shows the output for the session Toolkit Command Language (TCL) script:

```
Router# show call application voice session
Application session
    The script is compiled into the image
    It has 0 calls active.
    Interpreted by infrastructure version 2.0
The TCL Script is:
------------------
# app_session.tcl
#---------------------------------
# August 1999, Saravanan Shanmugham
#
# Copyright (c) 1998, 1999, 2000, 2001 by cisco Systems, Inc.
# All rights reserved.
#---------------------------------
#
# This tcl script mimics the default SESSION app
#
# If DID is configured, just place the call to the dnis
# Otherwise, output dial-tone and collect digits from the
# caller against the dial-plan.
#
# Then place the call. If successful, connect it up, otherwise
# the caller should hear a busy or congested signal.
# The main routine just establishes the statemachine and then exits.
# From then on the system drives the statemachine depending on the
# events it recieves and calls the appropriate tcl procedure
#--------------------------------
#   Example Script
#--------------------------------
proc init { } {
    global param
    set param(interruptPrompt) true
    set param(abortKey) *
    set param(terminationKey) #
}
proc act_Setup { } {
    global dest
    global beep
    set beep 0
    if { [infotag get leg_isdid] } {
        set dest [infotag get leg_dnis]
```

```
            leg proceeding leg_incoming
            leg setup $dest callInfo leg_incoming
            fsm setstate PLACECALL
        } else {
            leg setupack leg_incoming
            playtone leg_incoming tn_dial
            set param(dialPlan) true
            leg collectdigits leg_incoming param
        }
    }
    proc act_GotDest { } {
        global dest
        set status [infotag get evt_status]
        if {  $status == "cd_004" } {
            set dest [infotag get evt_dcdigits]
            leg proceeding leg_incoming
            leg setup $dest callInfo leg_incoming
        } else {
            puts "\nCall [infotag get con_all] got event $status collecting destina"
            call close
        }
    }
    proc act_CallSetupDone { } {
        global beep
        set status [infotag get evt_status]
        if { $status == "ls_000"} {
            set  creditTimeLeft [infotag get leg_settlement_time leg_all]
            if { ($creditTimeLeft == "unlimited") ||
                 ($creditTimeLeft == "uninitialized") } {
                puts "\n Unlimited Time"
            } else {
                # start the timer for ...
                if { $creditTimeLeft < 10 } {
                    set beep 1
                    set delay $creditTimeLeft
                } else {
                    set delay [expr $creditTimeLeft - 10]
                }
                timer start leg_timer $delay leg_incoming
            }
        } else {
            puts "Call [infotag get con_all] got event $status collecting destinati"
            call close
        }
    }
    proc act_Timer { } {
        global beep
        global incoming
        global outgoing
        set incoming [infotag get leg_incoming]
        set outgoing [infotag get leg_outgoing]
        if { $beep == 0 } {
            #insert a beep ...to the caller
            connection destroy con_all
            set beep 1
        } else {
            connection destroy con_all
            fsm setstate LASTWARN
        }
    }
    proc act_LastWarn { } {
        media play leg_incoming flash:out_of_time.au
    }
    proc act_Destroy { } {
        media play leg_incoming flash:beep.au
    }
    proc act_Beeped { } {
        global incoming
        global outgoing
        connection create $incoming $outgoing
    }
    proc act_ConnectedAgain { } {
        timer start leg_timer 10 leg_incoming
```

```
}
proc act_Ignore { } {
# Dummy
    puts "Event Capture"
}
proc act_Cleanup { } {
    call close
}
init
#----------------------------------
#   State Machine
#------------------
  set fsm(any_state,ev_disconnected)  "act_Cleanup              same_state"
  set fsm(CALL_INIT,ev_setup_indication) "act_Setup             GETDEST"
  set fsm(GETDEST,ev_collectdigits_done)  "act_GotDest           PLACECALL"
  set fsm(PLACECALL,ev_setup_done)    "act_CallSetupDone    CALLACTIVE"
  set fsm(CALLACTIVE,ev_leg_timer)    "act_Timer            INSERTBEEP"
  set fsm(INSERTBEEP,ev_destroy_done) "act_Destroy          same_state"
  set fsm(INSERTBEEP,ev_media_done)   "act_Beeped           same_state"
  set fsm(INSERTBEEP,ev_create_done)    "act_ConnectedAgain   CALLACTIVE"
  set fsm(LASTWARN,ev_destroy_done)     "act_LastWarn        CALLDISCONNECT"
  set fsm(CALLACTIVE,ev_disconnected)   "act_Cleanup         CALLDISCONNECT"
  set fsm(CALLDISCONNECT,ev_disconnected)   "act_Cleanup         same_state"
  set fsm(CALLDISCONNECT,ev_media_done)     "act_Cleanup         same_state"
  set fsm(CALLDISCONNECT,ev_disconnect_done) "act_Cleanup         same_state"
  set fsm(CALLDISCONNECT,ev_leg_timer)      "act_Cleanup         same_state"
  fsm define fsm CALL_INIT
```

The following is sample output for the **summary** keyword:

```
Router# show call application voice summary
name                 description
session              Basic app to do DID, or supply dialtone.
fax_hop_on           Script to talk to a fax redialer
clid_authen          Authenticate with (ani, dnis)
clid_authen_collect  Authenticate with (ani, dnis), collect if that fails
clid_authen_npw      Authenticate with (ani, NULL)
clid_authen_col_npw  Authenticate with (ani, NULL), collect if that fails
clid_col_npw_3       Authenticate with (ani, NULL), and 3 tries collecting
clid_col_npw_npw     Authenticate with (ani, NULL) and 3 tries without pw
DEFAULT              Default system session application
lib_off_app          Libretto Offramp
TCL Script Version 2.0 supported.
TCL Script Version 1.1 supported.
Voice Browser Version 2.0 for VoiceXML 1.0 & 2.0 supported.
```

The following is sample output for the **summary** keyword when STCAPP is enabled:

```
Router# show call application voice summary

 SERVICES (standalone applications):
  name                      type            description
  ipsla-responder           Tcl Script      builtin:app_test_rcvr_script.tcl
  clid_authen               Tcl Script      builtin:app_clid_authen_script.tcl
  clid_col_npw_npw          Tcl Script      builtin:app_clid_col_npw_npw_script.tcl
  DEFAULT                   C Script        builtin:Session_Service.C
  CTAPP                     C Script        builtin:CallTreatment_Service.C
  clid_authen_col_npw       Tcl Script      builtin:app_clid_authen_col_npw_script.tcl
  fax_hop_on                Tcl Script      builtin:app_fax_hop_on_script.tcl
  ipsla-testcall            Tcl Script      builtin:app_test_place_script.tcl
  clid_authen_npw           Tcl Script      builtin:app_clid_authen_npw_script.tcl
  session                   Tcl Script      builtin:app_session_script.tcl
  clid_authen_collect       Tcl Script      builtin:app_clid_authen_collect_script.tcl
  clid_col_npw_3            Tcl Script      builtin:app_clid_col_npw_3_script.tcl
  lib_off_app               CCAPI           Libretto Offramp
  DEFAULT.C.OLD             CCAPI           Obsolete system session application
  stcapp                    CCAPI           SCCP Call Control Application
  MGCPAPP                   CCAPI           MGCP Application
```

The following is sample output for the *stcapp* keyword when the STCAPP is enabled:

```
Router# show call application voice stcapp
```

```
App Status:              Active
CCM Status:              UP
CCM Group:               2
Registration Mode:       CCM
Total Devices:           5
Total Calls in Progress: 0
Total Call Legs in Use:  0
```

The following is sample output from the show call application voice command for a VoiceXML application named vapptest1:

```
Router# show call application voice vapptest1
VXML Application vapptest1
    URL=flash:demo0.vxml
    Security not trusted
    No languages configured
    It has: 0 calls active.
          0 incoming calls
          0 calls handed off to it
          0 call transfers initiated
          0 pages loaded,  0 successful
          0 prompts played
          0 recorded messages
    Interpreted by Voice Browser Version 2.0 for VoiceXML 1.0 & 2.0.
The VXML Script is:
------------------
<?xml version="1.0"?>
<vxml version="1.0">
  <form>
     <block>
      <audio src="flash:demo0.au"/>
    </block>
  </form>
</vxml>
```

The table below describes the fields shown in the **show call application voice** display:

Table 23: show call application voice Field Descriptions

Field	Description
URL	Location of the document used by the application.
It has: *n* calls active.	Number of calls that are using this application.
incoming calls	Number of incoming public switched telephone network (PSTN) or IP calls that invoked this application.
calls handed off to it	Number of calls that were handed off to this application by another TCL or VoiceXML application.
call transfers initiated	Number of call transfers that were initiated by this application.
pages loaded	Number of VoiceXML pages that were loaded by the application.
successful	Number of VoiceXML pages that were completed.

Field	Description
prompts played	Number of audio prompts that were played by the application.
recorded messages	Number of audio recordings made by the VoiceXML application.
Interpreted by	Programming language used by the application.
The TCL or VoiceXML Script is	Content of the VoiceXML document or TCL script.

Related Commands

Command	Description
call application voice	Defines the name to be used for an application and indicates the location of the appropriate IVR script to be used with the application.
call application voice load	Reloads the designated TCL script or VoiceXML document.

show call fallback cache

To display the current Calculated Planning Impairment Factor (ICPIF) estimates for all IP addresses in cache, use the **show call fallback cache** command in EXEC mode.

show call fallback cache [*ip-address*]

Syntax Description

ip -address	(Optional) Specific IP address.

Command Modes

EXEC (#)

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

Use this command to clear all entries in the cache.

Examples

The following example displays output from this command:

```
Router# show call fallback cache
Probe   IP Address      Codec   Delay   Loss    ICPIF   Reject  Accept
-----   ----------      -----   -----   ----    -----   ------  ------
1       1.1.1.4         g729r8  40      0     0       0       9
2       122.24.56.25    g729r8  148     10      5       1       4
2 active probes
Field                      Description
-------                    ------------
Probe                      Probe number
IP Address                 IP Address to which the probe is sent
Codec                      Codec Type of the probe
Delay                      Delay in milliseconds that the probe incurred
Loss                       Loss in % that the probe incurred
ICPIF                      Computed ICPIF value for the probe
Reject                     Number of times that calls of Codec Type <Codec>
                           were rejected to the IP Address
Accept                     Number of times that calls of Codec Type <Codec>
                           were accepted to the IP Address
active probes              Number of destinations being probed
Router# show call fallback cache 10.14.115.53
Probe   IP Address      Codec           ICPIF   Reject  Accept
-----   ----------      -----           -----   ------  ------
1       10.14.115.53     g729r8          0       0       2
1 active probes
```

Field descriptions should be self-explanatory.

Related Commands

Command	Description
show call fallback stats	Displays call fallback statistics.

show call fallback config

To display the call fallback configuration, use the **show call fallback config** command in EXEC mode.

show call fallback config

Syntax Description

This command has no arguments or keywords.

Command Modes

EXEC (#)

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Examples

The following example displays output from the **show call fallback config** command:

```
Router# show call fallback config
VoIP fallback config:
Fallback is ON
Using ICPIF threshold:
        ICPIF value timeout:20 seconds
        ICPIF threshold:20
Number of packets in a probe:20
IP precedence of probe packets:2
Fallback cache size:2 entries
Fallback cache timeout:240 seconds
Instantaneous value weight:65
MD5 Keychain:secret
```

The table below describes the fields shown in the show call fallback config display.

Table 24: show call fallback config Field Descriptions

Field	Description
Fallback is	Lists enabled/disabled state of call fallback.
Using ICPIF threshold	ICPIF is configured to determine network traffic.
ICPIF value timeout	Lists probe timeout for collecting ICPIF information.
ICPIF threshold	Lists configured ICPIF threshold.
Number of packets in a probe	Lists number of configured packets per probe.

Field	Description
IP precedence of probe packets	Lists configured IP precedence for probes.
Fallback cache size	Number of allowed entries in call fallback cache.
Fallback cache timeout	Length of cache timeout, in seconds.
Instantaneous value weight	Lists weight configured for calculating cache entry based on new probe and last entry.
MD5 Keychain	MD5 authentication has been configured with a keychain of secret.

Related Commands

Command	Description
call fallback monitor	Enables the monitoring of destinations without fallback to alternate dial peers.
show voice trunk-conditioning signaling	Enables fallback to alternate dial peers in case of network congestion.

show call fallback stats

To display the call fallback statistics, use the **show call fallback stats** command in EXEC mode.

show call fallback stats

Syntax Description

This command has no arguments or keywords.

Command Modes

EXEC (#)

Command History

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

To remove all values, use the **clear call fallback stats** command.

Examples

The following example displays output from the **show call fallback stats** command:

```
Router# show call fallback stats
VOIP Fallback Stats:
Total accepted calls:3
Total rejected calls:1
Total cache overflows:1
Field                      Description
-------                    ------------
Total accepted calls       Number of times that calls were successful over IP.
Total rejected calls       Number of times that calls were rejected over IP.
Total cache overflows      Number of times that the fallback cache overflowed and required
 pruning.
The table below describes the fields shown in the show call fallback stats
display
```

Table 25: show call fallback stats Fields with Descriptions

Field	Description
Total accepted calls	Number of times that calls were successful over IP.
Total rejected calls	Number of times that calls were rejected over IP.
Total cache overflows	Number of times that the fallback cache overflowed and required pruning.

Related Commands

Command	Description
clear call fallback stats	Clears the call fallback statistics.
show call fallback cache	Displays the current ICPIF estimates for all IP addresses in the cache.

show call filter components

To display the components used for filtering calls, use the show call filter components command in privileged EXEC mode.

show call filter components

Command Default

No default behavior or values

Command Modes

Privileged EXEC (#)

Command History

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

Examples

The following example shows the output from running the show call filter components command. The GCFM is the generic call filter module, which is the internal module that controls which components are filtered:

```
Router# show call filter components
 The following components registered in GCFM:
     ISDN
     VTSP
     CCAPI
     TGRM
     DIAL-PEER
     NUMBER-TRANSLATION
     SSAPP
     VOICE-IVR-V2
     H323
     SIP
     CRM
```

The table below describes the significant fields shown in the display.

Table 26: show call filter components Field Descriptions

Field	Description
The following components registered in GCFM:	Shows which components are filtered in the generic call filter module.

Related Commands

Command	Description
call filter match-list voice	Create a call filter match list for debugging voice calls.

Command	Description
debug call filter inout	Display the debug trace inside the GCFM.
debug condition match-list	Run a filtered debug on a voice call.
outgoing port	Configure debug filtering for the outgoing port.
show call filter match-list	Display call filter match lists.

show call filter match-list

To display call filter match lists, use the show call filter match-list command in privileged EXEC mode.

show call filter match-list *tag*

Syntax Description

tag	Numeric label that uniquely identifies the match list.

Command Default

No default behavior or values

Command Modes

Privileged EXEC (#)

Command History

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

Examples

The following example shows an output from the show call filter match-list command:

```
Router# show call filter match-list

*********************************************
call filter match-list 9 voice
*********************************************
  incoming calling-number 50200
  incoming called-number 50201
  incoming signal local ipv4 10.0.101.22
  incoming signal remote ipv4  10.0.101.21
  incoming media local ipv4  10.0.101.22
  incoming media remote ipv4  10.0.101.21
  incoming dialpeer 502
  outgoing calling-number 50200
  outgoing called-number 50201
  outgoing port 6/0:D
  outgoing dialpeer 501
 debug condition match-list is set to EXACT_MATCH
*********************************************
call filter match-list 10 voice
*********************************************
  incoming calling-number 50300
  incoming called-number 50301
  incoming signal local ipv4 10.0.101.22
  incoming signal remote ipv4  10.0.101.21
  incoming media local ipv4  10.0.101.22
  incoming media remote ipv4  10.0.101.21
  incoming dialpeer 504
  outgoing calling-number 50300
  outgoing called-number 50301
  outgoing port 6/1:D
  outgoing dialpeer 503
 debug condition match-list is set to EXACT_MATCH
```

The table below describes the significant fields shown in the display.

Table 27: show call filter match-list Field Descriptions

Field	Description
call filter match-list 9 voice	Shows which match list is being displayed.
debug condition match-list is set to EXACT_MATCH	Shows whether the debug condition is set for exact match or partial match.

Related Commands

Command	Description
call filter match-list voice	Create a call filter match list for debugging voice calls.
debug call filter inout	Display the debug trace inside the GCFM.
debug condition match-list	Run a filtered debug on a voice call.
show call filter components	Display the components used for filtering calls.