Cisco IOS Voice Commands: Sh

This chapter contains commands to configure and maintain Cisco IOS voice applications. The commands are presented in alphabetical order. Some commands required for configuring voice may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

For detailed information on how to configure these applications and features, refer to the *Cisco IOS Voice Configuration Guide*.

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 MC3810 multiservice access concentrator and 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
```

Table 28 describes significant fields shown in this output.

Table 28 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 in privileged 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

On a Cisco MC3810, 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
```

Related Commands

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

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".

The status is either Group-OutOfService (no session in the group has been established) or Group-Inservice (at least one session in the group has been established).

The status (use) is 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 declared its intent yet.

```
Router# show backhaul-session-manager group status all

Session-Group
Group Name   :grp1
   Set Name     :set1
   Status       :Group-OutOfService
   Status (use) :Group-None
```

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 is supported on the Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release.

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
```

Table 29 describes significant fields shown in this output.

Table 29 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.
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
```

Table 30 describes the fields shown in the **show call accounting-template voice** display.

Table 30 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** *time* | **more** *time*}] | **id** *identifier*]

Syntax Description

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

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	Support for 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(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
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.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release.

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 that are currently connected. This command applies to 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]
RemoteIPAddress=0.0.0.0
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]
TxDuration=24010 ms
FaxTxDuration=10910 ms
FaxRate=14400
NoiseLevel=-1
ACOMLevel=-1
OutSignalLevel=0
InSignalLevel=0
InfoActivity=0
ERLLevel=-1
SessionTarget=
ImgPages=0
```

Table 31 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 31 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.
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. 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.
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.

Table 31 show call active fax Field Descriptions (continued)

Field	Description
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.
GENERIC	Generic or common parameters, that is, parameters that are common for Voice over IP (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.
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.
Last Buffer Drain/Fill Event	Elapsed time since the last jitter buffer drain or fill event, in seconds.
LogicalIfIndex	Index number of the logical interface for this call.
LoWaterPlayoutDelay	Low-water-mark Voice Playout FIFO Delay during this call.
LowerIFName	Physical lower interface information. Appears only if the medium is ATM, FR, or 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 (for Frame Relay), or HDLC (for High-Level Data Link Control).
Modem passthrough signaling method in use	Indicates that this is a modem pass-through call and that named service 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.

Table 31 show call active fax Field Descriptions (continued)

Field	Description
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.
Percent Packet Loss	Total percent packet loss.
ReceiveBytes	Number of bytes received by the peer during this call.
ReceiveDelay	Average Playout FIFO Delay plus the Decoder Delay during this voice call.
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 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 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 SIP call legs for which call records are available.
Telephony call-legs	Total telephony call legs for which call records are available.
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. 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
```

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 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** [**id** *identifier*] | **compact** [**duration** {**less** *time* | **more** *time*}] | **echo-canceller** *call-id* | **id** *identifier* | **redirect** {**rtpvt** | **tbct**}]

Syntax Description

brief	(Optional) Displays a truncated version.
compact	(Optional) Displays a compact version.
duration	(Optional) Displays active calls that are longer or shorter than a specified *time*. The arguments and keywords are as follows: • **less**—Displays calls shorter than *time*. • **more**—Displays calls longer than *time*. • *time*—Elapsed time, in seconds. Range is from 1 to 2147483647. There is no default value.
echo-canceller *call-id*	(Optional) Displays information about the state of the extended echo canceller (EC). To query the echo state, you need to know the hex ID in advance. To find the hex ID, enter the **show call active voice brief** command or use the **show voice call status** command. Range is from 0 to FFFFFFFF.
id *identifier*	(Optional) Displays only the call with the specified *identifier*. Range is a hex value from 1 to FFFF.
redirect	(Optional) Displays information about active calls that are being redirected using Release-to-Pivot (RTPvt) or Two B-Channel Transfer (TBCT). The keywords are: • **rtpvt**—Displays information about RTPvt calls. • **tbct**—Displays information about TBCT calls.

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	Support for 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(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.1(3)T	This command was implemented for modem pass-through over VoIP on the Cisco AS5300.

Release	Modification
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 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.
12.2(13)T	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)	The **redirect** keyword was 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 voice calls and fax relay calls that are currently connected through the router.

When using the **show call active voice** command to view output related to fax relay calls, refer to the field descriptions for the **show call active fax** command in Table 31 on page 1301 and for the **show call history fax command** in Table 38 on page 1330.

When the extended EC is present, the **show call active voice** command displays the contents of the Ditech EC_CHAN_CTRL structure. Table 32 contains name and descriptions of the fields in the EC_CHAN_CTRL structure.

Table 32 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 ms. 1 = 32 ms. 2 = 48 ms. 3 = 64 ms. **Note** This field should be set just higher 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 ear” when talking). 2 = Reserved. 3 = Generate comfort noise (default).
FRZ0	h-register hold	1 = Freezes H-register; used for testing.

Table 32 EC_CHAN_CTRL Field Descriptions (continued)

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 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.

See Table 31 for a listing of the information types associated with this command.

Use the **show call active voice redirect** command to monitor any currently active calls that are implementing RTPvt or TBCT.

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

Examples

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

```
Router# show call active voice

Total call-legs:2

 GENERIC:
SetupTime=7587246 ms
Index=1
PeerAddress=
PeerSubAddress=
PeerId=0
PeerIfIndex=0
LogicalIfIndex=0
ConnectTime=7587506
CallDuration=00:00:11
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=2
TransmitPackets=101
TransmitBytes=1991
ReceivePackets=550
```

```
ReceiveBytes=11000
VOIP:
ConnectionId[0x7F8D82A4 0x928E11D5 0x8094FCFB 0x1C38F0FA]
IncomingConnectionId[0x7F8D82A4 0x928E11D5 0x8094FCFB 0x1C38F0FA]
RemoteIPAddress=172.29.248.111
RemoteUDPPort=17394
RoundTripDelay=4 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
FastConnect=TRUE

AnnexE=FALSE

Separate H245 Connection=FALSE

H245 Tunneling=FALSE

SessionProtocol=cisco
SessionTarget=
OnTimeRvPlayout=10300
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=70 ms
LoWaterPlayoutDelay=69 ms
ReceiveDelay=69 ms
LostPackets=0
EarlyPackets=0
LatePackets=0
VAD = enabled
CoderTypeRate=g729r8
CodecBytes=20
SignalingType=ext-signal
CallerName=
CallerIDBlocked=False
 GENERIC:
SetupTime=7587246 ms
Index=2
PeerAddress=133001
PeerSubAddress=
PeerId=133001
PeerIfIndex=8
LogicalIfIndex=7
ConnectTime=7587505
CallDuration=00:00:56
CallState=4
CallOrigin=1
ChargedUnits=0
InfoType=2
TransmitPackets=2801
TransmitBytes=56020
ReceivePackets=162
ReceiveBytes=3192
TELE:
ConnectionId=[0x7F8D82A4 0x928E11D5 0x8094FCFB 0x1C38F0FA]
IncomingConnectionId=[0x7F8D82A4 0x928E11D5 0x8094FCFB 0x1C38F0FA]
TxDuration=56030 ms
VoiceTxDuration=3210 ms
FaxTxDuration=0 ms
CoderTypeRate=g729r8
NoiseLevel=-44
ACOMLevel=-13
OutSignalLevel=-45
```

```
InSignalLevel=-45
InfoActivity=2
ERLLevel=7
EchoCancellerMaxReflector=64
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
```

Note Table 33 on page 1311 describes 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=1.7.95.3
RemoteUDPPort=16610
RemoteSignallingIPAddress=1.7.95.3
RemoteSignallingPort=1720
RemoteMediaIPAddress=1.7.95.3
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=ipv4:1.7.95.3
```

```
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=4085452930
GwReceivedCallingOctet3=0x0
GwReceivedCallingOctet3a=0x80
GwOutpulsedCallingNumber=4085550130
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
```

Note Table 31 on page 1301, Table 33 on page 1311, and Table 38 on page 1330 describe fields in the display.

Table 33 show call active voice Field Descriptions

Field	Description
ACOM Level	Current ACOM level for this call. This value is the sum of the echo return loss, echo return loss enhancement, and nonlinear processing loss for this call.
CallOrigin	Call origin: answer or originate.
CallState	Current state of the call.
CoderTypeRate	Negotiated coder transmit rate of voice or fax compression during this call.
ConnectionId	Global call identifier for this gateway call.
ConnectTime	Time at which the call was connected.
Delay	Amount of receiver delay. This is the current amount of de-jitter delay plus the processing delay.
Dial-Peer	Tag of the dial peer that is transmitting this call.
Early	Count of number of early packets received. Early packet is defined as a packet which arrives with the lowest delay so far. This event causes the DSP channel to recompute the minimum delay value. The adaptive de-jitter algorithm uses the minimum delay for computing jitter and setting the appropriate delay in the de-jitter buffer. This stat doesn't add a lot of value in determining voice quality.
EchoCancellerMaxReflector=64	The location of the largest reflector, in milliseconds. 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.
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 the voice signal played out with the 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 the 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 or frame-concealment strategies in G.729 and G.723.1 compression algorithms.
GapFillWithRedundancy	Duration of the voice signal played out with the signal synthesized from redundancy parameters available because voice data was lost or not received in time from the voice gateway for this call.
GapFillWith Silence	Duration of the 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.
Index	Dial-peer identification number.
InfoActivity	Active information transfer activity state for this call.
InfoType	Information type for this call.

Table 33 show call active voice Field Descriptions (continued)

Field	Description
InSignalLevel	Active input signal level from the telephony interface used by this call.
LogicalIfIndex	Index number of the logical interface for this call.
Lost	Number of lost RTP packets. This is computed by Cisco IOS software using the sequence numbers in the RTP header.
LoWaterPlayoutDelay	Low-water-mark Voice Playout FIFO Delay during this call.
NoiseLevel	Active noise level for this call.
OnTimeRvPlayout	Duration of the voice playout from data received in time for this call. You can derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values.
OutSignalLevel	Active output signal level to telephony interface used by this call.
PeerAddress	Destination pattern 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.
PeerSubaddress	Subaddress to which 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 call.
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 transmitted.
RoundTripDelay	Voice packet round-trip delay between the local and remote system on the IP backbone during this call.
SelectedQoS	Selected Resource Reservation Protocol. Protocol (RSVP) quality of service (QoS) for this call.
SessionProtocol	Session protocol used for an Internet call between the local and remote router via the IP backbone.
SessionTarget	Session target of the peer used for this call.
SetupTime	Value of the system UpTime when the call associated with this entry was started.
TransmitBytes	Number of bytes transmitted from this peer during this call.
TransmitPackets	Number of packets transmitted from this peer during this call.
TxDuration	Duration of transmit path open from this peer to the voice gateway for this call.
VADEnable	Whether voice activity detection (VAD) was enabled for this call.
VoiceTxDuration	Duration of voice transmission from this peer to the voice gateway for this call. You can 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 brief** command:

```
Router# show call active voice brief

<ID>:<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>
  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>
  sig:<on/off> <codec> (payload size)
ATM <protocol> [int vpi/vci 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
MODEMRELAY info:<rcvd>/<sent>/<resent> xid:<rcvd>/<sent>
total:<rcvd>/<sent>/<drops>
 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 example above) 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 hex if you want to use the **show call active voice echo-canceller x** command (x = call ID converted to hex).

The following are call ID examples converted to hex (generally incremented by 2):

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

Alternatively, you can use the **show voice call status** command to obtain the call ID. The call ID output is already in hex 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 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
```

Table 34 describes significant fields shown in the **show call active voice redirect** display.

Table 34 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 of TBCT calls currently being monitored	Total number of currently active TBCT calls.
ctrl name	Name of the T1 controller where the call originated.
tag	Call tag number that identifies the call.

Table 34 show call active voice redirect Field Descriptions (continued)

Field	Description
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 T.37 store-and-forward 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 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.

Defaults

If both the *name* argument and **summary** keyword are omitted, command output displays detailed information about all 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.

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 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 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 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>
```

Table 35 describes the fields shown in the **show call application voice** display:

Table 35 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 currently using this application.
incoming calls	Number of incoming 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 successfully completed.
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

To clear all entries in the cache, use the **clear call fallback cache** command.

Examples

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

```
Router# show call fallback cache

Probe   IP Address      Codec   Delay   Loss    ICPIF   Reject  Accept
-----   ----------      -----   -----   ----    -----   ------  ------
1       1.1.1.4         g729r8  40      00       0       9
2       122.24.56.25    g729r8  14810      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
```

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
```

Table 36 describes the fields shown in the **show call fallback config** display.

Table 36 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.
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.

Table 36 show call fallback config Field Descriptions (continued)

Field	Description
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.
```

Table 37 describes the fields shown in the **show call fallback stats** display.

Table 37 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 history fax

To display the call history table for fax transmissions, use the **show call history fax** command in user EXEC or privileged EXEC mode.

show call history fax [**brief** [**id** *identifier*] | **compact** [**duration** {**less** *time* | **more** *time*}] | [**id** *identifier*] | **last** *number*]

Syntax Description

brief	(Optional) Displays a truncated version of the call history table.
compact	(Optional) Displays a compact version.
duration *time*	(Optional) Displays history information for calls that are longer or shorter than a specified *time*. The arguments and keywords are as follows: • **less**—Displays calls shorter than *time*. • **more**—Displays calls longer than *time*. • *time*—Elapsed time, in seconds. Range is from 1 to 2147483647.
id *identifier*	(Optional) Displays only the call with the specified *identifier*. Range is a hex value from 1 to FFFF.
last *number*	(Optional) Displays the last calls connected, where the number of calls that appear is defined by the *number* argument. Range is from 1 to 100.

Command Modes

User EXEC
Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
12.0(3)XG	This command was implemented for Voice over Frame Relay (VoFR) on the Cisco 2600 series and Cisco 3600 series.
12.0(4)XJ	This command was modified for store-and-forward fax.
12.0(4)T	The **brief** keyword was added, and the command was implemented on the Cisco 7200 series.
12.0(7)XK	The **brief** keyword was implemented on the Cisco MC3810.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.1(5)XM	This command was implemented on the Cisco AS5800.
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XA	The output of this command was modified to indicate whether the call in question has been established using Annex E.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	Support for the Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 is not included in this release.

Release	Modification
12.2(11)T	This command is supported on the Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 in this release.
12.3(1)	The following fields were added: FaxRelayMaxJitterBufDepth, FaxRelayJitterBufOverFlow, FaxRelayHSmodulation, and FaxRelayNumberOfPages.

Usage Guidelines

This command displays a call-history table that contains a list of fax calls connected through the router in descending time order. The maximum number of calls contained in the table can be set to a number between 0 and 500 using the **dial-control-mib** command in global configuration mode. The default maximum number of table entries is 50. Each call record is aged out of the table after a configurable number of minutes has elapsed, also specified by the **dial-control-mib** command. The default timer value is 15 minutes.

You can display subsets of the call history table by using specific keywords. To display the last calls connected through this router, use the keyword **last**, and define the number of calls to be displayed with the *number* argument.

To display a truncated version of the call history table, use the **brief** keyword.

This command applies to both on-ramp and off-ramp store-and-forward fax functions.

Examples

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

```
Router# show call history fax

Telephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 0
MGCP call-legs: 0
Total call-legs: 1


GENERIC:
SetupTime=590180 ms
Index=2
PeerAddress=4085452930
PeerSubAddress=
PeerId=81
PeerIfIndex=221
LogicalIfIndex=145
DisconnectCause=10
DisconnectText=normal call clearing (16)
ConnectTime=59389
DisconnectTime=68204
CallDuration=00:01:28
CallOrigin=2
ReleaseSource=1
ChargedUnits=0
InfoType=fax
TransmitPackets=295
TransmitBytes=5292
ReceivePackets=2967
ReceiveBytes=82110
TELE:
ConnectionId=[0xD9ACDFF1 0x9F5D11D7 0x8002CF18 0xB9C3632]
IncomingConnectionId=[0xD9ACDFF1 0x9F5D11D7 0x8002CF18 0xB9C3632]
```

```
TxDuration=28960 ms
VoiceTxDuration=0 ms
FaxTxDuration=28960 ms
FaxRate=voice bps
FaxRelayMaxJitterBufDepth  = 0 ms
FaxRelayJitterBufOverFlow = 0
FaxRelayHSmodulation = 0
FaxRelayNumberOfPages = 0
NoiseLevel=-120
ACOMLevel=127
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=4085550130
OriginalCallingOctet=0x0
OriginalCalledNumber=52930
OriginalCalledOctet=0xE9
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0xFF
TranslatedCallingNumber=4085550130
TranslatedCallingOctet=0x0
TranslatedCalledNumber=52930
TranslatedCalledOctet=0xE9
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwReceivedCalledNumber=52930
GwReceivedCalledOctet3=0xE9
GwReceivedCallingNumber=4085550130
GwReceivedCallingOctet3=0x0
GwReceivedCallingOctet3a=0x80
```

Note Table 31 on page 1301 describes the significant fields shown in the display.

Table 38 describes the additional fields not shown in Table 31.

Table 38 show call history fax Field Descriptions

Field	Description
CallerName	Voice port station name string.
DisconnectCause	Cause code for the reason this call was disconnected.
DisconnectText	Descriptive text explaining the reason for the disconnect.
DisconnectTime	Time, in milliseconds, when this call was disconnected.
FaxRelayMaxJitterBufDepth	Maximum depth of jitter buffer (in msec)
FaxRelayJitterBufOverFlow	Count of number of network jitter buffer overflows (# of packets). These packets are equivelent to lost packets in their effect.
FaxRelayHSmodulation	Most recent high-speed modulation used.
FaxRelayNumberOfPages	Number of pages transmitted.
GwReceivedCalledNumber, GwReceivedCalledOctet3, GwReceivedCallingNumber, GwReceivedCallingOctet3, GwReceivedCallingOctet3a	Call information received at the gateway.

Table 38 show call history fax Field Descriptions (continued)

Field	Description
ImgPages	The fax pages that have been processed.
Incoming ConnectionId	The incoming_GUID. It can be different with ConnectionId (GUID) when there is long_pound or blast_call feature involved. In those cases, incoming_GUID is unique for all the subcalls that have been generated, and GUID is different for each subcall.
OriginalCallingNumber, OriginalCalling Octet, OriginalCalledNumber, OriginalCalledOctet, OriginalRedirectCalledNumber, OriginalRedirectCalledOctet	Original call information regarding calling, called, and redirect numbers, as well as 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.
TranslatedCallingNumber, TranslatedCallingOctet, TranslatedCalledNumber, TranslatedCalledOctet, TranslatedRedirectCalledNumber, TranslatedRedirectCalledOctet	Translated call information.

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

```
Router# show call history fax brief

<ID>: <start>hs.<index> +<connect> +<disc> pid:<peer_id> <direction> <addr>
 tx:<packets>/<bytes> rx:<packets>/<bytes> <disc-cause>(<text>)
 IP <ip>:<udp> rtt:<time>ms pl:<play>/<gap>ms lost:<lost>/<early>/<late>
  delay:<last>/<min>/<max>ms <codec>
 Telephony <int>: tx:<tot>/<voice>/<fax>ms <codec> noise:<lvl>dBm acom:<lvl>dBm

2    : 5996450hs.25 +-1 +3802 pid:100 Answer 408
 tx:0/0 rx:0/0 1F  (T30 T1 EOM timeout)
 Telephony : tx:38020/38020/0ms g729r8 noise:0dBm acom:0dBm

2    : 5996752hs.26 +-1 +3500 pid:110 Originate uut1@linux2.allegro.com
 tx:0/0 rx:0/0 3F  (The e-mail was not sent correctly. Remote SMTP server said: 354 )
 IP 14.0.0.1 AcceptedMime:0 DiscardedMime:0

3    : 6447851hs.27 +1111 +3616 pid:310 Originate 576341.
 tx:11/14419 rx:0/0 10  (Normal connection)
 Telephony : tx:36160/11110/25050ms g729r8 noise:115dBm acom:-14dBm

3    : 6447780hs.28 +1182 +4516 pid:0 Answer
 tx:0/0 rx:0/0 10  (normal call clearing.)
 IP 0.0.0.0 AcceptedMime:0 DiscardedMime:0

4    : 6464816hs.29 +1050 +3555 pid:310 Originate 576341.
 tx:11/14413 rx:0/0 10  (Normal connection)
 Telephony : tx:35550/10500/25050ms g729r8 noise:115dBm acom:-14dBm

4    : 6464748hs.30 +1118 +4517 pid:0 Answer
 tx:0/0 rx:0/0 10  (normal call clearing.)
 IP 0.0.0.0 AcceptedMime:0 DiscardedMime:0

5    : 6507900hs.31 +1158 +2392 pid:100 Answer 4085763413
 tx:0/0 rx:3/3224 10  (Normal connection)
```

```
 Telephony : tx:23920/11580/12340ms g729r8 noise:0dBm acom:0dBm

5    : 6508152hs.32 +1727 +2140 pid:110 Originate uut1@linux2.allegro.com
 tx:0/2754 rx:0/0 3F  (service or option not available, unspecified)
 IP 14.0.0.4 AcceptedMime:0 DiscardedMime:0

6    : 6517176hs.33 +1079 +3571 pid:310 Originate 576341.
 tx:11/14447 rx:0/0 10  (Normal connection)
 Telephony : tx:35710/10790/24920ms g729r8 noise:115dBm acom:-14dBm

6    : 6517106hs.34 +1149 +4517 pid:0 Answer
 tx:0/0 rx:0/0 10  (normal call clearing.)
 IP 0.0.0.0 AcceptedMime:0 DiscardedMime:0

7    : 6567382hs.35 +1054 +3550 pid:310 Originate 576341.
 tx:11/14411 rx:0/0 10  (Normal connection)
 Telephony : tx:35500/10540/24960ms g729r8 noise:115dBm acom:-14dBm

7    : 6567308hs.36 +1128 +4517 pid:0 Answer
tx:0/0 rx:0/0 10  (normal call clearing.)
 IP 0.0.0.0 AcceptedMime:0 DiscardedMime:0
```

Related Commands

Command	Description
show call active fax	Displays call information for T.37 store-and-forward fax transmissions that are in progress.
show call active voice	Displays call information for voice calls that are in progress.
show call history voice	Displays the call history table for voice calls.
show dial-peer voice	Displays configuration information for dial peers.
show num-exp	Displays how the number expansions are configured in VoIP.
show voice port	Displays configuration information about a specific voice port.

show call history video record

To display information about incoming and outgoing video calls, use the **show call history video record** command in privileged EXEC mode.

show call history video record

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.

Examples

On a Cisco MC3810, the following example displays information about two video calls:

```
Router# show call history video record

CallId = 4
CalledNumber = 221
CallDuration = 39006 seconds
DisconnectText = remote hangup
SVC: call ID = 8598630
Remote NSAP = 47.0091810000000002F26D4901.00107B09C645.C8
Local NSAP = 47.0091810000000002F26D4901.00107B4832E1.C8
vcd = 414, vpi = 0, vci = 158
SerialPort = Serial0
VideoSlot = 1, VideoPort = 0
CallId = 3
CalledNumber = 221
CallDuration = 557 seconds
DisconnectText = local hangup
SVC: call ID = 8598581
Remote NSAP = 47.0091810000000002F26D4901.00107B09C645.C8
Local NSAP = 47.0091810000000002F26D4901.00107B4832E1.C8
vcd = 364, vpi = 0, vci = 108
SerialPort = Serial0
VideoSlot = 1, VideoPort = 0
```

show call history voice

To display the call history table for voice calls, use the **show call history voice** command in user EXEC or privileged EXEC mode.

show call history voice [**brief** [**id** *identifier*] | **compact** [**duration** {**less** *time* | **more** *time*}] | [**id** *identifier*] | **last** *number* | **redirect** {**rtpvt** | **tbct**}]

Syntax Description

brief	(Optional) Displays a truncated version of the call history table.
compact	(Optional) Displays a compact version.
duration *time*	(Optional) Displays history information for calls that are longer or shorter than a specified *time*. The arguments and keywords are as follows: • **less**—Displays calls shorter than *time*. • **more**—Displays calls longer than *time*. • *time*—Elapsed time, in seconds. Range is from 1 to 2147483647.
id *identifier*	(Optional) Displays only the call with the specified *identifier*. Range is from 1 to FFFF.
last *number*	(Optional) Displays the last calls connected, where the number of calls that appear is defined by the *number* argument. Range is from 1 to 100.
redirect	(Optional) Displays information about calls that were redirected using Release-to-Pivot (RTPvt) or Two B-Channel Transfer (TBCT). The keywords are as follows: • **rtpvt**—Displays information about RTPvt calls. • **tbct**—Displays information about TBCT calls.

Command Modes

User EXEC
Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
12.0(3)XG	This command was implemented for Voice over Frame Relay (VoFR) on the Cisco 2600 series and Cisco 3600 series.
12.0(4)XJ	This command was modified for store-and-forward fax.
12.0(4)T	The **brief** keyword was added, and the command was implemented on the Cisco 7200 series.
12.0(5)XK	This command was introduced on Cisco MC3810 multiservice access concentrators.
12.0(7)XK	The **brief** keyword was implemented on the Cisco MC3810.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0(7)T.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.1(5)XM	This command was implemented on the Cisco AS5800.

Release	Modification
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XA	The output of this command was modified to indicate whether the call in question has been established using Annex E.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	Support for the Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 is not included in this release.
12.2(11)T	This command is supported on the Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 in this release.
12.2(13)T	The ReleaseSource field was added to the Field Description table, and the word **record** was deleted from the command name.
12.3(1)	The **redirect** keyword was added.

Usage Guidelines

This command displays a call-history table that contains a list of voice calls connected through the router in descending time order. The maximum number of calls contained in the table can be set to a number between 0 and 500 using the **dial-control-mib** command in global configuration mode. The default maximum number of table entries is 50. Each call record is aged out of the table after a configurable number of minutes has elapsed, also specified by the **dial-control-mib** command. The default timer value is 15 minutes.

You can display subsets of the call history table by using specific keywords. To display the last calls connected through this router, use the keyword **last**, and define the number of calls to be displayed with the *number* argument.

To display a truncated version of the call history table, use the **brief** keyword.

Use the **show call active voice redirect** command to review records for calls that implemented RTPvt or TBCT.

When a call is active, you can view its statistics by using the **show call active voice** command.

Examples

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

```
Router# show call history voice

GENERIC:
SetupTime=104648 ms
Index=1
PeerAddress=55240
PeerSubAddress=
PeerId=2
PeerIfIndex=105
LogicalIfIndex=0
DisconnectCause=10
DisconnectText=normal call clearing.
ConnectTime=104964
DisconectTime=143329
CallDuration=00:06:23
CallOrigin=1
ChargedUnits=0
InfoType=speech
TransmitPackets=37668
TransmitBytes=6157536
```

```
ReceivePackets=37717
ReceiveBytes=6158452
VOIP:
ConnectionId[0x4B091A27 0x3EDD0003 0x0 0xFEFD4]
RemoteIPAddress=1.14.82.14
RemoteUDPPort=18202
RoundTripDelay=2 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
FastConnect=TRUE

SessionProtocol=cisco
SessionTarget=ipv4:1.14.82.14
OnTimeRvPlayout=40
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=67 ms
LoWaterPlayoutDelay=67 ms
ReceiveDelay=67 ms
LostPackets=0 ms
EarlyPackets=0 ms
LatePackets=0 ms
VAD = enabled
CoderTypeRate=g729r8
CodecBytes=20
cvVoIPCallHistoryIcpif=0
SignalingType=cas

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 = 373sec
Time between Buffer Drain/Fills = Min 0sec Max 0sec

GENERIC:
SetupTime=104443 ms
Index=2
PeerAddress=50110
PeerSubAddress=
PeerId=100
PeerIfIndex=104
LogicalIfIndex=10
DisconnectCause=10
DisconnectText=normal call clearing.
ConnectTime=104964
DisconectTime=143330
CallDuration=00:06:23
CallOrigin=2
ChargedUnits=0
InfoType=speech
TransmitPackets=37717
TransmitBytes=5706436
ReceivePackets=37668
ReceiveBytes=6609552
TELE:
ConnectionId=[0x4B091A27 0x3EDD0003 0x0 0xFEFD4]
TxDuration=375300 ms
VoiceTxDuration=375300 ms
FaxTxDuration=0 ms
```

```
CoderTypeRate=g711ulaw
NoiseLevel=-75
ACOMLevel=11
SessionTarget=
ImgPages=0
```

The following example from a Cisco AS5350 router displays a sample of voice call history records showing release source information:

```
Router# show call history voice

Telephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 1
Total call-legs: 2

GENERIC:
SetupTime=85975291 ms
.
.
.
DisconnectCause=10
DisconnectText=normal call clearing (16)
ConnectTime=85975335
DisconnectTime=85979339
CallDuration=00:00:40
CallOrigin=1
ReleaseSource=1
.

.
.
DisconnectCause=10
DisconnectText=normal call clearing (16)
ConnectTime=85975335
DisconnectTime=85979339
CallDuration=00:00:40
CallOrigin=1
ReleaseSource=1
.
.
.
VOIP:
ConnectionId[0x2868AD84 0x375B11D4 0x8012F7A5 0x74DE971E]
.
.
.
GENERIC:
SetupTime=85975290 ms
.
.
.
DisconnectCause=10
DisconnectText=normal call clearing (16)
ConnectTime=85975336
DisconnectTime=85979340
CallDuration=00:00:40
CallOrigin=2
ReleaseSource=1
.
.
.
TELE:
ConnectionId=[0x2868AD84 0x375B11D4 0x8012F7A5 0x74DE971E]
```

Note Table 31 and Table 38 describe the significant fields shown in the display.

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

```
Router# show call history voice brief

<ID>: <start>hs.<index> +<connect> +<disc> pid:<peer_id> <direction> <addr>
  dur hh:mm:ss tx:<packets>/<bytes> rx:<packets>/<bytes> <disc-cause>(<text>)
 IP <ip>:<udp> rtt:<time>ms pl:<play>/<gap>ms lost:<lost>/<early>/<late>
  delay:<last>/<min>/<max>ms <codec>
  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>
  sig:<on/off> <codec> (payload size)
 ATM <protocol> [int vpi/vci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
  sig:<on/off> <codec> (payload size)
 Telephony <int>: tx:<tot>/<voice>/<fax>ms <codec> noise:<lvl>dBm acom:<lvl>dBm
```

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

```
Router# show call history voice redirect tbct

index=2, xfr=tbct-notify, status=redirect_success, start_time=*00:12:25.981 UTC Mon Mar 1
1993, ctrl name=T1-2/0,  tag=13
index=3, xfr=tbct-notify, status=redirect_success, start_time=*00:12:25.981 UTC Mon Mar 1
1993, ctrl name=T1-2/0,  tag=13
index=4, xfr=tbct-notify, status=redirect_success, start_time=*00:13:07.091 UTC Mon Mar 1
1993, ctrl name=T1-2/0,  tag=12
index=5, xfr=tbct-notify, status=redirect_success, start_time=*00:13:07.091 UTC Mon Mar 1
1993, ctrl name=T1-2/0,  tag=12

Number of call-legs redirected using tbct with notify:4
```

Table 39 describes significant fields shown in the **show call history voice redirect tbct** display.

Table 39 show call history voice redirect Field Descriptions

Field	Description
index	Index number of the record in the history file.
xfr	Whether TBCT or TBCT with notify has been invoked.
status	Status of the redirect request.
start_time	Time, in hours, minutes, and seconds when the redirected call began.
ctrl name	Name of the T1 controller where the call originated.
tag	Call tag number that identifies the call.
Number of call-legs redirected using tbct with notify	Total number of call legs that were redirected using TBCT with notify.

Related Commands

Command	Description
show call active fax	Displays call information for fax transmissions that are in progress.
show call active voice	Displays call information for voice calls that are in progress.
show call history fax	Displays the call history table for fax transmissions.
show dial-peer voice	Displays configuration information for dial peers.
show num-exp	Displays how the number expansions are configured in VoIP.
show voice port	Displays configuration information about a specific voice port.

show call language voice

To display a summary of languages configured and the URLs of the corresponding Tool Command Language (TCL) modules for the languages that are not built-in languages, use the **show call language voice** command in EXEC mode.

show call language voice [*language* | **summary**]

Syntax Description

language	(Optional) Two-character prefix configured with the **call language voice** command in global configuration mode, either for a prefix for a built-in language or one that you have defined; for example, “en” for English or “ru” for Russian.
summary	(Optional) Summary of all the languages configured and the URLs for the TCL modules other than built-in languages.

Command Modes

EXEC

Command History

Release	Modification
12.2(2)T	This command was introduced.

Usage Guidelines

This command is similar to the **show call application voice** command. If a language is built in, the URL listed reads “fixed.” If you decide to overwrite the built-in language with your own language, the word “fixed” in the URL column changes to the actual URL where your new application lives.

Examples

The following command displays a summary of the configured languages:

```
Router# show call language voice summary

name       url
sp         fixed
ch         fixed
en         fixed
ru         tftp://dirt/fwarlau/scripts/multilag/ru_translate.tcl
```

The following command displays information about Russian-language configuration:

```
Router# show call language voice ru

ru_translate.tcl
ru_translate.tcl~
singapore.cfg
test.tcl
people% more ru_translate.tcl
# Script Locked by: farmerj
# Script Version: 1.1.0.0
# Script Lock Date: Sept 24 2000
# ca_translate.tcl
#-----------------------------------------------------------------
# Sept 24, 2000 Farmer Joe
#
```

```
# Copyright (c) 2000 by Cisco Systems, Inc.
# All rights reserved.
#------------------------------------------------------------------
#<snip>...
...set prefix ""
#puts "argc"

#foreach arg $argv {
#puts "$arg"

#    translates $arg

#    puts "\t\t**** $prompt RETURNED"
#}
```

Related Commands

Command	Description
call language voice	Configures a TCL module.
call language voice load	Loads or reloads a TCL module from the configured URL location.
debug voip ivr	Specifies the type of VoIP IVR debug output that you want to view.
show call application voice	Shows and describes applications.

show call resource voice stats

To display resource statistics for an H.323 gateway, use the **show call resource voice stats** command in privileged EXEC mode.

show call resource voice stats

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(5)T	This command was introduced on the Cisco AS5300.
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T. This command is not operational on the Cisco AS5300, Cisco AS5350, and Cisco AS5400 in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	Support for the Cisco AS5300,Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 is not included in this release.
12.2(11)T	This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 in this release.

Usage Guidelines

This command displays the H.323 resources that are monitored when the **resource threshold** command is used to configure and enable resource threshold reporting.

Examples

The following example shows the resource statistics for an H.323 gateway:

```
Router# show call resource voice stats

Resource Monitor -  Dial-up Resource Statistics Information:

DSP Statistics:

Utilization: 0 percent
Total channels: 48
Inuse channels: 0
Disabled channels 0:
Pending channels: 0
Free channels: 48

DS0 Statistics:

Total channels: 0
Addressable channels: 0
Inuse channels: 0
Disabled channels: 0
```

```
Free channels: 0
```

Table 40 describes significant fields shown in this output.

Table 40 show call resource voice stats Field Descriptions

Statistic	Definition
Total channels	Number of channels physically configured for the resource.
Addressable channels	Number of channels that can be used for a specific type of dialup service, such as H.323, which includes all the DS0 resources that have been associated with a voice plain old telephone service (POTS) dial plan profile.
Inuse channels	Number of addressable channels that are in use. This value includes all channels that either have active calls or have been reserved for testing.
Free channels	Number of addressable channels that are free.
Pending channels	Number of addressable channels that are pending in loadware download.
Disabled channels	Number of addressable channels that are physically down or that have been disabled administratively with the **shutdown** or **busyout** command.

Related Commands

Command	Description
resource threshold	Configures a gateway to report H.323 resource availability to the gatekeeper of the gateway.
show call resource voice threshold	Displays the threshold configuration settings and status for an H.323 gateway.

show call resource voice threshold

To display the threshold configuration settings and status for an H.323 gateway, use the **show call resource voice threshold** command in privileged EXEC mode.

show call resource voice threshold

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(5)T	This command was introduced on the Cisco AS5300.
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(4)T	This command does not support the Cisco AS5300, Cisco AS5350, and Cisco AS5400 in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	Support for the Cisco AS5300,Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release.

Usage Guidelines

This command displays the H.323 resource thresholds that are configured with the **resource threshold** command.

Examples

The following example shows the resource threshold settings and status for an H.323 gateway:

```
Router# show call resource voice threshold

Resource Monitor -  Dial-up Resource Threshold Information:

DS0 Threshold:

Client Type: h323
High Water Mark: 70
Low Water Mark: 60
Threshold State: init
DSP Threshold:

Client Type: h323
High Water Mark: 70
Low Water Mark: 60
Threshold State: low_threshold_hit
```

Related Commands

Command	Description
resource threshold	Configures a gateway to report H.323 resource availability to the gatekeeper of the gateway.
show call resource voice stats	Displays resource statistics for an H.323 gateway.

show call rsvp-sync conf

To display the configuration settings for Resource Reservation Protocol (RSVP) synchronization, use the **show call rsvp-sync conf** command in privileged EXEC mode.

show call rsvp-sync conf

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(3)XI1	This command was introduced on the Cisco 2600 series, Cisco 3600 series, Cisco 7200, Cisco MC3810, Cisco AS5300, and Cisco AS5800.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	Support for the Cisco AS5300,Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 is not included in this release.
12.2(11)T	This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 in this release.

Examples

The following example shows sample output from this command:

```
Router# show call rsvp-sync conf

VoIP QoS: RSVP/Voice Signaling Synchronization config:

Overture Synchronization is ON
Reservation Timer is set to 10 seconds
```

Table 41 describes significant fields shown in this output.

Table 41 show call rsvp-sync conf Field Descriptions

Field	Description
Overture Synchronization is ON	Indicates whether RSVP synchronization is enabled.
Reservation Timer is set to xx seconds	Number of seconds for which the RSVP reservation timer is configured.

Related Commands

Command	Description
call rsvp-sync	Enables synchronization between RSVP and the H.323 voice signaling protocol.
call rsvp-sync resv-timer	Sets the timer for RSVP reservation setup.
debug call rsvp-sync events	Displays the events that occur during RSVP synchronization.
show call rsvp-sync stats	Displays statistics for calls that attempted RSVP reservation.

show call rsvp-sync stats

To display statistics for calls that attempted Resource Reservation Protocol (RSVP) reservation, use the **show call rsvp-sync stats** command in privileged EXEC mode.

show call rsvp-sync stats

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(3)XI1	This command was introduced.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Examples

The following example shows sample output from this command:

```
Router# show call rsvp-sync stats

VoIP QoS:Statistics Information:
Number of calls for which QoS was initiated   : 18478
Number of calls for which QoS was torn down   : 18478
Number of calls for which Reservation Success was notified : 0
Total Number of PATH Errors encountered : 0
Total Number of RESV Errors encountered : 0
Total Number of Reservation Timeouts encountered : 0
```

Table 42 describes significant fields shown in this output.

Table 42 show call rsvp-sync stats Field Descriptions

Field	Description
Number of calls for which QoS was initiated	Number of calls for which RSVP setup was attempted.
Number of calls for which QoS was torn down	Number of calls for which an established RSVP reservation was released.
Number of calls for which Reservation Success was notified	Number of calls for which an RSVP reservation was successfully established.
Total Number of PATH Errors encountered	Number of path errors that occurred.

Table 42 show call rsvp-sync stats Field Descriptions (continued)

Field	Description
Total Number of RESV Errors encountered	Number of reservation errors that occurred.
Total Number of Reservation Timeouts encountered	Number of calls in which the reservation setup was not complete before the reservation timer expired.

Related Commands

Command	Description
call rsvp-sync	Enables synchronization between RSVP and the H.323 voice signaling protocol.
call rsvp-sync resv-timer	Sets the timer for RSVP reservation setup.
debug call rsvp-sync events	Displays the events that occur during RSVP synchronization.
show call rsvp-sync conf	Displays the RSVP synchronization configuration.

show call spike status

To display the configured call spike threshold and statistics for incoming calls, use the **show call spike status** command in privileged EXEC mode.

show call spike status

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T. This command was not supported on the Cisco AS5300, Cisco AS5350, and Cisco AS5400 in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on the Cisco 1750 and Cisco 1751. This command was not supported on any other platforms in this release.
12.2(8)T	This command was implemented on the Cisco 7200 series. Supported 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 is sample output from this command:

```
Router# show call spike status

 Call Spiking:Configured
 Call spiking :NOT TRIGGERED
 total call count in sliding window ::20
```

Table 43 describes significant fields shown in this output.

Table 43 show call spike status Field Descriptions

Field	Description
Call Spiking:	Current enabled state of call spiking.
Call Spiking	Details if call spiking limit has been triggered.
total call count in sliding window	Number of calls during spiking interval.

Related Commands

Command	Description
call spike	Configures the limit for the number of incoming calls in a short period of time.

show call threshold

To display enabled triggers, current values for configured triggers, and number of application programming interface (API) calls that were made to global and interface resources, use the **show call threshold** command in privileged EXEC mode.

show call threshold {**config** | **status** [**unavailable**] | **stats**}

Syntax Description

config	Displays the current threshold configuration.
status	Displays the status of all configured triggers and whether or not the CPU is available.
unavailable	(Optional) Displays the status for all unavailable resources.
stats	Displays statistics for API calls; that is, the resource-based measurement.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T. This command was not supported on the Cisco AS5300, Cisco AS5350, and Cisco AS5400 platforms in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on the Cisco 1750 and Cisco 1751. This command was not supported on any other platforms in this release.
12.2(8)T	This command was implemented on the Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release.

Examples

The following is sample output from the **show call threshold config** command:

```
Router# show call threshold config

Some resource polling interval:
  CPU_AVG interval: 60
  Memory interval:  5

IF              Type            Value   Low     High    Enable
-----           ----            -----   ----    ----    ------
Serial3/1:23    int-calls       0       107     107     N/A
N/A             cpu-avg         0        70      90     busy&treat
```

The following is sample output from the **show call threshold status** command:

```
Router# show call threshold status

Status  IF              Type            Value   Low     High    Enable
------  ---             ------          ----    ----    ----    -----
Avail   N/A             total-calls     0       5       5000    busyout
Avail   N/A             cpu-avg         0       5       65      busyout
```

The following is sample output from the **show call threshold status unavailable** command:

```
Router# show call threshold status unavailable

Unavailable configured resources at the current time:
IF              Type            Value   Low     High    Enable
----            -----           -----   ----    ----    -----
```

The following is sample output from the **show call threshold stats** command:

```
Router# show call threshold stats

Total resource check: 0
 successful: 0
 failed:   0
```

Table 44 describes significant fields shown in this output.

Table 44 show call threshold Field Descriptions

Field	Description
CPU_AVG interval	Interval of configured trigger CPU_AVG.
Memory interval	Interval of configured trigger Memory.
IF	Interface.
Type	Type of resource.
Value	Value of call to be matched against low and high thresholds.
Low	Low threshold.
High	High threshold.
Enable	Shows if busyout and the **call treatment** command are enabled.

Related Commands

Command	Description
call threshold	Enables a resource and defines associated parameters.
call threshold poll-interval	Enables a polling interval threshold for CPU or memory.
clear call threshold	Clears enabled triggers and their associated parameters.

show call treatment

To display the call-treatment configuration and statistics for handling the calls on the basis of resource availability, use the **show call treatment** command in privileged EXEC mode.

show call treatment {**config** | **stats**}

Syntax Description

config	Displays the call treatment configuration.
stats	Displays statistics for handling the calls on the basis of resource availability.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T. This command was not supported on the Cisco AS5300, Cisco AS5350, and Cisco AS5400 in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on the Cisco 1750 and Cisco 1751. This command was not supported on any other platforms in this release.
12.2(8)T	This command was implemented on the Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release.

Examples

The following is sample output from this command:

```
Router# show call treatment config

Call Treatment Config
---------------------

Call treatment is OFF.
Call treatment action is: Reject
Call treatment disconnect cause is: no-resource
Call treatment ISDN reject cause-code is: 41
```

Table 45 describes significant fields shown in this output.

Table 45 show call treatment config Field Descriptions

Field	Description
Call treatment is:	State of call treatment, either ON or OFF.
Call treatment action is:	Action trigger assigned for call treatment.

Table 45 show call treatment config Field Descriptions

Call treatment disconnect cause is:	Reason for disconnect.
Call treatment ISDN reject cause-code is:	Reject code number assigned.

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

```
Router# show call treatment stats

Call Treatment Statistics
-------------------------

Total Calls by call treatment: 0
Calls accepted by call treatment: 0
Calls rejected by call treatment: 0
Reason          Num. of calls rejected
------          ----------------------
cpu-5sec:       0
cpu-avg:        0
total-mem:      0
io-mem:         0
proc-mem:       0
total-calls:    0
```

Table 46 describes significant fields shown in this output.

Table 46 show call treatment stats Field Descriptions

Field	Description
Total Calls by call treatment:	Number of calls received and treated.
Calls accepted by call treatment:	Calls that passed treatment parameters.
Calls rejected by call treatment:	Calls that failed treatment parameters.
cpu-5sec	Number of calls rejected for failing the cpu-5sec parameter.
cpu-avg	Number of calls rejected for failing the cpu-avg parameter.
total-mem	Number of calls rejected for failing the total-mem parameter.
io-mem	Number of calls rejected for failing the io-mem parameter.
proc-mem	Number of calls rejected for failing the proc-mem parameter.
total-calls	Number of calls rejected for failing the total-calls parameter.

Related Commands

Command	Description
call treatment	Configures how calls should be processed when local resources are unavailable.
clear call treatment stats	Clears the call-treatment statistics.

show call-manager-fallback all

To display the detailed configuration of all Cisco IP phones, voice ports, and dial peers in your network while Cisco CallManager fallback is active with Survivable Remote Site (SRS) Telephony, use the **show call-manager-fallback all** command in EXEC mode.

show call-manager-fallback all

Syntax Description

This command has no arguments or keywords.

Command Modes

EXEC

Command History

Release	Modification
12.1(5)YD	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	This command was implemented on the Cisco 1760.

Examples

The following is sample output from the **show call-manager-fallback all** command:

```
Router# show call-manager-fallback all

CONFIG
======
ip source-address 20.0.0.1 port 2000
max-ephones 10
max-dn 10
no huntstop
translate called 1
call-forward busy 5001
call-forward noan 5001 timeout 8
cor incoming allowall 1 5001-5010
cor outgoing allow1800 2 5010-5020
alias 2 3... to 5555
keepalive 30

ephone-dn 1
number 4444
no huntstop
translate called 1

ephone-dn 2
number 3333
no huntstop
translate called 1

ephone-dn 3
```

```
number 5555 secondary 3...
no huntstop
translate called 1

ephone-dn 4
no huntstop
translate called 1
.
.
.
voice-port 50/0/1
 station-id number 4444
 timeout ringing 8
 translate called 1
!
voice-port 50/0/2
 station-id number 3333
 timeout ringing 8
 translate called 1
!
voice-port 50/0/3
 station-id number 5555
 timeout ringing 8
 translate called 1
!
voice-port 50/0/4
 timeout ringing 8
 translate called 1
.
.
.
dial-peer voice 20046 pots
 destination-pattern 4444
 call-forward busy 5001
 call-forward noan 5001
 port 50/0/1

dial-peer voice 20047 pots
 destination-pattern 3333
 call-forward busy 5001
 call-forward noan 5001
 port 50/0/2

dial-peer voice 20048 pots
 destination-pattern 5555
 call-forward busy 5001
 call-forward noan 5001
 port 50/0/3

dial-peer voice 20049 pots
preference 9
 destination-pattern 3...
 call-forward busy 5001
 call-forward noan 5001
 port 50/0/3
```

Table 47 describes the fields in the sample output, in alphabetical order.

Table 47 show call-manager-fallback all Field Descriptions

Field	Description
destination-pattern	Destination pattern (telephone number) configured for this dial peer.
dial-peer voice	Voice dial peer.
ephone-dn	Cisco IP phone directory number.
huntstop	Huntstop is set.
ip source address	IP address used by the Cisco IP phones to register with the router for service.
keepalive	Cisco IP phone keepalive period, in seconds.
max-ephones	Maximum number of Cisco IP phones.
max-dn	Maximum number of directory numbers or virtual-voice ports.
port	TCP port number used by the Cisco IP phones to communicate with the router.
station-id number	Number used for caller-ID purposes when calls are made using the line.
voicemail	A voice-mail (speed-dial) number is set.
voice-port	(Virtual) voice port designator.

Related Commands

Command	Description
show call-manager fallback dial-peer	Displays output for the dial peers in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.
show call-manager fallback ephone-dn	Displays output for the Cisco IP phone directory numbers or virtual voice ports in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.
show call-manager fallback voice-port	Displays output for the voice ports in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.

show call-manager-fallback dial-peer

To display output for the dial peers while Cisco CallManager fallback is active with Survivable Remote Site (SRS) Telephony, use the **show call-manager-fallback dial-peer** command in EXEC mode.

show call-manager-fallback dial-peer

Syntax Description

This command has no arguments or keywords.

Command Modes

EXEC

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	This command was implemented on the Cisco 1760.

Examples

The following is sample output from this command:

```
Router# show call-manager-fallback dial-peer

dial-peer voice 20046 pots
 destination-pattern 4444
 call-forward busy 5001
 call-forward noan 5001
 port 50/0/1

dial-peer voice 20047 pots
 destination-pattern 3333
 call-forward busy 5001
 call-forward noan 5001
 port 50/0/2

dial-peer voice 20048 pots
 destination-pattern 5555
 call-forward busy 5001
 call-forward noan 5001
 port 50/0/3

dial-peer voice 20049 pots
 preference 9
 destination-pattern 3...
 call-forward busy 5001
 call-forward noan 5001
 port 50/0/3
```

Table 48 provides an alphabetical listing of the command fields in the sample output.

Table 48 show call-manager-fallback dial-peer Field Descriptions

Field	Description
destination pattern	Destination pattern (telephone number) configured for this dial peer.
dial-peer voice	Voice dial peer.
huntstop	Huntstop is set.
port	(Virtual) voice port designator.

Related Commands

Command	Description
show call-manager fallback all	Displays the detailed configuration of all the Cisco IP phones, voice ports, and dial peers in your network in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.
show call-manager fallback ephone-dn	Displays output for the Cisco IP phone directory numbers or virtual voice ports in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.
show call-manager fallback voice-port	Displays output for the voice ports in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.

show call-manager-fallback ephone-dn

To display output for the Cisco IP phone directory numbers or virtual voice ports in Cisco CallManager fallback mode when the Survivable Remote Site (SRS) Telephony feature is enabled, use the **show call-manager-fallback ephone-dn** command in EXEC mode.

show call-manager-fallback ephone-dn

Syntax Description

This command has no arguments or keywords.

Command Modes

EXEC

Command History

Release	Modification
12.1(5)YD	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	This command was implemented on the Cisco 1760.

Examples

The following is sample output from the **show call-manager-fallback ephone-dn** command:

```
Router# show call-manager-fallback ephone-dn

ephone-dn 1
number 4444
no huntstop
translate called 1

ephone-dn 2
number 3333
no huntstop
translate called 1

ephone-dn 3
number 5555 secondary 3...
no huntstop
translate called 1

ephone-dn 4
no huntstop
translate called 1
```

Table 49 provides an alphabetical listing of the command fields in the sample output.

Table 49 show call-manager-fallback ephone-dn Field Descriptions

Field	Description
ephone-dn	Cisco IP phone directory number.
huntstop	Huntstop is set.
number	Cisco IP phone number.

Related Commands

Command	Description
show call-manager fallback all	Displays the detailed configuration of all the Cisco IP phones, voice ports, and dial peers in a network in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.
show call-manager fallback dial-peer	Displays output for the dial peers in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.
show call-manager fallback voice-port	Displays output for the voice ports in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.

show call-manager-fallback voice-port

To display output for the voice ports while Cisco CallManager fallback is active with Survivable Remote Site (SRS) Telephony, use the **show call-manager-fallback voice-port** command in EXEC mode.

show call-manager-fallback voice-port

Syntax Description

This command has no arguments or keywords.

Command Modes

EXEC

Command History

Release	Modification
12.1(5)YD	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	This command was implemented on the Cisco 1760.

Examples

The following is sample output from the **show call-manager-fallback voice-port** command:

```
Router# show call-manager-fallback voice-port

voice-port 50/0/1
 station-id number 4444
 timeout ringing 8
 translate called 1
!
voice-port 50/0/2
 station-id number 3333
 timeout ringing 8
 translate called 1
!
voice-port 50/0/3
 station-id number 5555
 timeout ringing 8
 translate called 1
!
voice-port 50/0/4
 timeout ringing 8
 translate called 1
!
```

Table 50 describes significant fields shown in this output.

Table 50 show call-manager-fallback voice-port Field Descriptions

Field	Description
voice-port	(Virtual) voice port.
station-id number	Phone number used for caller-ID purposes for calls made from this voice port.

Related Commands

Command	Description
show call-manager fallback all	Displays the detailed configuration of all the Cisco IP phones, voice ports, and dial peers in your network in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.
show call-manager fallback dial-peer	Displays fallback output for the dial peers in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.
show call-manager fallback ephone-dn	Displays fallback output for the Cisco IP phone directory numbers or virtual voice ports in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.

show call-router routes

To display the routes cached in the current border element (BE), use the **show call-router routes** in EXEC mode.

show call-router routes [**static | dynamic | all**]

Syntax Description

static	(Optional) Displays the descriptors provisioned on the border element.
dynamic	(Optional) Displays dynamically learned descriptors.
all	(Optional) Displays both static and dynamic descriptors.

Defaults

All

Command Modes

EXEC

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T.
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 is sample output from this command.

```
Router# show call-router routes

Static Routes:
==============
  DescriptorID= 6561676C650000000000000000000000A
  lastChanged = 19930301063311
   IP addr         :port       Prefix
   172.18.195.64   :2099       5553122

Dynamic Routes:
===============
  DescriptorID= 506174726F6E6F757300000000000002
  lastChanged = 19930228190012
   IP addr         :port       Prefix
   172.18.195.65   :2099       310

  DescriptorID= 506174726F6E6F757300000000000003
  lastChanged = 19930228190012
   IP addr         :port       Prefix
   172.18.195.65   :2099       555301

  DescriptorID= 506174726F6E6F757300000000000004
  lastChanged = 19930228190012
   IP addr         :port       Prefix
   172.18.195.65   :2099       555302
```

```
DescriptorID= 506174726F6E6F757300000000000005
  lastChanged = 19930228190012
   IP addr         :port       Prefix
   172.18.195.65   :2099       818

  DescriptorID= 506174726F6E6F757300000000000001
  lastChanged = 19930228190012
   IP addr         :port       Prefix
   172.18.195.65   :2099       1005
```

Related Commands

Command	Description
show call-router active	Displays active call information for a voice call in progress.
show call-router history	Displays the VoIP call-history table.
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 port	Displays configuration information about a specific voice port.

show call-router status

To display the Annex G border element status, use the **show call-router status** command in user EXEC mode.

show call-router status [**neighbors**]

Syntax Description

neighbors	(Optional) Displays the neighbor border element status.

Command Modes

User EXEC

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and modified to add the **neighbors** keyword.

Examples

The following example displays the Annex G border element status. Note that the example shows the status for two neighbors.:

```
Router# show call-router status neighbors

ANNEX-G CALL ROUTER STATUS:
  ============================
    Border Element ID Tag   : Celine
    Domain Name             : Celine-Domain
    Border Element State    : UP
    Border Element Local IP : 172.18.193.31:2099
    Advertise Policy        : STATIC descriptors
    Hopcount Value          : 7
    Descriptor TTL          : 3180
    Access Policy           : Neighbors only
    Current Active Calls    : 0
    Current Calls in Cache  : 0
    Cumulative Active Calls : 0
    Usage Ind Messages Sent : 0
    Usage Ind Cfm Rcvd      : 0
    IRRs Received           : 0
    DRQs Received           : 0
    Usage Ind Send Retrys   : 0

  NEIGHBOR INFORMATION:
  =====================
    Local Neighbor ID : (none)
    Remote Element ID : (unknown)
    Remote Domain ID  : (unknown)
    IP Addr           : 1.2.3.4:2099
    Status            : DOWN
    Caching           : OFF
    Query Interval    : 30 MIN (querying disabled)
```

```
Usage Indications :
  Current Active Calls : 0
  Retry Period         : 600 SEC
  Retry Window         : 3600 MIN
Service Relationship Status: ACTIVE
  Inbound Service Relationship  : DOWN
    Service ID     : (none)
    TTL            : 1200 SEC
  Outbound Service Relationship : DOWN
    Service ID     : (none)
    TTL            : (none)
    Retry interval : 120 SEC (0 until next attempt)
```

Table 51 describes significant fields shown in this output.

Table 51 show call-router status Field Descriptions

Field	Description
Border Element ID Tag	Identifier for the border element.
Border Element State	Indicates if the border element is running.
Border Element Local IP	Local IP address of the border element.
Advertise Policy	Type of descriptors that the border element advertises to its neighbors. Default is **static**. Other values are **dynamic** and **all**.
Hopcount Value	Maximum number of border element hops through which an address resolution request can be forwarded. Default is 7.
Descriptor TTL	Time-to-live value, or the amount of time, in seconds, for which a route from a neighbor is considered valid. Range is from 1 to 2147483647. Default is 1800 (30 minutes).
Access Policy	Requires that a neighbor be explicitly configured for requests to be accepted.
Local Neighbor ID	Domain name reported in service relationships.
Service Relationship Status	Service relationship between two border elements is active.
Inbound Service Relationship	Inbound time-to-Live (TTL) value in number of seconds. Range is from 1 to 4294967295.
Outbound Service Relationship	Specifies the amount of time, in seconds, to establish the outbound relationship. Range is from 1 to 65535.
Retry interval	Retry value between delivery attempts, in number of seconds. Range is from 1 to 3600.

Related Commands

Command	Description
advertise	Controls the type of descriptors that the border element advertises to its neighbors.
call-router	Enables the Annex G border element configuration commands.
hopcount	Specifies the maximum number of border element hops through which an address resolution request can be forwarded.
local	Defines the local domain, including the IP address and port border elements that the border element should use for interacting with remote border elements.

Command	Description
shutdown	Shuts down the Annex G border element.
ttl	Sets the expiration timer for advertisements.

show ccm-manager

To display a list of Cisco CallManager servers and their current status and availability, use the **show ccm-manager** command in privileged EXEC mode.

show ccm-manager [**backhaul** | **config-download** | **fallback-mgcp** | **hosts** | **music-on-hold** | **redundancy**]

Syntax Description

backhaul	(Optional) Displays PRI backhaul link information only.
config-download	(Optional) Displays information about the status of Media Gateway Control Protocol (MGCP) configuration download.
fallback-mgcp	(Optional) Displays the status of the MGCP gateway fallback feature.
hosts	(Optional) Displays a list of each configured Cisco CallManager server in the network, together with its operational status and host IP address.
music-on-hold	(Optional) Displays information about all the multicast music-on-hold (MOH) sessions in the gateway at any given point in time.
redundancy	(Optional) Displays failover mode and status information for hosts, including the redundant link port, failover interval, keepalive interval, MGCP traffic time, switchover time, and switchback mode.

Defaults

Displays information related to all keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(3)T	This feature was introduced on the Cisco CallManager Version 3.0 and Cisco VG200.
12.2(2)XA	This command was implemented on the Cisco 2600 series and Cisco 3600 series.
12.2(2)XN	This command was modified to provide enhanced MGCP voice gateway interoperability to Cisco CallManager Version 3.1 for the Cisco 2600 series, Cisco 3600 series, and Cisco VG200.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(2)11 and the Cisco CallManager Version 3.2. It was implemented on the Cisco IAD2420 series.

Usage Guidelines

Use this command to determine whether your primary or backup Cisco CallManager server is down, idle, or polling the backup Cisco CallManager server.

Examples

The following is sample output from the **show ccm-manager** command for displaying the status and availability of both the primary and the backup Cisco CallManager server.

```
Router# show ccm-manager

MGCP Domain Name: c3660A.cisco.com
Priority        Status                  Host
============================================================
Primary         Registered              IOS-38 (172.20.71.38)
First Backup    Backup Ready            ios-44 (172.20.71.44)
Second Backup   None

Current active Call Manager:    172.20.71.38
Backhaul/Redundant link port:   2428
Failover Interval:              30 seconds
Keepalive Interval:             15 seconds
Last keepalive sent:            03:06:24 (elapsed time: 00:00:06)
Last MGCP traffic time:         03:06:24 (elapsed time: 00:00:06)
Last failover time:             02:56:35 from (172.20.71.44)
Switchback mode:                Graceful
MGCP Fallback mode:             Enabled/OFF
Last MGCP Fallback start time:  00:00:00
Last MGCP Fallback end time:    00:00:00

PRI Backhaul Link info:
    Link Protocol:      TCP
    Remote Port Number: 2428
    Remote IP Address:  172.20.71.38
    Current Link State: OPEN
    Statistics:
        Packets recvd:   1
        Recv failures:   0
        Packets xmitted: 3
        Xmit failures:   0
    PRI Ports being backhauled:
       Slot 1, port 1
MGCP Download Tones:             Enabled

Configuration Auto-Download Information
=======================================
Current version-id: {1645327B-F59A-4417-8E01-7312C61216AE}
Last config-downloaded:00:00:49
Current state: Waiting for commands
Configuration Download statistics:
        Download Attempted             : 6
          Download Successful          : 6
          Download Failed              : 0
        Configuration Attempted        : 1
          Configuration Successful     : 1
          Configuration Failed(Parsing): 0
          Configuration Failed(config) : 0
Last config download command: New Registration
Configuration Error History:
FAX mode: cisco
```

Table 52 describes significant fields shown in this output.

Table 52 show ccm-manager Field Descriptions

Field	Description
MGCP Domain Name (*system*)	System used in the Internet for translating domain names of network nodes into IP addresses.
Priority	Priority of the Cisco CallManager servers present in the network. Possible priorities are primary, first backup, and second backup.
Status	Current usage of the Cisco CallManager server. Possible values are registered, idle, backup polling, and undefined.
Host	Host IP address of the Cisco CallManager server.
Current active Call Manager	Active Cisco CallManager server. This field can be any one of the following: primary, first backup, and second backup.
Backhaul/Redundant link port	Port that the Cisco CallManager server is to use.
Failover Interval	Maximum amount of time that can elapse without the gateway receiving messages from the currently active Cisco Call Manager before the gateway switches to the backup Cisco Call Manager.
Keepalive Interval	If the gateway has not received any messages from the currently active Cisco CallManager server within the specified amount of time, the gateway sends a keepalive message to the Cisco CallManager server to determine if it is operational.
Last keepalive sent	Time at which the last keepalive message was sent.
Last MGCP traffic time	Time at which the last MGCP traffic message was sent.
Switchback mode	Displays the switchback mode configuration that determines when the primary Cisco CallManager server is used if it becomes available again while a backup Cisco CallManager server is being used. Values that can appear in this field include graceful, immediate, schedule-time, and uptime-delay.
MGCP Fallback mode	Displays the MGCP fallback mode configuration. If “Not Selected” displays, then fallback is not configured. If “Enabled/OFF” displays, then fallback is configured but not in effect. If “Enabled/ON” displays, then fallback is configured and in effect.
Last MGCP Fallback start time	Start time stamp of the last fallback.
Lasts MGCP Fallback end time	End time stamp of the last fallback.

The following is sample output from the **show ccm-manager config-download** command:

```
Router# show ccm-manager config-download

Configuration Auto-Download Information
=======================================
Current version-id:{4171F93A-D8FC-49D8-B1C4-CE33FA8095BF}
Last config-downloaded:00:00:47
Current state:Waiting for commands
Configuration Download statistics:
        Download Attempted             :6
          Download Successful          :6
          Download Failed              :0
        Configuration Attempted        :1
          Configuration Successful     :1
          Configuration Failed(Parsing):0
          Configuration Failed(config) :0
Last config download command:New Registration
```

Table 53 describes significant fields shown in this output.

Table 53 show ccm-manager config-download Field Descriptions

Field	Description
Current state	Current configuration state.
Download Attempted	Number of times the gateway has tried to download the configuration file. The number of successes and failures is displayed.
Configuration Attempted	Number of times the gateway has tried to configure the gateway based on the configuration file. The number of successes and failures is displayed.

The following is sample output from the **show ccm-manager fallback-mgcp** command:

```
Router# show ccm-manager fallback-mgcp

Current active Call Manager:   172.20.71.38
MGCP Fallback mode:            Enabled/OFF
Last MGCP Fallback start time: 00:14:35
Last MGCP Fallback end time:   00:17:25
```

Table 54 describes significant fields shown in this output.

Table 54 show ccm-manager fallback-mgcp Field Descriptions

Field	Description
MGCP Fallback mode	The following are displayed: • Not Selected—Fallback is not configured. • Enabled/OFF—Fallback is configured but not in effect. • Enabled/ON—Fallback is configured and in effect.
Last MGCP Fallback start time	Start time stamp of the last fallback.
Last MGCP Fallback end time	End time stamp of the last fallback.

The following is sample output from the **show ccm-manager music-on-hold** command:

```
Router# show ccm-manager music-on-hold

Current active multicast sessions :1
 Multicast       RTP port   Packets       Call   Codec    Incoming
 Address         number     in/out        id              Interface
=====================================================================
172.20.71.38     2428        5/5          99      g711
```

Table 55 describes significant fields shown in this output.

Table 55 show ccm-manager music-on-hold Field Descriptions

Field	Description
Current active multicast sessions	Number of active calls on hold.
Multicast Address	Valid class D address from which the gateway is getting the RTP streams.
RTP port number	Valid RTP port number on which the gateway receives the RTP packets.
Packets in/out	Number of RTP packets received and sent to the digital signal processor (DSP).
Call id	Call ID of the call that is on hold.
Codec	Codec number.
Incoming Interface	Interface through which the gateway is receiving the RTP stream.

Related Commands

Command	Description
show mgcp	Displays the MGCP configuration information.

show cdapi

To display the Call Distributor Application Programming Interface (CDAPI), use the **show cdapi** command in privileged EXEC mode.

show cdapi

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(7)T	This command was introduced on the Cisco AS5300.

Usage Guidelines

CDAPI is the internal application programming interface (API) that provides an interface between signaling stacks and applications.

Examples

The following is sample output from the **show cdapi** command:

```
Router# show cdapi

Registered CDAPI Applications/Stacks
====================================
Application TSP CDAPI Application
        Application Type(s)  Voice Facility Signaling
        Application Level    Tunnel
        Application Mode     Enbloc
Signaling Stack ISDN
        Interface Se023
Signaling Stack ISDN
        Interface Se123
Active CDAPI Calls
==================
Interface Se023
        No active calls.
Interface Se123
        Call ID = 0x39, Call Type = VOICE, Application = TSP CDAPI Application
CDAPI Message Buffers
=====================
Used Msg Buffers 0, Free Msg Buffers 1600
Used Raw Buffers 1, Free Raw Buffers 799
Used Large-Raw Buffers 0, Free Large-Raw Buffers 80
scarlatti1#
```

Related Commands

Command	Description
isdn protocol-emulate	Configures the Layer 2 and Layer 3 port protocol of a BRI voice port or a PRI interface to emulate NT (network) or TE (user) functionality.
isdn switch type	Configures the Cisco AS5300 universal access server PRI interface to support Q.SIG signaling.
pri-group nec-fusion	Configures your NEC PBX to support FCCS.
show rawmsg	Displays the raw messages owned by the required component.

show ces clock-select

To display the setting of the network clock for the specified port, use the **show ces clock-select** command in privileged EXEC mode.

show ces *slot/port clock-select*

Syntax Description

slot	Backplane slot number.
/port	Interface port number. The slash must be entered.

Command Modes

Privileged EXEC

Command History

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

Examples

The following is sample output from this command for slot 1, port 0:

```
Router# show ces 1/0 clock-select

Priority 1 clock source:not configured
Priority 2 clock source:not configured
Priority 3 clock source:ATM1/0 UP
Priority 4 clock source:Local oscillator
Current clock source:ATM1/0, priority:3
```

Related Commands

Command	Description
clock-select	Establishes the sources and priorities of the requisite clocking signals for the OC-3/STM-1 ATM Circuit Emulation Service network module.

show connect

To display configuration information about drop-and-insert connections that have been configured on a router, enter the **show connect** command in privileged EXEC mode.

show connect {**all** | **elements** | **name** | **id** | **port** {**T1** | **E**1} *slot/port*}}

Syntax Description

all	Displays a table of all configured connections.
elements	Displays registered hardware or software interworking elements.
name	Displays a connection that has been named by using the **connect** global configuration command. The name you enter is case sensitive and must match the configured name exactly.
id	Displays the status of a connection that you specify by an identification number or range of identification numbers. The router assigns these IDs automatically in the order in which they were created, beginning with 1. The **show connect all** command displays these IDs.
port	Displays the status of a connection that you specify by indicating the type of controller (T1 or E1) and location of the interface.
T1	Specifies a T1 controller.
E1	Specifies an E1 controller.
slot/port	Location of the T1 or E1 controller port whose connection status you want to see. Valid values for *slot* and *port* are 0 and 1. The slash must be entered.

Defaults

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(5)XK	This command was introduced on the Cisco 2600 series and Cisco 3600 series.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0(7)T.

Usage Guidelines

This command shows drop-and-insert connections on Cisco 2600 series and Cisco 3600 series routers. It displays different information in different formats, depending on the keyword that you use.

Examples

The following examples show how the same tabular information appears when you enter different keywords:

```
Router# show connect all

ID   Name               Segment 1            Segment 2           State
======================================================================
1    Test              -T1 1/0 01           -T1 1/1 02           ADMIN UP
```

```
2    Test2             -T1 1/0 03           -T1 1/1 04           ADMIN UP

Router# show connect id 1-2

ID   Name               Segment 1            Segment 2           State
============================================================================
1    Test              -T1 1/0 01           -T1 1/1 02           ADMIN UP
2    Test2             -T1 1/0 03           -T1 1/1 04           ADMIN UP

Router# show connect port t1 1/1

ID   Name               Segment 1            Segment 2           State
============================================================================
1    Test              -T1 1/0 01           -T1 1/1 02           ADMIN UP
2    Test2             -T1 1/0 03           -T1 1/1 04           ADMIN UP
```

The following examples show details about specific connections, including the number of time slots in use and the switching elements:

```
Router# show connect id 2

Connection: 2 - Test2
 Current State: ADMIN UP
 Segment 1: -T1 1/0 03
  TDM timeslots in use: 14-18 (5 total)
 Segment 2: -T1 1/1 04
  TDM timeslots in use: 14-18
Internal Switching Elements: VIC TDM Switch

Router# show connect name Test

 Connection: 1 - Test
 Current State: ADMIN UP
 Segment 1: -T1 1/0 01
  TDM timeslots in use: 1-13 (13 total)
 Segment 2: -T1 1/1 02
  TDM timeslots in use: 1-13
Internal Switching Elements: VIC TDM Switch
```

Related Commands

Command	Description
connect	Defines connections between T1 or E1 controller ports for Drop and Insert.
tdm-group	Configures a list of time slots for creating clear channel groups (pass-through) for TDM cross-connect.

show controllers rs366

To display information about the RS-366 video interface on the video dialing module (VDM), use the **show controllers rs366** command in privileged EXEC mode.

show controllers rs366 *slot port*

Syntax Description

slot	Slot location of the VDM module. On the Cisco MC3810, this value is either 1 or 2. If you do not enter the correct location, the command is rejected.
port	Port location of the EIA/TIA-366 interface in the VDM module. On the Cisco MC3810, this value is 0.

Defaults

No default behavior or values

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.

Examples

On a Cisco MC3810, the following example displays information about the RS-366 controller:

```
Router# show controllers rs366 0 1

RS366:driver is initialized in slot 1, port 0:

STATUS STATE LSR  LCR  ICSR EXT  T1     T2     T3     T4     T5
0x02   0x01  0x00 0x50 0xE0 0x00 5000   5000   5000   20000  10000
Dial string:
121C
```

Table 56 describes significant fields shown in this output.

Table 56 show controllers rs366 Field Descriptions

Field	Description
STATUS	Last interrupt status.
STATE	Current state of the state machine.
LSR	Line status register of the VDM.
LCR	Line control register of the VDM.
ICSR	Interrupt control and status register of the VDM.
EXT	Extended register of the VDM.

Table 56 show controllers rs366 Field Descriptions (continued)

Field	Description
T1 through T5	Timeouts 1 through 5 of the watchdog timer, in milliseconds.
Dial string	Most recently dialed number collected by the driver. 0xC at the end of the string indicates the EON (end of number) character.

show controllers timeslots

To display the channel-associated signaling (CAS) and ISDN PRI state in detail, use the **show controllers timeslots** command in privileged EXEC mode.

show controllers t1/e1 *controller-number* **timeslots** *timeslot-range*

Syntax Description

tl/e1	Type of interface.
controller-number	Controller number of CAS or ISDN PRI time slot. Range is from 0 to 7.
timeslots	Displays DS0 information.
timeslot-range	Timeslot. E1 range is from 1 to 31. T1 range is from 1 to 24.

Defaults

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
10.0	This command was introduced.
12.1(3)T	The **timeslots** keyword was added.
12.1(5)T	This command was implemented on the Cisco AS5400.

Usage Guidelines

Use this command to display the CAS and ISDN PRI channel state in detail. This command shows whether the DS0 channels of a controller are in idle, in-service, maintenance, or busyout states. Enter the **show controllers t1/e1** command to display statistics about the T1 or E1 links.

Examples

The following example shows that the CAS state is enabled on the Cisco AS5300 universal access server with a T1 PRI card:

```
Router# show controllers timeslots

T1 1 is up:
Loopback: NONE
DS0  Type       Modem    <->     Service       Channel       Rx          Tx
                                 State         State        A B C D     A B C D
-------------------------------------------------------------------------------
  1    cas-modem   1       in     insvc       connected    1  1  1  1    1  1  1  1
  2    cas         -       -      insvc       idle         0  0  0  0    0  0  0  0
  3    cas         -       -      insvc       idle         0  0  0  0    0  0  0  0
  4    cas         -       -      insvc       idle         0  0  0  0    0  0  0  0
  5    cas         -       -      insvc       idle         0  0  0  0    0  0  0  0
  6    cas         -       -      insvc       idle         0  0  0  0    0  0  0  0
  7    cas         -       -      insvc       idle         0  0  0  0    0  0  0  0
  8    cas         -       -      insvc       idle         0  0  0  0    0  0  0  0
  9    cas         -       -      insvc       idle         0  0  0  0    0  0  0  0
  10   cas         -       -      maint      static-bo     0  0  0  0    1  1  1  1
```

```
 11   cas         -       -      maint      static-bo     0  0  0  0    1  1  1  1
 12   cas         -       -      maint      static-bo     0  0  0  0    1  1  1  1
 13   cas         -       -      maint      static-bo     0  0  0  0    1  1  1  1
 14   cas         -       -      maint      static-bo     0  0  0  0    1  1  1  1
 15   cas         -       -      maint      static-bo     0  0  0  0    1  1  1  1
 16   cas         -       -      maint      static-bo     0  0  0  0    1  1  1  1
 17   cas         -       -      maint      static-bo     0  0  0  0    1  1  1  1
 18   cas         -       -      maint      static-bo     0  0  0  0    1  1  1  1
 19   cas         -       -      maint      dynamic-bo    0  0  0  0    1  1  1  1
 20   cas         -       -      maint      dynamic-bo    0  0  0  0    1  1  1  1
 21   cas         -       -      maint      dynamic-bo    0  0  0  0    1  1  1  1
 22   unused
 23   unused
 24   unused
```

The following example shows that the ISDN PRI state is enabled on the Cisco AS5300 universal access server with a T1 PRI card:

```
T1 2 is up:
Loopback: NONE
DS0 Type         Modem    <->  Service   Channel     Rx        Tx
                               State     State       A B C D   A B C D
--------------------------------------------------------------------------------
 1  pri          -        -    insvc     idle
 2  pri          -        -    insvc     idle
 3  pri          -        -    insvc     idle
 4  pri          -        -    insvc     idle
 5  pri          -        -    insvc     idle
 6  pri          -        -    insvc     idle
 7  pri          -        -    insvc     idle
 8  pri          -        -    insvc     idle
 9  pri          -        -    insvc     idle
10  pri          -        -    insvc     idle
11  pri          -        -    insvc     idle
12  pri          -        -    insvc     idle
13  pri          -        -    insvc     idle
14  pri          -        -    insvc     idle
15  pri          -        -    insvc     idle
16  pri          -        -    insvc     idle
17  pri          -        -    insvc     idle
18  pri          -        -    insvc     idle
19  pri          -        -    insvc     idle
20  pri          -        -    insvc     idle
21  pri-modem    2        in   insvc     busy
22  pri-modem    1        out  insvc     busy
23  pri-digi     -        in   insvc     busy
24  pri-sig      -        -    outofsvc  reserved
```