show call history fax through show debug condition

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**| **more**} *time*]| **id** *identifier*| **last** *number*]

Syntax Description

brief	(Optional) Displays a truncated version of the call history table.
id *identifier*	(Optional) Displays only the call with the specified identifier. Range is a hex value from 1 to FFFF.
compact	(Optional) Displays a compact version.
duration *time*	(Optional) Displays history information for calls that are longer or shorter than a specified *time* value. The arguments and keywords are as follows: • less--Displays calls shorter than the value in the *time* argument. • more--Displays calls longer than the value in the *time* argument. • *time* --Elapsed time, in seconds. Range is from 1 to 2147483647.
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 to100 .

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	This command was modified. The **brief** keyword was added, and the command was implemented on the Cisco 7200 series.

Release	Modification
12.0(7)XK	This command was modified. 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(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	This command was modified. 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	This command was integrated into Cisco IOS Release 12.2(8)T. Support for the Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 was not included in this release.
12.2(11)T	This command was implemented on the Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.
12.3(1)	This command was modified. The following fields were added: FaxRelayMaxJitterBufDepth, FaxRelayJitterBufOverFlow, FaxRelayHSmodulation, and FaxRelayNumberOfPages.
12.3(14)T	This command was modified. T.38 fax relay call statistics were made available to Call Detail Records (CDRs) through vendor-specific attributes (VSAs) and added to the call log.
12.4(15)T	This command was modified. The Port and BearerChannel display fields were added to the TELE call leg record of the command output.
12.4(16)	This command was modified. The Port and BearerChannel display fields were added to the TELE call leg record of the command output.
12.4(22)T	This command was modified. Command output was updated to show IPv6 information.

Usage Guidelines

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 from 0 to 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]
CallID=2
Port=3/0/0 (2)
BearerChannel=3/0/0.1
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
```

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

Table 1: show call history fax Field Descriptions

Field	Description
ACOM Level	Current ACOM level for this call. ACOM is the combined loss achieved by the echo canceler, which is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.
BearerChannel	Identification of the bearer channel carrying the call.
Buffer Drain Events	Total number of jitter buffer drain events.
Buffer Fill Events	Total number of jitter buffer fill events.
CallDuration	Length of the call, in hours, minutes, and seconds, hh:mm:ss.
CallerName	Voice port station name string.
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 (ms), at which the call was connected.
Consecutive-packets-lost Events	Total number of consecutive (two or more) packet-loss events.

Field	Description
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.
DisconnectCause	Cause code for the reason this call was disconnected.
DisconnectText	Descriptive text explaining the reason for the disconnect.
DisconnectTime	Time, in ms, when this call was disconnected.
EchoCancellerMaxReflector=64	The location of the largest reflector, in ms. The reflector size does not exceed the configured echo path capacity. For example, if 32 ms is configured, the reflector does not report beyond 32 ms.
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.
FaxRelayJitterBufOverFlow	Count of number of network jitter buffer overflows (number of packets). These packets are equivalent to lost packets.
FaxRelayMaxJitterBufDepth	Maximum depth of jitter buffer (in ms).
FaxRelayHSmodulation	Most recent high-speed modulation used.
FaxRelayNumberOfPages	Number of pages transmitted.
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.

Field	Description
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 VoIP and telephony call legs.
GwReceivedCalledNumber, GwReceivedCalledOctet3, GwReceivedCallingNumber, GwReceivedCallingOctet3, GwReceivedCallingOctet3a	Call information received at the gateway.
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.
ImgPages	The fax pages that have been processed.
Incoming ConnectionId	The incoming_GUID. It can be different with ConnectionId (GUID) when there is a 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.
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.

Field	Description
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, Frame Relay (FR), or High-Level Data Link Control (HDLC).
Media	Medium over which the call is carried. If the call is carried over the (telephone) access side, the entry is TELE. If the call is carried over the voice network side, the entry is either ATM, FR, or HDLC.
Modem passthrough signaling method in use	Indicates that this is a modem pass-through call and that named signaling events (NSEs)--a Cisco-proprietary version of named telephone events in RFC 2833--are used for signaling codec upspeed. The upspeed method is the method used to dynamically change the codec type and speed to meet network conditions. This means that you might move to a faster codec when you have both voice and data calls and then slow down when there is only voice traffic.
NoiseLevel	Active noise level for this call.
OnTimeRvPlayout	Duration of voice playout from data received on time for this call. Derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values.
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.
OutSignalLevel	Active output signal level to the telephony interface used by this call.
PeerAddress	Destination pattern or number associated with this peer.
PeerId	ID value of the peer table entry to which this call was made.

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

Field	Description
Time between Buffer Drain/Fills	Minimum and maximum durations between jitter buffer drain or fill events, in seconds.
TranslatedCallingNumber, TranslatedCallingOctet, TranslatedCalledNumber, TranslatedCalledOctet, TranslatedRedirectCalled Number, TranslatedRedirectCalledOctet	Translated call information.
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 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
```

The following example shows output for the **show call history fax** command with the T.38 Fax Relay statistics:

```
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=9872460 ms
Index=8
PeerAddress=41023
PeerSubAddress=
PeerId=1
PeerIfIndex=242
LogicalIfIndex=180
DisconnectCause=10
DisconnectText=normal call clearing (16)
ConnectTime=9875610 ms
DisconnectTime=9936000 ms
CallDuration=00:01:00 sec
CallOrigin=2
ReleaseSource=1
ChargedUnits=0
InfoType=fax
TransmitPackets=268
TransmitBytes=4477
ReceivePackets=1650
ReceiveBytes=66882
TELE:
ConnectionId=[0xD6635DD5 0x9FA411D8 0x8005000A 0xF4107CA0]
IncomingConnectionId=[0xD6635DD5 0x9FA411D8 0x8005000A 0xF4107CA0]
CallID=7
Port=3/0/0:0 (7)
BearerChannel=3/0/0.8
TxDuration=6170 ms
VoiceTxDuration=0 ms
FaxTxDuration=0 ms
FaxRate=disable bps
FaxRelayMaxJitterBufDepth=560 ms
FaxRelayJitterBufOverFlow=0
FaxRelayMostRecentHSmodulation=V.17/short/14400
FaxRelayNumberOfPages=1
FaxRelayInitHSmodulation=V.17/long/14400
FaxRelayDirection=Transmit
FaxRelayPktLossConceal=0
FaxRelayEcmStatus=ENABLED
FaxRelayEncapProtocol=T.38 (UDPTL)
FaxRelayNsfCountryCode=Japan
FaxRelayNsfManufCode=0031B8EE80C48511DD0D0000DDDD0000DDDD000000000000000022ED00B0A400
FaxRelayFaxSuccess=Success
NoiseLevel=0
ACOMLevel=0
SessionTarget=
ImgPages=0
CallerName=Analog 41023
CallerIDBlocked=False
OriginalCallingNumber=
OriginalCallingOctet=0x80
OriginalCalledNumber=41021
OriginalCalledOctet=0xA1
OriginalRedirectCalledNumber=
```

```
OriginalRedirectCalledOctet=0xFF
TranslatedCallingNumber=41023
TranslatedCallingOctet=0x80
TranslatedCalledNumber=41021
TranslatedCalledOctet=0xA1
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwReceivedCalledNumber=41021
GwReceivedCalledOctet3=0xA1
```

The table below describes the fields not shown in the table above.

Table 2: show call history fax Field Descriptions

Field	Description
FaxRelayDirection	Direction of fax relay.
FaxRelayEcmStatus	Fax relay error correction mode status.
FaxRelayEncapProtocol	Fax relay encapsulation protocol.
FaxRelayFaxSuccess	Fax relay success.
FaxRelayInitHSmodulation	Fax relay initial high speed modulation.
FaxRelayMostRecentHSmodulation	Fax relay most recent high speed modulation.
FaxRelayNsfCountryCode	Fax relay Nonstandard Facilities (NSF) country code.
FaxRelayNsfManufCode	Fax relay NSF manufacturers code.
FaxRelayPktLossConceal	Fax relay packet loss conceal.

Related Commands

Command	Description
dial-control-mib	Specifies attributes for the call history table.
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 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.

Command	Description
show voice port	Displays configuration information about a specific voice port.

show call history media

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

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

Syntax Description

brief	(Optional) Displays a truncated version of the call history table.
id *identifier*	(Optional) Displays only the call with the specified *identifier*. The range is from 1 to FFFF.
compact	(Optional) Displays a compact version of the call history table.
duration	(Optional) Displays the call history for the specified time duration.
less	Displays the call history for shorter duration calls.
more	Displays the call history for longer duration calls.
seconds	Time, in seconds. The range is from 1 to 2147483647.
last *number*	(Optional) Displays the last calls connected, where the number of calls that appear is defined by the *number*argument. The range is from 1 to100 .

Command Modes

User EXEC (>) Privileged EXEC (#)

Command History

Release	Modification
12.4(15)T	This command was introduced.

Usage Guidelines

This command displays a call-history table that contains a list of media calls connected through the router in descending time order. The maximum number of calls contained in the table can be set to a number from 0 to 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 **last** keyword, 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.

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

Examples

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

```
Router# show call history media
Telephony call-legs: 0
SIP call-legs: 0
H323 call-legs: 0
Call agent controlled call-legs: 0
Media call-legs: 4
Total call-legs: 4
GENERIC:
SetupTime=308530 ms
Index=4
PeerAddress=sip:mrcpv2ASRServer@10.5.18.224:5060
PeerSubAddress=
PeerId=2234
PeerIfIndex=184
LogicalIfIndex=0
DisconnectCause=10
DisconnectText=normal call clearing (16)
ConnectTime=309440 ms
DisconnectTime=320100 ms
CallDuration=00:00:10 sec
CallOrigin=1
ReleaseSource=7
ChargedUnits=0
InfoType=speech
TransmitPackets=237
TransmitBytes=37920
ReceivePackets=0
ReceiveBytes=0
VOIP:
ConnectionId[0x2FB5B737 0xC3511DB 0x8005000B 0x5FDA0EF4]
IncomingConnectionId[0x2FB5B737 0xC3511DB 0x8005000B 0x5FDA0EF4]
CallID=14
RemoteIPAddress=10.5.18.224
RemoteUDPPort=10002
RemoteSignallingIPAddress=10.5.18.224
RemoteSignallingPort=5060
RemoteMediaIPAddress=10.5.18.224
RemoteMediaPort=10002
SRTP = off
TextRelay = off
Fallback Icpif=0
Fallback Loss=0
Fallback Delay=0
RoundTripDelay=0 ms
SelectedQoS=best-effort
tx_DtmfRelay=rtp-nte
FastConnect=FALSE
AnnexE=FALSE
Separate H245 Connection=FALSE
H245 Tunneling=FALSE
SessionProtocol=sipv2
ProtocolCallId=2FBDA670-C3511DB-8015C48C-6A894889@10.5.14.2
SessionTarget=10.5.18.224
OnTimeRvPlayout=3000
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=2740 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=100 ms
```

```
LoWaterPlayoutDelay=40 ms
Source tg label=test5
ReceiveDelay=90 ms
LostPackets=0
EarlyPackets=0
LatePackets=0
VAD = disabled
CoderTypeRate=g711ulaw
CodecBytes=160
cvVoIPCallHistoryIcpif=16
MediaSetting=flow-around
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=
OriginalCalledOctet=0x0
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=555-0100
TranslatedCallingOctet=0x21
TranslatedCalledNumber=
TranslatedCalledOctet=0xC1
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwOutpulsedCallingNumber=555-0101
GwOutpulsedCallingOctet3=0x21
GwOutpulsedCallingOctet3a=0x81
MediaInactiveDetected=no
MediaInactiveTimestamp=
MediaControlReceived=
LongDurationCallDetected=no
LongDurationCallTimerStamp=
LongDurationCallDuration=
Username=
GENERIC:
SetupTime=308520 ms
Index=5
PeerAddress=sip:mrcpv2TTSServer@10.5.18.224:5060
PeerSubAddress=
PeerId=2235
PeerIfIndex=185
LogicalIfIndex=0
DisconnectCause=10
DisconnectText=normal call clearing (16)
ConnectTime=309370 ms
DisconnectTime=320100 ms
CallDuration=00:00:10 sec
CallOrigin=1
ReleaseSource=7
ChargedUnits=0
InfoType=speech
TransmitPackets=0
TransmitBytes=0
ReceivePackets=551
ReceiveBytes=88160
VOIP:
ConnectionId[0x2FB5B737 0xC3511DB 0x8005000B 0x5FDA0EF4]
IncomingConnectionId[0x2FB5B737 0xC3511DB 0x8005000B 0x5FDA0EF4]
CallID=13
RemoteIPAddress=10.5.18.224
RemoteUDPPort=10000
RemoteSignallingIPAddress=10.5.18.224
RemoteSignallingPort=5060
RemoteMediaIPAddress=10.5.18.224
RemoteMediaPort=10000
SRTP = off
TextRelay = off
Fallback Icpif=0
Fallback Loss=0
Fallback Delay=0
RoundTripDelay=0 ms
SelectedQoS=best-effort
```

```
tx_DtmfRelay=rtp-nte
FastConnect=FALSE
AnnexE=FALSE
Separate H245 Connection=FALSE
H245 Tunneling=FALSE
SessionProtocol=sipv2
ProtocolCallId=2FBC6E20-C3511DB-8013C48C-6A894889@10.5.14.2
SessionTarget=10.5.18.224
OnTimeRvPlayout=7000
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=2740 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=100 ms
LoWaterPlayoutDelay=40 ms
Source tg label=test5
ReceiveDelay=95 ms
LostPackets=0
EarlyPackets=0
LatePackets=0
VAD = disabled
CoderTypeRate=g711ulaw
CodecBytes=160
cvVoIPCallHistoryIcpif=16
MediaSetting=flow-around
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=
OriginalCalledOctet=0x0
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=555-0102
TranslatedCallingOctet=0x21
TranslatedCalledNumber=
TranslatedCalledOctet=0xC1
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwOutpulsedCallingNumber=555-0103
GwOutpulsedCallingOctet3=0x21
GwOutpulsedCallingOctet3a=0x81
MediaInactiveDetected=no
MediaInactiveTimestamp=
MediaControlReceived=
LongDurationCallDetected=no
LongDurationCallTimerStamp=
LongDurationCallDuration=
Username=
GENERIC:
SetupTime=408050 ms
Index=7
PeerAddress=sip:mrcpv2ASRServer@10.5.18.224:5060
PeerSubAddress=
PeerId=2234
PeerIfIndex=184
LogicalIfIndex=0
DisconnectCause=10
DisconnectText=normal call clearing (16)
ConnectTime=408160 ms
DisconnectTime=426260 ms
CallDuration=00:00:18 sec
CallOrigin=1
ReleaseSource=7
ChargedUnits=0
InfoType=speech
TransmitPackets=598
TransmitBytes=95680
ReceivePackets=0
ReceiveBytes=0
VOIP:
ConnectionId[0x6B02FC0C 0xC3511DB 0x8006000B 0x5FDA0EF4]
IncomingConnectionId[0x6B02FC0C 0xC3511DB 0x8006000B 0x5FDA0EF4]
```

```
CallID=19
RemoteIPAddress=10.5.18.224
RemoteUDPPort=10002
RemoteSignallingIPAddress=10.5.18.224
RemoteSignallingPort=5060
RemoteMediaIPAddress=10.5.18.224
RemoteMediaPort=10002
SRTP = off
TextRelay = off
Fallback Icpif=0
Fallback Loss=0
Fallback Delay=0
RoundTripDelay=0 ms
SelectedQoS=best-effort
tx_DtmfRelay=rtp-nte
FastConnect=FALSE
AnnexE=FALSE
Separate H245 Connection=FALSE
H245 Tunneling=FALSE
SessionProtocol=sipv2
ProtocolCallId=6B0E94CD-C3511DB-801DC48C-6A894889@10.5.14.2
SessionTarget=10.5.18.224
OnTimeRvPlayout=11000
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=9560 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=100 ms
LoWaterPlayoutDelay=55 ms
Source tg label=test5
ReceiveDelay=100 ms
LostPackets=0
EarlyPackets=0
LatePackets=0
VAD = disabled
CoderTypeRate=g711ulaw
CodecBytes=160
cvVoIPCallHistoryIcpif=16
MediaSetting=flow-around
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=
OriginalCalledOctet=0x0
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=555-0100
TranslatedCallingOctet=0x21
TranslatedCalledNumber=
TranslatedCalledOctet=0xC1
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwOutpulsedCallingNumber=555-0101
GwOutpulsedCallingOctet3=0x21
GwOutpulsedCallingOctet3a=0x81
MediaInactiveDetected=no
MediaInactiveTimestamp=
MediaControlReceived=
LongDurationCallDetected=no
LongDurationCallTimerStamp=
LongDurationCallDuration=
Username=
GENERIC:
SetupTime=408040 ms
Index=8
PeerAddress=sip:mrcpv2TTSServer@10.5.18.224:5060
PeerSubAddress=
PeerId=2235
PeerIfIndex=185
LogicalIfIndex=0
DisconnectCause=10
DisconnectText=normal call clearing (16)
```

```
ConnectTime=408130 ms
DisconnectTime=426260 ms
CallDuration=00:00:18 sec
CallOrigin=1
ReleaseSource=7
ChargedUnits=0
InfoType=speech
TransmitPackets=0
TransmitBytes=0
ReceivePackets=911
ReceiveBytes=145760
VOIP:
ConnectionId[0x6B02FC0C 0xC3511DB 0x8006000B 0x5FDA0EF4]
IncomingConnectionId[0x6B02FC0C 0xC3511DB 0x8006000B 0x5FDA0EF4]
CallID=18
RemoteIPAddress=10.5.18.224
RemoteUDPPort=10000
RemoteSignallingIPAddress=10.5.18.224
RemoteSignallingPort=5060
RemoteMediaIPAddress=10.5.18.224
RemoteMediaPort=10000
SRTP = off
TextRelay = off
Fallback Icpif=0
Fallback Loss=0
Fallback Delay=0
RoundTripDelay=0 ms
SelectedQoS=best-effort
tx_DtmfRelay=rtp-nte
FastConnect=FALSE
AnnexE=FALSE
Separate H245 Connection=FALSE
H245 Tunneling=FALSE
SessionProtocol=sipv2
ProtocolCallId=6B0CC055-C3511DB-801BC48C-6A894889@10.5.14.2
SessionTarget=10.5.18.224
OnTimeRvPlayout=9000
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=9560 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=100 ms
LoWaterPlayoutDelay=55 ms
Source tg label=test5
ReceiveDelay=100 ms
LostPackets=0
EarlyPackets=0
LatePackets=0
VAD = disabled
CoderTypeRate=g711ulaw
CodecBytes=160
cvVoIPCallHistoryIcpif=16
MediaSetting=flow-around
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=
OriginalCalledOctet=0x0
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=555-0100
TranslatedCallingOctet=0x21
TranslatedCalledNumber=
TranslatedCalledOctet=0xC1
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwOutpulsedCallingNumber=555-0101
GwOutpulsedCallingOctet3=0x21
GwOutpulsedCallingOctet3a=0x81
MediaInactiveDetected=no
MediaInactiveTimestamp=
MediaControlReceived=
```

```
LongDurationCallDetected=no
LongDurationCallTimerStamp=
LongDurationCallDuration=
Username=
```

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

Table 3: show call history media Field Descriptions

Field	Description
CallDuration	Length of the call, in hours, minutes, and seconds, hh:mm:ss.
CallOrigin	Call origin: not answer or originate.
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 ms, during which the call was connected.
GapFillWithInterpolation	Duration, in ms, of a voice signal played out with a signal synthesized from parameters, or samples of data preceding and following in time because voice data was lost or not received in time from the voice gateway for this call.
GapFillWithRedundancy	Duration, in ms, of a voice signal played out with a signal synthesized from available redundancy parameters because voice data was lost or not received in time from the voice gateway for this call.
GapFillWithPrediction	Duration, in ms, of the voice signal played out with a signal synthesized from parameters, or samples of data preceding in time, because voice data was lost or not received in time from the voice gateway for this call. Examples of such pullout are frame-eraser and frame-concealment strategies in G.729 and G.723.1 compression algorithms.
GapFillWithSilence	Duration, in ms, of a voice signal replaced with silence because voice data was lost or not received in time for this call.

Field	Description
GENERIC	Generic or common parameters; that is, parameters that are common for VoIP and telephony call legs.
H323 call-legs	Total H.323 call legs for which call records are available.
HiWaterPlayoutDelay	High-water-mark voice playout first in first out (FIFO) Delay during this call, in ms.
Index	Dial peer identification number.
InfoType	Information type for this call; for example, voice, speech, or fax.
LogicalIfIndex	Index number of the logical interface for this call.
LoWaterPlayoutDelay	Low-water-mark voice playout FIFO delay during this call, in ms.
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.
PeerAddress	Destination pattern or number associated with this peer.
PeerId	ID value of the peer table entry to which this call was made.
PeerIfIndex	Voice port index number for this peer. For ISDN media, this would be the index number of the B channel used for this call.
PeerSubAddress	Subaddress when this call is connected.
ReceiveBytes	Number of bytes received by the peer during this call.
ReceiveDelay	Average playout FIFO delay plus the decoder delay during this voice call, in ms.
ReceivePackets	Number of packets received by this peer during this call.
ReleaseSource	Number value of the release source.
RemoteIPAddress	Remote system IP address for the VoIP call.

Field	Description
RemoteUDPPort	Remote system User Datagram Protocol (UDP) listener port to which voice packets are sent.
RoundTripDelay	Voice packet round-trip delay, in ms, between the local and remote systems on the IP backbone for this call.
SelectedQoS	Selected Resource Reservation Protocol (RSVP) quality of service (QoS) for this call.
SessionProtocol	Session protocol used for an Internet call between the local and remote routers through the IP backbone.
SessionTarget	Session target of the peer used for this call.
SetupTime	Value of the system UpTime, in ms, when the call associated with this entry was started.
SIP call-legs	Total Session Initiation Protocol (SIP) call legs for which call records are available.
Telephony call-legs	Total telephony call legs for which call records are available.
TransmitBytes	Number of bytes sent by this peer during this call.
TransmitPackets	Number of packets sent by this peer during this call.
VAD	Whether voice activation detection (VAD) was enabled for this call.

Related Commands

Command	Description
dial-control-mib	Sets the maximum number of calls contained in the table.
show call active media	Displays call information for media calls in progress.

show call history video

To display call history information for signaling connection control protocol (SCCP) video calls, use the **show call history video** command in user EXEC or privileged EXEC mode.

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

Syntax Description

brief	(Optional) Displays a truncated version of video call history information.
id *identifier*	(Optional) Displays only the video call history with the specified identifier. Range is a hexadecimal value from 1 to FFFF.
compact	(Optional) Displays a compact version of video call history information.
duration	(Optional) Displays the call history for the specified time duration.
less	Displays the call history for shorter duration calls.
more	Displays the call history for longer duration calls.
seconds	Time, in seconds. The range is from 1 to 2147483647.
last *number*	(Optional) Displays the last calls connected, where the number of calls that appear is defined by the *number*argument. The range is from 1 to100 .

Command Modes

User EXEC (>) Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(16); 12.4(15)T	Cisco Unified CME 4.0	This command was modified. The Port and BearerChannel display fields were added to the TELE call leg record of the command output.

Examples

The following is sample output from the **show call history video** command with the **compact** option:

```
Router# show call history video compact
      <callID>  A/O FAX T<sec> Codec       type        Peer Address       IP R<ip>:<udp>
Total call-legs: 2
       241      ANS     T17    g729r8      VOIP        P555-0100          192.0.2.0:16926
       242      ORG     T17    g729r8      TELE-VIDEO  P555-0101
```

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

Table 4: show call history video Field Descriptions

Field	Description
callID	Unique identifier for the call leg.
A/O	Call leg was an answer (ANS) or an originator (ORG).
FAX	Fax number for the call leg.
T<sec>	Duration in seconds.
Codec	Codec used for this call leg.
type	Call type for this call leg.
Peer Address	Called or calling number of the remote peer.
IP R<ip>:<udp>	IP address and port number
Total call-legs	Total number of call legs for this call.

Related Commands

Command	Description
show call active video	Displays call information for SCCP video calls in progress.

show call history video record

To display information about incoming and outgoing video calls, use the **s how 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

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**| **more**} *seconds*]| **dest-route-string** *tag*| **id** *identifier*| **last** *number*| **redirect** {**rtpvt**| **tbct**}| **stats**]

Syntax Description

brief	(Optional) Displays a truncated version of the call history table.
id *identifier*	(Optional) Displays only the call with the specified identifier. Range is from 1 to FFFF.
compact	(Optional) Displays a compact version of the call history table.
dest-route-string *tag*	(Optional) Displays only the call with the specified destination route *tag* value. The range is from 1 to 10000.
duration *seconds*	(Optional) Displays history information for calls that are longer or shorter than the value of the specified *seconds* argument. The arguments and keywords are as follows: • **less** --Displays calls shorter than the *seconds* value. • **more** --Displays calls longer than the *seconds* value. • *seconds* --Elapsed time, in seconds. Range is from 1 to 2147483647.
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 to100 .
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.

stats	(Optional) Displays information about digital signal processing (DSP) voice quality metrics.

Command Modes

User EXEC (>) Privileged EXEC (#)

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
12.0(3)XG	Support was added 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 implemented on the Cisco MC3810.
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.
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 a specified call 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	This command was integrated into Cisco IOS Release 12.2(8)T. Support was not included for the Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.
12.2(11)T	Support was added for Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.
12.2(13)T	The ReleaseSource field was added to the Field Description table, and the **record** keyword was deleted from the command name.
12.3(1)	The **redirect** keyword was added.

Release	Modification
12.4(2)T	The LocalHostname display field was added to the VoIP call leg record.
12.4(11)XW	The **stats** keyword was added.
12.4(15)T	The Port and BearerChannel display fields were added to the TELE call leg record of the command output.
12.4(16)	The Port and BearerChannel display fields were added to the TELE call leg record of the command output.
12.4(22)T	Command output was updated to show IPv6 information.
15.3(3)M	This command was modified. The **dest-route-string** keyword was added.
Cisco IOS XE Release 3.10S	This command was integrated into Cisco IOS XE Release 3.10S.

Usage Guidelines

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 from 0 to 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. The timer value is 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 **last** keyword, 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 display its statistics by using the **show call active voice** command.

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

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]
CallID=2
RemoteIPAddress=10.14.82.14
RemoteUDPPort=18202
RoundTripDelay=2 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
FastConnect=TRUE
SessionProtocol=cisco
SessionTarget=ipv4:10.14.82.14
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]
CallID=3
Port=3/0/0 (3)
BearerChannel=3/0/0.1
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]
CallID=1
.
.
.
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]
CallID=2
Port=3/0/0 (2)
BearerChannel=3/0/0.1
```

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

```
Router# show call history voice brief
<ID>: <CallID> <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>
media inactive detected:<y/n> media cntrl rcvd:<y/n> timestamp:<time>
MODEMPASS <method> buf:<fills>/<drains> loss <overall%> <multipkt>/<corrected>
last <buf event time>s dur:<Min>/<Max>s
FR <protocol> [int dlci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
```

```
<codec> (payload size)
ATM <protocol> [int vpi/vci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
<codec> (payload size)
Telephony <int> (callID) [channel_id] tx:<tot>/<voice>/<fax>ms <codec> noise:<lvl>dBm
acom:<lvl>dBm
MODEMRELAY info:<rcvd>/<sent>/<resent> xid:<rcvd>/<sent> total:<rcvd>/<sent>/<drops>
disc:<cause code>
speeds(bps): local <rx>/<tx> remote <rx>/<tx>
Proxy <ip>:<audio udp>,<video udp>,<tcp0>,<tcp1>,<tcp2>,<tcp3> endpt: <type>/<manf>
bw: <req>/<act> codec: <audio>/<video>
tx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>
rx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>
```

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
```

The table below describes the significant fields shown in the **show call history voice redirect tbct** display.

Table 5: 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
dial-control-mib	Set the maximum number of calls contained in the table.
show call active fax	Displays call information for fax transmissions that are in progress.

Command	Description
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"
#}
```

Field descriptions should be self-explanatory.

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 leg

To display event logs and statistics for voice call legs, use the **show call leg**command in privileged EXEC mode.

show call leg {**active**| **history**} [**summary**| [**last** *number*| **leg-id** *leg-id*] [**event-log**| **info**]]

Syntax Description

active	Statistics or event logs for active call legs.
history	Statistics or event logs for terminated call legs.
summary	(Optional) A summary of each call leg.
last *number*	(Optional) Selected number of most recent call legs. Not available with **active** keyword.
leg-id *leg-id*	(Optional) A specific call leg. Output displays event logs or statistics for that call leg.
event-log	(Optional) Event logs for call legs.
info	(Optional) Statistics for call legs.

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

If you use the **leg-id** keyword, only statistics or event logs for that call leg display. To display event logs with this command, you must enable event logging with the **call leg event-log** command.

Examples

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

```
Router# show call leg active summary
G<id>  L<id>     Elog A/O FAX T<sec> Codec       type  Peer Address       IP R<ip>:<udp>
G11DC  L A        Y   ANS     T2     None        TELE  P4085550198
Total call-legs: 1
Router# show call leg active event-log

Event log for call leg ID: A        Connection ID: 11DC
buf_size=4K, log_lvl=INFO
<ctx_id>:<timestamp>:<seq_no>:<severity>:<msg_body>
A:1057277701:71:INFO: Call setup indication received, called = 4085550198,  calling = 52927,
```

```
 echo canceller = enable, direct inward dialing
A:1057277701:72:INFO: Dialpeer = 1
A:1057277701:77:INFO: Digit collection
A:1057277701:78:INFO: Call connected using codec None
Total call-legs: 1
Router# show call leg active info

Information for call leg ID: A        Connection ID: 11DC
 GENERIC:
SetupTime=3012940 ms
Index=1
PeerAddress=4085550198
PeerSubAddress=
PeerId=1
PeerIfIndex=329
LogicalIfIndex=253
ConnectTime=301295
CallDuration=00:00:20
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=2
TransmitPackets=412
TransmitBytes=98880
ReceivePackets=0
ReceiveBytes=0
TELE:
ConnectionId=[0x632D2CAB 0xACEB11D7 0x80050030 0x96F8006E]
IncomingConnectionId=[0x632D2CAB 0xACEB11D7 0x80050030 0x96F8006E]
TxDuration=20685 ms
VoiceTxDuration=0 ms
FaxTxDuration=0 ms
CoderTypeRate=None
NoiseLevel=-120
ACOMLevel=90
OutSignalLevel=-50
InSignalLevel=-41
InfoActivity=0
ERLLevel=38
EchoCancellerMaxReflector=16685
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=4085550198
OriginalCallingOctet=0x0
OriginalCalledNumber=52927
OriginalCalledOctet=0xE9
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0xFF
TranslatedCallingNumber=4085550198
TranslatedCallingOctet=0x0
TranslatedCalledNumber=52927
TranslatedCalledOctet=0xE9
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwReceivedCalledNumber=52927
GwReceivedCalledOctet3=0xE9
GwReceivedCallingNumber=4085550198
GwReceivedCallingOctet3=0x0
GwReceivedCallingOctet3a=0x81
Total call-legs: 1
```

For a description of the call leg statistics, see the description for the **show call active voice** command.

```
Router# show call leg active leg-id A

Call Information - Connection ID: 11DC , Call Leg ID: A
 GENERIC:
SetupTime=3012940 ms
Index=1
PeerAddress=4085550198
PeerSubAddress=
```

```
PeerId=1
PeerIfIndex=329
LogicalIfIndex=253
ConnectTime=301295
CallDuration=00:00:40
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=2
TransmitPackets=824
TransmitBytes=197760
ReceivePackets=0
ReceiveBytes=0
TELE:
ConnectionId=[0x632D2CAB 0xACEB11D7 0x80050030 0x96F8006E]
IncomingConnectionId=[0x632D2CAB 0xACEB11D7 0x80050030 0x96F8006E]
TxDuration=20685 ms
VoiceTxDuration=0 ms
FaxTxDuration=0 ms
CoderTypeRate=None
NoiseLevel=-120
ACOMLevel=90
OutSignalLevel=-50
InSignalLevel=-41
InfoActivity=0
ERLLevel=38
EchoCancellerMaxReflector=16685
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=4085550198
OriginalCallingOctet=0x0
OriginalCalledNumber=52927
OriginalCalledOctet=0xE9
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0xFF
TranslatedCallingNumber=4085550198
TranslatedCallingOctet=0x0
TranslatedCalledNumber=52927
TranslatedCalledOctet=0xE9
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwReceivedCalledNumber=52927
GwReceivedCalledOctet3=0xE9
GwReceivedCallingNumber=4085550198
GwReceivedCallingOctet3=0x0
GwReceivedCallingOctet3a=0x81
Call Event Log - Connection ID: 11DC , Call Leg ID: A
buf_size=4K, log_lvl=INFO
<ctx_id>:<timestamp>:<seq_no>:<severity>:<msg_body>
A:1057277701:71:INFO: Call setup indication received, called = 4085550198,  calling = 52927,
 echo canceller = enable, direct inward dialing
A:1057277701:72:INFO: Dialpeer = 1
A:1057277701:77:INFO: Digit collection
A:1057277701:78:INFO: Call connected using codec None
Call-leg found: 1
Router# show call leg active leg-id A event-log

Call Event Log - Connection ID: 11DC , Call Leg ID: A
buf_size=4K, log_lvl=INFO
<ctx_id>:<timestamp>:<seq_no>:<severity>:<msg_body>
A:1057277701:71:INFO: Call setup indication received, called = 4085550198,  calling = 52927,
 echo canceller = enable, direct inward dialing
A:1057277701:72:INFO: Dialpeer = 1
A:1057277701:77:INFO: Digit collection
A:1057277701:78:INFO: Call connected using codec None
Call-leg found: 1
Router# show call leg history summary

G<id>  L<id>     Elog A/O FAX T<sec> Codec       type  Peer Address       IP R<ip>:<udp>
disc-cause
G11DB  L 7        Y   ANS     T24    None        TELE  P4085550198 D10
```

```
G11DC  L A        Y   ANS     T159   None        TELE  P4085550198 D10
Total call-legs: 2
Router# show call leg history last 1
Call Information - Connection ID: 11DC , Call Leg ID: A
GENERIC:
SetupTime=3012940 ms
Index=4
PeerAddress=4085550198
PeerSubAddress=
PeerId=1
PeerIfIndex=329
LogicalIfIndex=253
DisconnectCause=10
DisconnectText=normal call clearing (16)
ConnectTime=301295
DisconnectTime=317235
CallDuration=00:02:39
CallOrigin=2
ReleaseSource=1
ChargedUnits=0
InfoType=speech
TransmitPackets=2940
TransmitBytes=705600
ReceivePackets=0
ReceiveBytes=0
TELE:
ConnectionId=[0x632D2CAB 0xACEB11D7 0x80050030 0x96F8006E]
IncomingConnectionId=[0x632D2CAB 0xACEB11D7 0x80050030 0x96F8006E]
TxDuration=20685 ms
VoiceTxDuration=0 ms
FaxTxDuration=0 ms
CoderTypeRate=None
NoiseLevel=-120
ACOMLevel=90
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=4085550198
OriginalCallingOctet=0x0
OriginalCalledNumber=52927
OriginalCalledOctet=0xE9
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0xFF
TranslatedCallingNumber=4085550198
TranslatedCallingOctet=0x0
TranslatedCalledNumber=52927
TranslatedCalledOctet=0xE9
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwReceivedCalledNumber=52927
GwReceivedCalledOctet3=0xE9
GwReceivedCallingNumber=4085550198
GwReceivedCallingOctet3=0x0
GwReceivedCallingOctet3a=0x81
Call Event Log - Connection ID: 11DC , Call Leg ID: A
buf_size=4K, log_lvl=INFO
<ctx_id>:<timestamp>:<seq_no>:<severity>:<msg_body>
A:1057277701:71:INFO: Call setup indication received, called = 4085550198,  calling = 52927,
 echo canceller = enable, direct inward dialing
A:1057277701:72:INFO: Dialpeer = 1
A:1057277701:77:INFO: Digit collection
A:1057277701:78:INFO: Call connected using codec None
A:1057277860:150:INFO: Inform application call disconnected (cause = normal call clearing
(16))
A:1057277860:154:INFO: Call disconnected (cause = normal call clearing (16))
A:1057277860:155:INFO: Call released
Total call-legs: 1
Total call-legs with event log: 1
Router# show call leg history leg-id A event-log

Call Event Log - Connection ID: 11DC , Call Leg ID: A
buf_size=4K, log_lvl=INFO
```

```
<ctx_id>:<timestamp>:<seq_no>:<severity>:<msg_body>
A:1057277701:71:INFO: Call setup indication received, called = 4085550198,  calling = 52927,
 echo canceller = enable, direct inward dialing
A:1057277701:72:INFO: Dialpeer = 1
A:1057277701:77:INFO: Digit collection
A:1057277701:78:INFO: Call connected using codec None
A:1057277860:150:INFO: Inform application call disconnected (cause = normal call clearing
(16))
A:1057277860:154:INFO: Call disconnected (cause = normal call clearing (16))
A:1057277860:155:INFO: Call released
Call-leg matched ID found: 1
Call-legs matched ID with event log: 1
```

Field descriptions should be self-explanatory.

Related Commands

Command	Description
call leg event-log	Enables event logging for voice, fax, and modem call legs.
call leg event-log dump ftp	Enables the voice gateway to write the contents of the call-leg event log buffer to an external file.
call leg event-log error-only	Restricts event logging to error events only for voice call legs.
call leg event-log max-buffer-size	Sets the maximum size of the event log buffer for each call leg.
call leg history event-log save-exception-only	Saves to history only event logs for call legs that had at least one error.
monitor call leg event-log	Displays the event log for an active call leg in real-time.

show call media forking

To display currently active media forking sessions, use the **show call media forking** command in user EXEC or privileged EXEC mode.

show call media forking

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC (>) Privileged EXEC (#)

Command History

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

Usage Guidelines

Use this command to verify that media forking was successful for relevant anchor legs.

Examples

The following example is a sample output from the s**how call media forking** command..

```
Router# show call media forking
Warning: Output may be truncated if sessions are added/removed concurrently!
Session Call n/f Destination (port address)
7 6 far 1234 1.5.35.254
8 6 near 5678 1.5.35.254
```

The table below describes the fields that are displayed in the output.

Field	Description
Session	Session Identifier.
Call	Call Leg identifier in hexadecimal. It must match the Call ID from the show call leg active command.
n/f	Direction (Near End or Far End) of the voice stream that was forked.
Destination (port address)	Destination for the forked packets. It consists of the folllowing: • RTP Port • IP Address

show callmon

To display call monitor information, use the **show callmon** command in user EXEC or privileged EXEC mode.

show callmon {**call**| **gcid**| **subscription**| **trace** {**all**| **event** {**all**| **call**| **connection**}| **exec**| **server**| **subscription**| **trigger**}}

Syntax Description

call	Displays the active call monitor calls.
gcid	Displays the active global call ID information.
subscription	Displays the subscription information.
trace	Displays the trace information.
all	Displays all types of traces based on time.
event	Displays the event trace information. • **all** --Displays all event traces. • **call** --Displays event traces related to a call. • **connection** --Displays the event traces related to a connection.
exec	Displays all critical execution traces.
server	Displays all session server up or down traces.
subscription	Displays all subscription traces.
trigger	Displays the entire trigger structure by index.

Command Modes

User EXEC (>) Privileged EXEC (#))

Command History

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

Examples

The following sample output from the **show callmon call** command shows active call monitor calls:

```
Router# show callmon call
 line dn     sub_id   number of call instance
6401,     1
      callID 2038(19D7),  *cg = 6401, cd = 6601
6601,     1
      callID 2039(19D7),  cg = 6401, *cd = 6601
```

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

Table 6: show callmon call Field Descriptions

Field	Description
dn	Directory number.
number of call	Number of call instances.
instance	Contents of the call instance.

The following sample output from the **show callmon gcid** command shows the active global call ID information:

```
Router# show callmon gcid
 GCID                              callIDs(active_entry_id)
AE48ECBC-D89311DB-87FC996E-115FF692
 isConfGcid:FALSE        gcid_conf:00000000-00000000-00000000-00000000
, 2038(19D7), 2039(19D7)
```

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

Table 7: show callmon gcid Field Descriptions

Field	Description
GCID	Global call ID.
CallIDs	Active call IDs.

Related Commands

Command	Description
callmonitor	Enables call monitoring messaging functionality on a SIP endpoint in a VoIP network.

show call prompt-mem-usage

To display the amount of memory used by prompts, use the **show call prompt-mem-usage**command in privileged EXEC mode.

show call prompt-mem-usage [detail]

Syntax Description

detail	(Optional) Displays details about memory usage and names of tones used.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.2(15)T	This command was introduced.
12.3(7)T	The **detail** keyword was added.

Usage Guidelines

Use this command to display the number of prompts loaded into the gateway, the amount of memory used by the prompts, the number of prompts currently being played, and the status of prompt loads.

For calls transferred by a Cisco CallManager Express (Cisco CME) system, the ringback tone generation for commit-at-alerting uses an interactive voice response (IVR) prompt playback mechanism. Ringback tone is played to the transferred party by the Cisco CME system associated with the transferring party.

The system automatically generates tone prompts as needed on the basis of the network-locale setting made in the Cisco CME system.

Examples

The following sample output shows details about the memory usage of the prompts that are used.

```
Router# show call prompt-mem-usage
Prompt memory usage:
             config'd      wait    active      free     mc total    ms total
    file(s)      0200      0010      0001     00189        00011       00002
    memory   02097152  00081259  00055536  01960357     00136795
 Prompt load counts: (counters reset 0)
    success 11(1st try) 0(2nd try), failure 0
 Other mem block usage:
            mcDynamic   mcReader
    gauge       00001      00001
Number of prompts playing: 1
Number of start delays   : 0
MCs in the ivr MC sharing table
===============================
Media Content: NoPrompt (0x83C64554)
  URL:
  cid=0, status=MC_READY size=24184 coding=g711ulaw refCount=0
```

```
Media Content: tone://GB_g729_tone_ringback (0x83266EC8)
  URL: tone://GB_g729_tone_ringback
```

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

Table 8: show call prompt-mem-usage Field Descriptions

Field	Description
file(s)	Number of prompts in different queues.
file(s) - config'd	Maximum number of configured prompts that can be simultaneously available in memory. In the sample output, the value of 200 in this field means that loading the 201st prompt results in the oldest prompts being removed.
file(s) -wait	Number of prompts in the wait queue that are not being used in any call and are ready to be deleted when there is no space for a new prompt. This field lists older prompts that can be deleted.
file(s) - active	Number of prompts that are being used in active calls. These prompts cannot be deleted.
file(s) - free	Number of prompts that can be loaded without deleting any prompt from the wait queue. This is the number of configured prompts (listed under config'd) minus the total number of prompts in the wait and active states.
file(s) - mc total	Total number of prompts in the wait and active states.
ms total	Number of media streams that are currently active. One media stream is used for playing INBOX prompts. A prompt is considered an INBOX prompt if its URL is either flash:, http:, ram:, or tftp:.
memory	Displays the memory used by prompts, in bytes.
memory - config'd	Maximum amount of memory configured to be available for prompts.
memory - wait	Total amount of memory used by prompts in the wait list.
memory - active	Total amount of memory used by prompts in the active list.
memory - free	Amount of available memory. This is the amount of configured prompts (listed under config'd) memory minus the total amount of memory used by the prompts in the wait and active lists.

Field	Description
memory - mc total	Total amount of memory used by prompts in the wait and active lists.
Prompt load counts	Number of successful attempts to load a prompt on the first try and on the second try, and the number of attempts to load a prompt that failed.
mcDynamic	Number of dynamic element queues that are active. A dynamic element queue is a list of prompts that are played together.
mcReader	Number of mcReaders that are active. An mcReader is used for playing one mcDynamic queue of prompts. An mcReader is used only if the mcDynamic contains prompts that are associated with one of the following types of URL: flash:, http:, ram:, or tftp:.
Number of prompts playing	Number of prompts that are currently playing.
Number of start delays	Number of times that prompts failed to start and have subsequently restarted.
MCs in the ivr MC sharing table	The fields below this line of text refer to each media content (prompt) currently cached in memory. In the sample output, the only cached prompt is the built-in default prompt named "NoPrompt."
Media Content	Name of the prompt, which is derived from the audio file URL (the characters after the last "/" in the URL). The address in parentheses is the memory location of the prompt.
URL	Location of the file for the prompt that is playing. In the case of the default prompt, NoPrompt, no URL is given.
cid	Call identification number of the call that initiated the loading of the prompt.

Field	Description
status	Status of the media content. The following values are possible: • MC_NOT_READY--Initial status for media content. When the media content is successfully loaded, the status will change to MC_READY. • MC_READY--Media content is loaded into memory and ready for use. • MC_LOAD_FAIL--Media content failed to load.
size	Size of the media content, in bytes.
coding	Type of encoding used by the media content.
refCount=0	Number of calls to which this media content is currently being streamed.

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 [**ds0**| **dsp**]

Syntax Description

ds0	(Optional) Specifies the voice digital signal level zero (DS0) resource statistics information.
dsp	(Optional) Specifies the voice digital signal processor (DSP) resource statistics information.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.0(5)T	This command was introduced.
12.1(5)XM2	This command was integrated into Cisco IOS Release 12.1(5)XM2
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T.
12.2(2)XB1	This command was integrated into Cisco IOS Release into 12.2(2)XB1.
12.2(8)T	This command was modified. Support for the Cisco AS5300,Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 series routers is not included in this release.
12.4(22)T	This command was modified. The **ds0** and **dsp** keywords were added.

Usage Guidelines

The **show call resource voice stats** command displays the H.323 resources that are monitored when the **resource threshold** command is used to configure resource threshold reporting.

Examples

The following is sample output from the show call resource voice stats command, which 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
```

The table below describes significant fields shown in this output.

Table 9: show call resource voice stats Field Descriptions

Statistic	Definition
Total channels	Number of channels physically configured for the resource.
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.
Disabled channels	Number of addressable channels that are physically down or that have been disabled administratively with the **shutdown** or **busyout** command.
Pending channels	Number of addressable channels that are pending in loadware download.
Free channels	Number of addressable channels that are free.
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.

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 [**ds0**| **dsp**]

Syntax Description

ds0	(Optional) Specifies the voice digital signal level zero (DS0) resource statistics information.
dsp	(Optional) Specifies the voice digital signal processor (DSP) resource statistics information.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.0(5)T	This command was introduced.
12.1(5)XM2	This command was integrated into Cisco IOS Release 12.1(5)XM2
12.2(2)XB1	This command was integrated into Cisco IOS Release into 12.2(2)XB1.
12.4(22)T	This command was modified. The **ds0** and **dsp** keywords were added.

Usage Guidelines

The **show call resource voice threshold** command displays the H.323 resource thresholds that are configured with the **resource threshold** command.

Examples

The following is sample output from the show call resource voice threshold command, which 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
```

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

Table 10: show call resource voice threshold Field Descriptions

Field	Description
High Water Mark	Resource-utilization level that triggers a message indicating that H.323 resource use is high. The range is 1 to 100. A value of 100 indicates that the resource is unavailable.The default is 90.
Low Water Mark	Resource-utilization level that triggers a message indicating that H.323 resource use has dropped below the high-usage level. The range is 1 to 100. The default is 90.

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
```

The table below describes significant fields shown in this output.

Table 11: 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
```

The table below describes significant fields shown in this output.

Table 12: 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.

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 [**dial-peer** *tag*]

Syntax Description

dial-peer	(Optional) Displays configuration information for a dial peer.
tag	(Optional) Specifies the dial peer identifying number. Range is from 1 to 2147483647.

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 was not included in this release.
12.2(11)T	Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 was added in this release.
15.1(3)T	This command was modified. The output fields of the command were modified to include the output at the dial peer level.

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
```

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

Table 13: show call spike status Field Descriptions

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

```
Router# show call spike status dial-peer 400
TAG        CONFIG    SPIKED TOTAL REJECTED CALLS    REJECTED CALLS
400        YES       NO        4                      0
```

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

Table 14: show call spike status (dial peer) Field Descriptions

Field	Description
TAG	Dial peer tag.
CONFIG	Displays if the **call spike** command has been configured.
SPIKED	Details if the call spiking limit has been triggered.
TOTAL REJECTED CALLS	Displays the number of calls rejected due to a call spike in the dial peer.
REJECTED CALLS	Displays the number of calls rejected when the call spike was triggered until the call spike control was released.

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 the 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 is 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 is 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 was implemented on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850.
15.2(2)T	This command was modified. The output was modified to display the configured bandwidth threshold, bandwidth availability, and call admission control statistics.

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
```

The table below describes significant fields shown in this output.

Table 15: 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 and number.
Type	Type of resource.
Value	Value of a call that is 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 thiscommand:

```
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
```

The table below describes significant fields shown in this output.

Table 16: 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.
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
```

The table below describes significant fields shown in this output.

Table 17: 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.

Field	Description
total-calls	Number of calls rejected for failing the total-calls parameter.

Related Commands

Command	Description
call treatment on	Enables call treatment to process calls when local resources are unavailable.
call treatment action	Configures the action that the router takes when local resources are unavailable.
call treatment cause-code	Specifies the reason for the disconnection to the caller when local resources are unavailable.
call treatment isdn-reject	Specifies the rejection cause-code for ISDN calls when local resources are unavailable.
clear call treatment stats	Clears the call-treatment statistics.

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	Descriptors provisioned on the border element.
dynamic	Dynamically learned descriptors.
all	Both static and dynamic descriptors.

Command Default

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= 6561676C650000000000000000000A
  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
```

Field descriptions should be self-explanatory.

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)
```

The table below describes significant fields shown in this output.

Table 18: 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.

Field	Description
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.
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**| **download-tones** [**c1**| **c2**]]

Syntax Description

backhaul	(Optional) Displays information about the backhaul link.
config-download	(Optional) Displays information about the status of Media Gateway Control Protocol (MGCP) and Skinny Client Control Protocol (SCCP) 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.
download-tones c1 | c2	(Optional) Displays custom tones downloaded to the gateway. The custom tone value of c1 or c2 specifies which tone information to display.

Command Default

If none of the optional keywords is specified, information related to all keywords is displayed.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.1(3)T	This command 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(11) and the Cisco CallManager Version 3.2. It was implemented on the Cisco IAD2420 series.
12.2(15)ZJ	The download-tones [c1 | c2] keywords were added for the following platforms: Cisco 2610XM, Cisco 2611XM, Cisco 2620XM, Cisco 2621XM, Cisco 2650XM, Cisco 2651XM, Cisco 2691, Cisco 3640A, Cisco 3660, Cisco 3725, and Cisco 3745.
12.3(4)T	The keywords were integrated into Cisco IOS Release 12.3(4)T.
12.3(14)T	New output was added relating to SCCP autoconfiguration.
12.4(15)XY	The display output was modified to include the number of TFTP download failures allowed.

Usage Guidelines

Use the **show ccm-manager config-download** command to determine the status of Cisco Unified Communications Manager servers and the automatic download information and statistics.

Examples

The following sample output shows the configured amplitudes, frequencies, and cadences of custom tone 1, Hong Kong:

```
Router# show ccm-manager download-tones c1
!
Custom Tone 1 : Hong Kong
Pulse dial:normal, Percent make:35%, DTMF low Amp.= 65424, high Amp.= 65446, Pcm:u-Law
                                      FXS  FXO  E&M  FXS  FXO  E&M
Dual Tone DR NF FOF FOS AOF AOF AOF AOS AOS AOS ONTF OFTF ONTS OFTS ONTT OFTT ONT4 OFT4
(optional) FOF2 FOS2 FOF3 FOS3 FOF4 FOS4 FOT FO4 AOT AO4 RCT1 RCT2 RCT3 RCT4
BUSY 0  2 480 620 -120 -120 -120 -120 -120 -120 500 500 0 0 0 0 0 0
RING_BACK           0  2   440   520 -120 -120 -120 -120 -120 -120   400   200   400  3000

CONGESTION          0  2   480   620 -200 -200 -200 -240 -240 -240   250   250     0     0
 NUMBER_UNOBTAINABLE 0  2   480   620 -120 -120 -120 -120 -120 -120 65535     0     0
0 DIAL_TONE           0  2   350   440 -150 -150 -150 -150 -150 -150 65535     0     0
 0 DIAL_TONE2          0  2   350   440 -150 -150 -150 -150 -150 -150 65535     0     0
  0 OUT_OF_SERVICE      0  1   950     0 -150 -150 -150    0    0    0   330   330     0
   0 ADDR_ACK            0  1   600     0 -240 -240 -240    0    0    0   125   125   125
65535 DISCONNECT          0  1   600     0 -150 -150 -150    0    0    0   330   330   330
 65535 OFF_HOOK_NOTICE     0  2  1400  2040 -240 -240 -240 -240 -240 -240   100   100
0     0 OFF_HOOK_ALERT      0  2  1400  2040 -240 -240 -240 -240 -240 -240   100   100
 0     0 WAITING             0  0     0     0    0    0    0    0    0    0     0     0
```

```
  0      0 CONFIRM            0  0      0      0     0     0     0     0     0     0      0      0
   0     0
CNFWRN_J             0  1   950      0 -170 -170 -190     0     0     0   100    100    100 65535
 CNFWRN_D             0  1   600      0 -170 -170 -190     0     0     0   100    100    100 65535
 STUTT_DIALTONE       0  2   350   440 -150 -150 -150 -150 -150 -150   100    100    100    100
   100   100 65535      0
PERM_SIG_TONE        0  1   480      0 -170 -170 -170     0     0     0 65535      0      0      0
WAITING1             0  0     0      0     0     0     0     0     0     0      0      0      0      0
 WAITING2             0  0     0      0     0     0     0     0     0     0      0      0      0
0 WAITING3             0  0     0      0     0     0     0     0     0     0      0      0      0
 0 WAITING4             0  0     0      0     0     0     0     0     0     0      0      0      0
  0 MSGWAIT_IND          0  0     0      0     0     0     0     0     0     0      0      0      0
   0 OFF_HOOK_WARN        0  0     0      0     0     0     0     0     0     0      0      0      0
    0
Sequence Tone       DR NF F1C1  F2C1   AOF   AOS  C1ONT C1OFT C2ONT C2OFT C3ONT C3OFT C4ONT
 C4OFT  F1C2  F2C2  F1C3  F2C3  F1C4  F2C4
INTERCEPT            0  0     0      0     0     0      0     0     0     0     0     0
TONE_ON_HOLD         0  0     0      0     0     0      0     0     0     0     0     0
NO_CIRCUIT           0  0     0      0     0     0      0     0     0     0     0     0
Legend:
DR: direction NF: number of frequency FO<F,S,T,4>: frequency of<1st,2nd,3rd,4th>  AO<F,S,T,4>:
 amplitude of<1st,2nd,3rd,4th>
FOF<1-4>: frequency of 1st, cadence<1-4> FOS<1-4>: frequency of 2nd, cadence<1-4>
RCT<1-4>: repeat count for cadence<1-4>  F(1-4>C<1-4> : frequency<1-4> of cadence<1-4>
C<1-4>ONT: cadence<1-4> on time C<1-4>OFT: cadence<1-4> off time
```

The three tables below and give descriptions of significant fields once the tones are automatically downloaded to the gateway.

Table 19: show ccm-manager download-tones Significant Output Fields

Field	Description
Percent make	Pulse ratio in percentage of make.
DTMF low Amp.	Low frequency level.
high Amp.	High frequency level.
Pcm	Pulse Code Modulation (mu-law or a-law).

Table 20: show ccm-manager download-tones Output Fields for Dual Tones

Field of Dual Tone	Description
DR	Direction to PSTN (0) or Packet Network (1).
NF	Number of Frequency (from 1 to 4).
FOF	Frequency of First component (in Hz).
FXS AOF	Amplitude of First component (from 1 to 65535 = +3 dBm0) for the foreign exchange station (FXS).
FXO AOF	Amplitude of First component (from 1 to 65535 = +3 dBm0) for the foreign exchange office (FXO).

Field of Dual Tone	Description
E&M AOF	Amplitude of First component (from 1 to 65535 = +3 dBm0) for the recEive and transMit (E&M).
FXS AOS	Amplitude of Second component (from 1 to 65535 = +3 dBm0) for the FXS.
FXO AOS	Amplitude of Second component (from 1 to 65535 = +3 dBm0) for the FXO.
E&M AOS	Amplitude of Second component (from 1 to 65535 = +3 dBm0) for the E&M.
ONTF	On time; time the tone is generated (milliseconds) for the first frequency.
OFTF	Off time; silence time (milliseconds) for the first frequency.
ONTS	On time; time the tone is generated (milliseconds) for the second frequency.
OFTS	Off time; silence time (milliseconds) for the second frequency.
ONTT	On time; time the tone is generated (milliseconds) for the third frequency.
OFTT	Off time; silence time (milliseconds) for the third frequency.
ONT4	On time; time the tone is generated (milliseconds) for the fourth frequency.
OFT4	Off time; silence time (milliseconds) for the fourth frequency.
FOF2	Frequency of First component for the second cadence.
FOS2	Frequency of Second component for the second cadence.
FOF3	Frequency of First component for the third cadence.
FOS3	Frequency of Second component for the third cadence.
FOF4	Frequency of First component for the fourth cadence.

Field of Dual Tone	Description
FOS4	Frequency of Second component for the fourth cadence.
FOT	Frequency of Third component (in Hertz).
FO4	Frequency of Fourth component (in Hertz).
AOT	Amplitude of Third component (from 1 to 65535 = +3 dBm0).
AO4	Amplitude of Fourth component (from 1 to 65535 = +3 dBm0).
RCT1	Number of repeat for the first cadence.
RCT2	Number of repeat for the second cadence.
RCT3	Number of repeat for the third cadence.
RCT4	Number of repeat for the fourth cadence.

Table 21: show ccm-manager download-tones Output Fields for Sequence Tones

Field of Sequence Tone	Description
DR	Direction to PSTN (0) or Packet Network (1).
NF	Number of Frequency (from 1 to 4).
F1C1	Frequency 1 of Cadence 1.
F2C1	Frequency 2 of Cadence 1.
AOF	Amplitude of First component (from 1 to 65535).
AOS	Amplitude of Second component (from 1 to 65535).
C1ONT	Cadence 1 On Time.
C1OFT	Cadence 1 Off Time.
C2ONT	Cadence 2 On Time.
C2OFT	Cadence 2 Off Time.
C3ONT	Cadence 3 On Time.

Field of Sequence Tone	Description
C3OFT	Cadence 3 Off Time.
C4ONT	Cadence 4 On Time.
C4OFT	Cadence 4 Off Time.
F1C2	Frequency 1 of Cadence 2.
F2C2	Frequency 2 of Cadence 2.
F1C3	Frequency 1 of Cadence 3.
F2C3	Frequency 2 of Cadence 3.
F1C4	Frequency 1 of Cadence 4.
F2C4	Frequency 2 of Cadence 4.

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 Unified Communications Manager server:

```
Router# show ccm-manager
MGCP Domain Name: Router2821.cisco.com
Priority        Status                   Host
============================================================
Primary         Registered               10.78.236.222
First Backup    None
Second Backup   None
Current active Call Manager:    10.78.236.222
Backhaul/Redundant link port:   2428
Failover Interval:              30 seconds
Keepalive Interval:             15 seconds
Last keepalive sent:            21:48:37 UTC Nov 4 2007 (elapsed time: 00:00:15)
Last MGCP traffic time:         21:48:51 UTC Nov 4 2007 (elapsed time: 00:00:02)
Last failover time:             None
Last switchback time:           None
Switchback mode:                Graceful
MGCP Fallback mode:             Not Selected
Last MGCP Fallback start time:  None
Last MGCP Fallback end time:    None
MGCP Download Tones:            Disabled
TFTP retry count to shut Ports: 3
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
```

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

Table 22: 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 Unified Communications Manager server. Values are Registered, Idle, Backup Polling, and Undefined.
Host	Host IP address of the Cisco CallManager server.
Current active Call Manager	IP address of the active Cisco Communications Manager server. This field can be the IP address of any one of the following Cisco Communications Manager servers: 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	Interval following which, if the gateway has not received any messages from the currently active Cisco Communications Manager server within the specified amount of time, the gateway sends a keepalive message to the Cisco Communications Manager server to determine if it is operational.
Last keepalive sent	Time in hours (military format), minutes and seconds at which the last keepalive message was sent.

Field	Description
Last MGCP traffic time	Time in hours (military format), minutes and seconds 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 are 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 in hours (military format), minutes and seconds of the last fallback.
Lasts MGCP Fallback end time	End time stamp in hours (military format), minutes and seconds of the last fallback.
MGCP Download Tones	Displays if the customized tone download is enabled.
TFTP retry count to shut Ports	Number of TFTP download failures allowed before endpoints are shutdown.

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

```
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
SCCP auto-configuration status
===============================================================
Registered with Call Manager: No
Local interface: FastEthernet0/0 (000c.8522.6910)
Current version-id: {D3A886A2-9BC9-41F8-9DB2-0E565CF51E5A}
Current config applied at: 04:44:45 EST Jan 9 2003
Gateway downloads succeeded: 1
Gateway download attempts: 1
Last gateway download attempt:  04:44:45 EST Jan 9 2003
```

```
Last successful gateway download: 04:44:45 EST Jan 9 2003
Current TFTP server: 10.2.6.101
Gateway resets: 0
Gateway restarts: 0
Managed endpoints: 6
Endpoint downloads succeeded: 6
Endpoint download attempts: 6
Last endpoint download attempt:  04:44:45 EST Jan 9 2003
Last successful endpoint download: 04:44:45 EST Jan 9 2003
Endpoint resets: 0
Endpoint restarts: 0
Configuration Error History:
sccp ccm CCM-PUB7 identifier 1
end
controller T1 2/0no shut
controller T1 2/0no shut
controller T1 2/0no shut
isdn switch-type primary-ni
end
```

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

Table 23: 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.
Managed endpoints	Number of SSCP-controlled endpoints (analog and BRI phones).
Endpoint downloads succeeded	Number of times the gateway has successfully downloaded the configuration files for SCCP-controlled endpoints.
Endpoint download attempts	Number of times the gateway has tried to download the configuration files for SCCP-controlled endpoints.
Endpoint resets	Number of SCCP gateway resets.
Endpoint restarts	Number of SCCP gateway restarts.
Configuration Error History	Displays SCCP autoconfiguration errors.

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
```

The table below displays te mode. Modes are as follows:

Table 24: show ccm-manager fallback-mgcp modes

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 in hh:mm:ss of the last fallback.
Last MGCP Fallback end time	End time stamp in hh:mm:ss 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
```

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

Table 25: 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
ccm-manager config	Supplies the local MGCP voice gateway with the IP address or logical name of the TFTP server from which to download XML configuration files and enable the download of the configuration.
debug ccm-manager	Displays debugging information about the Cisco CallManager.
show ccm-manager	Displays a list of Cisco CallManager servers, their current status, and their availability.
show ccm-manager fallback-mgcp	Displays the status of the MGCP gateway fallback feature.
show isdn status	Displays the Cisco IOS gateway ISDN interface status.
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.
12.3(4)T	This command was enhanced to display V.120 call types registering with the modem.

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. The output displays the following information:

- Signaling stacks that register with CDAPI
- Applications that register with CDAPI
- Active calls
- Call type of each active call
- Message buffers in use

```
Router# show cdapi
Registered CDAPI Applications/Stacks
====================================
Signaling Stack: ISDN
        Interface: Se6/0:23
Application: TSP CDAPI Application Voice
        Application Type(s) : Voice Data Facility Signaling V110 V120
        Application Level   : Tunnel
        Application Mode    : Enbloc
Application: TSP CDAPI Application COT
        Application Type(s) : Cot
        Application Level   : Tunnel
        Application Mode    : Enbloc
Application: CSM
        Application Type(s) : Modem V110 V120
        Application Level   : Basic
        Application Mode    : Enbloc
```

```
Signaling Stack: XCSP
Application: dialer
        Application Type(s) : Data
        Application Level   : Basic
        Application Mode    : Enbloc
Active CDAPI Calls
==================
        Se7/7:23 Call ID = 0x7717, Call Type = V.120, Application = CSM
CDAPI Message Buffers
=====================
Free Msg Buffers: 320
Free Raw Buffers: 320
Free Large-Raw Buffers: 120
```

Field descriptions should be self-explanatory. However, the following information may be of help:

- Enbloc is the mode where all call-establishment information is sent in the setup message (opposite of overlap mode, where additional messages are needed to establish the call).
- Cot is the Continuity Test (COT) subsystem that supports the continuity test required by the Signaling System 7 (SS7) network to conduct loopback and tone check testing on the path before a circuit is established.

Related Commands

Command	Description
debug cdapi	Displays information about the CDAPI.

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
```

Field descriptions should be self-explanatory.

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, use the **show connect**command in privileged EXEC mode.

show connect {**all**| **elements**| **name**| **id**| **port** {**T1**| **E1**}*slot/port*}

Syntax Description

all	Information for all configured connections.
elements	Information for registered hardware or software interworking elements.
name	Information for 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	Information for 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	Information for a connection that you specify by indicating the type of controller (T1 or E1) and location of the interface.
T1	T1 controller.
E1	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.

Command Default

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 modular access routers that support drop-and-insert. 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
```

Field descriptions should be self-explanatory.

Related Commands

Command	Description
connect	Defines connections between T1 or E1 controller ports for Drop and Insert.

Command	Description
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. Valid entries are 1 or 2.
port	Port location of the EIA/TIA-366 interface in the VDM module.

Command Default

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

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
```

The table below describes significant fields shown in this output.

Table 26: 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.

Field	Description
LCR	Line control register of the VDM.
ICSR	Interrupt control and status register of the VDM.
EXT	Extended register of the VDM.
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 *controller -number*	Controller number of CAS or ISDN PRI time slot. Range is from 0 to 7.
timeslots *timeslot -range*	Timeslot. E1 range is from 1 to 31. T1 range is from 1 to 24.

Command Default

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. The command shows whether the DS0 channels of a controller are in idle, in-service, maintenance, or busyout states. Use the **show controllers e1** or **show controllers t1** command to display statistics about the E1 or T1 links.

Examples

The following example shows that the CAS state is enabled on the Cisco AS5300 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 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
```

Field descriptions should be self-explanatory.

Related Commands

Command	Description
show controllers e1	Displays information about E1 links.
show controllers t1	Displays information about T1 links.

show controllers voice

To display information about voice-related hardware, use the **show controllers voice** command inprivileged EXEC mode.

show controllers voice

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.0(5)XQ	This command was introduced on the Cisco 1750.

Usage Guidelines

This command displays interface status information that is specific to voice-related hardware, such as the registers of the TDM switch, the host port interface of the digital signal processor (DSP), and the DSP firmware versions. The information displayed is generally useful only for diagnostic tasks performed by technical support.

Examples

The following is sample output from this command:

```
Router# show controllers voice
EPIC Switch registers:
STDA 0xFF STDB 0xFF SARA 0xAD SARB 0xFF SAXA 0xFF SAXB 0x0 STCR 0x3F
MFAIR 0x3F
STAR 0x65 OMDR 0xE2 VNSR 0x0 PMOD 0x4C PBNR 0xFF POFD 0xF0 POFU 0x18
PCSR 0x1 PICM 0x0 CMD1 0xA0 CMD2 0x70 CBNR 0xFF CTAR 0x2 CBSR 0x20 CSCR
0x0
DSP 0 Host Port Interface:
HPI Control Register 0x202
InterfaceStatus 0x2A MaxMessageSize 0x80
RxRingBufferSize 0x6 TxRingBufferSize 0x9
pInsertRx 0x4 pRemoveRx 0x4 pInsertTx 0x6 pRemoveTx 0x6
Rx Message 0:
packet_length 100 channel_id 2 packet_id 0 process id 0x1
0000:   0000 4AC7 5F08 91D1 0000 0000 7DF1 69E5 63E1 63E2
0020:   6E7C ED67 DE5D DB5C DC60 EC7E 6BE1 58D3 50CD 4DCE
0040:   50D2 5AE5 7868 DA52 CE4A C746 C647 C94B D25A EAF4
0060:   5DD7 4FCD 4ACA 4ACC 4FD3 5DE8 F769 DC58 D352 D253
0080:   D65B E573 6CDF 59D3 4ECF 4FD0
Rx Message 1:
packet_length 100 channel_id 1 packet_id 0 process id 0x1
0000:   0000 1CDD 3E48 3B74 0000 0000 3437 3D4C F0C8 BBB5
0020:   B2B3 B7BF D25B 4138 3331 3339 435F CFBD B6B2 B1B4
0040:   BBC8 7E48 3B34 3131 363D 4FDE C3B9 B3B1 B3B8 C2DB
0060:   533F 3833 3235 3B48 71CC BDB7 B4B5 B8BF CF67 483D
0080:   3836 383C 455B DAC6 BDB9 B9BB
```

```
Rx Message 2:
packet_length 100 channel_id 2 packet_id 0 process id 0x1
0000:   0000 4AC8 5F08 9221 0000 0000 54DA 61F5 EF60 DA53
0020:   CF4F CD4E D256 DB63 FCEE 5FDA 55D1 50CF 4FD3 56D8
0040:   5DE1 6E7C EC60 DC59 D655 D456 D85D DF6A F4F4 69E2
0060:   5CDD 5BDC 5BDE 61E9 6DF1 FF76 F16D E96A E566 EA6A
0080:   EB6F F16D EF79 F776 F5F5 73F0
Rx Message 3:
packet_length 100 channel_id 1 packet_id 0 process id 0x1
0000:   0000 1CDE 3E48 3BC4 0000 0000 C0CC EC54 453E 3C3C
0020:   3F47 56F3 D1C7 C1BF C0C6 CEE1 6752 4A46 4648 4E59
0040:   6FE4 D6CF CDCE D2DA E57E 675E 5B5B 5E62 6B76 FCF6
0060:   F6FA 7D75 7373 7BF5 EAE1 DCDA DADD E6FE 6559 514D
0080:   4D4E 5563 EFD9 CDC8 C5C6 CAD1
Rx Message 4:
packet_length 100 channel_id 2 packet_id 0 process id 0x1
0000:   0000 4AC6 5F08 9181 0000 0000 DD5B DC5E E161 E468
0020:   FAFD 6CE1 5AD3 53D1 53D7 61EC EA59 CF4A C644 C344
0040:   CA4E D86C 60D0 48C2 3EBD 3CBD 3EC0 47CF 5976 DF4F
0060:   C945 C242 C146 C94E D668 73DB 54CE 4DCC 4DCE 53DB
0080:   64F9 ED63 DC59 DA58 DC5D E46C
Rx Message 5:
packet_length 100 channel_id 1 packet_id 0 process id 0x1
0000:   0000 1CDC 3E48 3B24 0000 0000 5B5B 5D62 6A76 FCF5
0020:   F5F9 7D78 7374 7CF5 EAE1 DDDA DBDD E7FE 6559 514E
0040:   4D4F 5663 EFD8 CDC8 C6C6 CAD1 E760 4E46 403F 4047
0060:   5173 D5C7 BFBC BCBE C5D4 6D4C 3F3B 3939 3D46 5ADB
0080:   C5BC B7B6 B8BD C8E8 4F3F 3835
Tx Message 0:
packet_length 100 channel_id 1 packet_id 0 process id 0x1
0000:   0000 4AC6 5F08 9181 0000 003C DD5B DC5E E161 E468
0020:   FAFD 6CE1 5AD3 53D1 53D7 61EC EA59 CF4A C644 C344
0040:   CA4E D86C 60D0 48C2 3EBD 3CBD 3EC0 47CF 5976 DF4F
0060:   C945 C242 C146 C94E D668 73DB 54CE 4DCC 4DCE 53DB
0080:   64F9 ED63 DC59 DA58 DC5D E46C
Tx Message 1:
packet_length 100 channel_id 2 packet_id 0 process id 0x1
0000:   0000 1CDC 3E48 3B24 0000 003C 5B5B 5D62 6A76 FCF5
0020:   F5F9 7D78 7374 7CF5 EAE1 DDDA DBDD E7FE 6559 514E
0040:   4D4F 5663 EFD8 CDC8 C6C6 CAD1 E760 4E46 403F 4047
0060:   5173 D5C7 BFBC BCBE C5D4 6D4C 3F3B 3939 3D46 5ADB
0080:   C5BC B7B6 B8BD C8E8 4F3F 3835
Tx Message 2:
packet_length 100 channel_id 1 packet_id 0 process id 0x1
0000:   0000 4AC7 5F08 91D1 0000 003C 7DF1 69E5 63E1 63E2
0020:   6E7C ED67 DE5D DB5C DC60 EC7E 6BE1 58D3 50CD 4DCE
0040:   50D2 5AE5 7868 DA52 CE4A C746 C647 C94B D25A EAF4
0060:   5DD7 4FCD 4ACA 4ACC 4FD3 5DE8 F769 DC58 D352 D253
0080:   D65B E573 6CDF 59D3 4ECF 4FD0
Tx Message 3:
packet_length 100 channel_id 2 packet_id 0 process id 0x1
0000:   0000 1CDD 3E48 3B74 0000 003C 3437 3D4C F0C8 BBB5
0020:   B2B3 B7BF D25B 4138 3331 3339 435F CFBD B6B2 B1B4
0040:   BBC8 7E48 3B34 3131 363D 4FDE C3B9 B3B1 B3B8 C2DB
0060:   533F 3833 3235 3B48 71CC BDB7 B4B5 B8BF CF67 483D
0080:   3836 383C 455B DAC6 BDB9 B9BB
Tx Message 4:
packet_length 100 channel_id 1 packet_id 0 process id 0x1
0000:   0000 4AC8 5F08 9221 0000 003C 54DA 61F5 EF60 DA53
0020:   CF4F CD4E D256 DB63 FCEE 5FDA 55D1 50CF 4FD3 56D8
0040:   5DE1 6E7C EC60 DC59 D655 D456 D85D DF6A F4F4 69E2
0060:   5CDD 5BDC 5BDE 61E9 6DF1 FF76 F16D E96A E566 EA6A
0080:   EB6F F16D EF79 F776 F5F5 73F0
Tx Message 5:
packet_length 100 channel_id 2 packet_id 0 process id 0x1
0000:   0000 1CDE 3E48 3BC4 0000 003C C0CC EC54 453E 3C3C
0020:   3F47 56F3 D1C7 C1BF C0C6 CEE1 6752 4A46 4648 4E59
0040:   6FE4 D6CF CDCE D2DA E57E 675E 5B5B 5E62 6B76 FCF6
0060:   F6FA 7D75 7373 7BF5 EAE1 DCDA DADD E6FE 6559 514D
0080:   4D4E 5563 EFD9 CDC8 C5C6 CAD1
Tx Message 6:
packet_length 100 channel_id 2 packet_id 0 process id 0x1
0000:   0000 1CDA 3E48 3A84 0000 003C E75F 4E46 403F 4147
```

```
0020:   5174 D5C7 BFBC BCBE C5D4 6C4C 3F3B 3939 3D46 5BDA
0040:   C5BC B7B6 B8BD C8E9 4F3F 3834 3437 3D4C EEC8 BBB5
0060:   B2B3 B8BF D35A 4138 3331 3339 435F CEBD B6B1 B1B4
0080:   BBC9 7C48 3B34 3131 363D 4FDE
Tx Message 7:
packet_length 100 channel_id 1 packet_id 0 process id 0x1
0000:   0000 4AC5 5F08 9131 0000 003C 66DE 66EB 67EE FE6E
0020:   F7E7 6B68 E068 EE6A DF5C DF62 EDF1 6FF2 7A78 67DC
0040:   5EDF 62E7 64E6 66E0 7071 EA69 F86E E260 DE5D E665
0060:   EB75 F0FB 6DE9 64E4 69E3 66EA 67E9 6DF9 F177 EC6E
0080:   EB6E F876 F875 7D6E E966 E05D
Tx Message 8:
packet_length 100 channel_id 2 packet_id 0 process id 0x1
0000:   0000 1CDB 3E48 3AD4 0000 003C C2B9 B3B1 B3B8 C2DC
0020:   523F 3733 3235 3C49 72CB BDB7 B4B5 B8BF CF67 483C
0040:   3836 373C 455C DAC6 BDB9 B9BB C0CC EE54 453E 3C3C
0060:   3F47 56F1 D1C7 C1BF C0C6 CEE1 6651 4A46 4648 4D59
0080:   70E3 D6CF CDCE D2D9 E67E 675E
Bootloader 1.8, Appn 3.1
Application firmware 3.1.8, Built by claux on Thu Jun 17 11:00:05 1999
VIC Interface Foreign Exchange Station 0/0, DSP instance (0x19543C0)
Singalling channel num 128 Signalling proxy 0x0 Signaling dsp 0x19543C0
tx outstanding 0, max tx outstanding 32
ptr 0x0, length 0x0, max length 0x0
dsp_number 0, Channel ID 1
received 0 packets, 0 bytes, 0 gaint packets
0 drops, 0 no buffers, 0 input errors 0 input overruns
650070 bytes output, 4976 frames output, 0 output errors, 0 output
underrun
0 unaligned frames
VIC Interface Foreign Exchange Station 0/1, DSP instance (0x1954604)
Singalling channel num 129 Signalling proxy 0x0 Signaling dsp 0x1954604
tx outstanding 0, max tx outstanding 32
ptr 0x0, length 0x0, max length 0x0
dsp_number 0, Channel ID 2
received 0 packets, 0 bytes, 0 gaint packets
0 drops, 0 no buffers, 0 input errors 0 input overruns
393976 bytes output, 3982 frames output, 0 output errors, 0 output
underrun
0 unaligned frames
```

Field descriptions are hardware-dependent and are meant for use by trained technical support.

Related Commands

Command	Description
show dial-peer voice	Displays configuration information and call statistics for dial peers.
show interface dspfarm	Displays hardware information including DRAM, SRAM, and the revision-level information on the line card.
show voice dsp	Displays the current status of all DSP voice channels.
show voice port	Displays configuration information about a specific voice port.

show crm

To display the carrier call capacities statistics, use the **show crm** command in privileged EXEC mode.

show crm

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

Both the **show trunk group**commandand the **show crm** command display values for the maximum number of calls. These values originate from different configuration procedures:

- In the **show trunk group** command, the Max Calls value originates from the **max-calls** command in the trunk group configuration.
- In the **show crm** command, Max calls indicates the maximum number of available channels after the carrier ID or trunk group label is assigned to an interface using the **trunk-group** (interface) command.

Examples

The following example illustrates the carrier call capacities statistics:

```
Router# show crm
Carrier:1411
    Max calls:4
    Max Voice (in) :      4    Cur Voice (in) :      0
    Max Voice (out):      4    Cur Voice (out):      0
    Max Data (in)  :      4    Cur Data (in)  :      0
    Max Data (out) :      4    Cur Data (out) :      0
Trunk Group Label: 100
    Max calls:6
    Max Voice (in) :      6    Cur Voice (in) :      0
    Max Voice (out):      6    Cur Voice (out):      0
    Max Data (in)  :      6    Cur Data (in)  :      0
    Max Data (out) :      6    Cur Data (out) :      0
```

The table below describes the fields shown in this output, in alphabetical order.

Table 27: show crm Field Descriptions

Field	Description
Carrier	ID of the carrier that handles the calls.
Cur Data (in)	Current number of incoming data calls that are handled by the carrier or trunk group.
Cur Data (out)	Current number of outgoing data calls that are handled by the carrier or trunk group.
Cur Voice (in)	Current number of incoming voice calls that are handled by the carrier or trunk group.
Cur Voice (out)	Current number of outgoing voice calls that are handled by the carrier or trunk group.
Max Calls	Maximum number of calls that are handled by the carrier or trunk group.
Max Data (in)	Maximum number of incoming data calls that are handled by the carrier or trunk group.
Max Data (out)	Maximum number of outgoing data calls that are handled by the carrier or trunk group.
Max Voice (in)	Maximum number of incoming voice calls that are handled by the carrier or trunk group.
Max Voice (out)	Maximum number of outgoing voice calls that are handled by the carrier or trunk group.
Trunk Group Label	Label of the trunk group that handles the calls.

Related Commands

Command	Description
carrier-id (dial-peer)	Specifies the carrier associated with VoIP calls.
max-calls	Specifies the maximum number of calls handled by a trunk group.
show trunk group	Displays the configuration parameters for one or more trunk groups.
trunk-group (interface)	Assigns an interface to a trunk group.
trunk-group-label (dial-peer)	Specifies the trunk group associated with VoIP calls.

show csm

To display the call switching module (CSM) statistics for a particular digital signal processor (DSP) channel, all DSP channels, or a specific modem or DSP channel, use the **show csm**command in privileged EXEC mode.

Cisco AS5300 Universal Access Server

show csm {**call-rate [table]| callre-source**| **modem** [*slot*/*port*| **group** *modem-group-number*]| **signaling-channel**}

Cisco AS5400Series Router

show csm {**call rate [table]| call-resource**| **modem** [*slotport*| **group** *modem-group-number*]**| signaling-channel**| **voice** *slot*/*port*}

Syntax Description

call-rate	Displays the incoming and outgoing call switching rate.
table	(Optional) Displays the incoming and outgoing call switching rate in the form of numerical table.
call-resource	Displays statistics about the CSM call resource.
modem	Displays CSM call statistics for modems.
slot / *port*	(Optional) Location (and thereby identity) of a specific modem.
group	(Optional) Displays modem group information.
modem -group-number	(Optional) Location of a particular dial peer. Range: 1 to 32767.
signaling-channel	Displays CSM signaling channel Information.
voice	Displays CSM call statistics for DSP channels.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
11.3 NA	This command was introduced.

Release	Modification
12.0(3)T	This command was modified. Port-specific values for the Cisco AS5300 were added.
12.0(7)T	This command was modified. Port-specific values for the Cisco AS5800 were added.

Usage Guidelines

This command shows the information related to CSM, which includes the DSP channel, the start time of the call, the end time of the call, and the channel on the controller used by the call.

Use the **show csm modem** command to display the CSM call statistics for a specific modem, for a group of modems, or for all modems. If a *slot* / *port* argument is specified, then CSM call statistics are displayed for the specified modem. If the *modem-group-number*argument is specified, the CSM call statistics for all of the modems associated with that modem group are displayed. If no keyword is specified, CSM call statistics for all modems on the Cisco AS5300 universal access server are displayed.

Use the **show csm voice** command to display CSM statistics for a particular DSP channel. If the *slot* / *dspm* / *dsp* / *dsp-channel* or *shelf* / *slot* / *port*argumentis specified, the CSM call statistics for calls using the identified DSP channel are displayed. If no argument is specified, all CSM call statistics for all DSP channels are displayed.

Examples

The following is sample output from the **show csm** command for the Cisco AS5300 universal access server:

```
Router# show csm voice 2/4/4/0
 slot 2, dspm 4, dsp 4, dsp channel 0,
 slot 2, port 56, tone, device_status(0x0002): VDEV_STATUS_ACTIVE_CALL.
csm_state(0x0406)=CSM_OC6_CONNECTED, csm_event_proc=0x600E2678, current call thru PRI line
invalid_event_count=0, wdt_timeout_count=0
wdt_timestamp_started is not activated
wait_for_dialing:False, wait_for_bchan:False
pri_chnl=TDM_PRI_STREAM(s0, u0, c22), tdm_chnl=TDM_DSP_STREAM(s2, c27)
dchan_idb_start_index=0, dchan_idb_index=0, call_id=0xA003, bchan_num=22
csm_event=CSM_EVENT_ISDN_CONNECTED, cause=0x0000
ring_no_answer=0, ic_failure=0, ic_complete=0
dial_failure=0, oc_failure=0, oc_complete=3
oc_busy=0, oc_no_dial_tone=0, oc_dial_timeout=0
remote_link_disc=0, stat_busyout=0
oobp_failure=0
call_duration_started=00:06:53, call_duration_ended=00:00:00, total_call_duration=00:00:44
The calling party phone number = 408
The called party phone number  = 5271086
total_free_rbs_timeslot = 0, total_busy_rbs_timeslot = 0, total_dynamic_busy_rbs_timeslot
= 0, total_static_busy_rbs_timeslot = 0,
total_sw56_rbs_timeslot = 0, total_sw56_rbs_static_bo_ts = 0,
total_free_isdn_channels = 21, total_busy_isdn_channels = 0,total_auto_busy_isdn_channels
= 0,
min_free_device_threshold = 0
```

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

Table 28: show csm voice Field Descriptions

Field	Description
slot	Slot where the VFC resides.

Field	Description
dsp	DSP through which this call is established.
slot/port	Logical port number for the device. This is equivalent to the DSP channel number. The port number is derived as follows: • (max_number_of_dsp_channels per dspm=12) * the dspm # (0-based) + • (max_number_of_dsp_channels per dsp=2) * the dsp # (0-based) + the dsp channel number (0-based).
tone	Which signaling tone is being used (DTMF, MF, R2). This only applies to CAS calls. Possible values are as follows: • mf • dtmf • r2-compelled • r2-semi-compelled • r2-non-compelled

Field	Description
device_status	Status of the device. Possible values are as follows: • VDEV_STATUS_UNLOCKED--Device is unlocked (meaning that it is available for new calls). • VDEV_STATUS_ACTIVE_WDT--Device is allocated for a call and the watchdog timer is set to time the connection response from the central office. • VDEV_STATUS_ACTIVE_CALL--Device is engaged in an active, connected call. • VDEV_STATUS_BUSYOUT_REQ--Device is requested to busyout; does not apply to voice devices. • VDEV_STATUS_BAD--Device is marked as bad and not usable for processing calls. • VDEV_STATUS_BACK2BACK_TEST--Modem is performing back-to-back testing (for modem calls only). • VDEV_STATUS_RESET--Modem needs to be reset (for modem only). • VDEV_STATUS_DOWNLOAD_FILE--Modem is downloading a file (for modem only). • VDEV_STATUS_DOWNLOAD_FAIL--Modem has failed during downloading a file (for modem only). • VDEV_STATUS_SHUTDOWN--Modem is not powered up (for modem only). • VDEV_STATUS_BUSY--Modem is busy (for modem only). • VDEV_STATUS_DOWNLOAD_REQ--Modem is requesting connection (for modem only).

Field	Description
csm_state	

Field	Description
	CSM call state of the current call (PRI line) associated with this device. Possible values are as follows: • CSM_IDLE_STATE--Device is idle. • CSM_IC_STATE--A device has been assigned to an incoming call. • CSM_IC1_COLLECT_ADDR_INFO--A device has been selected to perform ANI/DNIS address collection for this call. ANI/DNIS address information collection is in progress. The ANI/DNIS is used to decide whether the call should be processed by a modem or a voice DSP. • CSM_IC2_RINGING--The device assigned to this incoming call has been told to get ready for the call. • CSM_IC3_WAIT_FOR_SWITCH_OVER--A new device is selected to take over this incoming call from the device collecting the ANI/DNIS address information. • CSM_IC4_WAIT_FOR_CARRIER--This call is waiting for the CONNECT message from the carrier. • CSM_IC5_CONNECTED--This incoming call is connected to the central office. • CSM_IC6_DISCONNECTING--This incoming call is waiting for a DISCONNECT message from the VTSP module to complete the disconnect process. • CSM_OC_STATE --An outgoing call is initiated. • CSM_OC1_REQUEST_DIGIT--The device is requesting the first digit for the dial-out number. • CSM_OC2_COLLECT_1ST_DIGIT--The first digit for the dial-out number has been collected. • CSM_OC3_COLLECT_ALL_DIGIT--All the digits for the dial-out number have been collected. • CSM_OC4_DIALING--This call is waiting for a dsx0 (B channel) to be available for dialing out. • CSM_OC5_WAIT_FOR_CARRIER--This

Field	Description
	(outgoing) call is waiting for the central office to connect. • CSM_OC6_CONNECTED--This (outgoing) call is connected. • CSM_OC7_BUSY_ERROR--A busy tone has been sent to the device (for VoIP call, no busy tone is sent; just a DISCONNECT INDICATION message is sent to the VTSP module), and this call is waiting for a DISCONNECT message from the VTSP module (or ONHOOK message from the modem) to complete the disconnect process. • CSM_OC8_DISCONNECTING--The central office has disconnected this (outgoing) call, and the call is waiting for a DISCONNECT message from the VTSP module to complete the disconnect process.
csm_state: invalid_event_count	Number of invalid events received by the CSM state machine.
wdt_timeout_count	Number of times the watchdog timer is activated for this call.
wdt_timestamp_started	Whether the watchdog timer is activated for this call.
wait_for_dialing	Whether this (outgoing) call is waiting for a free digit collector to become available to dial out the outgoing digits.
wait_for_bchan	Whether this (outgoing) call is waiting for a B channel to send the call out on.
pri_chnl	Which type of TDM stream is used for the PRI connection. For PRI and CAS calls, it is always TDM_PRI_STREAM.
tdm_chnl	Which type of TDM stream is used for the connection to the device used to process this call. In the case of a VoIP call, this is always set to TDM_DSP_STREAM.
dchan_idb_start_index	First index to use when searching for the next IDB of a free D channel.
dchan_idb_index	Index of the currently available IDB of a free D channel.

Field	Description
csm_event	Event just passed to the CSM state machine.
cause	Event cause.
ring_no_answer	Number of times a call failed because there was no response.
ic_failure	Number of failed incoming calls.
ic_complete	Number of successful incoming calls.
dial_failure	Number of times a connection failed because there was no dial tone.
oc_failure	Number of failed outgoing calls.
oc_complete	Number of successful outgoing calls.
oc_busy	Number of outgoing calls whose connection failed because there was a busy signal.
oc_no_dial_tone	Number of outgoing calls whose connection failed because there was no dial tone.
oc_dial_timeout	Number of outgoing calls whose connection failed because the timeout value was exceeded.
call_duration_started	Start of this call.
call_duration_ended	End of this call.
total_call_duration	Duration of this call.
The calling party phone number	Calling party number as given to CSM by ISDN.
The called party phone number	Called party number as given to CSM by ISDN.
total_free_rbs_time slot	Total number of free RBS (CAS) time slots available for the whole system.
total_busy_rbs_time slot	Total number of RBS (CAS) time slots that have been busied-out. This includes both dynamically and statically busied out RBS time slots.
total_dynamic_busy_rbs_time slot	Total number of RBS (CAS) time slots that have been dynamically busied out.

Field	Description
total_static_busy_rbs_time slot	Total number of RBS (CAS) time slots that have been statically busied out (that is, they are busied out using the CLI command).
total_free_isdn_channels	Total number of free ISDN channels.
total_busy_isdn_channels	Total number of busy ISDN channels.
total_auto_busy_isdn_channels	Total number of ISDN channels that are automatically busied out.

Related Commands

Command	Description
show call active voice	Displays the contents of the active call table.
show call history voice	Displays the contents of the call history table.
show num-exp	Displays how number expansions are configured.
show voice port	Displays configuration information about a specific voice port.

show csm call

To view the call switching module (CSM) call statistics, use the **show csm call** command in privileged EXEC mode

show csm call {**failed**| **rate**| **total**}

Syntax Description

failed	CSM call fail/reject rate for the last 60 seconds, 60 minutes, and 72 hours.
rate	CSM call rate for the last 60 seconds, 60 minutes, and 72 hours.
total	Total number of CSM calls for the last 60 seconds, 60 minutes, and 72 hours.

Command Default

No default behavior or values.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.3(2)T	This command was introduced on the Cisco AS5850.

Usage Guidelines

Use this command to understand CSM call volume.

Examples

The following examples show the CSM call statistics for the last 60 seconds:

```
Router# show csm call rate

  15
  14
  13
  12
  11
  10
   9
   8
   7
   6
   5
   4
   3
   2
```

```
  1
    0....5....1....1....2....2....3....3....4....4....5....5....
              0    5    0    5    0    5    0    5    0    5
          CSM call switching rate per second (last 60 seconds)
          # = calls entering the module per second
```

```
Router# show csm call failed
```

```
 15
 14
 13
 12
 11
 10
  9
  8
  7
  6
  5
  4
  3
  2
  1
    0....5....1....1....2....2....3....3....4....4....5....5....
              0    5    0    5    0    5    0    5    0    5
        CSM call fail/reject rate per second (last 60 seconds)
          # = calls failing per second
```

```
Router# sh csm call total
```

```
1344
1244
1144
1044
 944
 844
 744
 644
 544
 444
 344
 244
 144
  44
    0....5....1....1....2....2....3....3....4....4....5....5....
              0    5    0    5    0    5    0    5    0    5
                      CSM total calls (last 60 seconds)
```

 # = number of calls

Field descriptions should be self-explanatory.

show cube status

To display the Cisco Unified Border Element (Cisco UBE) status, the software version, the license capacity, the image version, and the platform name of the device, use the **show cube status** command in user EXEC or privileged EXEC mode.

show cube status

Syntax Description

This command has no arguments or keywords.

Command Default

Cisco UBE status is not displayed.

Command Modes

User EXEC (>)

Privileged EXEC (#)

Command History

Release	Modification
15.2(1)T	This command was introduced.
15.1(3)S1	This command was modified. The output was modified to have only token characters (an alphanumeric character, hyphen [-], dot [.], exclamation mark [!], percent [%], asterisk [*], underscore [_], plus sign [+], grave [`], apostrophe ['], or a tilde [~]) in server and user-agent Session Initiation Protocol (SIP) headers. The nontoken characters present in the image name is replaced by a dot[.].

Usage Guidelines

The display of Cisco UBE status-related information is supported by the implementation of the CISCO-UBE-MIB. This MIB also provides Simple Network Management Protocol (SNMP) support for the Cisco UBE status:

The Cisco UBE status display is enabled only if the **mode border-element** command is configured with call license capacity. The **show cube status** command displays the following message if the license capacity is not configured.

```
Cisco Unified Border Element (CUBE) application is not enabled
```

Examples

The following example configures the **mode border-element** command with call license capacity and enables the display of Cisco UBE status on the Cisco 3845 router:

```
Device(config)# voice service voip
Device(conf-voi-serv)# mode border-element license capacity 200
```

After saving the configuration and reloading the device:

```
Device> show cube status
```

```
CUBE-Version : 8.8
SW-Version : 15.2(1)T, Platform 3845
HA-Type : none
Licensed-Capacity : 200
```

In Cisco IOS Release 15.1(3)S1 and later releases, the output is as follows:

```
Device> show cube status

CUBE-Version : 8.8
SW-Version : 15.2.1.T, Platform 3845
HA-Type : none
Licensed-Capacity : 200
```

The table below describes the fields shown in the display.

Table 29: show cube status Field Descriptions

Field	Description
CUBE-Version	Version of the Cisco UBE application running on the device.
SW-Version	Image version and platform name of the device running the Cisco UBE application. This matches the image version and platform name returned by the **show version** command.
HA-Type	The type of High Availability (HA) feature configured and running on the device. The following HA types are supported: • none: Cisco UBE does not support HA. • cold-standby-chassis-to-chassis: Device-to-device cold standby support. • hot-standby-chassis-to-chassis: Device-to-device hot standby support.
Licensed-Capacity	Number of SIP call legs that Cisco UBE is licensed to use. The range is from 0 to 999999. This number matches the number of licenses configured using the **mode border-element license capacity** command. **Note** The number of SIP call legs that Cisco UBE can use is platform-dependent and is not affected by the specified value for the **capacity** keyword in Cisco IOS Release 15.2(1)T.

Related Commands

Command	Description
mode border-element	Enables the set of commands used in the border-element configuration on the Cisco 2900 and Cisco 3900 series platforms.

show debug condition

To display the debugging filters that have been enabled for VoiceXML applications, ATM-enabled interfaces, or Frame Relay interfaces, use the **show debug condition** command in privileged EXEC mode.

show debug condition

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.2(11)T	This command was introduced on the Cisco 3640, Cisco 3660, Cisco AS5300, Cisco AS5350, and Cisco AS5400.
12.0(28)S	This command was integrated into Cisco IOS Release 12.0(28)S and was enhanced to include debugging for ATM-enabled and Frame Relay-enabled interfaces.
12.2(25)S	This command was integrated into Cisco IOS Release 12.2(25)S.
12.2(27)SBC	This command was integrated into Cisco IOS Release 12.2(27)SBC.
12.2(28)SB	This command was integrated into Cisco IOS Release 12.2(28)SB.
12.4(9)T	This command was enhanced to include debugging for ATM-enabled and Frame Relay-enabled interfaces.

Usage Guidelines

This command displays the debugging filter conditions that have been set for VoiceXML applications by using the **debug condition application voice** command.

Examples

The following is sample output from this command when it is used with the VoiceXML application:

```
Router# show debug condition
Condition 1: application voice vmail (1 flags triggered)
       Flags: vmail
Condition 2: application voice myapp1 (1 flags triggered)
       Flags: myapp1
```

The following is sample output from this command when an ATM interface is being debugged:

```
Router# show debug condition

Condition 1: atm-vc 0/56784 AT2/0 (0 flags triggered)
Condition 2: atm-vc 255/45546 AT2/0 (0 flags triggered)
Condition 3: atm-vc 0/266 AT6/0 (1 flags triggered)
```

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

Table 30: show debug condition Field Descriptions

Field	Description
Condition 1	Sequential number identifying the filter condition that was set for the specified command.
Flags	Name of the voice application for which the condition was set.
at2/0	Interface number of the ATM interface that has the debug condition applied.
atm-vc 0/56784	Virtual channel identifier (VCI). Alternatively, virtual path identifier/virtual channel identifier (VCI/VPI) pair.

Related Commands

Command	Description
debug condition application voice	Filters out debugging messages for all VoiceXML applications except the specified application.
debug http client	Displays debugging messages for the HTTP client.
debug vxml	Displays debugging messages for VoiceXML features.