show voice trace through shutdown (voice-port)

show voice trace

To display the call trace information about a specified port, use the **show voice trace** command in privileged EXEC mode.

show voice trace *interface-slot* **[detail]**

Syntax Description

interface-slot	Voice interface slot.
detail	(Optional) Displays detailed statistics of the specified port.

Command Default

Privileged EXEC (#)

Command History

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

Usage Guidelines

Use the **show voice trace**commandto display the call trace information about specified port. The field descriptions are self-explanatory.

Examples

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

```
Router# show voice trace 1/1/1 detail

1/1/1 Stack 0:
State Transitions: timestamp (state, event) -> (state, event) ...
96.732 (S_OPEN_PEND, E_DSP_INTERFACE_INFO) ->
96.732 (S_DOWN, E_HTSP_IF_INSERVICE) ->
97.092 (S_OPEN_PEND, E_HTSP_GO_UP) ->
Event Counts (zeros not shown): (event, count)
(E_HTSP_IF_INSERVICE, 1) :(E_HTSP_GO_UP, 1) :(E_DSP_INTERFACE_INFO, 1) :
State Counts (zeros not shown): (state, count)
(S_OPEN_PEND, 2) :(S_DOWN, 1) :
Stack 1:
State Transitions: timestamp (state, event) -> (state, event) ...
97.092 (DID_NULL, E_DSP_SIG_0100) ->
97.092 (DID_INIT, E_HTSP_INSERVE) ->
97.092 (DID_PENDING, E_DSP_SIG_0100) ->
Event Counts (zeros not shown): (event, count)
(E_HTSP_INIT, 1) :(E_HTSP_INSERVE, 1) :(E_DSP_SIG_0100, 2) :
State Counts (zeros not shown): (state, count)
(DID_NULL, 2) :(DID_INIT, 1) :(DID_PENDING, 1) :
```

show voice translation-profile

To display one or more translation profiles, use the **show voice translation-profile** command in privileged EXEC mode.

show voice translation-profile [*name*| **sort** [**ascending**| **descending**]]

Syntax Description

name	Name of the translation profile to display.
sort [**ascending** | **descending**	Display order of the translation profiles by *name*.

Command Default

Ascending order

Command Modes

Privileged EXEC (#)

Command History

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

Examples

The following sample output displays all the voice translation profiles in ascending order:

```
Router# show voice translation-profile sort ascending
Translation Profile: 1
 Rule for Calling number:
 Rule for Called number: 1
 Rule for Redirect number:
Translation Profile: 2
 Rule for Calling number:1
 Rule for Called number: 2
 Rule for Redirect number:
Translation Profile: 6
 Rule for Calling number:1
 Rule for Called number: 6
 Rule for Redirect number:2
```

The table below describes the fields shown in this output.

Table 1: show voice translation-profile Field Descriptions

Field	Description
Translation Profile	Name of the translation profile.
Rule for Called number	Number of the rule used for translating called numbers. If the field is blank, this translation profile does not have a rule assigned to that number type.

Field	Description
Rule for Calling number	Number of the rule used for translating calling numbers. If the field is blank, this translation profile does not have a rule assigned to that number type.
Rule for Redirect number	Number of the rule used for translating redirect numbers. If the field is blank, this translation profile does not have a rule assigned to that number type.

Related Commands

Command	Description
voice translation-profile	Initiates a voice translation-profile definition.
voice translation-rule	Initiates a voice translation-rule definition.

show voice translation-rule

To display one or more translation rules, use the **show voice translation-rule** command in privileged EXEC mode.

show voice translation-rule [*number*| **sort** [**ascending**| **descending**]]

Syntax Description

number	Number of the translation rule to display. Valid values are from 1 to 2147483647.
sort [**ascending** | **descending**	Display order of the translation rules by *number*.

Command Default

Ascending order

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

Under each translation rule are numbered subrules.

Examples

The following sample output displays the translation rule number 6:

```
Router# show voice translation-rule 6
Translation-rule tag: 6
   Rule 1:
   Match pattern: 65088801..
   Replace pattern: 6508880101
   Match type: none   Replace type: none
   Match plan: none   Replace plan: none
```

The following sample output displays all the translation rules in ascending order:

```
Router# show voice translation-rule sort ascending
Translation-rule tag: 1
   Rule 3:
   Match pattern: 5108880...
   Replace pattern: 5108880101
   Match type: none   Replace type: none
   Match plan: none   Replace plan: none
   Rule 4:
   Match pattern: 510890....
   Replace pattern: 5108880101
   Match type: none   Replace type: none
   Match plan: none   Replace plan: none
Translation-rule tag: 2
```

```
Rule 1:
Match pattern: 51088802..
Replace pattern: 5108880101
Match type: none   Replace type: none
Match plan: none   Replace plan: none
Rule 2:
Match pattern: 51088803..
Replace pattern: 5108880101
Match type: none   Replace type: none
Match plan: none   Replace plan: none
Rule 3:
Match pattern: 510889....
Replace pattern: 5108880101
Match type: none  Replace type: none
Match plan: none   Replace plan: none
Rule 4:
Match pattern: 510890....
Replace pattern: 5108880101
Match type: none   Replace type: none
Match plan: none   Replace plan: none
```

The table below describes the fields shown in this output.

Table 2: show voice translation-rule Field Descriptions

Field	Description
Translation-rule tag	Number of the translation rule.
Rule	Number of the rule defined within the translation rule.
Match pattern	SED-like expression used to match incoming call information.
Replace pattern	SED-like expression used to replace *match-pattern* in the call information.
Match type	Type of incoming calls to match.
Replace type	Type to replace Match type.
Match plan	Plan of incoming calls to match.
Replace plan	Plan to replace Match plan.

Related Commands

Command	Description
rule (voice translation-rule)	Defines the SED expressions for translating calls.
test voice translation-rule	Tests the rules in a translation-rule definition.
voice translation-rule	Initiates a voice translation-rule definition.
voice translation-profile	Initiates a voice translation-profile definition.

show voice trunk-conditioning signaling

To display the status of trunk-conditioning signaling and timing parameters for a voice port, use the **show voice trunk-conditioning signaling** command in user EXEC or privileged EXEC mode.

show voice trunk-conditioning signaling [**summary**| *voice-port*]

Syntax Description

summary	(Optional) Displays a summary of the status for all voice ports on the router or concentrator.
voice -port	(Optional) Displays a detailed report for a specified voice port.

Command Modes

User EXEC (>) Privileged EXEC (#)

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810 as the **show voice permanent-call** command.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.0(7)XK	This command was renamed **show voice trunk-conditioning signaling**.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.1(3)T	This command was implemented on the Cisco 2600 series and Cisco 3600 series.

Usage Guidelines

This command displays the trunk signaling status for analog and digital voice ports on the Cisco 2600 series and the Cisco 3600 series routers.

Examples

The following is sample output from the **show voice trunk-conditioning signaling summary** command:

```
Router# show voice trunk-conditioning signaling summary
2/0/0 is shutdown
2/0/1 is shutdown
3/0:0 8  is shutdown
3/0:1 1  is shutdown
3/0:2 2  is shutdown
3/0:3 3  is shutdown
3/0:5 5  is shutdown
3/0:6(6) :
 status :
```

```
3/0:7 7  is shutdown
3/1:0 8  is shutdown
3/1:1 1  is shutdown
3/1:3 3  is shutdown
3/1:5 5  is shutdown
3/1:7 7  is shutdown
```

The following is sample output from the **show voice trunk-conditioning signaling** command for voice port 3/0:6:

```
Router# show voice trunk-conditioning signaling 3/0:6
hardware-state ACTIVE signal type is NorthamericanCAS
status :
forced playout pattern = STOPPED
trunk_down_timer = 0, rx_ais_duration = 0, idle_timer = 0
```

The table below describes significant fields in these outputs.

Table 3: show voice trunk-conditioning signaling Field Descriptions

Field	Description
current timer	Time since last signaling packets were received.
forced playout pattern	Which forced playout pattern is sent to PBX: • 0 = no forced playout pattern is sent • 1 = receive IDLE playout pattern is sent • 2 = receive OOS playout pattern is sent
hardware-state	Hardware state based on received IDLE pattern: • IDLE = both sides are idle • ACTIVE = at least one side is active
signal type	Signaling type used by lower level driver: northamerica, melcas, transparent, or external.
idle timer	Time the hardware on both sides has been in idle state.
last-ABCD	Last received or transmitted signal bit pattern.
max inter-arrival time	Maximum interval between received signaling packets.
missing	Number of missed signal packets.
mode	Signaling packet generation frequency: • Fast mode = every 4 milliseconds • Slow mode = same frequency as keepalive timer
out of seq	Number of out-of-sequence signal packets.

Field	Description
playout depth	Number of packets in playout buffer.
prev-seq#	Sequence number of previous signaling packet.
refill count	Number of packets created to maintain nominal length of playout packet buffer.
rx_ais_duration	Time since receipt of AIS indicator.
seq#	Sequence number of signaling packet.
sig pkt cnt	Number of transmitted or received signaling packets.
signal path	Status of signaling path.
signaling playout history	Signaling bits received in last 60 milliseconds.
trunk_down_timer	Time since last signaling packets were received.
tx_oos_timer	Time since PBX started sending OOS signaling pattern defined by **signal pattern oos transmit**.
very late	Number of very late signaling packets.

Related Commands

Command	Description
show dial-peer voice	Displays the configuration for all VoIP and POTS dial peers configured on the router.
show voice dsp	Shows the current status of all DSP voice channels.
show voice port	Displays configuration information about a specific voice port.
show voice trunk-conditioning supervisory	Displays the status of trunk supervision and configuration parameters for voice ports.

show voice trunk-conditioning supervisory

To display the status of trunk supervision and configuration parameters for a voice port, use the **show voice trunk-conditioning supervisory** command in user EXEC or privileged EXEC mode.

show voice trunk-conditioning supervisory [**summary**| *voice-port*]

Syntax Description

summary	(Optional) Displays a summary of the status for all voice ports on the router or concentrator.
voice -port	(Optional) Detailed report for a specified voice port.

Command Modes

User EXEC (>) Privileged EXEC (#)

Command History

Release	Modification
12.0(7)XK	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810 platforms.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.1(3)T	This command was implemented on the Cisco 2600 series and Cisco 3600 series.
12.4(15)T10	The output of thiscommand was modified to report values configured by the **signal timing idle suppress-voice** command. The values for the **suppress-voice** and **resume-voice** keywords are shown as the "idle = *seconds*" and "idle_off = *milliseconds*" fields, respectively.

Usage Guidelines

This command displays the trunk supervision and configuration status for analog and digital voice ports.

Examples

The following is sample output from the **show voice trunk-conditioning supervisory summary** command for all voice ports:

```
Router# show voice trunk-conditioning supervisory summary
2/0/0 is shutdown
2/0/1 is shutdown
3/0:0 8  is shutdown
3/0:1 1  is shutdown
3/0:2 2  is shutdown
3/0:3 3  is shutdown
3/0:5 5  is shutdown
3/0:6(6) : state : TRUNK_SC_CONNECT, voice : on , signal : on ,master
3/0:7(7) : state : TRUNK_SC_CONNECT, voice : on , signal : on ,master
3/1:0(8) : state : TRUNK_SC_CONNECT, voice : on , signal : on ,master
3/1:1(1) : state : TRUNK_SC_CONNECT, voice : on , signal : on ,master
```

```
3/1:3(3) : state : TRUNK_SC_CONNECT, voice : on , signal : on ,master
3/1:5(5) is shutdown
3/1:7(7) is shutdown
```

The following is sample output from the **show voice trunk-conditioning supervisory** command for voice port 3/0:6:

```
Router# show voice trunk-conditioning supervisory 3/0:6
3/0:6(6) : state : TRUNK_SC_CONNECT, voice : on, signal : on, master
 status: trunk connected
 sequence oos : idle and oos
 pattern :rx_idle = 0x0 rx_oos = 0xF
 timing : idle = 0, restart = 0, standby = 0, timeout = 40
 supp_all = 0, supp_voice = 0, keep_alive = 5
 timer: oos_ais_timer = 0, timer = 0
```

The following shows a sample trunk conditioning setting for the **voice class permanent** command and sample output from the **s how voice trunk-conditioning supervisory** command that shows the values for the timeout timing field:

```
!
voice class permanent 1
  signal pattern idle transmit 0101
  signal pattern idle receive 0101
  signal pattern oos transmit 1111
  signal pattern oos receive 0101
  signal timing idle suppress-voice 10 resume-voice 150
!
Router# show voice trunk-conditioning supervisory

SLOW SCAN
0/0/0:0(1) : state : TRUNK_SC_CONNECT, voice : off , signal : on ,slave
 status: rcv IDLE, trunk connected
 sequence oos : idle and oos
 pattern :rx_idle = 0101 rx_oos = 0101 tx_idle = 0101 tx_oos =  1111
 timeout timing : idle = 10, idle_off = 150, restart = 0, standby = 0, timeout = 30
 supp_all = 0, supp_voice = 0, keep_alive = 5
 timer: oos_ais_timer = 0, timer = 0
```

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

Table 4: show voice trunk-conditioning supervisory Field Descriptions

Field	Description
idle	Timer setting (in seconds) configured by the **suppress-voice** option of the **signal timing idle suppress-voice** command.
idle_off	Timer setting (in milliseconds) configured by the **resume-voice** option of the **signal timing idle suppress-voice** command.
keep_alive	Signaling packets periodically sent to the far end, even if there is no signal change. These signaling packets function as keep alive messages.
master	Voice port configured as "connect trunk *xxxx* ."
oos_ais_timer	Time since the signaling packet with alarm indication signal (AIS) indicator was received.

Field	Description
pattern	4-bit signaling pattern.
restart	Restart timeout after far end is out-of-service (OOS).
rx-idle	Signaling bit pattern indicating that the far end is idle.
rx-oos	Signaling bit pattern sent to the PBX indicating that the network is OOS.
standby	Time before the slave side goes back to standby after the far end goes OOS.
supp_all	Timeout before suppressing transmission of voice and signaling packets to the far end after detection of PBX OOS.
supp_voice	Timeout before suppressing transmission of voice packet to the far end after detection of PBX OOS.
timeout	Timeout for nonreceipt of keepalive packets before the far end is considered to be OOS.
timeout timing	Delay between the detection of incoming seizure and when the digital signal processor (DSP)-to-Cisco IOS interaction to open up the audio path is initiated.
TRUNK_SC_CONNECT	Trunk conditioning supervisory component status.

Related Commands

Command	Description
show dial-peer voice	Displays the configuration for all VoIP and POTS dial peers configured on the router.
show voice dsp	Displays the current status of all DSP voice channels.
show voice port	Displays configuration information about a specific voice port.
show voice trunk-conditioning signaling	Displays the status of trunk-conditioning signaling and timing parameters for a voice port.
voice-class permanent	Assigns a previously configured voice class for a Cisco trunk or FRF.11 trunk to a voice port.

show voice vtsp

To display information about the voice port configuration and Voice Telephony Service Provider (VTSP), use the **show voice vtsp** command in privileged EXEC mode.

show voice vtsp {**call** [**dspstats**| **fsm**| **log** [*call-ID*]| **verbose**]| **fork dsp-status**} [*call ID*]

Syntax Description

call	Displays the call control block information.
dspstats	(Optional) Displays the selective statistics of digital signal processor (DSP) voice channels.
fsm	(Optional) Displays information about the Finite State Machine Dump (FSM).
log *call-ID*	(Optional) Displays the call related logs. If a call ID is specified, this command displays the status of a specific call. The call ID value range is from 1 to 4294967295
verbose	(Optional) Displays the verbose output.
fork	Displays the media forking information.
dsp-status	Displays the status of media forking in the DSP.
call-ID	(Optional) Displays the status of the call. The value range is from 0x0 to 0xFFFFFFFF. >

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.4(24)T	This command was introduced in a release earlier than Cisco IOS Release 12.4(24)T.

Usage Guidelines

Use the **show voice vtsp** command to display information about the voice port configuration.

Examples

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

```
Router# show voice vtsp call dspstats 0x833
```

```
***DSP VOICE TX STATISTICS***
Tx Vox/Fax Pkts: 1337, Tx Sig Pkts: 0, Tx Comfort Pkts: 181
Tx Dur(ms): 46840, Tx Vox Dur(ms): 26740, Tx Fax Dur(ms): 0
        ***DSP VOICE RX STATISTICS***
Rx Vox/Fax Pkts: 1347, Rx Signal Pkts: 0, Rx Comfort Pkts: 180
Rx Dur(ms): 46840, Rx Vox Dur(ms): 23300, Rx Fax Dur(ms): 0
Rx Non-seq Pkts: 0, Rx Bad Hdr Pkts: 0
Rx Early Pkts: 0, Rx Late Pkts: 0
        ***DSP VOICE VP_DELAY STATISTICS***
Clk Offset(ms): 80, Rx Delay Est(ms): 50
Rx Delay Lo Water Mark(ms): 50, Rx Delay Hi Water Mark(ms): 70
        ***DSP VOICE VP_ERROR STATISTICS***
Predict Conceal(ms): 0, Interpolate Conceal(ms): 0
Silence Conceal(ms): 0, Retroact Mem Update(ms): 0
Buf Overflow Discard(ms): 0, Talkspurt Endpoint Detect Err: 0
        ***DSP LEVELS***
TDM Bus Levels(dBm0): Rx -68.5 from PBX/Phone, Tx -4.4 to PBX/Phone
TDM ACOM Levels(dBm0): +64.1, TDM ERL Level(dBm0): +10.0
TDM Bgd Levels(dBm0): -80.0, with activity being silence
        ***DSP VOICE ERROR STATISTICS***
Rx Pkt Drops(Invalid Header): 0, Tx Pkt Drops(HPI SAM Overflow): 0
        ***DSP VOICE GSMAMR-NB STATISTICS***
EncodingRate: 7 DecodingRate: 7
numEncodeChanges: 0 numDecodeChanges: 0
numCRCFail: 0 numFrameBadQuality: 0
numInvalidCMR: 0 numInvalidFrameType: 0
```

Related Commands

Command	Description
debug vtsp	Displays the state of the gateway and the call events.

show voip debug version

To display the current version of the Voice over IP debug structure, use the **show voip debug version**command in privileged EXEC mode.

show voip debug version

Command Default

No default behavior or values

Command Modes

Privileged EXEC (#)

Command History

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

Examples

The following example shows output from the **show voip debug version** command:

```
Router# show voip debug version
voip debug version 1.0
```

The table below describes significant fields shown in the display.

Table 5: show voip debug version Field Descriptions

Field	Description
voip debug version 1.0	Shows the version of the debug structure.

Related Commands

Command	Description
show voip rtp connections	Displays RTP named event packets.

show voip fpi call-rate

To display the average call rates at the forwarding plane interface, use the **show voip fpi call-rate** command in privileged EXEC mode.

show voip fpi call-rate interval*seconds* **history** *seconds*

Syntax Description

interval	Displays the message rates at the FPI interface
seconds	The number of seconds for the interval. The range is from 1 to 300
history	Specifies how far back information is kept and displayed.
seconds	The number of seconds that will be displayed. The range is from 1 to 86400.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
Cisco IOS XE Release 3.9S	This command was introduced.

Usage Guidelines

This command displays the call-rate data that is collected on the forwarding plane interface when the **debug voip fpi call-rate** is enabled .

Examples

The following shows the output for the **show voip fpi call-rate** command

```
Router# show voip fpi call-rate interval 1 history 1
------ ------ ------ ------ ------ ------ ------
Sec ADD MOD DEL EVT_UP EVT_DN CPU 5S
------ ------ ------ ------ ------ ------ ------
67 0 0 0 0 0 0
------ ------ ------ ------ ------ ------ ------
```

show voip fpi calls

To display call information for TDM and IVR calls in the Forwarding Plane Interface (FPI), use the **show voip fpi calls** command in privileged EXEC mode.

show voip fpi calls[**all** | **confID** *identifier*| **callID** *identifier*| **correlator** *identifier*]

Syntax Description

all	(Optional) Displays the detailed statistics for all calls in the FPI where the collection processes have been enabled.
confID *identifier*	(Optional) Displays detailed call information for a call based on the conference ID.
callID *identifier*	(Optional) Displays detailed call information for a call based on the call ID.
correlator *identifier*	(Optional) Displays detailed call information for a call based on the correlator ID.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
Cisco IOS XE Release 3.9S	This command was introduced.

Usage Guidelines

Examples

The following are sample output from the **show voip fpi calls** command

```
Router# show voip fpi calls
Number of Calls : 2
---------- ---------- ---------- ----------- --------------- ---------------
    confID correlator    AcallID     BcallID           state           event
---------- ---------- ---------- ----------- --------------- ---------------
        20         20         87          88       ALLOCATED DETAIL_STAT_RSP
        21         21         89          90       ALLOCATED DETAIL_STAT_RSP


Router# show voip fpi calls confID 20
----------------------------------------------------------------------------
VoIP-FPI call entry details:
----------------------------------------------------------------------------
Call Type         :            IP_IP     confID            :              20
correlator        :               20     call_state        :       ALLOCATED
last_event        :  DETAIL_STAT_RSP     alloc_start_time  :      2737426765
modify_start_time:                 0     delete_start_time:                0
Media Type(SideA):               RTP     Media Type(SideB):              RTP
----------------------------------------------------------------------------
```

```
FPI State Machine Stats:
------------------------
create_req_call_entry_inserted            :        1
call_create_req_fsm_successful            :        1
call_provision_rsp_ok                     :        1
call_provision_rsp_fsm_successful         :        1
event_ind_media_up_to_app                 :        2
------------------------------------------------------------------------------------------
SIDE_A  RTP details   -   gccb=0x7FE69FA11C08
------------------------------------------------------------------------------------------
confID           :        20    fpi_user_data    :        20
callID           :        87    dstCallID        :        88    mainstcallID     :
87
srcport          :     16552    dstport          :     16580    DP add_sent      :
 1
dp_add_fail      :         0    dp_add_pending   :         0    dp_delete_sent   :
 0
dp_delete_waiting:         0    dp_delete_done   :         0    final_stats_pend :
 0
ha_create_sent   :         1    is_video         :         0    media_type       :
 0
is dspfarm xcode :        No    is conference    :        No    stream_type      :     VOICE
rtp_type         :  SENDRECV
------------------------------------------------------------------------------------------
SIDE_B  RTP details   -   gccb=0x7FE6A9B5A960
------------------------------------------------------------------------------------------
confID           :        20    fpi_user_data    :        20
callID           :        88    dstCallID        :        87    mainstcallID     :
88
srcport          :     16554    dstport          :     16400    DP add_sent      :
 1
dp_add_fail      :         0    dp_add_pending   :         0    dp_delete_sent   :
 0
dp_delete_waiting:         0    dp_delete_done   :         0    final_stats_pend :
 0
ha_create_sent   :         1    is_video         :         0    media_type       :
 0
is dspfarm xcode :        No    is conference    :        No    stream_type      :     VOICE
rtp_type         :  SENDRECV
------------------------------------------------------------------------------------------
Detailed Stats from DataPlane:
------------------------------
mgm_handle      : 20
-------------------------------------------------------------------
Call Present in :    FMAN RP   FMAN FP       CPP
                 ----------------------------------------
                        YES       YES       YES
-------------------------------------------------------------------
                     Field              sideA              sideB
-------------------------------------------------------------------
          dtmf_payload_type                  0                  0
   redundant_data_pyld_type                255                255
                   tos_mask                  0                  0
                 dtmf_flags                  0                  0
                ucode_flags                  5                  5
                 local_port              16552              16554
             remote_port_tx              16580              16400
             remote_port_rx              16580              16400
                 session_id         0x30000050         0x30000052
  hairpin_prtnr_null(ucode)               NULL               NULL
       hairpin_prtnr_callid                  0                  0
         dsp_interface_null               NULL               NULL
-------------------------------------------------------------------

 DSP Resource Used : No


Router# show voip fpi calls callid 87
---------------------------------------------------------------------------
VoIP-FPI call entry details:
---------------------------------------------------------------------------
Call Type        :           IP_IP     confID           :             20
```

```
correlator       :                20     call_state       :        ALLOCATED
last_event       :  DETAIL_STAT_RSP      alloc_start_time :       2737426765
modify_start_time:                 0     delete_start_time:                0
Media Type(SideA):               RTP     Media Type(SideB):              RTP
---------------------------------------------------------------------------
FPI State Machine Stats:
------------------------
create_req_call_entry_inserted            :         1
call_create_req_fsm_successful            :         1
call_provision_rsp_ok                     :         1
call_provision_rsp_fsm_successful         :         1
event_ind_media_up_to_app                 :         2
------------------------------------------------------------------------------------------
SIDE_A  RTP details   -   gccb=0x7FE69FA11C08
------------------------------------------------------------------------------------------
confID           :        20    fpi_user_data    :        20
callID           :        87    dstCallID        :        88    mainstcallID     :
87
srcport          :     16552    dstport          :     16580    DP add_sent      :
 1
dp_add_fail      :         0    dp_add_pending   :         0    dp_delete_sent   :
 0
dp_delete_waiting:         0    dp_delete_done   :         0    final_stats_pend :
 0
ha_create_sent   :         1    is_video         :         0    media_type       :
 0
is dspfarm xcode :        No   is conference    :        No    stream_type      :    VOICE
rtp_type         :  SENDRECV
------------------------------------------------------------------------------------------
SIDE_B  RTP details   -   gccb=0x7FE6A9B5A960
------------------------------------------------------------------------------------------
confID           :        20    fpi_user_data    :        20
callID           :        88    dstCallID        :        87    mainstcallID     :
88
srcport          :     16554    dstport          :     16400    DP add_sent      :
 1
dp_add_fail      :         0    dp_add_pending   :         0    dp_delete_sent   :
 0
dp_delete_waiting:         0    dp_delete_done   :         0    final_stats_pend :
 0
ha_create_sent   :         1    is_video         :         0    media_type       :
 0
is dspfarm xcode :        No   is conference    :        No    stream_type      :    VOICE
rtp_type         :  SENDRECV
------------------------------------------------------------------------------------------
Detailed Stats from DataPlane:
------------------------------
mgm_handle      : 20
-------------------------------------------------------------------
Call Present in :    FMAN RP   FMAN FP       CPP
                 ----------------------------------------
                          YES       YES       YES
-------------------------------------------------------------------
                      Field                 sideA               sideB
-------------------------------------------------------------------
          dtmf_payload_type                     0                   0
   redundant_data_pyld_type                   255                 255
                   tos_mask                     0                   0
                 dtmf_flags                     0                   0
                ucode_flags                     5                   5
                 local_port                 16552               16554
             remote_port_tx                 16580               16400
             remote_port_rx                 16580               16400
                 session_id            0x30000050          0x30000052
  hairpin_prtnr_null(ucode)                  NULL                NULL
       hairpin_prtnr_callid                     0                   0
        dsp_interface_null                   NULL                NULL
-------------------------------------------------------------------

 DSP Resource Used : No
```

```
Router# show voip fpi calls all

Number of Calls : 2
---------------------------------------------------------------------------
VoIP-FPI call entry details:
---------------------------------------------------------------------------
Call Type        :           IP_IP     confID           :             24
correlator       :              24     call_state       :      ALLOCATED
last_event       :  DETAIL_STAT_RSP    alloc_start_time :     2902404766
modify_start_time:               0     delete_start_time:              0
Media Type(SideA):             RTP     Media Type(SideB):            RTP
---------------------------------------------------------------------------
FPI State Machine Stats:
------------------------
create_req_call_entry_inserted              :         1
call_create_req_fsm_successful              :         1
call_provision_rsp_ok                       :         1
call_provision_rsp_fsm_successful           :         1
event_ind_media_up_to_app                   :         2
------------------------------------------------------------------------------------------
SIDE_A  RTP details   -   gccb=0x7FE69FA11C08
------------------------------------------------------------------------------------------
confID           :        24    fpi_user_data    :        24
callID           :        95    dstCallID        :        96    mainstcallID     :
95
srcport          :     16568    dstport          :     16580    DP add_sent      :
 1
dp_add_fail      :         0    dp_add_pending   :         0    dp_delete_sent   :
 0
dp_delete_waiting:         0    dp_delete_done   :         0    final_stats_pend :
 0
ha_create_sent   :         1    is_video         :         0    media_type       :
 0
is dspfarm xcode :        No   is conference    :        No   stream_type      :    VOICE
rtp_type         :  SENDRECV
------------------------------------------------------------------------------------------
SIDE_B  RTP details   -   gccb=0x7FE6A9B5A960
------------------------------------------------------------------------------------------
confID           :        24    fpi_user_data    :        24
callID           :        96    dstCallID        :        95    mainstcallID     :
96
srcport          :     16570    dstport          :     16400    DP add_sent      :
 1
dp_add_fail      :         0    dp_add_pending   :         0    dp_delete_sent   :
 0
dp_delete_waiting:         0    dp_delete_done   :         0    final_stats_pend :
 0
ha_create_sent   :         1    is_video         :         0    media_type       :
 0
is dspfarm xcode :        No   is conference    :        No   stream_type      :    VOICE
rtp_type         :  SENDRECV
------------------------------------------------------------------------------------------
Detailed Stats from DataPlane:
------------------------------
mgm_handle      : 24
-------------------------------------------------------------------
Call Present in :    FMAN RP   FMAN FP       CPP
                 ----------------------------------------
                         YES       YES       YES
-------------------------------------------------------------------
                      Field               sideA              sideB
-------------------------------------------------------------------
          dtmf_payload_type                   0                  0
   redundant_data_pyld_type                 255                255
                   tos_mask                   0                  0
                 dtmf_flags                   0                  0
                ucode_flags                   5                  5
                 local_port               16568              16570
             remote_port_tx               16580              16400
             remote_port_rx               16580              16400
```

```
                 session_id          0x30000060          0x30000062
   hairpin_prtnr_null(ucode)               NULL                NULL
        hairpin_prtnr_callid                  0                   0
          dsp_interface_null               NULL                NULL
-------------------------------------------------------------------

 DSP Resource Used : No
------------------------------------------------------------
---------------------------------------------------------------------------
VoIP-FPI call entry details:
---------------------------------------------------------------------------
Call Type         :            IP_IP     confID           :             25
correlator        :               25     call_state       :      ALLOCATED
last_event        :  DETAIL_STAT_RSP     alloc_start_time :     2902505765
modify_start_time:                 0     delete_start_time:              0
Media Type(SideA):               RTP     Media Type(SideB):            RTP
---------------------------------------------------------------------------
FPI State Machine Stats:
------------------------
create_req_call_entry_inserted              :         1
call_create_req_fsm_successful              :         1
call_provision_rsp_ok                       :         1
call_provision_rsp_fsm_successful           :         1
event_ind_media_up_to_app                   :         1
------------------------------------------------------------------------------------------
SIDE_A  RTP details   -   gccb=0x7FE6A9B9CFA8
------------------------------------------------------------------------------------------
confID            :        25    fpi_user_data    :        25
callID            :        97    dstCallID        :        98    mainstcallID     :
97
srcport           :     16572    dstport          :     16584    DP add_sent      :
 1
dp_add_fail       :         0    dp_add_pending   :         0    dp_delete_sent   :
 0
dp_delete_waiting:          0    dp_delete_done   :         0    final_stats_pend :
 0
ha_create_sent    :         0    is_video         :         0    media_type       :
 0
is dspfarm xcode :        No    is conference    :        No    stream_type      :     VOICE
rtp_type          :  SENDRECV
------------------------------------------------------------------------------------------
SIDE_B  RTP details   -   gccb=0x7FE69FA132F8
------------------------------------------------------------------------------------------
confID            :        25    fpi_user_data    :        25
callID            :        98    dstCallID        :        97    mainstcallID     :
98
srcport           :     16574    dstport          :     16404    DP add_sent      :
 1
dp_add_fail       :         0    dp_add_pending   :         0    dp_delete_sent   :
 0
dp_delete_waiting:          0    dp_delete_done   :         0    final_stats_pend :
 0
ha_create_sent    :         1    is_video         :         0    media_type       :
 0
is dspfarm xcode :        No    is conference    :        No    stream_type      :     VOICE
rtp_type          :  SENDRECV
------------------------------------------------------------------------------------------
Detailed Stats from DataPlane:
------------------------------
mgm_handle       : 25
-------------------------------------------------------------------
Call Present in :    FMAN RP   FMAN FP       CPP
                 ----------------------------------------
                         YES       YES       YES
-------------------------------------------------------------------
                      Field              sideA               sideB
-------------------------------------------------------------------
          dtmf_payload_type                  0                   0
   redundant_data_pyld_type                255                 255
                   tos_mask                  0                   0
                 dtmf_flags                  0                   0
                ucode_flags                  0                   5
                 local_port              16572               16574
```

```
             remote_port_tx                16584                16404
             remote_port_rx                16584                16404
                 session_id           0x30000064           0x30000066
  hairpin_prtnr_null(ucode)                 NULL                 NULL
       hairpin_prtnr_callid                    0                    0
         dsp_interface_null                 NULL                 NULL
-------------------------------------------------------------------

 DSP Resource Used : No
```

show voip fpi stats

To display the TDM and IVR statistics and error counters in the Forwarding Plane Interface (FPI), use the **show voip fpi stats** command in privileged EXEC mode.

show voip fpi stats [fsm]

Syntax Description

fsm	(Optional) Displays the finite state machine (FSM) events.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
Cisco IOS XE Release 3.9S	This command was introduced.

Examples

The following is a sample output from the **show voip fpi stats** command:

```
Router# show voip fpi stats

*********************** ACTIVE *******************************
                           IDLE      ALLOCATING       ALLOCATED       MODIFYING
     CREATE_REQ             25               0               0               0
     MODIFY_REQ              0               0              21              17
     DELETE_REQ              0               0              22               3
  GET_STATS_REQ              0               0               0               0
    PROV_RSP_OK              0              25               0              18
  PROV_RSP_FAIL              0               0               0               0
     DELETE_RSP              0               0               0               0
  GET_STATS_RSP              0               0               0               0
  STATS_TMR_EXP              0               0               0               0
     TMR_EXPIRY              0               0               0               0
CREATE_STRM_REQ              0               0               0               0
MODIFY_STRM_REQ              0               0               0               0
DELETE_STRM_REQ              0               0               0               0
DETAIL_STAT_REQ              0               0             631               0
DETAIL_STAT_RSP              0               0             631               0
DT_STAT_TMR_EXP              0               0               0               0
                       DELETING  ALLOC_MOD_PEND MODIFY_MOD_PEND  DELETE_PENDING
     CREATE_REQ              0               0               0               0
     MODIFY_REQ              0               0               8               0
     DELETE_REQ              0               0               0               0
  GET_STATS_REQ              0               0               0               0
    PROV_RSP_OK              0               0              17               3
  PROV_RSP_FAIL              0               0               0               0
     DELETE_RSP             25               0               0               0
  GET_STATS_RSP              0               0               0               0
  STATS_TMR_EXP              0               0               0               0
     TMR_EXPIRY              0               0               0               0
CREATE_STRM_REQ              0               0               0               0
MODIFY_STRM_REQ              0               0               0               0
DELETE_STRM_REQ              0               0               0               0
DETAIL_STAT_REQ              0               0               0               0
DETAIL_STAT_RSP              0               0               0               0
```

```
DT_STAT_TMR_EXP             0               0               0               0
********************* END ACTIVE ***************************


********************** STANDBY *****************************
                         IDLE      ALLOCATING       ALLOCATED       MODIFYING
     CREATE_REQ             0               0               0               0
     MODIFY_REQ             0               0               0               0
     DELETE_REQ             0               0               0               0
  GET_STATS_REQ             0               0               0               0
    PROV_RSP_OK             0               0               0               0
  PROV_RSP_FAIL             0               0               0               0
     DELETE_RSP             0               0               0               0
  GET_STATS_RSP             0               0               0               0
  STATS_TMR_EXP             0               0               0               0
     TMR_EXPIRY             0               0               0               0
CREATE_STRM_REQ             0               0               0               0
MODIFY_STRM_REQ             0               0               0               0
DELETE_STRM_REQ             0               0               0               0
DETAIL_STAT_REQ             0               0               0               0
DETAIL_STAT_RSP             0               0               0               0
DT_STAT_TMR_EXP             0               0               0               0
                     DELETING  ALLOC_MOD_PEND MODIFY_MOD_PEND  DELETE_PENDING
     CREATE_REQ             0               0               0               0
     MODIFY_REQ             0               0               0               0
     DELETE_REQ             0               0               0               0
  GET_STATS_REQ             0               0               0               0
    PROV_RSP_OK             0               0               0               0
  PROV_RSP_FAIL             0               0               0               0
     DELETE_RSP             0               0               0               0
  GET_STATS_RSP             0               0               0               0
  STATS_TMR_EXP             0               0               0               0
     TMR_EXPIRY             0               0               0               0
CREATE_STRM_REQ             0               0               0               0
MODIFY_STRM_REQ             0               0               0               0
DELETE_STRM_REQ             0               0               0               0
DETAIL_STAT_REQ             0               0               0               0
DETAIL_STAT_RSP             0               0               0               0
DT_STAT_TMR_EXP             0               0               0               0
********************* END STANDBY ***************************
```

show voip htsp

To display the voip and hybrid transport switching protocol (HTSP) connections active in the router, use the **show voip htsp**command in privileged EXEC mode.

show voip htsp info [**controller** [**T1** *slot-number*]]

Syntax Description

info	Displays htsp related information.
controller	(Optional) Displays information about controllers such as DS3,T1,and E1.
T1	(Optional) Displays information about T1 controller.
slot-number	(Optional) controller slot number.

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

Use the**show voip htsp command to**display the voip and hybrid transport switching protocol (HTSP) connections active in the router.

Examples

The following is sample output from the **show voip htsp** command:

```
Router# show voip htsp
NOTE: '-' means Not Applicable for that signalling type
  SLOT/                                              TSP        TDM      TDM
 PORT/                                  TSP         BEAR     CONNECT   CROSS
 CHANNEL    HTSPINFO    VTSP_CDB        CDB         CHAN_T     DONE   CONNECT
=========  ==========  ==========  ==========  ==========  =======  =======
 02/00/01  0x677371E8  0x68905A48  0x67757AA4  0x677371E8     y        y
 02/00/02  0x67737780  0x00000000  0x00000000  0x00000000     n        n
 02/00/03  0x67737D18  0x68906548  0x67757584  0x67737D18     y        y
 02/00/04  0x677382B0  0x68904C88  0x677572F4  0x677382B0     y        y
 02/00/05  0x67738848  0x00000000  0x00000000  0x00000000     n        n
 02/00/06  0x67738DE0  0x00000000  0x00000000  0x00000000     n        n
 02/00/07  0x67739378  0x689054C8  0x67756B44  0x67739378     y        y
 02/00/08  0x67739910  0x68907888  0x677568B4  0x67739910     y        y
 02/00/09  0x67739EA8  0x00000000  0x00000000  0x00000000     n        n
 02/00/10  0x6773A440  0x00000000  0x00000000  0x00000000     n        n
 02/00/11  0x6773A9D8  0x68906D88  0x67756104  0x6773A9D8     y        y
 02/00/12  0x6773AF70  0x68908388  0x67755E74  0x6773AF70     y        y
```

```
02/00/13  0x6773B508  0x00000000  0x00000000  0x00000000      n        n
02/00/14  0x6773BAA0  0x00000000  0x00000000  0x00000000      n        n
02/00/15  0x6773C038  0x689096C8  0x677556C4  0x6773C038      y        y
02/00/17  0x6773C5D0  0x68909148  0x67755434  0x6773C5D0      y        y
02/00/18  0x6773CB68  0x00000000  0x00000000  0x00000000      n        n
02/00/19  0x6773D100  0x00000000  0x00000000  0x00000000      n        n
02/00/20  0x6773D698  0x68905788  0x67754C84  0x6773D698      y        y
02/00/21  0x6773DC30  0x68905D08  0x677549F4  0x6773DC30      y        y
02/00/22  0x6773E1C8  0x00000000  0x00000000  0x00000000      n        n
02/00/23  0x6773E760  0x00000000  0x00000000  0x00000000      n        n
02/00/24  0x6773ECF8  0x68906AC8  0x67754244  0x6773ECF8      y        y
02/00/25  0x6773F290  0x68907308  0x67753FB4  0x6773F290      y        y
02/00/26  0x6773F828  0x00000000  0x00000000  0x00000000      n        n
02/00/27  0x6773FDC0  0x00000000  0x00000000  0x00000000      n        n
02/00/28  0x67740358  0x689080C8  0x67753804  0x67740358      y        y
02/00/29  0x677408F0  0x68908908  0x67753574  0x677408F0      y        y
02/00/30  0x67740E88  0x00000000  0x00000000  0x00000000      n        n
02/00/31  0x67741420  0x68909408  0x67753054  0x67741420      y        y
02/02/01  0x67B88824  0x00000000  0x00000000      -           -        n
02/02/02  0x67B88DBC  0x00000000  0x00000000      -           -        n
02/02/03  0x67B89354  0x00000000  0x00000000      -           -        n
02/02/04  0x67B898EC  0x00000000  0x00000000      -           -        n
02/02/05  0x67B89E84  0x00000000  0x00000000      -           -        n
02/02/06  0x67B8A41C  0x00000000  0x00000000      -           -        n
02/02/07  0x67B8A9B4  0x00000000  0x00000000      -           -        n
02/02/08  0x67B8AF4C  0x00000000  0x00000000      -           -        n
02/02/09  0x67B8B4E4  0x00000000  0x00000000      -           -        n
```

Related Commands

Command	Description
debug voip vtsp	Displays information about the voice telephony service provider (VTSP).

show voip recmsp session

To display active recording Media Service Provider (MSP) session information, use the **show voip recmsp session** command in privileged EXEC mode.

show voip recmsp session [**detail call-id** *callid*]

Syntax Description

detail	(Optional) Displays detailed active session information.
call-id *callid*	(Optional) Specifies the recording MSP call ID. The range is from 0 to 65535.

Command Default

Displays brief information about recorded calls that have the anchor call ID, forked call ID, and MSP call ID.

Command Modes

Privileged EXEC (#)

Command History

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

Usage Guidelines

Use the **show voip recmsp session** command to display MSP-related information about the recorder, for example, the way the recording MSP views the recording session.

The **show voip recmsp session detail call-id** *callid* command provides detailed information about each recording session. It provides details about the anchor leg and nonanchor leg. It also shows how the anchor and nonanchor streams are mapped to the forked leg Real-Time Transport Protocol (RTP) streams.

Examples

The following is sample output from the **show voip recmsp session detail call-id** command. Fields in the display are self-explanatory.

```
Router# show voip recmsp session detail call-id
 140
RECMSP active sessions:
Detailed Information
=========================
Recording MSP Leg Details:
Call ID: 143
GUID : 7C5946D38ECD
AnchorLeg Details:
Call ID: 141
Forking Stream type: voice-nearend
Participant: 708090
Non-anchor Leg Details:
Call ID: 140
```

```
Forking Stream type: voice-farend
Participant: 10000
Forked Leg Details:
Call ID: 145
Near End Stream CallID 145
Stream State ACTIVE
Far End stream CallID 146
Stream State ACTIVE
Found 1 active sessions
```

Related Commands

Command	Description
media-recording	Configures voice class recording parameters.

show voip rtp connections

To display Real-Time Transport Protocol (RTP) named event packets, use the **show voip rtp connections**command in privileged EXEC mode.

show voip rtp connections [detail]

Syntax Description

detail	(Optional) Displays the called-party and calling-party numbers associated with a call.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.0	This command was introduced.
12.3(7)T	The **detail** keyword was added.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.
12.4(22)T	Command output was updated to show IPv6 information.

Usage Guidelines

This command displays information about RTP named event packets, such as caller ID number, IP address, and port for both the local and remote endpoints. The output from this command provides an overview of all the connections in the system, and this information can be used to narrow the criteria for debugging. The **debug voip rtp** command floods the console with voice packet information. You can use the **show voip rtp connections** command to get caller ID, remote IP address, or remote port identifiers that you can use to limit the output from the **debug voip rtp** command.

The **detail** keyword allows you to identify the phone or phones that have connected two RTP call legs to create VoIP-to-VoIP or VoIP-to-POTS hairpins. If the **detail** keyword is omitted, the output does not display calls that are connected by hairpin call routing.

Examples

The table below describes the significant fields shown in the examples. Each line of output under "VoIP RTP active connections" shows information for one call leg. A phone call normally consists of two call legs, one connected to the calling party and one connected to the called party. The router joins (or bridges) the two call legs to make a call. The **show voip rtp connections** command shows the RTP information for H.323 and Session Initiation Protocol (SIP) calls only; it does not directly show the POTS call legs. The information for the IP phone can be seen using the **show ephone offhook** command.

The following sample output shows an incoming H.323 call that is being directed to an IP phone attached to a Cisco CallManager Express (CME) system.

```
Router# show voip rtp connections
VoIP RTP active connections :
No. CallId  dstCallId  LocalRTP RmtRTP LocalIP          RemoteIP
1   21      22         16996    18174  10.4.204.37      10.4.204.24
Found 1 active RTP connections
```

The following sample output shows the same call as in the previous example, but using the **detail** keyword with the command. The sample output shows the called number (1509) and calling number (8108) on both call legs (21 and 22); the called and calling numbers are the same on both legs for a simple A-to-B call. Leg 21 is the H.323 segment of the and leg 22 is the POTS segment that goes to the IP phone.

```
Router#
show voip rtp connections detail
VoIP RTP active connections :
No. CallId  dstCallId  LocalRTP RmtRTP LocalIP          RemoteIP
1   21      22         16996    18174  10.4.204.37      10.4.204.24
  callId 21 (dir=1):called=1509 calling=8108 redirect=
     dest callId 22:called=1509 calling=8108 redirect=
   1 context 64FB3358 xmitFunc 6032E8B4
Found 1 active RTP connections
```

The following example shows the call from the previous example being transferred by extension 1509 to extension 1514. Notice that the dstCallId changed from 22 to 24, but the original call leg (21) for the transferred party is still present. This implies that H.450.2 capability was disabled for this particular call, because if H.450.2 was being used for the transfer, the transfer would have caused the incoming H.323 call leg to be replaced with a new call.

```
Router# show voip rtp connections

VoIP RTP active connections :
No. CallId  dstCallId  LocalRTP RmtRTP LocalIP         RemoteIP
1   21      24         16996    18174  10.4.204.37      10.4.204.24
Found 1 active RTP connections
```

The following example shows the detailed output for the same transfer as shown in the previous example. The original incoming call leg is still present (21) and still has the original called and calling numbers. The transferred call leg (24) shows 1509 (the transferring party) as the calling party and 1514 (the transfer destination) as the called party.

```
Router# show voip rtp connections detail
VoIP RTP active connections :
No. CallId  dstCallId  LocalRTP RmtRTP LocalIP         RemoteIP
1   21      24         16996    18174  10.4.204.37      10.4.204.24
  callId 21 (dir=1):called=1509 calling=8108 redirect=
     dest callId 24:called=1514 calling=1509 redirect=
   1 context 6466E810 xmitFunc 6032E8B4
Found 1 active RTP connections
```

The following sample output shows a cross-linked call with two H.323 call legs. The first line of output shows that the CallID for the first call leg is 7 and that this call leg is associated with another call leg that has a destination CallId of 8. The next line shows that the CallID for the leg is 8 and that it is associated with another call leg that has a destination CallId of 7. This cross-linkage between CallIds 7 and 8 shows that the first call leg is related to the second call leg (and vice versa). From this you can infer that the two call legs are actually part of the same phone call.

In an active system you can expect many lines of output that you would have to sort through to see which ones have this cross-linkage relationship. The lines showing two related call legs are not necessarily listed in adjacent order.

```
Router# show voip rtp connections
VoIP RTP active connections :
```

```
No. CallId   dstCallId          LocalRTP     RmtRTP         LocalIP         RemoteIP
1        7           8             16586      22346      172.27.82.2      172.29.82.2
2        8           7             17010      16590      172.27.82.2      192.168.1.29

Found 2 active RTP connections
```

The following example shows RTP information with IPv6 local and remote addresses:

```
Router# show voip rtp connections
VoIP RTP active connections :
No.  CallId   dstCallId  LocalRTP  RmtRTP   LocalIP                              RemoteIP
1     11      9          17424     18282    2001:DB8:C18:1:218:FEFF:FE71:2AB6
2001:DB8:C18:1:218:FEFF:FE71:2AB6
2     12      10         18282     17424    2001:DB8:C18:1:218:FEFF:FE71:2AB6
2001:DB8:C18:1:218:FEFF:FE71:2AB6
Found 2 active RTP connections
```

Table 6: show voip rtp connections Field Descriptions

Field	Description
No.	Identifier of an RTP connection in this output.
CallId	Internal call identifier of a telephony call leg (RTP connection).
dstCallId	Internal call identifier of a VoIP call leg.
LocalRTP	RTP port of the media stream for the local entity.
RmtRTP	RTP port of the media stream for the remote entity.
LocalIP	IPv4 or IPv6 address of the media stream for the local entity.
RemoteIP	IPv4 or IPv6 address of the media stream for the remote entity.
dir	0 indicates an outgoing call. 1 indicates an incoming call.
called	Extension that received the call.
calling	Extension that made the call.
redirect	Original called number if the incoming call was forwarded.
context	Internal memory address for the control block associated with the call.
xmitFunc	Internal memory address for the transmit function to which incoming RTP packets (on the H.323 and SIP side) are sent; the address for the function that delivers the packets to the ephone.

Related Commands

Command	Description
debug voip rtp	Enables debugging for RTP named event packets.
show ephone offhook	Displays information and packet counts for phones that are currently off hook.

show voip rtp forking

To display the Real-Time Transport Protocol (RTP) media-forking connections, use the **show voip rtp forking** command in privileged EXEC mode.

show voip rtp forking

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.4(24)T	This command was introduced in a release earlier than Cisco IOS Release 12.4(24)T.

Usage Guidelines

The **show voip rtp forking** command displays information about RTP named event packets, such as type of stream, IP address, and port for both the local and remote endpoints. The output from this command provides an overview of all the media-forking connections in the system, and this information can be used to narrow the criteria for debugging. The **debug voip rtp** command floods the console with voice packet information. You can use the **show voip rtp forking** command to display the remote IP address, or remote port identifiers that you can use to limit the output from the **debug voip rtp** command.

Examples

The following is sample output from the **show voip rtp forking** command:

```
Router# show voip rtp forking
VoIP RTP active forks :
 Fork 1
   stream type voice-only (0): count 1
     remote ip 9.13.36.101,  remote port 20590,  local port 17596
       codec g711alaw,  logical ssrc 0x60
       packets sent 237,  packets received 413
   stream type voice+dtmf (1): count 0
   stream type dtmf-only (2): count 0
   stream type voice-nearend (3): count 1
     remote ip 9.13.36.102,  remote port 18226,  local port 17434
       codec g729r8,  logical ssrc 0x103
       packets sent 39,  packets received 0
   stream type voice+dtmf-nearend (4): count 0
   stream type voice-farend (5): count 1
     remote ip 9.13.36.120,  remote port 16912,  local port 21098
       codec g729r8,  logical ssrc 0x105
       packets sent 39,  packets received 0
   stream type voice+dtmf-farend (6): count 0
   stream type video (7): count 0
```

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

Table 7: show voip rtp forking Field Descriptions

Field	Description
stream type	Indicates the type of stream.
count	Number of packets in the specified type of stream.
remote ip	IPv4 or IPv6 address of the media stream for the remote entity.
remote port	RTP port of the media stream for the remote entity.
local port	RTP port of the media stream for the local entity.
codec	Codec supported on the specified channel.
logical ssrc	Indicates the logical synchronization source (SSRC) for the specified channel.
packets sent	Total number of packets sent from the channel.
packets received	Total number of packets received by the channel.

Related Commands

Command	Description
debug voip rtp	Enables debugging for RTP named event packets.

show voip trunk group

To display the internal list of voip trunk groups, use the **show voip trunk group** command in user EXEC or privileged EXEC mode.

show voip trunk group

Syntax Description

This command has no arguments or keywords.

Command Default

Command Modes

User EXEC (>) Privileged EXEC (#)

Command History

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

Usage Guidelines

Use this command to display VOIP trunk groups.

Examples

The following example is a sample output from the **show voip trunk group** command.

```
Router# show voip trunk group

=====================================================
name: 1
protocol: cisco
ip: 1.3.45.2
xsvc: TRUE
```

Related Commands

Command	Description
voip trunk group	Specifies a VOIP trunk group.

show vrm active_calls

To display active-only voice calls either for a specific voice feature card (VFC) or for all VFCs, use the **show vrm active_calls**command in privileged EXEC mode.

show vrm active_calls {*dial-shelf-slot-number*| **all**}

Syntax Description

dial -shelf-slot-number	Slot number of the dial shelf. Range is from 0 to 13.
all	Displays list of all active calls for VFC slots.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.0(7)T	This command was introduced on the Cisco AS5800.

Usage Guidelines

Use this command to display active-only voice calls either for a specific VFC or for all VFCs. Each active call occupies a block of information describing the call. This information provides basically the same information as the **show vrm vdevice** command.

Examples

The following is sample output from this command specifying a dial-shelf slot number:

```
Router# show vrm active_calls 6
slot =  6 virtual voice dev (tag) =  61 channel id = 2
capabilities list map = 9FFF
last/current codec loaded/used = None
TDM timeslot = 241
Resource (vdev_common) status = 401 means :active others
tot ingress data =  24
tot ingress control  = 1308
tot ingress data drops  = 0
tot ingress control drops  = 0
tot egress data  = 22051
tot egress control  = 1304
tot egress data drops  = 0
tot egress control drops  = 0
slot =  6 virtual voice dev (tag) =  40 channel id = 2
capabilities list map = 9FFF
last/current codec loaded/used = None
TDM timeslot = 157
Resource (vdev_common) status = 401 means :active others
```

The table below describes significant fields shown in this output.

Table 8: show vrm active_calls Field Descriptions

Field	Description
slot	Slot where the voice card is installed.
virtual voice dev (tag)	ID number of the virtual voice device.
channel id	ID number of the channel associated with this virtual voice device.
capability list map	Bitmaps for the codec supported on that DSP channel. Values are the following: • CC_CAP_CODEC_G711U: 0x1 • CC_CAP_CODEC_G711A: 0x2 • CC_CAP_CODEC_G729IETF: 0x4 • CC_CAP_CODEC_G729a: 0x8 • CC_CAP_CODEC_G726r16: 0x10 • CC_CAP_CODEC_G726r24: 0x20 • CC_CAP_CODEC_G726r32: 0x40 • CC_CAP_CODEC_G728: 0x80 • CC_CAP_CODEC_G723r63: 0x100 • CC_CAP_CODEC_G723r53: 0x200 • CC_CAP_CODEC_GSM: 0x400 • CC_CAP_CODEC_G729b: 0x800 • CC_CAP_CODEC_G729ab: 0x1000 • CC_CAP_CODEC_G723ar63: 0x2000 • CC_CAP_CODEC_G723ar53: 0x4000 • CC_CAP_CODEC_G729: 0x8000
last/current codec loaded/used	Last codec loaded or used.
TDM time slot	Time-division-multiplexing time slot.
Resource (vdev_common) status	Current status of the VFC.
tot ingress data	Total amount of data (number of packets) sent from the PSTN side of the connection to the VoIP side of the connection.

Field	Description
tot ingress control	Total number of control packets sent from the PSTN side of the connection to the VoIP side of the connection.
tot ingress data drops	Total number of data packets dropped from the PSTN side of the connection to the VoIP side of the connection.
tot ingress control drops	Total number of control packets dropped from the PSTN side of the connection to the VoIP side of the connection.
tot egress data	Total amount of data (number of packets) sent from the VoIP side of the connection to the PSTN side of the connection.
tot egress control	Total number of control packets sent from the VoIP side of the connection to the PSTN side of the connection.
tot egress data drops	Total number of data packets dropped from the VoIP side of the connection to the PSTN side of the connection.
tot egress control drops	Total number of control packets dropped from the VoIP side of the connection to the PSTN side of the connection.

Related Commands

Command	Description
show vrm vdevices	Displays detailed information for a specific DSP or a brief summary display for all VFCs.

show vrm vdevices

To display detailed information for a specific digital signal processor (DSP) or summary information for all voice feature cards (VFCs), use the **show vrm vdevices**command in privileged EXEC mode.

show vrm vdevices {**vfc-slot-number voice-device-number**| **alarms [vfc-slot-number-for-alarms]**| **summary**}

Syntax Description

vfc -slot-number	Slot number of the VFC. Range is from 0 to 11.
voice -device-number	DSP number. Range is from 1 to 96.
alarms	DSP alarm statistics for all DSPs on all slots or specified slots.
vfc -slot-number-for-alarms	(Optional) Slots for which you need alarm information. If no slots are specified, alarm information for all slots is displayed.
summary	Synopsis of voice feature card DSP mappings, capabilities, and resource states.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.0(7)T	This command was introduced on the Cisco AS5800.
12.2(11)T	The **alarms** keyword and *vfc-slot-number-for-alarms* argument were added.

Usage Guidelines

Use this command to display detailed information for a specific DSP or a brief summary for all VFCs. The display provides information such as the number of channels, channels per DSP, bitmap of digital signal processor modules (DSPMs), DSP alarm statistics, and version numbers. This information is useful in monitoring the current state of your VFCs.

The display for a specific DSP provides information on the codec that each channel is using, if active, or on the codec that was last used and whether the channel is not currently sending cells. It also displays the state of the resource. In most cases, if there is an active call on that channel, the resource should be marked active. If the resource is marked as reset or bad, this may be an indication of a response loss for the VFC on a reset request. If this condition persists, you might experience a problem with the communication link between the router shelf and the VFC.

Examples

The following is sample output from this command specifying dial-shelf slot number and DSP number. In this particular example, the call is active so the statistics displayed are for this active call. If no calls are currently active on the device, the statistics would be for the previous (or last active) call.

```
Router# show vrm vdevices 6 1
slot =  6 virtual voice dev (tag) =  1 channel id = 1
capabilities list map = 9FFF
last/current codec loaded/used = None
TDM timeslot = 0
Resource (vdev_common) status = 401 means :active others
tot ingress data =  101
tot ingress control  = 1194
tot ingress data drops  = 0
tot ingress control drops  = 0
tot egress data  = 39722
tot egress control  = 1209
tot egress data drops  = 0
tot egress control drops  = 0
slot =  6 virtual voice dev (tag) =  1 channel id = 2
capabilities list map = 9FFF
last/current codec loaded/used = None
TDM timeslot = 1
Resource (vdev_common) status = 401 means :active others
tot ingress data =  21
tot ingress control  = 1167
tot ingress data drops  = 0
tot ingress control drops  = 0
tot egress data  = 19476
tot egress control  = 1163
tot egress data drops  = 0
tot egress control drops  = 0
```

The table below describes significant fields shown in this output.

Table 9: show vrm vdevices Field Descriptions

Field	Description
slot	Slot in which the voice card is installed.
virtual voice dev (tag)	ID number of the virtual voice device.
channel id	ID number of the channel that is associated with this virtual voice device.

Field	Description
capabilities list map	Bitmaps for the codec supported on that DSP channel. Values are as follows: • CC_CAP_CODEC_G711U: 0x1 • CC_CAP_CODEC_G711A: 0x2 • CC_CAP_CODEC_G729IETF: 0x4 • CC_CAP_CODEC_G729a: 0x8 • CC_CAP_CODEC_G726r16: 0x10 • CC_CAP_CODEC_G726r24: 0x20 • CC_CAP_CODEC_G726r32: 0x40 • CC_CAP_CODEC_G728: 0x80 • CC_CAP_CODEC_G723r63: 0x100 • CC_CAP_CODEC_G723r53: 0x200 • CC_CAP_CODEC_GSM: 0x400 • CC_CAP_CODEC_G729b: 0x800 • CC_CAP_CODEC_G729ab: 0x1000 • CC_CAP_CODEC_G723ar63: 0x2000 • CC_CAP_CODEC_G723ar53: 0x4000 • CC_CAP_CODEC_G729: 0x8000 • CC_CAP_CODEC_GSMEFR: 0x40000 • CC_CAP_CODEC_T38FAX: 0x10000
last/current codec loaded/used	Last codec loaded or used.
TDM timeslot	Time-division-multiplexing time slot.

Field	Description
Resource (vdev_common) status	Current status of the VFC. Values are as follows: • FREE = 0x0000 • ACTIVE_CALL = 0x0001 • BUSYOUT_REQ = 0x0002 • BAD = 0x0004 • BACK2BACK_TEST = 0x0008 • RESET = 0x0010 • DOWNLOAD_FILE = 0x0020 • DOWNLOAD_FAIL = 0x0040 • SHUTDOWN = 0x0080 • BUSY = 0x0100 • OIR = 0x0200 • HASLOCK = 0x0400 /* vdev_pool has locked port */ • DOWNLOAD_REQ = 0x0800 • RECOVERY_REQ = 0x1000 • NEGOTIATED = 0x2000 • OOS = 0x4000
tot ingress data	Total amount of data (number of packets) sent from the public switched telephone network (PSTN) side of the connection to the VoIP side of the connection.
tot ingress control	Total number of control packets sent from the PSTN side of the connection to the VoIP side of the connection.
tot ingress data drops	Total number of data packets dropped from the PSTN side of the connection to the VoIP side of the connection.
tot ingress control drops	Total number of control packets dropped from the PSTN side of the connection to the VoIP side of the connection.
tot egress data	Total amount of data (number of packets) sent from the VoIP side of the connection to the PSTN side of the connection.

Field	Description
tot egress control	Total number of control packets sent from the VoIP side of the connection to the PSTN side of the connection.
tot egress data drops	Total number of data packets dropped from the VoIP side of the connection to the PSTN side of the connection.
tot egress control drops	Total number of control packets dropped from the VoIP side of the connection to the PSTN side of the connection.

The following sample output displays alarm statistics for slot 6 of the DSP.

```
Router# show vrm vdevices alarms 6
----------------------ALARM STATISTICS FOR SLOT 6 ------------------------
TAG Mod DSP Chn OperStat  AlmCnt   AlmTime     AlmCause     AlmText
--------------------------------------------------------------------------
1   1   1   1   READY CD  0        0           1
            2   READY CD  0        0           1
2   1   2   1   READY CD  0        0           1
            2   READY CD  0        0           1
3   1   3   1   READY CD  0        0           1
            2   READY CD  0        0           1
4   1   4   1   READY CD  0        0           1
            2   READY CD  0        0           1
5   1   5   1   READY CD  0        0           1
            2   READY CD  0        0           1
6   1   6   1   READY CD  0        0           1
            2   READY CD  0        0           1
++++++++++++++++++++++++++++++++++++++++++++++++++++++++++++++++++++++++++
7   2   1   1   READY CD  0        0           1
            2   READY CD  0        0           1
8   2   2   1   READY CD  0        0           1
            2   READY CD  0        0           1
9   2   3   1   READY CD  0        0           1
            2   READY CD  0        0           1
10  2   4   1   READY CD  0        0           1
!
94  16  4   1   READY CD  0        0           1
            2   READY CD  0        0           1
95  16  5   1   READY CD  0        0           1
            2   READY CD  0        0           1
96  16  6   1   READY CD  0        0           1
            2   READY CD  0        0           1
++++++++++++++++++++++++++++++++++++++++++++++++++++++++++++++++++++++++++
```

The table below describes significant fields shown in this output.

Table 10: show vrm vdevices alarms Field Descriptions

Field	Description
TAG	Logical tag number.
Mod	DSP module number.
DSP	DSP number within the module.

Field	Description
Chn	Channel number for the DSP within the module.
OperStat	Operational status of the channel.
AlmCnt	Alarm count since bootup on that channel.
AlmTime	Time at which last alarm message was received.
AlmCause	Cause of last alarm message received.
AlmText	Text message corresponding to the last alarm message.
Possible Values for the Operational Status of the Channel (OperStat)	
RESET	RESET state.
DOWN	DOWN state.
READY CR	CORE READY state.
READY CD	CODEC READY state.
IDLE V	VOICE IDLE state.
IDLE FAX	FAX IDLE state.
READY V	VOICE READY state.
READY FX	FAX READY state.
READY D	DTMF READY state.
UNKNOWN	UNKNOWN state.

The following is sample output from this command specifying a summary list. In the "Voice Device Mapping" area, the "C_Ac" column indicates the number of active calls for a specific DSP. If there are any nonzero numbers under the "C_Rst" and/or "C_Bad" column, a reset request was sent, but it was lost; this could mean a faulty DSP.

```
Router# show vrm vdevices summary
***********************************************************
******summary of voice devices for all voice cards*********
***********************************************************
slot = 6 major ver = 0 minor ver = 1 core type used = 2
number of modules = 16 number of voice devices (DSPs) = 96
chans per vdevice = 2 tot chans = 192 tot active calls = 178
module presense bit map = FFFF tdm mode = 1 num_of_tdm_timeslots = 384
auto recovery is on
```

```
number of default voice file (core type images) = 2
file 0 maj ver = 0 min ver = 0 core_type = 1
trough size = 2880 slop value = 0 built-in codec bitmap = 0
loadable codec bitmap = 0 fax codec bitmap = 0
file 1 maj ver = 3 min ver = 1 core_type = 2
trough size = 2880 slop value = 1440 built-in codec bitmap = 40B
loadable codec bitmap = BFC fax codec bitmap = 7E
-------------------Voice Device Mapping------------------------
Logical Device (Tag)  Module#  DSP#  C_Ac  C_Busy  C_Rst  C_Bad
---------------------------------------------------------------
1                     1        1     2     0       0      0
2                     1        2     2     0       0      0
3                     1        3     2     0       0      0
4                     1        4     2     0       0      0
5                     1        5     2     0       0      0
6                     1        6     2     0       0      0
+++++++++++++++++++++++++++++++++++++++++++++++++++++++++++
7                     2        1     2     0       0      0
8                     2        2     2     0       0      0
9                     2        3     2     0       0      0
10                    2        4     1     0       0      0
11                    2        5     2     0       0      0
12                    2        6     1     0       0      0
.
.
.
91                    16       1     2     0       0      0
92                    16       2     2     0       0      0
93                    16       3     1     0       0      0
94                    16       4     2     0       0      0
95                    16       5     2     0       0      0
96                    16       6     2     0       0      0
+++++++++++++++++++++++++++++++++++++++++++++++++++++++++++
Total active call channels = 178
Total busied out channels = 0
Total channels in reset = 0
Total bad channels = 0
Note :Channels could be in multiple states
```

The table below describes significant fields shown in this output.

Table 11: show vrm vdevices summary Field Descriptions

Field	Description
slot	Slot number in which the VFC is installed.
major ver	Major version of firmware running on the VFC.
minor ver	Minor version of firmware running on the VFC.
core type used	Type of DSPware in use. Values are as follows: • 1 = UBL (boot loader) • 2 = high complexity core • 3 = medium complexity core • 4 = low complexity core • 255 = invalid
number of modules	Number of modules on the VFC. Maximum number is 16.

Field	Description
number of voice devices (DSP)s	Number of possible DSPs. Maximum number is 96.
chans per vdevice	Number of channels (meaning calls) that each DSP can handle.
tot chans	Total number of channels.
tot active calls	Total number of active calls on this VFC.
module presense bit map	Indicates a 16-bit bitmap, each bit representing a module.
tdm mode	Time-division-multiplex bus mode. Values are as follows: • 0 = VFC is in classic mode. • 1 = VFC is in plus mode. This field should always be 1.
num_of_tdm_timeslots	Total number of calls that can be handled by the VFC.
auto recovery	Whether auto recovery is enabled. When autorecovery is enabled, the VRM tries to recover a DSP by resetting it if, for some reason, the DSP stops responding.
number of default voice file (core type images)	Number of DSPware files in use.
number of default voice file (maj ver)	Major version of the DSPware in use.
min ver	Minor version of the DSPware in use.
core_type	Type of DSPware in use. Values are as follows: • 1 = boot loader • 2 = high complexity core • 3 = medium complexity core • 4 = low complexity core
trough size	Indirect representation of the complexity of the DSPware in use. **Note** Effective with Cisco IOS Release 12.1(5)XM, this value is no longer displayed.

Field	Description
slop value	Indirect representation of the complexity of the DSPware in use. **Note** Effective with Cisco IOS Release 12.1(5)XM, this value is no longer displayed.
built-in codec bitmap	Bitmap of the codec built into the DSP firmware. Values are as follows: • CC_CAP_CODEC_G711U: 0x0001 • CC_CAP_CODEC_G711A: 0x0002 • CC_CAP_CODEC_G729IETF: 0x0004 • CC_CAP_CODEC_G729a: 0x0008 • CC_CAP_CODEC_G726r16: 0x0010 • CC_CAP_CODEC_G726r24: 0x0020 • CC_CAP_CODEC_G726r32: 0x0040 • CC_CAP_CODEC_G728: 0x0080 • CC_CAP_CODEC_G723r63: 0x0100 • CC_CAP_CODEC_G723r53: 0x0200 • CC_CAP_CODEC_GSM: 0x0400 • CC_CAP_CODEC_G729b: 0x0800 • CC_CAP_CODEC_G729ab: 0x1000 • CC_CAP_CODEC_G723ar63: 0x2000 • CC_CAP_CODEC_G723ar53: 0x4000 • CC_CAP_CODEC_G729: 0x8000 • CC_CAP_CODEC_GSMEFR: 0x40000 • CC_CAP_CODEC_T38FAX: 0x10000

Field	Description
loadable codec bitmap	Loadable codec bitmap for the loadable codecs. Values are as follows: • CC_CAP_CODEC_G711U: 0x0001 • CC_CAP_CODEC_G711A: 0x0002 • CC_CAP_CODEC_G729IETF: 0x0004 • CC_CAP_CODEC_G729a: 0x0008 • CC_CAP_CODEC_G726r16: 0x0010 • CC_CAP_CODEC_G726r24: 0x0020 • CC_CAP_CODEC_G726r32: 0x0040 • CC_CAP_CODEC_G728: 0x0080 • CC_CAP_CODEC_G723r63: 0x0100 • CC_CAP_CODEC_G723r53: 0x0200 • CC_CAP_CODEC_GSM: 0x0400 • CC_CAP_CODEC_G729b: 0x0800 • CC_CAP_CODEC_G729: = 0x1000 • CC_CAP_CODEC_G723ar63: 0x2000 • CC_CAP_CODEC_G723ar53: 0x4000 • CC_CAP_CODEC_G729: 0x8000 • CC_CAP_CODEC_GSMEFR: 0x40000 • CC_CAP_CODEC_T38FAX: 0x10000
fax codec bitmap	Fax codec bitmap. Values are as follows: • FAX_NONE = 0x1 • FAX_VOICE = 0x2 • FAX_144 = 0x80 • FAX_120 = 0x40 • FAX_96 = 0x20 • FAX_72 = 0x10 • FAX_48 = 0x08 • FAX_24 = 0x04
Logical Device (Tag)	Tag number or DSP number on the VFC.

Field	Description
Module#	Number identifying the module associated with a specific logical device.
DSP#	Number identifying the DSP on the VFC.
C_Ac	Number of active calls on the identified DSP.
C_Busy	Number of busied-out channels associated with the identified DSP.
C_Rst	Number of channels in the reset state associated with the identified DSP.
C_Bad	Number of defective ("bad") channels associated with the identified DSP.
Total active call channels	Total number of active calls.
Total busied out channels	Total number of busied-out channels.
Total channels in reset	Total number of channels in the reset state.
Total bad channels	Total number of defective channels.

Related Commands

Command	Description
show vrm active_calls	Displays active-only voice calls either for a specific VFC or for all VFCs.

show vsp

To display cumulative information about voice streaming processing (VSP) sessions, use the **show vsp** command in privileged EXEC mode.

show vsp {**all**| **debug**| **session**| **statistics**}

Syntax Description

all	Displays all available information on VSP sessions, including the information specified by the other keywords listed in this table.
debug	Displays the type of debugging information that is enabled by using the **debug vsp** command.
session	Displays cumulative statistics about active VSP sessions.
statistics	Displays statistics about active VSP sessions, including memory statistics.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.2(11)T	This command was introduced on the Cisco 3640, Cisco 3660, Cisco AS5300, Cisco AS5350, and Cisco AS5400.

Usage Guidelines

Use the **clear vsp statistics** command to reset the counters to 0 for the **show vsp** command.

Examples

The following is sample output from the **show vsp debug** command:

```
Router# show vsp debug
         VSP:<1>[0x62291660](0x62291660) debug_flag=0x7FF
```

The following is sample output from the **show vsp session** command:

```
Router# show vsp session
VSP_STATS:Session Statistics -
        sessions total=0; max_active=0, current=0
        session_duration last=0; max=0, min=0 ms
        pre_stream_wait last=0; max=0, min=0 ms
        stream_duration last=0; max=0, min=0 ms
        post_stream_wait last=0; max=0, min=0 ms
        stream_size last=0; max=0, min=0 bytes
        streaming_rate last=0; max=0, min=0 bytes/sec
```

```
        total_packet_count last=0; max=0, min=0 packets
        drop_packet_count last=0; max=0, min=0 packets
        particle_packet_count last=0; max=0, min=0 packets
```

The following is sample output from the **show vsp statistics** command:

```
Router# show vsp statistics
VSP_STATS:Session Statistics -
        sessions total=0; max_active=0, current=0
        session_duration last=0; max=0, min=0 ms
        pre_stream_wait last=0; max=0, min=0 ms
        stream_duration last=0; max=0, min=0 ms
        post_stream_wait last=0; max=0, min=0 ms
        stream_size last=0; max=0, min=0 bytes
        streaming_rate last=0; max=0, min=0 bytes/sec
        total_packet_count last=0; max=0, min=0 packets
        drop_packet_count last=0; max=0, min=0 packets
        particle_packet_count last=0; max=0, min=0 packets
VSP_STATS: Format Statistics -
        au_format_count=20
        wav_format_count=3
        other_format_count=0
VSP_STATS: Codec Statistics -
        codec_g729_count=4
        codec_g726_count=10
        codec_g711_count=0
        codec_g728_count=2
        codec_g723_count=5
        codec_gsm_count=2
        codec_other_count=0
VSP_STATS: Media Statistics -
        ram_count=23
        http_count=0
        smtp_count=0
        rtsp_count=0
        other_count=0
VSP_STATS:RTP Statistics -
        ts_gap_samples max=76800, min=80 samples
        [Unexpected SSRC Change (USC)]
                usc_count last=0; total=0, max=0, min=0
        [Out of sequence packet (OOSP)]
                oosp_count last=0; total=0, max=0, min=0
        [Unexpected timestamp gap (UTG)]
                max_utg_count last=0; total=0, max=0, min=0
        [Comfort Noise (CN)]
                max_cn_count last=4; total=70, max=8, min=4
        [Unexpected payload type or size (UPTS)]
                upt_count last=0; total=0, max=0, min=0; last_type=0
                ups_count last=0; total=198, max=61, min=0; last_size=2 bytes
        [Data exceeds limit (DEL)]
                del_count last=0; total=2, max=1, min=0
        [Silence exceeds timeout (SET)]
                set_count last=0; total=0, max=0, min=0
VSP_STATS:Packet Statistics -
        [Silence patching total (SPT)]
                spt_count last=296; total=7230, max=889, min=290
        [Concealment patching total (CPT)]
                cpt_count last=0; total=34, max=18, min=0
        [Normal patching total (NPT)]
                npt_count last=171; total=4249, max=453, min=106
```

The table below describes the fields shown in this output.

Table 12: show vsp statistics Field Descriptions

Field	Description
Session Statistics	

Field	Description
sessions total; max_active, current	Total number of VSP sessions since router startup or since the **clear vsp statistics** command was used. The active value should always be 0.
session_duration last; max, min	Duration of the last (most recent) session, and of the longest and shortest sessions in msecs.
pre_stream_wait last; max, min	Msecs that elapsed before the arrival of the first packet. Values are shown for last session, and for the session with the longest and shortest waits.
stream_duration last; max, min	Msecs between first packet arrival and last packet flush. Values are shown for last session, and for the session with the longest and shortest durations.
post_stream_wait last; max, min	Msecs between last packet flush and close of session.
stream_size last; max, min	Data streaming size.
streaming_rate last; max, min	Data streaming rate.
total_packet_count last; max, min	Total packets processed.
drop_packet_count last; max, min	Total packets dropped. The difference between the total packet count and packets dropped is the number of packets that have been accepted.
particle_packet_count last; max, min	Total particle packets processed.
Format Statistics	
au_format_count	Number of VSP sessions that used audio files in .au format.
wav_format_count	Number of VSP sessions that used audio files in .wav format.
other_format_count	Number of VSP sessions that used audio files of an unknown format.
Codec Statistics	
codec_g729_count	Number of VSP sessions that used the G.729 codec.
codec_g726_count	Number of VSP sessions that used the G.726 codec.
codec_g711_count	Number of VSP sessions that used the G.711 codec.
codec_g728_count	Number of VSP sessions that used the G.728 codec.

Field	Description
codec_g723_count	Number of VSP sessions that used the G.723 codec.
codec_gsm_count	Number of VSP sessions that used the GSM codec.
codec_other_count	Number of VSP sessions that used an unknown codec.
Media Statistics	
ram_count	Total number of RAM recordings and playouts.
http_count	Total number of HTTP recordings and playouts.
smtp_count	Total number of SMTP recordings.
rtsp_count	Total number of RTSP recordings and playouts.
other_count	Should always be 0.
RTP Statistics	
ts_gap_samples max min	Permissible timestamp gap in samples.
[Unexpected SSRC Change (USC)]	
usc_count last; total, max, min	Number of times that the source of the streaming has changed.
[Out of sequence packet (OOSP)]	
oosp_count last; total, max, min	Number of out-of-sequence packets.
[Unexpected timestamp gap (UTG)]	
max_utg_count last; total, max, min	Number of packets with an unexpected timestamp gap.
[Unexpected payload type or size (UPTS)]	
upt_count last; total, max, min; last_type	Number of comfort noise packets.
ups_count last; total, max, min; last_size	Number of packets with unexpected nonvoice payload sizes.
[Data exceeds limit (DEL)]	
del_count last; total, max, min	Number of times that the total recording size is larger than the preset recording size.
[Silence exceeds timeout (SET)]	

Field	Description
set_count last; total, max, min	Number of times that the timestamp gap is larger than the preset timeout value.
Packet Statistics	
[Silence patching total (SPT)]	
spt_count last; total, max, min	Number of silence packets that have been inserted during recording.
[Concealment patching total (CPT)]	
cpt_count last; total, max, min	Number of concealment packets that have been inserted during recording.
[Normal patching total (NPT)]	
npt_count last; total, max, min	Number of normal packets that have been patched during recording.

Related Commands

Command	Description
clear vsp statistics	Clears the statistics for VSP sessions.

show wsapi

To display information on the Cisco Unified Communication IOS services, including registration, statistics, and route information, use the **show wsapi** command in user EXEC or privileged EXEC mode.

show wsapi{**http-client| http-server| registration**| **registration**{**all| xcc| xcdr| xsvc**}**| svcc route**}

Syntax Description

http-client	Displays the statistics that have been collected on the http client interface.
http-server	Displays the statistics that have been collected on the http server interface.
registration	Displays the currently registered applications on the WSAPI subsystem.
all	Displays all registered applications.
xcc	Displays the applications that are registered to the XCC provider.
xcdr	Displays the applications that are registered to the XCDR provider.
xsvc	Displays the applications that are registered to the XSVC provider.
xsvc route	Displays the internal route information in the XSVC provider.

Command Modes

User EXEC Privileged EXEC

Command History

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

Usage Guidelines

Use this command to display information on the Cisco Unified Communication IOS services.

Examples

The following example shows a sample output from the **show wsapi http-client** command.

```
Router# show wsapi http-client
```

```
WSAPI Outgoing Notify/Solicit Message Statistics
=========================================
wsapi_show_httpc_callback_context_invalid: 0
wsapi_show_httpc_callback_context_error: 0
wsapi_show_httpc_callback_no_reg: 5
wsapi_show_httpc_callback_notify_OK: 85
wsapi_show_httpc_callback_notify_error: 0
wsapi_show_httpc_callback_client_error: 0
wsapi_show_httpc_callback_error: 7
wsapi_show_httpc_callback_client_error: 0
wsapi_show_httpc_callback_decode_error: 28
wsapi_show_httpc_callback_no_txID: 0
wsapi_show_httpc_callback_OK: 655
wsapi_show_httpc_create_msg_error: 0
wsapi_show_httpc_context_active: 0
wsapi_tx_context_freeq depth: 4
```

The following example shows a sample output from the **show wsapi http-server** command.

```
Router# show wsapi http-server

WSAPI Incoming Request Message Statistics
=========================================
wsapi_show_https_urlhook: 23
wsapi_show_https_post_action: 23
wsapi_show_https_post_action_fail: 0
wsapi_show_https_xml_fault: 0
wsapi_show_https_post_action_done: 23
wsapi_show_https_service_timeout: 0
wsapi_show_https_send_error: 0
wsapi_show_https_invalid_context: 0
wsapi_show_https_data_active: 0
wsapi_https_data_q depth: 1
wsapi_show_https_internal_service_error: 0
wsapi_show_https_service_unavailable_503: 0
wsapi_show_https_not_found_404: 0
wsapi_show_https_registration_success: 9
wsapi_show_https_not_registered: 0
wsapi_show_https_registration_auth_fail: 1
wsapi_show_https_registration_fail: 0
wsapi_show_https_un_registered: 0
```

The following example shows a sample output from the **show wsapi registration** command.

```
Router# show wsapi registration

Provider XCC
=====================================================
registration
id: 4FA11CC:XCC:myapp:5
appUrl:http://sj22lab-as2:8090/xcc
appName: myapp
provUrl: http://10.1.1.1:8090/cisco_xcc
prober state: STEADY
connEventsFilter:
CREATED|AUTHORIZE_CALL|ADDRESS_ANALYZE|REDIRECTED|ALERTING|CONNECTED|TRANSFERRED|CALL_DELIVERY|DISCONNECTED|HANDOFF_JOIN|HANDOFF_LEAVE
mediaEventsFilter:
DTMF|MEDIA_ACTIVITY|MODE_CHANGE||TONE_DIAL|TONE_OUT_OF_SERVICE|TONE_RINGBACK|TONE_SECOND_DIAL
blockingEventTimeoutSec: 1
blockingTimeoutHandle: CONTINUE_PROCESSING

Provider XSVC
=====================================================
registration index: 2
id: 4FA0F8C:XSVC:myapp:3
appUrl:http://sj22lab-as2:8090/xsvc
appName: myapp
provUrl: http://10.1.1.1:8090/cisco_xsvc
prober state: STEADY
route filter:
event filter: off
```

```
Provider XCDR
=====================================================
registration index: 1
id: 4FA10A0:XCDR:myapp:1
appUrl:http://sj22lab-as2:8090/xcdr
appName: myapp
provUrl: http://10.1.1.1:8090/cisco_xcdr
prober state: STEADY
cdr format: COMPACT
event filter: off
```

The following example shows a sample output from the **show wsapi xsvc route** command.

```
Router# show wsapi xsvc route

Route SANJOSE_SIP
=====================================================
Type: VOIP
Description: OUT
Filter:
Trunk:
Trunk Name: 1.3.45.2
Trunk Type: SIPV2
Trunk Status: UP
Route SANJOSE_PRI
=====================================================
Type: PSTN
Description: IN
Filter:
Trunk:
Trunk Name: Se0/1/0:23
Trunk Type: ISDN PRI
Trunk Status: UP
Total channels 2
Channel bitmap 0x01FFFFFE 1-24
Link bitmap 0x00000006
Alarm 0x00000001
Time elapsed 516
Interval 92
CurrentData
0 Line Code Violations, 0 Path Code Violations
0 Slip Secs, 0 Fr Loss Secs, 0 Line Err Secs, 0 Degraded Mins
0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 0 Unavail Secs
TotalData
49 Line Code Violations, 7 Path Code Violations,
0 Slip Secs, 1 Fr Loss Secs, 1 Line Err Secs, 0 Degraded Mins,
0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 2 Unavail Secs
Trunk Name: Se0/1/1:23
Trunk Type: ISDN PRI
Trunk Status: UP
Total channels 2
Channel bitmap 0x01FFFFFE 1-24
Link bitmap 0x00000006
Alarm 0x00000001
Time elapsed 516
Interval 92
CurrentData
0 Line Code Violations, 0 Path Code Violations
0 Slip Secs, 0 Fr Loss Secs, 0 Line Err Secs, 0 Degraded Mins
0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 0 Unavail Secs
TotalData
42 Line Code Violations, 4 Path Code Violations,
0 Slip Secs, 1 Fr Loss Secs, 1 Line Err Secs, 0 Degraded Mins,
0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 2 Unavail Secs
```

Related Commands

Command	Description
provider	Enables a Cisco Unified Communicatoins IOS service provider.

show xcsp port

To display the status of a router port under the control of the external control service provider (XCSP) subsystem, use the **show xcsp port**command in privileged EXEC mode.

show xcsp port *slot-num port-num*

Syntax Description

slot -num	Slot number of the interface card. Values are as follows: • Cisco AS5350: From 0 to 3. • Cisco AS5400: From 0 to 7. • Cisco AS5850: From 0 to 5 and from 8 to 13. Slots 6 and 7 are reserved for the route switch controller (RSC).
port -num	Port number of the interface card. Values are as follows: • Cisco AS5350: For T1/E1, from 0 to 7. For T3, from 1 to 28. • Cisco AS5400: For T1/E1, from 0 to 7. For T3, from 1 to 28. • Cisco AS5850: For T1/E1, from 0 to 23. For T3, from 1 to 28.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(2)XB	This command was introduced.
12.2(11)T	The command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco AS5850.

Examples

The following is sample output from this command:

```
Router# show xcsp port 1 0
Slot 1 configured
Number of ports configured=1 slot state= Up
===================================================
```

```
Port 0 State= Up type = 5850 24 port T1
Channel states
  0  Idle
  1  Idle
  2  Idle
  3  Idle
  4  Idle
  .
  .
  .
  22  Idle
  23  Idle
```

The table below describes significant fields in this output.

Note To get the field description output, you must enter the *slot-num* and *port-num* arguments for the **show xcsp port** command.

Table 13: show xcsp port Field Descriptions

Field	Descriptions
Port	Port number. Range is from 1 to 28.
State	Port state; can be Up or Down.
type	T1 or E1 ports on the AS5400: 8. T1 or E1 ports on the AS5850: 24. T3 ports on the AS5400 and AS5850: 28.

Field	Descriptions
Channel states	Channel states. Values are as follows: • Blocked • Connection in progress • Cot Check In Progress • Cot Check Pending • Down • Idle • In Release in progress • In Use • Invalid • Loopback • Not Present • Out of Service • Out Release in progress • Playing Tone • Shutdown

Related Commands

Command	Description
show xcsp slot	Displays the status of XCSP slots.

show xcsp slot

To display the status of a router slot under the control of the external control service provider (XCSP) subsystem, use the **show xcsp slot** command in privileged EXEC mode.

show xcsp slot *slot-num*

Syntax Description

slot -num	The slot number of the T1 or E1 interface card. Values are as follows: • Cisco AS5350: From 0 to 3. • Cisco AS5400: From 1 to 7. • Cisco AS5850: From 0 to 5 and from 8 to 13. Slots 6 and 7 are reserved for the route switch controller (RSC).

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(2)XB	This command was introduced.
12.2(11)T	The command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco AS5850.

Examples

The following is sample output from this command:

```
Router# show xcsp slot 1
Slot 1 configured
Number of ports configured=1 slot state= Up
```

The table below describes significant fields shown in this output.

Table 14: show xcsp slot Field Descriptions

Field	Description
slot state	Slot state; can be either Up or Down.

Related Commands

Command	Description
show xcsp port	Displays the status of XCSP ports.

shut

To shut down a set of digital signal processors (DSPs) on the Cisco 7200 series router, use the **shut** command in DSP configuration mode. To put DSPs back in service, use the **no** form of this command.

shut *number*

no shut *number*

Syntax Description

number	Number of DSPs to be shut down.

Command Default

No shut

Command Modes

DSP configuration

Command History

Release	Modification
12.0(5)XE	This command was introduced on the Cisco 7200 series.
12.1(1)T	This command was modified to add information about DSP groups.

Usage Guidelines

This command applies to VoIP on the Cisco 7200 series routers.

Examples

The following example shuts down two sets of DSPs:

```
shut 2
```

shutdown (Annex G neighbor)

To disable the service relationships requirement for border elements, use the **shutdown** command in config-nxg-neigh-srvc mode. To enable the service relationship for border elements, use the **no**form of this command.

shutdown

no shutdown

Syntax Description

This command has no arguments or keywords.

Command Default

The Annex G neighbor is shut down.

Command Modes

Annex G neighbor service (config-nxg-neigh-svc)

Command History

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

Usage Guidelines

The **no shutdown** command verifies that a domain name has been configured and ensures that the border element has been configured to reject messages from unknown "stranger" border elements.

Examples

The following example enables the border element:

Router(config-nxg-neigh-srvc)# **no shutdown**

Related Commands

Command	Description
access -policy	Requires that a neighbor be explicitly configured.
inbound ttl	Sets the inbound time-to-live value.
outbound retry -interval	Defines the retry period for attempting to establish the outbound relationship between border elements.
retry interval	Defines the time between delivery attempts.
retry window	Defines the total time that a border element attempts delivery.

shutdown (Annex G)

To shut down the Annex G border element (BE), use the **shutdown** command in Annex G configuration mode. To reinstate the Annex G BE, use the no form of this command.

shutdown

no shutdown

Syntax Description

This command has no arguments or keywords.

Command Default

The Annex G border element is not shut down.

Command Modes

Annex G configuration (config-annexg)

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T. 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(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

While the Annex G BE is in shutdown state, all Annex G messages received from neighbors are ignored and the colocated gatekeeper does not use the Annex G BE for address resolution.

Examples

The following example shuts the BE down:

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

Related Commands

Command	Description
call -router	Enables the Annex G border element configuration commands.
show call -router status	Displays the Annex G BE status.

shutdown (dial-peer)

To change the administrative state of the selected dial peer from up to down, use the **shutdown** command in dial-peer configuration mode. To change the administrative state of this dial peer from down to up, use the **no** form of this command.

shutdown

no shutdown

Syntax Description

This command has no arguments or keywords.

Command Default

No shutdown

Command Modes

Dial-peer configuration (config-dial-peer)

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
12.1(1)	This command was modified for store-and-forward fax.

Usage Guidelines

When a dial peer is shut down, you cannot initiate calls to that peer.

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

Examples

The following example changes the administrative state of voice telephony (plain old telephone service [POTS]) dial peer 10 to down:

```
dial-peer voice 10 pots
shutdown
```

The following example changes the administrative state of voice telephony (POTS) dial peer 10 to up:

```
dial-peer voice 10 pots
no shutdown
```

Related Commands

Command	Description
dial -peer voice	Enters dial-peer configuration mode, defines the type of dial peer, and defines the dial-peer tag number.

shutdown (DSP Farm profile)

To disable the digital signal processor (DSP) farm profile, use the **shutdown** command in DSP farm profile configuration mode. To allocate DSP farm resources and associate with the application, use the **no** form of this command.

shutdown

no shutdown

Syntax Description

This command has no arguments or keywords.

Command Default

Disabled

Command Modes

DSP farm profile configuration (config-dspfarm-profile)

Command History

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

Usage Guidelines

It is essential that the profile be disabled by using the **shutdown** command before a DSP farm profile is updated.

Examples

The following example allocates DSP farm resources and associates with the application:

```
Router(config-dspfarm-profile)#
 no shutdown
```

Related Commands

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

shutdown (gatekeeper)

To disable the gatekeeper, use the **shutdown** command in gatekeeper configuration mode. To enable the gatekeeper, use the **no** form of this command.

shutdown

no shutdown

Syntax Description

This command has no arguments or keywords.

Command Default

Disabled (shut down)

Command Modes

Gatekeeper configuration (config-gk)

Command History

Release	Modification
11.3(2)NA	This command was introduced on the Cisco 2500 series and Cisco 3600 series.
12.0(3)T	The command was integrated into Cisco IOS Release 12.0(3)T and implemented on the Cisco MC3810.

Usage Guidelines

The gatekeeper does not have to be enabled before you can use the other gatekeeper configuration commands. In fact, it is recommended that you complete the gatekeeper configuration before bringing up the gatekeeper because some characteristics may be difficult to alter while the gatekeeper is running, as there may be active registrations or calls.

The no shutdown command enables the gatekeeper, but it does not make the gatekeeper operational. The two exceptions to this are as follows:

- If no local zones are configured, a **no shutdown** command places the gatekeeper in INACTIVE mode waiting for a local zone definition.
- If local zones are defined to use an HSRP virtual address, and the HSRP interface is in STANDBY mode, the gatekeeper goes into HSRP STANDBY mode. Only when the HSRP interface is ACTIVE does the gatekeeper go into the operational UP mode.

Examples

The following command disables a gatekeeper:

```
shutdown
```

Related Commands

Command	Description
shutdown (gateway)	Shuts down all VoIP call service on a gateway.

shutdown (gateway)

To shut down all VoIP call service on a gateway, use the **shutdown**commandin voice service configuration mode. To enable VoIP call service, use the **no** form of this command.

shutdown [forced]

no shutdown

Syntax Description

forced	(Optional) Forces the gateway to immediately terminate all in-progress calls.

Command Default

Call service is enabled

Command Modes

Voice service configuration (config-voi-serv)

Command History

Release	Modification
12.3(1)	This command was introduced.

Examples

The following example shows VoIP call service being shut down on a Cisco gateway:

```
voice service voip
shutdown
```

The following example shows VoIP call service being enabled on a Cisco gateway:

```
voice service voip
no shutdown
```

Related Commands

Command	Description
shutdown (gatekeeper)	Disables the gatekeeper.

shutdown (mediacard)

To disable a selected media card, use the **shutdown** command in mediacard configuration mode. To enable a selected media card, use the **no** form of this command.

shutdown

no shutdown

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

Media card configuration

Command History

Release	Modification
12.3(8)XY	This command was introduced on the Communication Media Module.
12.3(14)T	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(3)	This command was integrated into Cisco IOS Release 12.4(3).

Usage Guidelines

Use the **no shutdown** command at the end of media card configuration. If there are any active connections when you disable the media card, the Digital Signal Processor Resource Manager (DSPRM) displays a warning message indicating that the DSP resources allocated on other media cards for some of the resource pool in this media card will be removed or that there are active connections available in this resource pool and prompts you for a response. Profiles that use resources on this card must be brought up separately after using this command.

Examples

The following example shows how to enable a media card:

```
no shutdown
```

Related Commands

Command	Description
resource-pool	Creates a DSP resource pool on the selected media card.

shutdown (auto-config application)

To disable an auto-configuration application for download, use the **shutdown** command in auto-config application configuration mode. To enable an auto-configuration application for download, use the **no**form of this command.

shutdown

no shutdown

Syntax Description

This command has no keywords or arguments.

Command Default

Disabled

Command Modes

Auto-config application configuration (auto-config-app)

Command History

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

Examples

The following example shows the **shutdown** command used to enable an auto-configuration application for download:

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

Related Commands

Command	Description
auto-config	Enables auto-configuration or enters auto-config application configuration mode for the SCCP application.
show auto-config	Displays the current status of auto-configuration applications.

shutdown (RLM)

To shut down all of the links under the RLM group, use the **shutdown** command in RLM configuration mode. RLM does not try to reestablish those links until the command is negated. To disable this function, use the **no** form of this command.

shutdown

no shutdown

Syntax Description

This command has no arguments or keywords.

Command Default

Disabled

Command Modes

RLM configuration

Command History

Release	Modification
11.3(7)	This command was introduced.

Related Commands

Command	Description
clear interface	Resets the hardware logic on an interface.
clear rlm group	Clears all RLM group time stamps to zero.
interface	Defines the IP addresses of the server, configures an interface type, and enters interface configuration mode.
link (RLM)	Specifies the link preference.
protocol rlm port	Reconfigures the port number for the basic RLM connection for the whole rlm-group.
retry keepalive	Allows consecutive keepalive failures a certain amount of time before the link is declared down.
server (RLM)	Defines the IP addresses of the server.
show rlm group statistics	Displays the network latency of the RLM group.
show rlm group status	Displays the status of the RLM group.

Command	Description
show rlm group timer	Displays the current RLM group timer values.
timer	Overwrites the default setting of timeout values.

shutdown (settlement)

To deactivate the settlement provider, use the shutdown command in settlement configuration mode. To activate a settlement provider, use the no **shutdown** command.

shutdown

no shutdown

Syntax Description

This command has no arguments or keywords.

Command Default

The default status of a settlement provider is deactivated. The settlement provider is down.

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced on the Cisco 2500 series, Cisco 3600 series, and Cisco AS5300.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

Use this command at the end of the configuration of a settlement server to bring up the provider. This command activates the provider. Otherwise, transactions do not go through the provider to be audited and charged. Use the shutdown command to deactivate the provider.

Examples

The following example enables a settlement server:

```
settlement 0
 no shutdown
```

The following example disables a settlement server:

```
settlement 0
 shutdown
```

Related Commands

Command	Description
connection -timeout	Configures the time that a connection is maintained after completing a communication exchange.
customer -id	Identifies a carrier or ISP with a settlement provider.

Command	Description
device -id	Specifies a gateway associated with a settlement provider.
encryption	Sets the encryption method to be negotiated with the provider.
max -connection	Sets the maximum number of simultaneous connections to be used for communication with a settlement provider.
response -timeout	Configures the maximum time to wait for a response from a server.
retry -delay	Sets the time between attempts to connect with the settlement provider.
session -timeout	Sets the interval for closing the connection when there is no input or output traffic.
settlement	Enters settlement configuration mode and specifies the attributes specific to a settlement provider.
type	Configures an SAA-RTR operation type.

shutdown (voice-port)

To take the voice ports for a specific voice interface card offline, use the **shutdown** command in voice-port configuration mode. To put the ports back in service, use the **no** form of this command.

shutdown

no shutdown

Syntax Description

This command has no arguments or keywords.

Command Default

Shutdown

Command Modes

Voice-port configuration (config-voiceport)

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
12.4(22)T	Support for IPv6 was added.

Usage Guidelines

When you use this command, all ports on the voice interface card are disabled. When you use the **no**form of the command, all ports on the voice interface card become enabled. A telephone connected to an interface hears silence when a port is shut down.

Examples

The following example takes voice port 1/1/0 offline:

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

Related Commands

Command	Description
shutdown (port)	Disables a port.