show vdev

To display information about the digital signal processors (DSPs) on a specific card, use the **show vdev** command in privileged EXEC mode.

show vdev {*slot*/*port*}

Syntax Description

slot	Slot in which the voice card resides.
port	Port on the voice card.

Command Default

No default behavior or values.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

This command can be used on the standby and active route switch controller (RSC) to verify that dynamic and bulk synchronization have been performed correctly on a specified port.

Examples

The following example shows the output for the last port on a 324 universal port card.

```
Router# show vdev 2/323

flags = 0x0000
dev_status = 0x0000
service = 0x0000
service_type = 0x0
min_speed = 0, max_speed = 0
modulation = 0, err_correction = 0, compression = 0
csm_call_info = 0x0, csm_session = Invalid
vdev_p set to modem_info

DSPLIB information:
dsplib_state = 0x0
dsplib_next_action = 0x0

HDLC information:
call_id = 0x0
called_number =
speed = 0
ces = 0x0
spc = FALSE
d_idb = 0x0

Bulk sync reference = 2, Global bulk syncs = 2
```

Table 186 displays significant fields shown in the output.

Table 186 show vdev Field Descriptions

Field	Description
flags	Internal vdev flags
dev_status	Additional flags giving status of the resource
service	Service currently running on this DSP
service_type	Service type as passed in by RPM
min_speed	Minimum configured modem speed
max_speed	Maximum configured modem speed
modulation	Maximum modulation to be negotiated
err_correction	Error correction to be negotiated
compression	Compression to be negotiated
csm_call_info	Address of the associated csm_call_info structure
csm_session	Session ID as maintained by CSM
vdev_p	Address of the associated resource structure
dsplib_state	State of the resource as seen by the DSPLIB
dsplib_next_action	Next DSPLIB action that should be taken on this resource
call_id	Call identifier if this resource has a HDLC call
called_number	Called number if this resource has a HDLC call
speed	Speed of the connection if this resource has a HDLC call
ces	Circuit emulation service information
spc	True if semi permanent call link
d_idb	Address of the associated D channel idb, if this resource has a HDLC call
Bulk sync reference	Number of times that this resource has been bulk synchronized
Global bulk syncs	Number of bulk synchronizations that the VDEV High Availability client has performed

Related Commands

Command	Description
debug vdev	Turns on debugging for voice devices.
show redundancy	Displays current or historical status and related information on a redundant RSC.

show vfc

To see the entries in the host-name-and-address cache, use the **show vfc** command in privileged EXEC mode.

show vfc *slot-number* [**technology**]

Syntax Description

slot-number	VFC slot number.
technology	(Optional) Displays the technology type of the VFC.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced on the Cisco AS5300.
12.0(2)XH	The **technology** keyword was added.

Examples

The following is sample output from this command showing that the card in slot 1 is a C549 DSPM:

```
Router# show vfc 1 technology

Technology in VFC slot 1 is C549
```

Field descriptions should be self-explanatory.

Related Commands

Command	Description
voice-card	Configures a voice card and enters voice-card configuration mode.

show vfc cap-list

To show the current list of files on the capability list for this voice feature card (VFC), use the **show vfc cap-list** command in user EXEC mode.

show vfc *slot* **cap-list**

Syntax Description

slot	Slot where the VFC is installed. Range is from 0 to 2.

Command Modes

User EXEC

Command History

Release	Modification
11.3 NA	This command was introduced on the Cisco AS5300.

Examples

The following is sample output from this command:

```
Router# show vfc 1 cap-list

Capability List for VFC in slot 1:
1. fax-vfc-l.0.1.bin
2. bas-vfc-l.0.1.bin
3. cdc-g729-l.0.1.bin
4. cdc-g711-l.0.1.bin
5. cdc-g726-l.0.1.bin
6. cdc-g728-l.0.1.bin
7. cdc-gsmfr-l.0.1.bin
```

The first line in this output is a general description, stating that this is the capability list for the VFC residing in slot 1. Below this is a numbered list, each line of which identifies one currently installed in-service file.

Related Commands

Command	Description
show vfc default-file	Displays the default files included in the default file list for this VFC.
show vfc directory	Displays the list of all files residing on this VFC.
show vfc version	Displays the version of the software residing on this VFC.

show vfc default-file

To show the default files included in the default file list for a voice feature card (VFC), use the **show vfc default-file** command in user EXEC mode.

show vfc *slot* **default-file**

Syntax Description

slot	Slot where the VFC is installed. Range is from 0 to 2.

Command Modes

User EXEC

Command History

Release	Modification
11.3 NA	This command was introduced on the Cisco AS5300.

Examples

The following is sample output from this command:

```
Router# show vfc 1 default-file

Default List for VFC in slot 1:
1. btl-vfc-l.0.13.0.bin
2. cor-vfc-l.0.1.bin
3. bas-vfc-l.0.1.bin
4. cdc-g729-l.0.1.bin
5. fax-vfc-l.0.1.bin
6. jbc-vfc-l.0.13.0.bin
```

The first line in this output is a general description, stating that this is the default list for the VFC residing in slot 1. Below this is a numbered list, each line of which identifies one default file.

Related Commands

Command	Description
show vfc cap-list	Displays the current list of files on the capability list for this VFC.
show vfc directory	Displays the list of all files residing on this VFC.
show vfc version	Displays the version of the software residing on this VFC.

show vfc directory

To show the list of all files residing on a voice feature card (VFC), use the **show vfc directory** command in user EXEC mode.

show vfc *slot* **directory**

Syntax Description

slot	Slot where the VFC is installed. Range is from 0 to 2.

Command Modes

User EXEC

Command History

Release	Modification
11.3 NA	This command was introduced on the Cisco AS5300.

Usage Guidelines

Use this command to display a list of all of the files currently stored in Flash memory for a particular VFC.

Examples

The following is sample output from this command:

```
Router# show vfc 1 directory

Files in slot 1  VFC flash:
     File Name                        Size (Bytes)
1 .  vcw-vfc-mz.gsm.VCW               292628
2 .  btl-vfc-1.0.13.0.bin             4174
3 .  cor-vfc-1.0.1.bin                54560
4 .  jbc-vfc-1.0.13.0.bin             16760
5 .  fax-vfc-1.0.1.bin                64290
6 .  bas-vfc-1.0.1.bin                54452
7 .  cdc-g711-1.0.1.bin               190
8 .  cdc-g729-1.0.1.bin               21002
9 .  cdc-g726-1.0.1.bin               190
10.  cdc-g728-1.0.1.bin               22270
11.  cdc-gsmfr-1.0.1.bin              190
```

Table 187 describes significant fields in this output.

Table 187 show vfc directory Field Descriptions

Field	Description
File Name	Name of the file stored in Flash memory.
Size (Bytes)	Size of the file in bytes.

Related Commands

Command	Description
show vfc cap-list	Displays the current list of files on the capability list for this VFC.
show vfc default-file	Displays the default files included in the default file list for this VFC.
show vfc version	Displays the version of the software residing on this VFC.

show vfc version

To show the version of the software residing on a voice feature card (VFC), use the **show vfc version** command in user EXEC mode.

show vfc *slot* **version** {**dspware** | **vcware**}

Syntax Description

slot	Slot where the VFC is installed. Range is from 0 to 2.
dspware	Which DSPWare software to display.
vcware	Which VCWare software to display.

Command Modes

Privileged or user EXEC

Command History

Release	Modification
11.3 NA	This command was introduced on the Cisco AS5300.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T with changes to the command output.

Usage Guidelines

Use this command to display the version of the software currently installed in Flash memory on a VFC.

Examples

The following is sample output from this command:

```
Router# show vfc 0 version dspware

Version of Dspware in VFC slot 0 is 0.10
```

The output from this command is a simple declarative sentence stating the version number for the selected type of software (in this example, DSPWare) for the VFC residing in the selected slot number (in this example, slot 0).

Cisco IOS Release 12.2(13)T adds new information to the output of the **show vfc** *slot* **version vcware** and **show vfc** *slot* **version dspware** commands. Messages are output if the Cisco VCWare or DSPWare is not compatible with the Cisco IOS image. The new information is advisory only, so there is no action taken if the software is compatible or incompatible.

If the versions detected fall within the defined criteria and are compatible, nothing is output at bootup time. A confirmation line is output when the **show vfc version vcware** and **show vfc version dspware** commands are used:

```
Router# show vfc 1 version vcware

Voice Feature Card in Slot 1:
VCWare Version     : 7.35
ROM Monitor Version: 1.3
     DSPWare Version    : 3.4.46L
     Technology         : C549
VCWare/DSPWare version compatibility OK
```

Table 188 shows output field descriptions for the **show vfc version vcware** command with compatible firmware.

Table 188 show vfc version vcware Field Descriptions

Field	Description
Voice Feature Card in Slot	Slot in which the VFC is installed.
VCWare Version	Cisco VCWare version. Version 7.35 is the required minimum for Cisco IOS Release 12.2(11)T and higher.
ROM	ROM monitor version shows 1.3.
DSPWare Version	The DSPWare version shows 3.4.46L, which is the required minimum for Cisco IOS Release 12.2(11)T and higher.
Technology	The technology shows C549. C549 technology is available to support either medium-complexity codecs or high-complexity codecs.
VCWare/DSPWare version compatibility	The Cisco VCWare and DSPWare versions are compatible with Cisco IOS software. Cisco VCWare/DSPWare version compatibility is either OK or shows a mismatch. **Note** This option is available only with Cisco IOS Release 12.2(10) mainline and later release or Cisco IOS Release 12.2(11)T and later.

The following is sample outpou from this command.

```
Router# show vfc 1 version dspware

DSPWare version in VFC slot 1 is 3.4.46L
VCWare/DSPWare version compatibility OK
```

Table 189 shows output field descriptions for the **show vfc version dspware** command with compatible firmware.

Table 189 show vfc version dspware Field Descriptions

Field	Description
Voice Feature Card in Slot	Slot in which the VFC is installed.

Table 189 show vfc version dspware Field Descriptions (continued)

Field	Description
DSPWare Version	The DSPWare version shows 3.4.46L, which is the required minimum for Cisco IOS Release 12.2(10)T and higher.
VCWare/DSPWare version compatibility	The Cisco VCWare and DSPWare versions are compatible with Cisco IOS software. Cisco VCWare/DSPWare version compatibility is either OK or shows a mismatch. **Note** This option is available only with Cisco IOS Release 12.2(10) mainline and later or 12.2(11)T and later.

If the found versions are out of range or otherwise mismatched, a representative message is output when you boot up the router or is appended to the output of the **show vfc version vcware** and **show vfc version dspware** commands. Other than the output of these messages, the version check has no other effect, and the software functions normally. The following is an example of when a found version is out of range or mismatched at bootup:

```
...
Firmware version mismatch for bundle AS5300 VCWare
  - version found (6.04) is lower than minimum required (7.35)
Firmware version mismatch for bundle AS5300 C549
  - version found (3.3.10L) is lower than minimum required (3.4.46L)
```

If you were to enter an explicit request, and the software were incompatible, the following output would be displayed:

```
Router# show vfc 1 version vcware

Voice Feature Card in Slot 1:
VCWare Version     : 6.04
ROM Monitor Version: 1.3
     DSPWare Version    : 3.3.10L
     Technology         : C549
Firmware version mismatch for bundle AS5300 VCWare
  - version found (6.04) is lower than minimum required (7.14)
Firmware version mismatch for bundle AS5300 C549
  - version found (3.3.10L) is lower than minimum required (3.4.26L)

Router# show vfc 1 version dspware

DSPWare version in VFC slot 1 is 3.3.10L
Firmware version mismatch for bundle AS5300 VCWare
  - version found (6.04) is lower than minimum required (7.14)
Firmware version mismatch for bundle AS5300 C549
  - version found (3.3.10L) is lower than minimum required (3.4.26L)
```

Related Commands

Command	Description
show vfc cap-list	Displays the current list of files on the capability list for this VFC.
show vfc default-file	Displays the default files included in the default file list for this VFC.
show vfc directory	Displays the list of all files residing on this VFC.

show video call summary

To display summary information about video calls and the current status of the Video CallManager (ViCM), use the **show video call summary** command in privileged EXEC mode.

show video call summary

Syntax Description

There are no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use this command to quickly examine the status of current video calls. In Cisco IOS Release 12.0(5)XK and Release 12.0(7)T, there can be only one video call in progress.

Examples

The following example displays information about the ViCM when no call is in progress on the serial interface that connects to the local video codec:

```
Router# show video call summary

Serial0:ViCM = Idle, Codec Ready
```

The following output shows a call starting:

```
Router# show video call summary

Serial0:ViCM = Call Connected
```

The following output shows a call disconnecting:

```
Router# show video call summary

Serial0:ViCM = Idle
```

Related Commands

Command	Description
show call history video record	Displays information about video calls.

show voice accounting method

To display connectivity status information for accounting method lists, use the **show voice accounting method** command in privileged EXEC mode.

show voice accounting method [*method-list-name*]

Syntax Description

method-list-name	(Optional) Name of a specific method list. This option displays connectivity status information for a single method list identified by this argument.

Command Default

If no argument is specified, connectivity status information for all accounting method lists is displayed.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use the **show voice accounting method** command to display the history of status (reachable or unreachable), status transition time, and statistics of the accounting status for a specified accounting method list or all the accounting method lists. A maximum of ten status histories are displayed.

Examples

The following is sample output from the **show voice accounting method** command for a specific method list:

```
Router# show voice accounting method ml1

Accounting Method List [ml1]
======================
Current Status:
---------------
unreachable                 [21:52:39 gmt Dec 4 2002]
last record sent time       [23:14:59 gmt Dec 4 2002]
total probe sent out        [84]

Status History:
---------------
(2) unreachable             [21:52:39 gmt Dec 4 2002]
(1) reachable               [21:46:19 gmt Dec 4 2002]
                 SUCCESS                              FAILURE
Record  [Received    | Notified ] [Received    |  Notified   | Reported ]
Type    [from server| to client] [from server|  to client  | to call  ]
------  [-----------|----------] [-----------|------------|----------]
START   [      0     |      0    ] [      0     |      0      |     0    ]
UPDATE  [      0     |      0    ] [      0     |      0      |     0    ]
STOP    [      0     |      0    ] [      84    |      84     |     0    ]
ACCT_ON [      0     |      0    ] [      0     |      0      |     0    ]
------  [-----------|----------] [-----------|------------|----------]
```

```
TOTAL    [       0       |       0     ] [       84      |        84      |       0     ]
```

If there is no status history, as in the following example, no status history is displayed.

```
Router# show voice accounting method

Accounting Method List [ml1]
======================
Current Status:
---------------
reachable                   [21:52:39 gmt Dec 4 2002]
last record sent time       [23:14:59 gmt Dec 4 2002]
total probe sent out        [2]

                 SUCCESS                        FAILURE
Record   [Received    | Notified ] [Received    |  Notified    |  Reported ]
Type     [from server|  to client] [from server|   to client  |  to call  ]
------   [-----------|----------] [-----------|------------|----------]
START    [      0     |      0    ] [      0     |      0      |      0    ]
UPDATE   [      0     |      0    ] [      0     |      0      |      0    ]
STOP     [      0     |      0    ] [      2     |      2      |      0    ]
ACCT_ON  [      0     |      0    ] [      0     |      0      |      0    ]
------   [-----------|----------] [-----------|------------|----------]
TOTAL    [      0     |      0    ] [      2     |      2      |      0    ]
```

Table 190 describes the significant fields shown in the display.

Table 190 show voice accounting method Field Descriptions

Field	Description
Current Status: reachable or unreachable	Current status of the method list: reachable or unreachable and the time (in hh:mm:ss) and date the method list reached this status.
last record sent time	Time (in hh:mm:ss) and date the last accounting record was sent to the method list.
total probe sent out	Number of probe records sent up to the time of the show command.
SUCCESS: Received from server	Number of success status of the accounting records of this type received from the method list.
SUCCESS: Notified to client	Number of success status of the accounting records of this type for which notifications were sent to the GAS.
FAILURE: Received from server	Number of failure status of the accounting records of this type received from the method list.
FAILURE: Notified to client	Number of failure status of the accounting records of this type for which notifications were sent to the GAS.
FAILURE: Reported to call	Number of failure status of the accounting records of this type that were reported to the call application.

Related Commands

Command	Description
clear voice accounting method	Clears accounting status statistics for a particular accounting method list or all accounting method lists.

show voice accounting response pending

To display information regarding pending VoIP AAA accounting responses, use the **show voice accounting response pending** command in privileged EXEC mode.

show voice accounting response pending

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

The following example displays information regarding pending VoIP AAA accounting responses:

```
Router# show voice accounting response pending

Total num of acct sessions waiting for acct responses: 0
Total num of acct start responses pending:             0
Total num of acct interim update responses pending:     0
Total num of acct stop responses pending:              0
```

Table 191 lists and describes the significant output fields.

Table 191 show voice accounting response pending Field Descriptions

Field	Description
Total num of acct sessions waiting for acct responses	Number of accounting sessions that are waiting for accounting responses.
Total num of acct start responses pending	Number of accounting start responses that are pending.
Total num of acct interim update responses pending	Number of accounting interim update responses that are pending.
Total num of acct stop responses pending	Number of accounting stop responses that are pending.

show voice busyout

To display information about the voice-busyout state, use the **show voice busyout** command in privileged EXEC mode.

show voice busyout

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was implemented on the Cisco 2600 series and Cisco 3600 series.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

This command displays the following information:

- Interfaces that are being monitored for busyout events
- Voice ports currently in the busyout state and the reasons

Examples

The following is sample output from this command:

```
Router# show voice busyout

If following network interfaces are down, voice port will be put into busyout state
ATM0
Serial0
The following voice ports are in busyout state

1/1     is forced into busyout state
1/2     is in busyout state caused by network interfaces
1/3     is in busyout state caused by ATM0
1/4     is in busyout state caused by network interfaces
1/5     is in busyout state caused by Serial0
```

Field descriptions should be self-explanatory.

Related Commands

Command	Description
busyout forced	Forces a voice port into the busyout state.
busyout monitor	Places a voice port in the busyout monitor state.
busyout seize	Changes the busyout seize procedure from a voice port.
voice-port busyout	Places all voice ports associated with a serial or ATM interface in a busyout state.

show voice call

To display the call status for voice ports on the Cisco router, use the **show voice call** command in user EXEC or privileged EXEC mode.

Cisco 827, Cisco 1700 Series, and Cisco 7750 with Analog Voice Ports

show voice call [*slot***/***port* | **status** [*call-id*] [**sample** *seconds*] | **summary**]

Cisco 2600, Cisco 3600, Cisco 3700 Series with Analog Voice Ports

show voice call [*slot***/***subunit***/***port* | **status** [*call-id*] [**sample** *seconds*] | **summary**]

Cisco 2600, Cisco 3600, and Cisco 3700 Series with Digital Voice Ports (with T1 Packet Voice Trunk Network Modules)

show voice call [*slot***/***port***:***ds0-group* | **status** [*call-id*] [**sample** *seconds*] | **summary**]

Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5850, Cisco 7200 Series, and Cisco 7500 Series with Digital Voice Ports

show voice call [*slot***/***port***:***ds0-group* | **status** [*call-id*] [**sample** *seconds*] | **summary**]

:

Syntax Description

Cisco 827, Cisco 1700 Series, and Cisco 7750 with Analog Voice Ports	
*slot***/***port*	(Optional) A specific analog voice port: • *slot*—Physical slot in which the analog voice module (AVM) is installed. • **/***port*—Analog voice port number. Range is from 1 to 6. The slash mark is required.
status [*call-id*]	(Optional) Displays status of active calls. If *call-id* is specified, this command displays the status of a specific call.
sample *seconds*	(Optional) Displays status over a specified sampling interval, in seconds. Range is from 1 to 30. Default is 10.
summary	(Optional) Displays current settings and state of the voice port, regardless of port activity.

Cisco 2600 Series, Cisco 3600 Series, Cisco 3700 Series with Analog Voice Ports	
*slot***/***subunit***/***port*	(Optional) A specific analog voice port: • *slot*—Router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • **/***subunit*—Voice interface card (VIC) in which the voice port is located. Valid entries are 0 and 1. (The VIC fits into the voice network module.) The slash mark is required. • **/***port*—Analog voice port number. Valid entries are 0 and 1. The slash mark is required.
status [*call-id*]	(Optional) Displays status of active calls. If *call-id* is specified, this command displays the status of a specific call.

sample *seconds*	(Optional) Displays status over a specified sampling interval, in seconds. Range is from 1 to 30. Default is 10.
summary	(Optional) Displays current settings and state of the voice port, regardless of port activity.

Cisco 2600, Cisco 3600, and Cisco 3700 Series with Digital Voice Ports (with T1 Packet Voice Trunk Network Modules)	
*slot***/***port***:***ds0-group*	(Optional) A specific digital voice port: • *slot*—Router slot in which the packet voice trunk network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • **/***port*—T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. (One VWIC fits in an NM.) The slash mark is required. • **:***ds0-group*—T1 or E1 logical port number. Range is from 0 to 23 for T1 and from 0 to 30 for E1. The colon is required.
status [*call-id*]	(Optional) Displays status of active calls. If *call-id* is specified, this command shows the status of a specific call.
sample *seconds*	(Optional) Displays status over a specified sampling interval, in seconds. Range is from 1 to 30. Default is 10.
summary	(Optional) Displays current settings and state of the DSP port regardless of port activity.

Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5850, Cisco 7200 Series, and Cisco 7500 Series with Digital Voice Ports	
*slot***/***port***:***ds0-group*	(Optional) A specific digital voice port: • *slot*—Router slot in which the packet voice trunk network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • **/***port*—T1 or E1 physical port in the VWIC. Valid entries are 0 and 1. (One VWIC fits in an NM.) The slash mark is required. • **:***ds0-group*—T1 or E1 logical port number. Range is from 0 to 23 for T1 and from 0 to 30 for E1. The colon is required.
status [*call-id*]	(Optional) Displays status of active calls. If *call-id* is specified, this command shows the status of a specific call.
sample *seconds*	(Optional) Displays status over a specified sampling interval, in seconds. Range is from 1 to 30. Default is 10.
summary	(Optional) Displays current settings and state of the voice port regardless of port activity.

Command Modes

User EXEC
privileged EXEC

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was implemented on the Cisco 2600 series and Cisco 3600 series.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.2(13)T	This command was modified with the **status**, *call-id*, and **sample** *seconds* command options. This command is available on all voice platforms.
12.4(3d)	This command was modified to support the Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms for Non-Facility Associated Signaling (NFAS) configuration. Output was modified to provide accurate port information for NFAS configuration on these platforms.
15.1(3)T	This command was modified. The output of this command was enhanced to display the connection status of foreign exchange office (FXO) ports.

Usage Guidelines

This command works on Voice over Frame Relay, Voice over ATM, and Voice over IP by providing the status at the following levels of the call-handling module:

- Call-processing state machine
- End-to-end call manager
- Protocol state machine
- Tandem switch

Note This command is not supported in Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms for NFAS configuration before Cisco IOS Release 12.4(3d).

This command displays call-processing and protocol state-machine information for a voice port if the information is available. This command also shows information on the DSP channel associated with the voice port if the information is available. All real-time information in the DSP channel, such as jitter and buffer overrun, is queried to the DSP channel, and asynchronous responses are returned to the host side.

If no call is active on a voice port, the **show voice call summary** command displays only the VPM (shutdown) state. If a call is active on a voice port, the **show voice call summary** command displays voice telephony service provider (VTSP) state. For an on-net call or a local call without local bypass (not cross-connected), the codec and voice activity detection (VAD) fields are displayed. For an off-net call or a local call with local bypass, the codec and VAD fields are not displayed.

When a call is active on a voice port, the **show voice call summary** command displays the VTSP state. The VTSP state always shows the VTSP signaling state irrespective of the type of call: voice call or a fax call. A fax call does not display S_Fax. The following output is displayed:

```
PORT           CODEC    VAD VTSP STATE            VPM STATE
============== ======== === ==================== ======================
1/0:1.1         1         y  S_CONNECT             EM_CONNECT
```

Note Use the **show voice dsmp stream** command to display the current session of the voice Distributed Stream Media Processor (DSMP) media stream and its related applications.

The **show voice call** command does not display the codec and VAD fields because this information is in the summary display. If you use the **show voice call status** command by itself, an immediate list of all the active calls is shown. You can use the *call-id* argument to request that the DSP associated with the *call-id* be queried for run-time statistics twice, once immediately, and a second time after **sample** *seconds*.

The **sample** *seconds* is the number of seconds over which the status is to be determined. The results of the run-time statistic queries are then analyzed and presented in a one-line summary format.

When a call terminates during the specified sample period, the following output message is returned:

```
CallID call id cannot be queried
CallID call id second sample responses unavailable
```

Note The Voice Call Tuning feature is not supported on the Cisco AS5300.

Examples

The following is sample output from the **show voice call summary** command showing two local calls connected without local bypass:

```
Router# show voice call summary

PORT    CODEC    VAD VTSP STATE            VPM STATE
======= ======== === ===================== ========================
0:17.18                                     *shutdown*
0:18.19 g729ar8   n  S_CONNECT             FXOLS_OFFHOOK
0:19.20                                    FXOLS_ONHOOK
0:20.21                                    FXOLS_ONHOOK
0:21.22                                    FXOLS_ONHOOK
0:22.23                                    FXOLS_ONHOOK
0:23.24                                    EM_ONHOOK
1/1                                        FXSLS_ONHOOK
1/2                                        FXSLS_ONHOOK
1/3                                        EM_ONHOOK
1/4                                        EM_ONHOOK
1/5                                        FXOLS_ONHOOK
1/6     g729ar8   n  S_CONNECT             FXOLS_CONNECT
```

The following is sample output from the **show voice call summary** command showing two local calls connected with local bypass:

```
Router# show voice call summary

PORT    CODEC    VAD VTSP STATE            VPM STATE
======= ======== === ===================== ========================
0:17.18                                     *shutdown*
0:18.19              S_CONNECT             FXOLS_OFFHOOK
0:19.20                                    FXOLS_ONHOOK
0:20.21                                    FXOLS_ONHOOK
0:21.22                                    FXOLS_ONHOOK
0:22.23                                    FXOLS_ONHOOK
0:23.24                                    EM_ONHOOK
1/1                                        FXSLS_ONHOOK
1/2                                        FXSLS_ONHOOK
1/3                                        EM_ONHOOK
1/4                                        EM_ONHOOK
1/5                                        FXOLS_ONHOOK
1/6                  S_CONNECT             FXOLS_CONNECT
```

The following is sample output from the **show voice call summary** command in which the connected FXO port 0/2/0 shows status of "FXOLS_ONHOOK" whereas the FXO port 0/2/1, which is disconnected, shows a status of "FXOLS_BUSYOUT":

```
Router# show voice call summary

PORT           CODEC     VAD VTSP STATE            VPM STATE
============== ========= === ==================== ======================
0/0/0         -          -  -                     FXSLS_ONHOOK
0/0/1         -          -  -                     FXSLS_ONHOOK
0/3/0:23.1      -          -  -
0/3/0:23.2      -          -  -
.
.
.
0/3/0:23.23     -          -  -
0/1/0         -          -  -                     DID_ONHOOK
0/1/1         -          -  -                     DID_ONHOOK
0/2/0         -          -  -                     FXOLS_ONHOOK
0/2/1         -          -  -                     FXOLS_BUSYOUT
2/0/0         -          -  -                     FXSLS_ONHOOK
2/0/1         -          -  -                     FXSLS_ONHOOK
2/0/2         -          -  -                     FXSLS_ONHOOK
2/0/3         -          -  -                     FXSLS_ONHOOK
2/0/4         -          -  -                     FXSLS_ONHOOK
2/0/5         -          -  -                     FXSLS_ONHOOK
2/0/6         -          -  -                     FXSLS_ONHOOK
2/0/7         -          -  -                     FXSLS_ONHOOK
```

Note Beginning in Cisco IOS Release 15.1(3)T, there is improved status monitoring of FXO ports—any time an FXO port is connected or disconnected, a message is displayed to indicate the status change. For example, the following message is displayed to report that a cable has been connected, and the status is changed to "up" for FXO port 0/2/0:

```
000118: Jul 14 18:06:05.122 EST: %LINK-3-UPDOWN: Interface Foreign Exchange Office 0/2/0,
changed state to operational status up due to cable reconnection
```

The following is sample output from the **show voice call summary** command showing one regular PRI port and one NFAS PRI port on a Cisco AS5350, Cisco AS5400, or Cisco AS5850 platform. Port 3/2:D belongs to a regular PRI voice port with time slots 0 and 22. Port Se3/1 belongs to an NFAS PRI voice port with time slots 0,1, and 2 on T1 controller 3/1, which is a member of an NFAS group.

In the case of NFAS on Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms, the port is reported in terms of the serial interface associated with the T1 controller, and the time slot is counted from 0 (for example, 0, 1, 2, 3).

```
Router# show voice call summary

PORT           CODEC    VAD VTSP STATE            VPM STATE
============== ======== === ==================== ======================
3/2:D.0        None      y  S_ALERTING            S_TSP_INCALL
3/2:D.22       None      y  S_ALERTING            S_TSP_INCALL
Se3/1:0        None      y  S_CONNECT             S_TSP_CONNECT
Se3/1:1        None      y  S_CONNECT             S_TSP_CONNECT
Se3/1:2        None      y  S_CONNECT             S_TSP_CONNECT
```

Note The output from the **show voice call summary** command is slightly different in the PORT field on platforms other than the Cisco AS5350, Cisco AS5400, and Cisco AS5850. The contrast between platform types is as follows:

```
Platform       Regular PRI (T1)    NFAS PRI (T1)*
-------------------------------------------------------------
non-AS5xxx     3/0:23.TS           3/1:23.TS
AS5xxx         3/0:D.TS            Ser3/1:(TS-1)
```

* Assumes T1 3/1 is a member of an NFAS group with T1 3/0 as the primary NFAS member, and TS is the time slot counted from a base of 1 (for example 1, 2, 3).

The following is sample output from the **show voice call** command for analog voice ports:

```
Router# show voice call

1/1 vpm level 1 state = FXSLS_ONHOOK
vpm level 0 state = S_UP
1/2 vpm level 1 state = FXSLS_ONHOOK
vpm level 0 state = S_UP
1/3 is shutdown
1/4 vtsp level 0 state = S_CONNECT
vpm level 1 state = S_TRUNKED
vpm level 0 state = S_UP
1/5 vpm level 1 state = EM_ONHOOK
vpm level 0 state = S_UP
1/6 vpm level 1 state = EM_ONHOOK
vpm level 0 state = S_UP

Router# show voice call 1/4

1/4 vtsp level 0 state = S_CONNECT
vpm level 1 state = S_TRUNKED
vpm level 0 state = S_UP
router#    ***DSP VOICE VP_DELAY STATISTICS***
Clk Offset(ms): 1445779863, Rx Delay Est(ms): 95
Rx Delay Lo Water Mark(ms): 95, Rx Delay Hi Water Mark(ms): 125
        ***DSP VOICE VP_ERROR STATISTICS***
Predict Conceal(ms): 10, Interpolate Conceal(ms): 0
Silence Conceal(ms): 0, Retroact Mem Update(ms): 0
Buf Overflow Discard(ms): 20, Talkspurt Endpoint Detect Err: 0
        ***DSP VOICE RX STATISTICS***
Rx Vox/Fax Pkts: 537, Rx Signal Pkts: 0, Rx Comfort Pkts: 0
Rx Dur(ms): 50304730, Rx Vox Dur(ms): 16090, 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 TX STATISTICS***
Tx Vox/Fax Pkts: 567, Tx Sig Pkts: 0, Tx Comfort Pkts: 0
Tx Dur(ms): 50304730, Tx Vox Dur(ms): 17010, Tx Fax Dur(ms): 0
        ***DSP VOICE ERROR STATISTICS***
Rx Pkt Drops(Invalid Header): 0, Tx Pkt Drops(HPI SAM Overflow): 0
        ***DSP LEVELS***
TDM Bus Levels(dBm0): Rx -70.3 from PBX/Phone, Tx -68.0 to PBX/Phone
TDM ACOM Levels(dBm0): +2.0, TDM ERL Level(dBm0): +5.6
TDM Bgd Levels(dBm0): -71.4, with activity being voice
```

The following is sample output from the **show voice call** command for analog voice ports on a Cisco 7200 series. The output includes the DSPfarm, T1 interface, and DS0 or TLM slot configuration:

```
Router# show voice call 6/0:0

6/0:0 1 -         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 2 -         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 3 -         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 4 -         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 5 -         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 6 -         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 7 -         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 8 -         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 9 -         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 10-         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 11-         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 12-         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
```

The following is sample output from the **show voice call status** command on the Cisco 2600 series. You can use this command rather than the **show call active brief** command to obtain the caller ID; the caller ID output of the **show voice call status** command is already in hexadecimal form.

```
Router# show voice call status

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

Using the *call-id* argument is a generic means to identify active calls. If the *call-id* is omitted, the query shows all active voice calls. In the following example, a list of all active calls with relevant identifying information is shown:

```
Router# show voice call status

CallID    CID    ccVdb       Port      DSP/Ch  Called #   Codec    Dial-peers
0x3       11D4   0x62972834  1/0/0     1/1      10001     g711ulaw 1/2
0x4       11D4   0x62973AD0  1/0/1     2/1     *10001     g711ulaw 2/1
0xA       11DB   0x62FE9D68  1/1/0     3/1     *2692      g729r8   0/2692
2 active calls found
```

Note You can query only one call at a time. If you attempt queries from different ports (console and Telnet), and if a query is in progress on another port, the system asks you to wait for completion of that query. You can query any call from anywhere, at anytime, except during the sample interval for an enquiry that is already in progress. This simplifies the implementation significantly and does not reduce the usefulness of the command.

The following example shows echo-return-loss (ERL) reflector information where the call ID is 3 and the sample period is 10 seconds:

```
Router# show voice call status 3 sample 10

Gathering information (10 seconds)...
CallID     Port      DSP/Ch  Codec    Rx/Tx      ERL          Jitter
0x3        1/0/0     1/1     g711ulaw 742/154    5.6          50/15
```

In this example, ERL is the echo return loss (in dB) as reported by the DSP. Jitter values are the current delay and the jitter of the packets around that delay.

If the router is running the extended echo canceller, output looks similar to the following if you enter the same command. The output shows a new value under ERL/Reflctr: the time difference, in ms, between the original signal and the loudest echo (peak reflector) as detected by the echo canceller:

```
Router# show voice call status 3 sample 10

Gathering information (10 seconds)...
CallID     Port      DSP/Ch  Codec    Rx/Tx      ERL/Reflctr Jitter
0x3        1/0/0     1/1     g711ulaw 742/154    5.6/12       50/15
```

The following examples show output using the NextPort version of the standard echo canceller. (Time-slot information is also in the output for digital ports.)

```
Router# show voice call status

CallID      CID  ccVdb      Port      DSP/Ch  Called #    Codec    Dial-peers
0x97        12BB 0x641B0F68 3/0:D.1   1012/2  31001       g711ulaw 3/31000
0x99        12BE 0x641B0F68 3/0:D.2   1012/3  31002       g711ulaw 3/31000
2 active calls found

Router# show voice call status

CallID      CID  ccVdb       Port     DSP/Ch  Called #    Codec    Dial-peers
0x2         11D1 0x62FE6478 1/0/0     1/1      10001      g711ulaw 1/2
0x3         11D1 0x62FE80F0 1/0/1     2/1     *10001      g711ulaw 2/1
1 active call found
```

When using the **test call id** command, you must specify a call ID, which you can obtain by using the **show voice call status** command. The following is an example of how to obtain the call ID for use as the *call-id* argument. The first parameter displayed in the output is the call ID.

Note Do not use the 0x prefix in the *call-id* argument when you enter the resulting call ID in the **test call status** command.

The following shows keyword choices when using the **show voice call** command with the **|** (pipe) option:

```
Router# show voice call | ?

  append    Append redirected output to URL (URLs supporting append operation
            only)
  begin     Begin with the line that matches
  exclude   Exclude lines that match
  include   Include lines that match
  redirect  Redirect output to a URL
  tee       Copy output to a URL
```

Table 192 describes significant fields shown in the previous displays.

Table 192 show voice call Field Descriptions

Field (listed alphabetically)	Description
Called #	Called number. • No “*” before the number denotes an originating call leg. Two of the call legs in the example constitute one locally switched call and one network call, so the call legs refer to two active calls. • A “*” before the number denotes a destination call leg (for example, this number was called with Called #).
CallID	This hexadecimal number used for further query is the monotonically increasing number that call control maintains for each call leg (ccCallID_t).
ccVdb	Value that is displayed in many other debugs to identify these call legs.
CID	Conglomerate value derived from the GUID that appears in the **show call active brief** command.
Codec	Codec.
Dial-peers	Dial peer.
DSP/Ch	DSP and channel allocated to this call leg. The format of these values is platform dependent (particularly the Cisco AS5300, which shows the DSP number as a 3-digit number, <VFC#><DSPM#><DSP#>). Time-slot information is also in the output for digital ports. For example, if you are using a digital port, the time slot is also returned: dsp/ch/time slot.
ERL	Echo return loss (in dB).
ERL/Reflctr	Time difference, in ms, between the original signal and the loudest echo (peak reflector) as detected by the echo canceller.
Jitter	Current values of the delay and the jitter of the packets around that delay.
Port	Voice port.
Rx/Tx	Transmit and receive rates for the connection.
VAD	Voice-activity detection: y or n.
VPM STATE	Voice-port-module (VPM) state.
VTSP STATE	Voice-telephony-service-provider (VTSP) state.

For more information about the extended echo canceller, see *Extended ITU-T G.168 Echo Cancellation*.

Related Commands

Command	Description
show call active brief	Displays a summary of active call information.

show dial-peer voice	Displays the configuration for all VoIP and POTS dial peers configured on the router.
show voice dsmp stream	Displays the current session of the voice DSPM media stream.
show voice dsp	Displays the current status of all DSP voice channels.
show voice port	Displays configuration information about a specific voice port.
test call id	Manipulates the echo canceller and jitter buffer parameters in real time.

show voice cause-code

To display error category to Q.850 cause code mapping, use the **show voice cause-code** command in user EXEC mode.

show voice cause-code category-q850

Syntax Description

category q850	Displays error category to Q.850 cause code mapping.

Command Default

No default behavior or values.

Command Modes

User EXEC

Command History

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

Usage Guidelines

Use this command to display the internal error category to Q.850 cause code mapping table, and configured and default values, with category descriptions.

Examples

The following example displays Q.850 cause code mapping:

```
Router# show voice cause-code category-q850

The Internal Error Category to Q850 cause code mapping table:-

 Error Configured Default    Description
Category  Q850     Q850
   128     27        3  Destination address resolution failure
   129     38      102  Call setup timeout
   178     41       41  Internal Communication Error
   179     41       41  External communication Error
   180     47       47  Software Error
   181     47       47  Software Resources Unavailable
   182     47       47  Hardware Resources Unavailable
   183     41       41  Capability Exchange Failure
   184     49       49  QoS Error
   185     41       41  RTP/RTCP receive timer expired or bearer layer failure
   186     38       38  Signaling socket failure
   187     38       38  Gateway or signaling interface taken out of service
   228     50       50  User is denied access to this service
   278     65       65  Media Negotiation Failure due to non-existing Codec
```

Table 193 describes the significant fields shown in the display.

Table 193 show voice cause-code Field Descriptions

Field	Description
128	Destination address resolution failure
129	Call setup timeout
178	Internal communication error
179	External communication Error
180	Software error
181	Software resources unavailable
182	Hardware resources unavailable
183	Capability exchange failure
184	QoS error
185	RTP/RTCP receive timer expired or bearer layer failure
186	Signaling socket failure
187	Gateway or signaling interface taken out of service
228	User denied access to this service
278	Media negotiation failure due to non existing codec

Related Commands

Command	Description
error-category q850-cause	Specifies Q.850 cause code mapping

show voice class called-number

To display a specific voice class called-number, use the **show voice class called-number** command in privileged EXEC mode.

show voice class called-number [**inbound** | **outbound**] *tag*

Syntax Description

inbound	Displays the specified inbound voice class called-number.
outbound	Displays the specified outbound voice class called-number.
tag	Digits that identify this voice class called-number.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use this command to display a specific inbound or outbound voice class called-number.

Examples

The following is sample output from this command:

```
Router# show voice class called-number outbound 200
Called Number Outbound: 200
                index 1      4085550100
                index 2      4085550102
                index 3      4085550103
                index 4      4085550104
```

Table 194 describes significant fields shown in the display.

Table 194 show voice class called-number Field Descriptions

Field	Description
Called Number Inbound/Outbound	The tag for the specified inbound or outbound voice class called-number.
index *number*	The number or range of numbers for this voice class called number.

Related Commands

Command	Description
show voice class called-number-pool	Displays voice class called number pool configuration information.

show voice class called-number-pool

To display a voice class called-number pool, use the **show voice class called-number-pool** command in privileged EXEC mode.

show voice class called-number-pool tag [**detail**]

Syntax Description

tag	Digits that identify this voice class called-number-pool. Range is 1 to 10000.
detail	Displays idle called number and allocated called number information.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use this command to display the voice class called number pool configuration information. The **detail** keyword displays up to 16 idle called numbers, and up to 4 allocated called numbers for each allocated request.

Examples

The following sample output displays configuration information for voice class called-number-pool 100, including idle called numbers and allocated called numbers:

```
Router(config)# show voice class called-number-pool 100 detail

Called Number Pool: 100
index 1 100A11 - 100A20
index 2 200#55 - 200#77
index 3 5551111 - 6662333
index 99 123C11 - 123C99
All called numbers are generated from table: FALSE
No of idle called numbers: 16
List of idle called numbers:
100A11 100A12 .. Display up to 16 idle called number from the pool
100A13 100A14
100A15 100A16
100A17 100A18
100A19 100A20
200#55 200#56
200#57 200#58
200#59 200#60
No of alloc requests : 1
Ref Id Alloc PC Size
2 41F84190 16
List of alloc called numbers: .. Display the first 4 allocated called number for RefId 2
200#61 200#62
200#63 200#64
```

Table 195 describes significant fields shown in the display.

Table 195 show voice class called-number-pool Field Descriptions

Field	Description
Called Number Pool	Tag that identifies the called number pool.
index	Number or range of numbers for this called number pool.
All called numbers are generated from table	• FALSE—Numbers are not generated from called number table. • TRUE—Numbers are generated from called number table.
No. of idle called numbers	Number of idle called numbers in the called number pool.
List of idle called numbers	List of idle numbers in the called number pool.
No. of alloc requests	Number of requests for numbers from the called number pool.
Ref Id Alloc PC Size	Reference ID for a specific list of allocated numbers.
List of alloc called numbers	List of first four allocated numbers from the called number pool.

Related Commands

Command	Description
show voice class called-number	Displays a specific voice class called-number.

show voice class resource-group

To display the resource group configuration information for a specific resource group or all resource groups, use the **show voice class resource-group** command in privileged EXEC mode.

show voice class resource-group {*tag* | **all**}

Syntax Description

tag	Unique tag for the resource group.
all	Displays information for all voice resource groups.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
15.1(2)T	This command was introduced.

Usage Guidelines

You can use the **show voice class resource-group** command to display the parameters configured to monitor resources.

Examples

The following is sample output from the **show voice class resource-group** command:

```
Router> enable
Router# show voice class resource-group 2

Resource Availability Indicator status
Resource Index 2

Resource Type:SYSTEM
         Status: Low threshold
Resource Type: MEM Subtype: io-mem Low/High watermark: 2/5
         Status: Low threshold
Report Interval 34
------------------------------------------------------------
```

Table 196 describes the significant fields shown in the display.

Table 196 show voice class resource-group Field Descriptions

Field	Description
Resource Index	Unique index value to identify the resource group.
Resource Type	Type of the resource being monitored.
Status	Status of the resource.

Table 196 show voice class resource-group Field Descriptions (continued)

Field	Description
Subtype	Subtype of the resource being monitored.
Report Interval	Periodic reporting interval for the resource being monitored. The status of the resource being monitored is reported based on the preconfigured timer value.

Related Commands

Command	Description
debug rai	Enables debugging for Resource Allocation Indication (RAI).
rai target	Configures the SIP RAI mechanism.
resource (voice)	Configures parameters for monitoring resources, use the resource command in voice-class configuration mode.
periodic-report interval	Configures periodic reporting parameters for gateway resource entities.
voice class resource-group	Enters voice-class configuration mode and assigns an identification tag number for a resource group.

show voice class uri

To display summary or detailed information about configured uniform resource identifier (URI) voice classes, use the **show voice class uri** command in user EXEC or privileged EXEC mode.

show voice class uri [*tag* | **summary**]

Syntax Description

tag	(Optional) Specific URI voice class for which to display detailed information.
summary	(Optional) Displays a short summary of all URI voice classes.

Command Default

Detailed information about the configured URI voice classes is displayed.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Release	Modification
12.3(4)T	This command was introduced.
15.1(2)T	This command was modified. The command was enhanced to display the mutiple hosts in the configured URI classes.

Usage Guidelines

If both the *tag* argument and **summary** keyword are omitted, the output displays detailed information about all URI voice classes.

Examples

The following is sample output from this command:

```
Router# show voice class uri

Voice URI class: 100
    SNMP status = Active
    Schema = sip
    pattern = 12345

Voice URI class: 101
    SNMP status = Active
    Schema = sip
    pattern = 555....

Voice URI class: 102
    SNMP status = Active
    Schema = sip
    user-id = demo
    host = cisco
    phone context =

Voice URI class: 103
```

```
   SNMP status = Active
   Schema = tel
   phone number = 555....
   phone context =

Voice URI class: 700
    SNMP status = Active
    Schema = sip
    pattern = elmo@sip.tgw.com*

Voice URI class: 104
    SNMP status = Active
    Schema = tel
    pattern = 5550134

Voice URI class: 700
    SNMP status = Active
    Schema = sip
    user-id =
    host = exmp.example.com
    phone context =

    host instances:
     ipv4:192.168.0.1
     ipv6:[2001:0DB8:0:1:FFFF:1234::5]
     dns:ogw.example.com
```

The following is sample output from this command with the **summary** keyword:

```
Router# show voice class uri summary
_____________________________________________________________
Class Name                       Schema        SNMP
-------------------------------------------------------------
100                              sip           Active
101                              sip           Active
102                              sip           Active
103                              tel           Active
700                              sip           Active
104                              tel           Active
_____________________________________________________________
```

Table 197 describes the significant fields in the displays.

Table 197 show voice class uri Field Descriptions

Field	Description
Class Name	Tag that identifies the URI voice class.
Schema	Whether the voice class is used for SIP or TEL URIs.
pattern	Pattern used to match the entire SIP or TEL URI as configured with the **pattern** command.
user-id	Pattern used to match the user-id field in the SIP URI as configured with the **user-id** command.
host	Pattern used to match the host field in the SIP URI with the **host** command.
phone number	Pattern used to match the phone number field in a TEL URI as configured with the **phone number** command.
phone context	Pattern used to match the phone context field in a SIP or TEL URI as configured with the **phone context** command.

Related Commands

Command	Description
debug voice uri	Displays debugging messages related to URI voice classes.
show dialplan incall uri	Displays which dial peer is matched for a specific URI in an incoming call.
show dialplan uri	Displays which outbound dial peer is matched for a specific destination URI.
voice class uri	Creates or modifies a voice class for matching dial peers to calls containing a SIP or TEL URI.

show voice connectivity summary

To display the results of the last connectivity checks performed on all analog Foreign Exchange Station (FXS) ports on a router, use the **show voice connectivity summary** command in privileged EXEC mode.

show voice connectivity summary

Syntax Description

This command has no arguments or keywords.

Command Default

A summary of the last connectivity checks performed on all analog FXS ports on a router is displayed.

Command Modes

Privileged EXEC (#)

Command History

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

Examples

The following example shows how the **show voice connectivity summary** command is used:

```
Router> enable
Router# show voice connectivity summary
.
.
.
! The summary results include information such as the port address, type of connectivity
! check performed, result of connectivity check for each port
```

show voice data

To display the call control application programming interface (CCAPI) and Telephony Service Provider (VTSP) data structures, use the **show voice data** command in user EXEC or privileged EXEC mode.

show voice data {**ccapi** {**ccCallEntry** {*call-id* | **all**} | **ccCallInfo**} | **vtsp** {**ccCallInfo** | **vtsp_cdb** {*call-id* | **all**} | **vtsp_sdb** {*call-id* | **all**}}

Syntax Description

ccapi	Displays all the CCAPI calls.
ccCallEntry	Displays the call entry.
call-id	Call identifier (ID) in the range 1 to 4294967295.
all	Displays all the call entries.
ccCallInfo	Displays the call information.
vtsp	Displays all the VTSP calls.
vtsp_cdb	Displays all the VTSP call control back calls.
vtsp_sdb	Displays all the VTSP signalling data block calls.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Release	Modification
12.4(22)T	This command was introduced in a release earlier than Cisco IOS Release 12.4(22)T.

Examples

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

```
Router# show voice data ccapi ccCallEntry all

CallEntry=0x6B8051B0; CallID=7(0x7)::

element:{ 0x6B8051B0; 0x6B8051B4; 0x6B8051B8; } 7; <appReturnStack>; 1735408; 1;
0x6B8051D8; 7; 8; callInfo:{ 0; 112233; <NULL>; 889988; <NULL>; <NULL>; <NULL>; <NULL>;
<NULL>; <NULL>; <NULL>; FALSE; FALSE; TRUE; <NULL>; 0; 0; 0; <NULL>; RegularLine; Unknown;
D356CC33-E54B-11D7-8005-00169D6EE1AE; D356CC33-E54B-11D7-8005-00169D6EE1AE; 0; 0; 0; 0; 0;
998877; 0x6B80547C; 0; TRUE; FALSE; 0.0.0.0; 0.0.0.0; 0x6B8054A0; 0x6B8054A4; 0x6B8054A8;
0x6B8054AC; 0; FALSE; FALSE; 0x6B8054BC; 0; call_decode:{ redirect_info:{ 0xFF; 0xFF;
0xFF; 0xFF; 0xFF; 0xFF; 0x00; 0xFF; 255; <NULL>; <NULL>; 0x00; FALSE; FALSE; } 0x00; 0x80;
0x00; 0x80; 0; 0x00; <NULL>; 0; 0x00; <NULL>; FALSE; FALSE; FALSE; FALSE; -1; <NULL>;
TRUE; <transfer_info>; FALSE; 129; 40; 104; 0xFF; TRUE; } FALSE;
D357685B-E54B-11D7-8016-CB962D72A90A; 0; 0; 0; 0; 0; 0; 0x6B805634; FALSE; <NULL>; FALSE;
FALSE; FALSE; 0; 0; 0; <NULL>; ISDN 7/0:1:D; FALSE; FALSE; FALSE; 0x00; <NULL>; <NULL>;
0x6B80585C; 0; 0x6B805864; } 0x6B805914; 0x6B805918; 0x6B80591C; 0x6B805920;
<altAssocList>; FALSE; 0x6B80593C; 0x6B805940; 0x6B805944; FALSE; 0; 65535; TRUE; 0;
FALSE; 1; <disconnect_timer>; <inter_digit_timer>; 10000; <initial_timer_timestamp>;
10000; FALSE; 0; 0; -1; <NULL>; 0x6B8059F8; <evCategoryMask>; <evDetailMask>; 4294967295;
0x6B805C48; FALSE; 0; 0; TRUE; TRUE; TRUE; 0; 0; 0x6B805C6C; FALSE; 0; 4; 0; -1; FALSE;

CallEntry=0x6B805C90; CallID=8(0x8)::
```

```
element:{ 0x6B805C90; 0x6B805C94; 0x6B805C98; } 8; <appReturnStack>; 1735408; 2;
0x6B805CB8; 8; 7; callInfo:{ 0; 112233; <NULL>; 889988; <NULL>; 112233; 112233; <NULL>;
<NULL>; <NULL>; <NULL>; FALSE; FALSE; TRUE; <NULL>; 0; 0; 0; <NULL>; RegularLine; Unknown;
D356CC33-E54B-11D7-8005-00169D6EE1AE; D356CC33-E54B-11D7-8005-00169D6EE1AE; 7; 0; 0; 0; 2;
112233; 0x6B805F5C; 0; FALSE; FALSE; 0.0.0.0; 0.0.0.0; 0x6B805F80; 0x6B805F84; 0x6B805F88;
0x6B805F8C; 0; FALSE; FALSE; 0x6B805F9C; 0; call_decode:{ redirect_info:{ 0xFF; 0xFF;
0xFF; 0xFF; 0xFF; 0xFF; 0x00; 0xFF; 255; <NULL>; <NULL>; 0x00; FALSE; FALSE; } 0x00; 0x80;
0x00; 0x00; 0; 0x00; <NULL>; 0; 0x00; <NULL>; FALSE; FALSE; FALSE; FALSE; -1; <NULL>;
TRUE; <transfer_info>; FALSE; 129; 40; 104; 0xFF; TRUE; } FALSE;
D357685B-E54B-11D7-8016-CB962D72A90A; 0; 0; -1; 0; 0; 0; 0x6B806114; FALSE; <NULL>; FALSE;
FALSE; FALSE; 0; 0; 0; <NULL>; ISDN 7/0:1:D; TRUE; FALSE; FALSE; 0x00; <NULL>; <NULL>;
0x6B80633C; 0; 0x6B806344; } 0x6B8063F4; 0x6B8063F8; 0x6B8063FC; 0x6B806400;
<altAssocList>; FALSE; 0x6B80641C; 0x6B806420; 0x6B806424; FALSE; 0; 65535; FALSE; 0;
FALSE; 1; <disconnect_timer>; <inter_digit_timer>; 10000; <initial_timer_timestamp>;
10000; FALSE; 0; 0; -1; <NULL>; 0x6B8064D8; <evCategoryMask>; <evDetailMask>; 4294967295;
0x6B806728; FALSE; 0; 0; TRUE; TRUE; TRUE; 0; 0; 0x6B80674C; FALSE; 0; 4; 0; -1; FALSE;
```

Table 198 describes the significant fields shown in the display.

Table 198 show voice data Field Descriptions

Field	Description
CallEntry	Displays the call entry identification number used for the incoming call leg.
CallID	Displays the specified call identifier value.
element	Indicates the various configuration values for the service element.
callInfo	Displays the call informaton.
call_decode	Displays the status of the audio decoder.
redirect_info	Displays the forwarding request information when a call is being forwarded.
transfer_info	Displays the call transfer request information.
disconnect_timer	Displays the timeout value, in seconds, specified to disconnect the call.
inter_digit_timer	Displays the maximum allowable time, in seconds, between digits dialed by the user.

Related Commands

Command	Description
debug voip ccapi error	Traces error logs in the call control API.

show voice dnis-map

To display current dialed-number identification service (DNIS) map information, use the **show voice dnis-map** command in privileged EXEC mode.

show voice dnis-map [*dnis-map-name* | **summary**]

Syntax Description

dnis-map-name	(Optional) Name of a specific DNIS map.
summary	(Optional) Displays a short summary of each DNIS map.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

This command displays a detailed description of each configured DNIS map.

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

If the **summary** keyword is used, the command displays a one-line summary about each DNIS map.

If an asterisk is displayed next to a DNIS map name when the **summary** keyword is used, it means that the DNIS map is configured, but not running. Normally this is because the external text file was not successfully loaded, for example:

```
dnis-map            Entries    URL
--------            -------    ---
dmap1               1
*dmap4              0          http://dnismaps/dnismap4.txt
```

To create a DNIS map, use the **voice dnis-map** command. You can link to an external DNIS map text file or use the **dnis** command to add numbers to a DNIS map in Cisco IOS software.

To associate a DNIS map with a dial peer, use the **dnis-map** command.

Examples

The following is sample output from the **show voice dnis-map** command:

```
Router# show voice dnis-map

There are 2 dnis-maps configured

Dnis-map dmap1
-----------------------------------------
  It has 3 entries
  It is not populated from a file.
```

```
DNIS                  URL
----                  ---
 4085551212           tftp://global/tickets/movies.vxml
 4085551234           tftp://global/tickets/plays.vxml
 4085554321           tftp://global/tickets/games.vxml


Dnis-map dmap4
-----------------------------------------
  It has 0 entries
  It is populated from url http://dnismaps/dnismap4.txt


DNIS                URL
----                ---
```

Table 199 describes the fields shown in this output.

Table 199 show voice dnis-map Field Descriptions

Field	Description
Dnis-map	Name of a DNIS map that is configured on the gateway.
DNIS	Destination telephone number specified in this DNIS map.
URL	Location of the VoiceXML document to invoke for this DNIS number.

The following is sample output from the **show voice dnis-map summary** command:

```
Router# show voice dnis-map summary

There are 3 dnis-maps configured

dnis-map            Entries    URL
--------            -------    ---
dmap1               3
dmap4               0        http://dnismaps/dnismap4.txt
dmap6               8
```

Table 200 describes the fields shown in this output.

Table 200 show voice dnis-map summary Field Descriptions

Field	Description
dnis-map	Names of the DNIS maps that are configured on the gateway.
Entries	Number of entries in DNIS maps that reside on the gateway. This field displays 0 if the DNIS map is a text file stored on an external server.
URL	Location of externally stored DNIS maps.

Related Commands

Command	Description
dnis	Adds a DNIS number to a DNIS map.
dnis-map	Associates a DNIS map to a dial peer.
voice dnis-map	Enters DNIS map configuration mode to create a DNIS map.
voice dnis-map load	Reloads a DNIS map that has changed since the previous load.

show voice dsmp stream

To display the current session of voice Distributed Stream Media Processor (DSPM) media stream, the recent state transitions, and stream connection, use the **show voice dsmp stream** command in privileged EXEC mode.

show voice dsmp stream {*stream ID* | **leg**}

Syntax Description

stream ID	DSMP media stream identifier. Range: 1 to 4294967295.
leg	Call leg corresponding to a caller ID.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.3(14)T	This command was introduced.

Usage Guidelines

When the calls hang, use this command to get the current sessions of the DSMP media stream. You can look at the DSMP state transitions corresponding to the calls and find out the problems.

Examples

The following example shows an output of a typical DSMP session in a VoIP call. This call consists of four streams, two input streams and two output streams:

```
Router# show voice dsmp stream
Total number of streams in use is: 4


Stream information:: stream=1
Type: TDM, Direction: OUTPUT
Fax/Modem Type: voice
Xmit Function: 0x00000000
Xmit function is Enabled
Call ID: 4, Conference ID: -1


Session information:: session=0x658CA948 dsp_intf=0x642DDD8C dsp_name=1/9:3

connections=2 streams=4 (5 1 4 3 )
current state S_DSMP_VC_RUNNING current container simple_voice_container
State Transitions: timestamp (container, state) -- event -> (container, state)
367121.596 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367121.796 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367122.712 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367122.732 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367122.920 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
```

```
367122.940 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367123.112 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367123.152 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367124.432 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367124.632 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367124.732 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367124.932 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367125.032 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367125.232 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367126.140 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367126.160 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367126.340 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367126.380 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367126.548 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367126.568 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)

Session log information::
Regular Timer:
    Timer start operations:
             Timestamp    Duration(ms)        Caller
            367122.652            4000    0x6113397C
            367119.388            4000    0x6113397C
            367117.624           10000    0x6112ED88

    Timer stop operations:
             Timestamp    Duration(ms)        Caller
            367122.656               0    0x61133A98
            367119.392               0    0x61133A98
            367117.624               0    0x6112F060
            367117.624               0    0x6112EE24
    Number of overwritten entries: 2

Periodic Timer:
    Timer start operations:
    None
    Timer stop operations:
    None
Packet suppression is disabled


Stream information:: stream=3
Type: PACKET, Direction: OUTPUT
Fax/Modem Type: voice
Xmit Function: 0x6111D324
Xmit function is Enabled
Call ID: 3, Conference ID: 2
DSP Encap: 0x1
Codec Mask: 0x4; Codec Bytes: 20
Fax Rate Mask: 0x2; Fax Bytes: 20; T38 Disabled
```

```
VAD Mask: 0x2


Session information:: session=0x658CA948 dsp_intf=0x642DDD8C dsp_name=1/9:3

connections=2 streams=4 (5 1 4 3 )
current state S_DSMP_VC_RUNNING current container simple_voice_container
State Transitions: timestamp (container, state) -- event -> (container, state)
367128.452 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367128.652 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367129.556 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367129.588 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367129.756 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367129.796 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367129.968 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367129.988 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367131.276 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367131.472 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367131.572 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367131.772 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367131.872 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367132.072 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367132.980 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367133.000 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367133.180 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367133.220 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367133.400 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367133.420 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)

Session log information::
Regular Timer:
    Timer start operations:
             Timestamp    Duration(ms)        Caller
            367131.020            4000    0x6113397C
            367128.316            4000    0x6113397C
            367122.652            4000    0x6113397C
            367119.388            4000    0x6113397C
    Number of overwritten entries: 1

    Timer stop operations:
             Timestamp    Duration(ms)        Caller
            367131.024               0    0x61133A98
            367128.320               0    0x61133A98
            367122.656               0    0x61133A98
```

```
            367119.392               0    0x61133A98
    Number of overwritten entries: 4

Periodic Timer:
    Timer start operations:
    None
    Timer stop operations:
    None
Packet suppression is disabled


Stream information:: stream=4
Type: PACKET, Direction: INPUT
Fax/Modem Type: voice
Xmit Function: 0x61F2CA34
Xmit function is Enabled
Call ID: 3, Conference ID: 2
DSP Encap: 0x1
Codec Mask: 0x4; Codec Bytes: 20
Fax Rate Mask: 0x2; Fax Bytes: 20; T38 Disabled
VAD Mask: 0x2


Session information:: session=0x658CA948 dsp_intf=0x642DDD8C dsp_name=1/9:3

connections=2 streams=4 (5 1 4 3 )
current state S_DSMP_VC_RUNNING current container simple_voice_container
State Transitions: timestamp (container, state) -- event -> (container, state)
367133.400 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367133.420 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367134.692 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367134.892 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367134.992 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367135.192 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367135.292 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367135.492 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367136.400 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367136.432 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367136.600 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367136.640 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367136.812 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367136.840 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367138.112 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367138.312 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367138.412 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367138.612 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
```

```
367138.712 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367138.912 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)

Session log information::
Regular Timer:
    Timer start operations:
         Timestamp    Duration(ms)        Caller
        367137.648            4000    0x6113397C
        367134.440            4000    0x6113397C
        367131.020            4000    0x6113397C
        367128.316            4000    0x6113397C
    Number of overwritten entries: 3

    Timer stop operations:
         Timestamp    Duration(ms)        Caller
        367137.648               0    0x61133A98
        367134.440               0    0x61133A98
        367131.024               0    0x61133A98
        367128.320               0    0x61133A98
    Number of overwritten entries: 6

Periodic Timer:
    Timer start operations:
    None
    Timer stop operations:
    None
Packet suppression is disabled


Stream information:: stream=5
Type: TDM, Direction: INPUT
Fax/Modem Type: voice
Xmit Function: 0x00000000
Xmit function is Enabled
Call ID: 4, Conference ID: -1


Session information:: session=0x658CA948 dsp_intf=0x642DDD8C dsp_name=1/9:3

connections=2 streams=4 (5 1 4 3 )
current state S_DSMP_VC_RUNNING current container simple_voice_container
State Transitions: timestamp (container, state) -- event -> (container, state)
367138.712 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367138.912 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367139.824 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367139.844 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367140.024 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367140.064 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367140.244 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367140.252 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367141.536 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367141.736 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
```

```
367141.836 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367142.036 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367142.136 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367142.336 (simple_voice_container, S_DSMP_VC_RUNNING) -- E_DSMP_CC_PLAY_REQ ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367143.244 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367143.264 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367143.444 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367143.484 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)
367143.652 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_BEGIN
-> (simple_voice_container, CNFSM_NO_STATE_CHANGE)
367143.672 (simple_voice_container, CNFSM_CONTAINER_STATE) -- E_DSMP_DSP_DTMF_DIGIT_END ->
(simple_voice_container, CNFSM_NO_STATE_CHANGE)

Session log information::
Regular Timer:
    Timer start operations:
             Timestamp    Duration(ms)        Caller
            367137.648            4000    0x6113397C
            367134.440            4000    0x6113397C
            367131.020            4000    0x6113397C
            367128.316            4000    0x6113397C
    Number of overwritten entries: 3

    Timer stop operations:
             Timestamp    Duration(ms)        Caller
            367137.648               0    0x61133A98
            367134.440               0    0x61133A98
            367131.024               0    0x61133A98
            367128.320               0    0x61133A98
    Number of overwritten entries: 6

Periodic Timer:
    Timer start operations:
    None
    Timer stop operations:
    None
Packet suppression is disabled
```

Table 201 describes the significant fields shown in the display.

Table 201 show voice dsmp stream Field Descriptions

Field	Description
Stream information	Shows stream ID.
Type	Type of stream.
Direction	Direction of stream.
Fax/Modem Type	Type of fax or modem.
Xmit Function	Transmit function in use.
Call ID	Caller ID of call leg.
Conference ID	Conference ID.

Table 201 show voice dsmp stream Field Descriptions (continued)

Field	Description
Session information	Information about the associated session.
connections	Number of stream connections.
streams	Number of streams.
current state	Current state and container of the session.
State Transitions	State transitions of the associated session.
DSP Encap	Encapsulation associated with the session.
Codec Mask	Codec mask associated with the session.
Fax Rate Mask	Fax rates associated with the session.
Fax Bytes	Fax bytes associated with the session.
VAD Mask	VAD mask associated with the session.

Related Commands

Command	Description
show call active voice	Displays call information for voice calls in progress.
show voice call	Displays the call status for voice ports on the Cisco router.

show voice dsp

To display the current status or selective statistics of digital signal processor (DSP) voice channels, use the **show voice dsp** command in user EXEC or privileged EXEC mode.

show voice dsp [**active** [**slot** *slot-number* [*slot-number*]] | **capabilities slot** *slot-number* **dsp** *dsp-number* | **cpu-load slot** *slot-number* **dsp** *dsp-number* [**reset**] | **detailed** | **error** | [**group all** | **sorted-list**] **slot** *slot-number* | **signalling** | **voice** | **version** [*slot* | *slot/dsp*] [*slot* | *slot/dsp*]]

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show voice dsp [**active** [**slot** *slot-number*] | **capabilities slot** *slot-number* **dsp** *dsp-number* | **cpu-load slot** *slot-number* **dsp** *dsp-number* [**reset**] | **crash-dump** | **detailed** | **error** | **group** {**all** | **slot** *slot-number}* | **signalling** | **sorted-list slot** *slot-number* | **voice**]

Syntax Description

active	(Optional) Displays active channels.
slot *slot-number* [*slot-number*]	(Optional) Specifies either a single slot or the first slot in a range. To specify a range of slots, you can enter a second slot in the syntax of this argument. The second slot specifies the end of the range. All slots in the range are affected by the command.
capabilities	(Optional) Displays DSP capabilities.
dsp *dsp-number*	(Optional) Specifies the DSP on the slot.
cpu-load	(Optional) Displays DSP CPU load.
reset	(Optional) Resets the DSP CPU statistics.
crash-dump	(Optional) Displays the DSP crash dump status. **Note** To enable a DSP crash dump, set file limit to a non-zero number, and set the destination to a valid file name.
detailed	(Optional) Displays detailed information about DSP status.
error	(Optional) Displays DSP errors.
group	(Optional) Displays DSP group information.
all	(Optional) Displays all the DSP group details.
sorted-list	(Optional) Displays a DSP sorted list.
signaling	(Optional) Displays DSP signaling channel usage.
voice	(Optional) Displays DSP voice channel usage.
version	(Optional) Displays the DSP firmware version.
slot	(Optional) The first slot in a range. To specify a range of slots, you can enter a second slot in the syntax of this argument. The second slot specifies the end of the range. All slots in the range are affected by the command.
/dsp	(Optional) The first DSP in a range. To specify a range of DSPs, you can enter a second DSP in the syntax of this argument. The second DSP specifies the end of the range. All DSPs in the range are affected by the command. The slash mark is required.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was implemented on the Cisco 2600 series and Cisco 3600 series, and the display format was modified.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.3(14)T	The command was modified. Command output was enhanced to display status information for NM-HDV network module TI-549 DSPs.
12.4(4)T	The command was modified. Command output was enhanced to display the codec setting for modem relay operation.
12.4(4)XC	The command was modified. The **version** keyword was added and the command was implemented on the Cisco AS5350XM and Cisco AS5400XM platforms.
12.4(11)T	The command was modified. Command output was enhanced to display information about DSP H.320 channels.
Cisco IOS XE Release 2.5	This command was integrated into Cisco IOS XE Release 2.5.
Cisco IOS XE Release 3.2S	This command was implemented on the Cisco ASR 1000 Series Router.

Usage Guidelines

Use this command when abnormal behavior occurs in the DSP voice channels. The channel or channels should have an active voice call at the time the command is executed.

Examples

The following sample output shows the current status of the codec, set for modem relay, on channel 1:

```
Router# show voice dsp

----------------------------FLEX VOICE CARD 1 ------------------------------
                           *DSP VOICE CHANNELS*
DSP   DSP             DSPWARE CURR  BOOT                         PAK   TX/RX
TYPE  NUM CH CODEC    VERSION STATE STATE   RST AI VOICEPORT TS ABRT PACK COUNT
===== === == ======== ======= ===== ======= === == ========= == ==== ============
C5510 001 01 modem-re 4.5.909 busy  idle      0  0 1/1/0     05    0      298/353
                           *DSP SIGNALING CHANNELS*
DSP   DSP             DSPWARE CURR  BOOT                         PAK   TX/RX
TYPE  NUM CH CODEC    VERSION STATE STATE   RST AI VOICEPORT TS ABRT PACK COUNT
===== === == ======== ======= ===== ======= === == ========= == ==== ============
C5510 001 05 {flex}   4.5.909 alloc idle      0  0 1/1/3     02    0         15/0
C5510 001 06 {flex}   4.5.909 alloc idle      0  0 1/1/2     02    0         17/0
C5510 001 07 {flex}   4.5.909 alloc idle      0  0 1/1/1     06    0         31/0
C5510 001 08 {flex}   4.5.909 alloc idle      0  0 1/1/0     06    0        321/0
------------------------END OF FLEX VOICE CARD 1 ----------------------------
```

The following sample output shows the current status of all DSP voice channels:

```
Router# show voice dsp

DSP# 0, channel# 0 G729A BUSY
DSP# 0, channel# 1 G729A BUSY
DSP# 1, channel# 2 FAX IDLE
DSP# 1, channel# 3 FAX IDLE
DSP# 2, channel# 4 NONE BAD
DSP# 2, channel# 5 NONE BAD
DSP# 3, channel# 6 NONE BAD
```

```
DSP# 3, channel# 7 NONE BAD
DSP# 4, channel# 8 NONE BAD
DSP# 4, channel# 9 NONE BAD
DSP# 5, channel# 10 NONE BAD
DSP# 5, channel# 11 NONE BAD
```

The following is sample output from this command on a Cisco 1750 router:

```
Router# show voice dsp

DSP#0: state IN SERVICE, 2 channels allocated
channel#0: voice port 1/0, codec G711 ulaw, state UP
channel#1: voice port 1/1, codec G711 ulaw, state UP
DSP#1: state IN SERVICE, 2 channels allocated
channel#0: voice port 2/0, codec G711 ulaw, state UP
channel#1: voice port 2/1, codec G711 ulaw, state UP
DSP#2: state RESET, 0 channels allocated
```

The following is sample output from this command on a secure Survivable Remote Site Telephony (SRST) router with the NM-HDV network module and the TI-549 (C549) DSP installed:

```
Router# show voice dsp

DSP  DSP    DSPWARE  CURR     BOOT                                PAK   TX/RX
TYPE NUM CH CODEC    VERSION  STATE STATE  RST AI VOICEPORT TS ABORT PACK COUNT
==== === == ======== ======= ===== ======= === == ======== === ==== ===========
C549  1  01 {medium} 4.4.3    IDLE  idle     0  0   1/0:0    1   0    9357/9775
C549  1  02 {medium} 4.4.3    IDLE  idle     0      1/0:0    2   0    0/0
C549  2  01 {medium} 4.4.3    IDLE  idle     0  0   1/0:0    3   0    0/0
C549  2  02 {medium} 4.4.3    IDLE  idle     0      1/0:0    4   0    0/0
C549  3  01 {medium} 4.4.3    IDLE  idle     0  0   1/0:0    5   0    0/13
C549  3  02 {medium} 4.4.3    IDLE  idle     0      1/0:0    6   0    0/13
```

The following is sample output from this command for an H.320 network configured for video support:

```
Router# show voice dsp

DSP  DSP             DSPWARE CURR  BOOT                               PAK     TX/RX
TYPE NUM CH CODEC    VERSION STATE STATE   RST AI VOICEPORT TS ABORT  PACK COUNT
==== === == ======== ======= ===== ======= === == ========= == ===== ============ edsp 001
01 g711ulaw  0.1 IDLE  50/0/1.1 edsp 002 02 g711ulaw  0.1 IDLE  50/0/1.2 edsp 003 01
g729r8 p  0.1 IDLE  50/0/2.1 ----------------------------FLEX VOICE CARD 1
------------------------------
                           *DSP VOICE CHANNELS*
DSP   DSP             DSPWARE CURR  BOOT                               PAK   TX/RX
TYPE  NUM CH CODEC    VERSION STATE STATE   RST AI VOICEPORT TS ABRT PACK COUNT
===== === == ======== ======= ===== ======= === == ========= == ==== ============
C5510 001 05 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 001 06 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 001 07 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 001 08 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 001 09 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 001 10 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 001 11 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 001 12 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 001 13 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 001 14 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 001 15 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 001 16 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 01 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 02 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 03 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 04 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 05 None     9.0.105 idle  idle      0  0                  0          0/0
```

```
C5510 003 06 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 07 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 08 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 09 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 10 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 11 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 12 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 13 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 14 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 15 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 16 None     9.0.105 idle  idle      0  0                  0          0/0
                                    *DSP H.320 CHANNELS*
DSP   DSP     TX/RX       DSPWARE CURR                 PAK   TX/RX
TYPE  NUM CH  CODEC       VERSION STATE VOICEPORT TS ABRT PACK COUNT
===== === === =========== ======= ===== ========= == ==== ============
C5510 001 01  h320p(01)   9.0.105 busy  1/0/0:15  06
      001 02  h320s(02)   9.0.105 busy  1/0/0:15  07
      001 03  h320s(03)   9.0.105 busy  1/0/0:15  08
      001 04  h320s(04)   9.0.105 busy  1/0/0:15  09
      001 01a g711ulaw    9.0.105 busy                0  1013663/5083
                                                                  00
      001 01v h263 /h263  9.0.105 busy                0  104908/30911
                                                                 4
------------------------END OF FLEX VOICE CARD 1 ----------------------------
```

Table 202 describes the significant fields shown in the displays.

Table 202   show voice dsp Field Descriptions

Field	Description
DSP	Number of the DSP.
channel	Number of the channel and its status.
DSP TYPE	TI-549 (C549) DSP.
DSP NUM	Number of the DSP.
CH	Channel number.
CODEC	Complexity setting.
DSPWARE VERSION	Version of DSPware.
CURR STATE	Current status of the channel: alloc (allocated), busy, or idle.
BOOT STATE	DSP readiness, either idle or in service.
RST	Number of times the DSP has been reset or restarted.
AI	Alarm indication count on the channel.
VOICEPORT	Voice card number and slot.
TS	Time slot.
PAK ABORT	Number of dropped packets.
TX/RX PACK COUNT	Number of transmitted and received packets.

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The following sample output shows the DSP Type, DSP number, channel number, codecs running, DSP firmware version, and the current state of channels running on the DSP SPA inside the Cisco ASR 1000 Series Router:

```
Router# show voice dsp
----------------------------- SPA-DSP 1/1 ---------------------------

       *DSP INFORMATION*
DSP    DSP               DSPWARE CURR
TYPE   NUM CH CODEC      VERSION STATE RST AI
====== === == ========= ======== ===== === ==
SP2600 001    None      26.07.00 up    4   0
SP2600 002    None      26.07.00 up    3   0
SP2600 003    None      26.07.00 up    3   0
SP2600 004    None      26.07.00 up    1   0
SP2600 005    None      26.07.00 up    1   0
SP2600 006    None      26.07.00 up    1   0
SP2600 007    None      26.07.00 up    1   0
SP2600 008    None      26.07.00 up    1   0
SP2600 009    None      26.07.00 up    1   0
SP2600 010    None      26.07.00 up    1   0
SP2600 011    None      26.07.00 up    1   0
SP2600 012    None      26.07.00 up    1   0
SP2600 013    None      26.07.00 up    1   0
SP2600 014    None      26.07.00 up    1   0
SP2600 015    None      26.07.00 up    1   0
SP2600 016    None      26.07.00 up    1   0
SP2600 017    None      26.07.00 up    1   0
SP2600 018    None      26.07.00 up    1   0
SP2600 019    None      26.07.00 up    1   0
SP2600 020    None      26.07.00 up    1   0
SP2600 021    None      26.07.00 up    1   0

------------------------- END OF SPA-DSP 1/1 -------------------------------------
```

The following example shows the active channels on DSP SPA located in slot 1 on the Cisco ASR 1000 Series Router:

```
Router# show voice dsp active slot 1
------------------ SPA-DSP 1/1 ---------------------------
*DSP VOICE CHANNELS*
DSP    DSP               DSPWARE CURR
TYPE   NUM CH CODEC      VERSION STATE RST AI
====== === == ========= ======== ===== === ==
SP2600 001 01 g711ulaw  26.07.00 busy  4   0
SP2600 002 01 g711ulaw  26.07.00 busy  3   0
------------------------- END OF SPA-DSP 1/1 ------------
```

The following example shows the channel capabilities for different types of codecs on the Cisco ASR 1000 Series Router:

```
Router# show voice dsp capabilities slot 1

Card 1/1 DSP 1 Capabilities:
DSP Type: SP2600 - 43

    Credits   645 ,  G711Credits 15,  HC Credits  37,  MC Credits  23,
    FC Channel  43,  HC Channel  17,  MC Channel  28,
    Conference 8-party credits:
     G711  58 ,  G729  107,  G722  129,  ILBC  215
  Secure Credits:
```

```
    Sec LC Xcode  24,      Sec HC Xcode  64,
    Sec MC Xcode  35,      Sec G729 conf 161,
    Sec G722 conf 215,     Sec ILBC conf 322,
    Sec G711 conf 92 ,
  Max Conference Parties per DSP:
    G711 88,  G729 48,  G722 40,  ILBC 24,
    Sec G711 56,  Sec G729 32,
    Sec G722 24  Sec ILBC 16,
  Voice Channels:
    g711perdsp = 43,  g726perdsp = 28, g729perdsp = 17, g729aperdsp = 28,
    g723perdsp = 17,  g728perdsp = 17, g723perdsp = 17, gsmperdsp   = 28,
    gsmefrperdsp = 17, gsmamrnbperdsp = 17,
    ilbcperdsp = 17, isacperdsp = 8  modemrelayperdsp = 17,
    g72264Perdsp = 28, h324perdsp = 17,
    m_f_thruperdsp = 43, faxrelayperdsp = 28,
    maxchperdsp = 43, minchperdsp = 17,
    srtp_maxchperdsp = 27, srtp_minchperdsp = 14, faxrelay_srtp_perdsp =
4,
    g711_srtp_perdsp = 27, g729_srtp_perdsp = 14, g729a_srtp_perdsp = 24,----------
```

The following example shows the details of the DSP errors on the Cisco ASR 1000 Series Router.

Note The crash dump details must be enabled to display the crash dump for a DSP SPA. To enable a crash dump, set the destination of the crash dump file to a valid file name, and set the file limit to a non-zero number.

```
Router#show voice dsp crash-dump

Voice DSP Crash-dump status:
    Destination file url is <none>
    File limit is 0
DSP crash dump is currently disabled
To enable DSP crash dump, set file-limit to a non-zero number and set
destination to a valid file name
```

Related Commands

Command	Description
dsp services dspfarm	Enables the DSP-farm services.
dspfarm profile	Enters the DSP farm profile configuration mode and defines a profile for DSP farm services.
show dspfarm	Displays DSP farm service information, such as operational status, and DSP resource allocation for transcoding.

show voice dsp channel

To display the voice digital signal processor (DSP) channels, use the **show voice dsp channel** command in user EXEC or privileged EXEC mode.

show voice dsp channel {**operational-status** {*slot* | */dsp* | */channel*} [*slot* | */dsp* | */channel*] | **statistics** *slot-number* [*slot-number*] | **traffic** *slot-number* [*slot-number*]}

Syntax Description

operational-status	Displays the operational state for active sessions on a specific channel or range of channels.
slot	A single slot or the first slot in a range. To specify a range of slots, you can enter a second slot in the syntax of this argument. The second slot specifies the end of the range. All slots in the range are affected by the command.
/dsp	A single DSP on the slot or the first DSP in a range. To specify a range of DSPs, you can enter a second DSP in the syntax of this argument. The second DSP specifies the end of the range. All DSPs in the range are affected by the command. The slash mark is required.
/channel	A single DSP channel or the first DSP channel in a range. The second occurrence of this argument specifies either a single DSP channel or the last DSP channel in a range. The slash mark is required.
statistics	Displays DSP statistics for a specific channel or range of channels.
slot-number	A single slot or the first slot in a range. To specify a range of slots, you can enter a second slot in the syntax of this argument. The second slot specifies the end of the range. All slots in the range are affected by the command.
traffic	Displays traffic on a specific channel or range of channels.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Release	Modification
12.4(4)XC	The command was introduced on the Cisco AS5350XM and Cisco AS5400XM platforms.
12.4(11)T	The command was modified. Command output was enhanced to display information about DSP H.320 channels.

Usage Guidelines

Use this command when abnormal behavior occurs in the DSP voice channels. The channel or channels should have an active voice call at the time the command is executed.

Examples

The following is sample output from the **show voice dsp channel operational-status** command on slot 3/13/1:

```
Router# show voice dsp channel operational-status 3/13/1

 Operational status of Slot/DSP/Channel : 3/13/1
```

```
 Servicetype  : VOICE
 Codec Type  : gsmamr-nb
 Encapsulation : RTP
 Transmitted Packets : 346
 Transmitted Bytes : 11740
 Received Packets : 411
 Received Bytes : 11142
 Playout de-jitter mode :  None
 Playout de-jitter buffer minimum delay : 0 msec
 Playout de-jitter buffer initial delay : 0 msec
 Playout de-jitter buffer maximum delay : 0 msec
 Noise level : -5.0
 ERLLevel : 6
 ACOMLevel : 6
CodecPktPeriod=20 Milliseconds
CodecFrameFormat=bandwidth-efficient
CodecCrc=Disabled
CodecModes=3,6
CodecEncodeRate=6
CodecDecodeRate=6
CodecEncodeChanges=1
CodecDecodeChanges=0
CodecCrcFails=0
CodecBadFrameQuality=0
CodecInvalidCMRs=0
CodecInvalidFrameType=0
 Voice activity detection : Enabled
 Dtmf Relay : inband-voice
 ComfortNoisePak : 52
 TxVoiceDuration : 11560
 VoiceRxDuration : 3380
 Rx OutOfSeq Paks : 0
 Rx Late Paks : 0
 Rx Early Paks : 0
 Lost Packets : 0
 Playout Delay Current : 50
 Playout Delay Min : 50
 Playout Delay Max : 50
 Playout Delay ClockOffset : 80
 Playout Delay Jitter : 0
 Error Rx Drop : 0
 Error Tx Drop : 0
 Error Tx Control : 0
 Error Rx Control : 0
 Playout Error Predictive : 0
 Playout Error Interpolative : 0
 Playout Error Silence : 0
 Playout Error BufferOverFlow : 0
 Playout Error Retroactive : 0
 Playout Error Talkspurt : 0
```

Table 203 describes the significant fields shown in the display.

Table 203 show voice dsp channel Field Descriptions

Field	Description
DSP	Number of the DSP.
Channel	Number of the channel and its status.
Codec Type	Complexity setting.
TxVoiceDuration	Transmitted voice duration.

Related Commands

Command	Description
show voice dsp	Displays the current status or selective statistics of DSP voice channels,.

show voice dsp crash-dump

To display voice digital signal processor (DSP) crash dump information, use the **show voice dsp crash-dump** command in privileged EXEC configuration mode.

show voice dsp crash-dump

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

The following example checks your configuration:

```
Router# show voice dsp crash-dump

Voice DSP Crash-dump status:
    Destination file url is slot0:banjo-152-s
    File limit is 20
    Last DSP dump file written was
          tftp://112.29.248.12/tester/26-152-t2
    Next DSP dump file written will be slot0:banjo-152-s1
```

The following example shows that the crash dump feature is enabled:

```
Router# show voice dsp crash-dump

Voice DSP Crash-dump status:
    Destination file url is
          tftp://172.29.248.12/xxtir/dspdump6.bin
    File limit is 10
    Last DSP dump file written was
          tftp://172.29.248.12/xxtir/dspdump6.bin1
    Next DSP dump file written will be
          tftp://172.29.248.12/xxtir/dspdump6.bin2
```

The following example shows that the crash dump feature is disabled:

```
Router# show voice dsp crash-dump

Voice DSP Crash-dump status:
    Destination file url is
          tftp://172.29.248.12/xxtir/dspdump6.bin
    File limit is 0
    Last DSP dump file written was
          tftp://172.29.248.12/xxtir/dspdump6.bin1
DSP crash dump is currently disabled
To enable DSP crash dump, set file-limit to a non-zero number
```

Field descriptions should be self-explanatory.

Related Commands

Command	Description
debug voice dsp crash-dump	Displays crash dump debug information.
voice dsp crash-dump	Enables the crash dump feature and specifies the destination file and the file limit.

show voice dsp summary

To display the digital signal processor (DSP) summary, use the **show voice dsp summary** command in user EXEC or privileged EXEC mode.

show voice dsp summary [*slot* | *slot*/*dsp*] [*slot* | *slot*/*dsp*]

Syntax Description

slot	(Optional) A single slot or the first slot in a range. To specify a range of slots, you can enter a second slot in the syntax of this argument. The second slot specifies the end of the range. All slots in the range are affected by the command.
/*dsp*	(Optional) A single DSP on the slot or the first DSP in a range. To specify a range of DSPs, you can enter a second DSP in the syntax of this argument. The second DSP specifies the end of the range. All DSPs in the range are affected by the command. The slash mark is required.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Release	Modification
12.4(4)XC	This command was introduced. The command was implemented on the Cisco AS5350XM and Cisco AS5400XM platforms.
12.4(11)T	The command was modified. Command output was enhanced to display information about DSP H.320 channels.
12.4(19)	The command was modified. Command output was modified to accurately show the “Codectype” as “voice” rather than “fax” for T.38 calls.
12.4(18a)	The command was modified. Command output was modified to accurately show the “Codectype” as “voice” rather than “fax” for T.38 calls.
12.4(13f)	The command was modified. Command output was modified to accurately show the “Codectype” as “voice” rather than “fax” for T.38 calls.
12.4(15)T5	The command was modified. Command output was modified to accurately show the “Codectype” as “voice” rather than “fax” for T.38 calls.

Examples

The following sample output from the **show voice dsp summary** command shows summary information about DSPs:

```
Router# show voice dsp summary

Total number of DSPs = 48

Codectype       Calls    Codectype       Calls    Codectype       Calls
g729r8 pre-ietf     0    g729ar8             0    g726r16             0
g726r24             0    g726r32             0    g711ulaw            0
g711alaw            1    g728                0    g723r63             0
g723r53             0    gsmfr               0    gsmefr              0
g729br8             0    g729abr8            0    g723ar63            0
g723ar53            0    g729r8              0    t38                 0
clear-channel       0    vofr cisco          0    llcc                0
```

```
g726r40             0    transparent         0    modem-relay         0
cisco               0                        0                        0
pass-through        0    gsmamr-nb           0

Legend       :
======
Channel state: (s)shutdown   (a)active call (d)download pending
               (b)busiedout  (B)bad         (p)busyout pending
Call type    : (v)voice      (f)fax-relay   (_)not in use

Summary      :
=======
Channels     :  Total 768  In-Use    001
Calls        :  Total 001  Voice     001  Fax  000
             :  Free  713  Disabled  000

      DSP      DSP        DSP    Channel            Call
DSP#  State    Complexity Resets State              Type
2/1   ACTIVE   FLEXI      0      ________ ________  ________ ________
2/2   ACTIVE   FLEXI      0      ________ ________  ________ ________
2/3   ACTIVE   FLEXI      0      ________ ________  ________ ________
2/4   ACTIVE   FLEXI      0      ________ ________  ________ ________
2/5   ACTIVE   FLEXI      0      ________ ________  ________ ________
2/6   ACTIVE   FLEXI      0      ________ ________  ________ _________
```

Table 202 describes the significant fields shown in the display.

Table 204 show voice dsp summary Field Descriptions

Field	Description
DSP	Number of the DSP.
Codectype	Complexity setting.
Channels	Number of the channel and its status.
State	Status of the calls.

Related Commands

Command	Description
show voice dsp	Displays the current status or selective statistics of DSP voice channels,.

show voice eddri prefix

To show applicable prefixes for the event dispatcher and data repository interface (EDDRI), use the **show voice eddri prefix** command in privileged EXEC mode.

show voice eddri prefix [*prefix_number*]

Syntax Description

all	All neighbors
prefix_number	(Optional) Specified EDDRI prefix.

Command Default

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

If no prefix is specified, all configured prefixes appear.

The EDDRI notifies threaded grep (TGREP) when an attribute changes on some subsystems. EDDRI interacts with the dial-peer subsystem, trunk-group subsystems, call-control API (CCAPI) subsystem, and customer-relationship-management (CRM) subsystem to notify changes in particular attributes. EDDRI is responsible for creating the prefix database.

Examples

The following example shows output for the **show voice eddri prefix** command:

```
prefix 4 address family decimal
advertise flag 0x27 ac 24 tc 24 capacity timer 25 sec
AC_avg 24, FD_avg 0, SD_avg 0
succ_curr 0 tot_curr 0
succ_report 0 tot_report 0
changed 0 replacement position 0
trunk group castg2
dial peer tag 1001
```

Field descriptions should be self-explanatory.

Related Commands

Command	Description
debug voip eddri	Turns on debugging for the EDDRI.

show voice enum-match-table

To display the rules of an ENUM match table, use the **show voice enum-match-table** command in privileged EXEC mode.

show voice enum-match-table [*table-number* [**sort**]]

Syntax Description

table-number	(Optional) ENUM match table to display, by number. Range is from 1 to 15.
sort	(Optional) Sorts the output by ascending table number.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

This command displays the ENUM match table rules in the order in which they were defined. The **sort** keyword changes the display to list the rules from lowest to highest preference.

Examples

The following sample output displays the rules of ENUM match table number 3:

```
Router# show voice enum-match-table 3

voice enum_match_table 3
rule 1 5 /^9\(1.*\)/ /+\1/ cisco
rule 2 4 /^9011\(.*\)/ /+1408\1/ arpa
rule 10 1 /^(.*)/ /\1/ e164.cisco.com
```

The following sample output displays the ENUM match tables in ascending order by table number:

```
Router# show voice enum-match-table

voice enum-match-table 3
rule 1 5 /^9\(1.*\)/ /+\1/ cisco
rule 2 4 /^9011\(.*\)/ /+1408\1/ arpa
rule 10 1 /^(.*)/ /\1/ e164.cisco.com

voice enum-match-table 5
rule 2 4 /^9011\(.*\)/ /+1408\1/ arpa
rule 10 1 /^(.*)/ /\1/ e164.cisco.com
```

Field descriptions should be self-explanatory.

Related Commands

Command	Description
rule (ENUM configuration)	Defines the ENUM rule.
test enum	Tests the ENUM rule.
voice enum-match-table	Initiates the voice ENUM match table definition.

show voice hpi capture

To display capture status and statistics, use the **show voice hpi capture** command in privileged EXEC mode.

show voice hpi capture

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

This command displays the capture status and statistics. Use this command to confirm logger status and to examine the logger status output when the logger is running.

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

Note If you are experiencing problems with certain voice calls, the engineering team at Cisco might ask you to capture the control messages using the voice DSP logger. You can capture these messages by turning on the logger, repeating the problematic calls, and capturing the logs. Only Cisco engineers can determine if you should send the logs in for further review.

Examples

The following sample output shows capture statistics (HPI capture and logging) and status:

```
Router# show voice hpi capture

HPI Capture is on and is logging to URL ftp://172.23.184.216/d:\test_data.dat1 messages
sent to URL, 0 messages droppedMessage Buffer (total:inuse:free)  2134:0000:2134Buffer
Memory:699952 bytes, Message size:328 bytes
```

Field descriptions should be self-explanatory.

Related Commands

Command	Description
debug hpi	Enables debugging for HPI message events.
voice hpi capture	Allocates the Host Port Interface (HPI) capture buffer (size in bytes) and sets up or changes the destination URL for captured data.

show voice iec description

To display Internal Error Code (IEC) descriptions, use the **show voice iec description** command in user EXEC mode.

show voice iec description ***string***

Syntax Description

string	Six-part dotted decimal string that displays the definition of an internal error code.

Command Default

No default behavior or values.

Command Modes

User EXEC

Command History

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

Examples

The following example displays IEC descriptions:

```
Router# show voice iec description 1.1.180.2.21.4

    IEC Version: 1
    Entity: 1 (Gateway)
    Category: 180 (Software error)
    Subsystem: 2 (TCL IVR)
    Error: 21 (Script syntax)
    Diagnostic Code: 4
```

Table 205 describes significant fields shown in the display.

Table 205 show voice iec description Field Descriptions

Field	Description
IEC version	IEC version. A value of 1 indicates the Cisco IOS Release 12.3(4)T version.
Entity	Network physical entity (hardware system) that generated the IEC. The value 1 is assigned to the gateway.
Category	Error category, defined in terms of ITU-based Q.850 cause codes and VoIP network errors.
Subsystem	Specific subsystem within the physical entity where the IEC was generated.
Error Code	Error code within the subsystem.
Diagnostic Code	Cisco internal diagnostic value. Report this value to Cisco Technical Support.

Related Commands

Command	Description
show voice statistics iec	Displays IEC statistics.

show voice lmr

To display the Land Mobile Radio (LMR) related dynamic information and static information for LMR ports or a DS0 group, use the **show voice lmr** command in privileged EXEC mode.

show voice lmr [*slot*/*subunit*/*port* | *slot*/*port*:*ds0-group*] [**details** | **timing** [**warnings**]]

Syntax Description

slot/subunit/port	(Optional) Voice port that you specify with the *slot*/*subunit*/*port* designation. • *slot* specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • *subunit* specifies a voice interface card (VIC) in which the voice port is located. Valid entries are 0 and 1. • *port* specifies an analog voice port number. Valid entries are 0 and 1. The slash marks are required.
slot/port:*ds0-group*	(Optional) Voice port that you specify with the *slot*/*port*:*ds0-group* designation. • *slot* specifies a router slot in which the packet voice NM is installed. Valid entries are router slot numbers for the particular platform. • *port* specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. • *ds0-group* specifies a T1 or E1 logical port number. T1 range is from 0 to 23. E1 range is from 0 to 30. The colon is required.
details	(Optional) Displays more information. If this keyword is omitted, less information is displayed.
timing	(Optional) Displays the timing configuration for all LMR ports.
warnings	(Optional) Displays all LMR ports that are having suspicious timing configuration.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.4(24)T	This command was modified in a release earlier than Cisco IOS Release 12.4(24)T. The **timing** and **warnings** keywords were added.

Usage Guidelines

This command displays information for LMR voice ports only. If no voice port is specified, the command displays information for all ear and mouth (E&M) LMR voice ports.

When the **details** keyword is used, this command displays information about timeouts, timers, and injected tones and pauses, in addition to detailed voice port and active call information found in the **show voice port** and **show call active voice** commands.

Examples

The following is sample output from the **show voice lmr** command for an E&M LMR analog voice port on a Cisco 3745 router:

```
Router# show voice lmr 2/0/0

2/0/0
=========
 Connection type: n/a
 Out Attenuation = 0 db, In Gain = 0 dB
 E-lead capability is inactive, polarity = normal
 M-lead capability is inactive, polarity = normal
 voice-class tone-signal test
 state = LMR_CONNECT, e-lead = off, m-lead = off
 full duplex, voice path = rx
 Terminating side of the connection
 TransmitPackets=113, TransmitBytes=2241
 ReceivePackets=113, ReceiveBytes=2241
 CoderTypeRate=g729r8
 NoiseLevel=-65, ACOMLevel=22
 OutSignalLevel=-68, InSignalLevel=-79
 RemoteIPAddress=10.5.25.40, RemoteUDPPort=17272
 Remote SignallingIPAddress=10.5.25.40, Port=15418
 Remote MediaIPAddress=10.5.25.40, Port=17272
 RoundTripDelay=2 ms
 SessionProtocol=cisco
 VAD =enabled
```

The following is sample output from the **show voice lmr details** command for an E&M LMR analog voice port on a Cisco 3745 router:

```
Router# show voice lmr 2/0/0 details

2/0/0
=========
 Description:
 Connection type: n/a
 Out Attenuation = 0 db, In Gain = 0 dB
 Timing hangover: 500 ms
 E-lead capability is inactive, polarity = normal
 M-lead capability is inactive, polarity = normal
 Timing hookflash-in: 480
 Timing delay-voice: 470 ms
 Music On Hold Threshold: -38 dB, Noise Threshold: -62 dB
 E&M type: 1, Operation:  2-wire
 Impedance is set to 600r Ohm
 lmr tear down timeout is set to 1800 second
 lmr PTT transmit timeout is not set
 lmr PTT receive timeout is not set
 voice-class tone-signal test
         inject tone 1 1950 3 150
         inject tone 2 2000 0 60
         inject pause 3 60
         inject tone 4 2175 3 150
         inject tone 5 1000 0 50
         inject guard-tone 6 1950 -10
 state = LMR_CONNECT, e-lead = off, m-lead = off
 full duplex, voice path = rx
```

```
Terminating side of the connection
TransmitPackets=113, TransmitBytes=2241
ReceivePackets=113, ReceiveBytes=2241
CoderTypeRate=g729r8
NoiseLevel=-66, ACOMLevel=22
OutSignalLevel=-68, InSignalLevel=-79
PeerAddress=37200
PeerSubAddress=
PeerId=200
SessionTarget=

RemoteIPAddress=10.5.25.40, RemoteUDPPort=17272
Remote SignallingIPAddress=10.5.25.40, Port=15418
Remote MediaIPAddress=10.5.25.40, Port=17272
RoundTripDelay=0 ms
SessionProtocol=cisco
VAD =enabled
SelectedQoS=best-effort
ProtocolCallId=
SessionTarget=
```

Table 206 describes the significant fields shown in the output, in the order in which they appear.

Table 206 show voice lmr Field Descriptions

Field	Description
Connection type	Type of connection between LMR routers: private line, automatic ringdown (PLAR), trunk, or n/a
Out Attenuation	Output attenuation.
In Gain	Input gain.
E-lead capability	Active or inactive.
polarity	Polarity of the E&M voice port: normal or reverse.
M-lead capability	Active or inactive.
voice class tone-signal	Name of the tone-signal voice class.
state=	Signaling state.
e-lead =	On or off.
m-lead =	On or off.
full duplex	Voice path for the voice port is operating in full duplex mode.
half duplex	Voice path for the voice port is operating in half duplex mode.
voice path	Transmit or receive.
TransmitPackets	Number of packets sent by this peer during this call.
TransmitBytes	Number of bytes sent by this peer during this call.
ReceivePackets	Number of packets received by this peer during this call.
ReceiveBytes	Number of bytes received by the peer during this call.
CoderTypeRate	Negotiated coder rate. This value specifies the send rate of voice or fax compression to its associated call leg for this call.

Table 206 show voice lmr Field Descriptions (continued)

Field	Description
NoiseLevel	Active noise level for this call.
ACOMLevel	Current ACOM level for this call. ACOM is the combined loss achieved by the echo canceller, which is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.
OutSignalLevel	Active output signal level to the telephony interface used by this call.
InSignalLevel	Active input signal level from the telephony interface used by this call.
RemoteIPAddress	Remote system IP address for the VoIP call.
RemoteUDPPort	Remote system User Datagram Protocol (UDP) listener port to which voice packets are sent.
Remote SignallingIPAddress, Port	Call control server IP address and signaling port number.
Remote MediaIPAddress, Port	Remote side media server IP address and RTP port number.
RoundTripDelay	Voice packet round trip delay between the local and remote systems on the IP backbone for this call.
SessionProtocol	Session protocol used for an Internet call between the local and remote routers through the IP backbone.
VAD	Whether voice activation detection (VAD) is enabled.
Description	Description of what the port is connected to.
Timing hangover	Number of milliseconds of delay before the digital signal processor (DSP) tells Cisco IOS software to turn off the E-lead after the DSP detects that the voice stream has stopped.
Timing hookflash-in	Maximum duration of a hookflash for a Foreign Exchange Station (FXS) interface.
Timing delay-voice	Delay before a voice packet is played out.
Music On Hold Threshold	Decibel level of music played when calls are put on hold.
Noise Threshold	Noise threshold for incoming calls.
E&M type	E&M signaling type.
Operation	2-wire or 4-wire operation.
Impedance	Terminating impedance of the interface.
lmr tear down timeout	Time for which the voice port waits before tearing down an LMR connection after detecting no voice activity.
lmr PTT transmit timeout	Maximum time for transmitting a voice packet.
lmr PTT receive timeout	Maximum time for receiving a voice packet.

Table 206 show voice lmr Field Descriptions (continued)

Field	Description
inject pause	Pause injected before the voice packet is played out.
inject tone	Tone injected before the voice packet is played out.
inject guard-tone	Guard tone played out with the voice packet.
PeerAddress	Destination pattern or number associated with this peer.
PeerSubAddress	Subaddress when this call is connected.
PeerId	ID value of the peer table entry to which this call was made.
SessionTarget	Network-specific address to receive calls from the dial peer.
SelectedQoS	Selected RSVP quality of service (QoS) for this call.
ProtocolCallId	Voice signaling specific call ID.

Related Commands

Command	Description
show call active voice	Displays call information for voice calls in progress.
show voice port	Displays configuration information about a specific voice port.

show voice permanent-call

To display information about the permanent calls on a voice interface, use the **show voice permanent-call** command in user EXEC or privileged EXEC mode.

show voice permanent-call [*voice-port*] [**summary**]

Syntax Description

voice-port	(Optional) Slot number or slot/port number of the voice interface for which you wish to display permanent call information.
summary	(Optional) Displays summary information about VoFR and VoATM ports used for permanent connections.

Command Default

When no parameters are specified with this command, the output displays information for all ports containing permanent calls. When a specific interface is specified, information is displayed about the permanent calls for that interface only.

Command Modes

User EXEC
Privileged EXEC

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810.
12.0(4)T	The command was integrated into Cisco IOS Release 12.0(4)T.

Examples

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

```
Router# show voice permanent-call 1/1

1/1 state=connect coding=G729A payload size=30 vad=off
ec=8 (ms), cng=off fax=on digit_relay=on Seq num = off, VOFR Serial0,dlci = 550,cid = 6
TX INFO :slow-mode seq#= 25, sig pkt cnt= 19646, last-ABCD=1101
hardware-state ACTIVE signal type is CEPT/MELCAS
voice-gate CLOSED,network-path OPEN MASTER
 1101 1101 1101 1101 1101 1101 1101 1101 1101 1101
 1101 1101 1101 1101 1101 1101 1101 1101 1101 1101
 1101 1101 1101 1101 1101 1101 1101 1101 1101 1101
RX INFO :slow-mode, sig pkt cnt= 19648, under-run = 0, over-run = 0
missing = 0, out of seq = 0, very late = 0
playout depth = 0 (ms), refill count = 1
 prev-seq#= 25, last-ABCD=1101, slave standby timeout 25000 (ms)
max inter-arrival time 0 (ms), current timer 384 (ms)
max timeout timer 5016 (ms), restart timeout is 0 (ms)
signaling packet fast-mode inter-arrival times (ms)
16 24 16 24 16 24 16 24 16 24 16 24 16 24 16 24
16 24 16 24 16 24 16 24 0 0 0 0 0 0 0 0
```

```
signaling playout history
1101 1101 1101 1101 1101 1101 1101 1101 1101 1101
1101 1101 1101 1101 1101 1101 1101 1101 1101 1101
1101 1101 1101 1101 1101 1101 1101 1101 1101 1101
```

The following is sample output from the **show voice permanent-call summary** command:

```
Router# show voice permanent-call summary

1/1 state= connect, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 880,cid = 6
1/2 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 102
1/3 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 103
1/4 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 104
1/5 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 105
1/6 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 106
1/7 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 107
1/8 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 108
1/9 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 109
1/10 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 110
1/11 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 111
1/12 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 112
1/13 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 113
1/14 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 114
1/15 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 115
1/17 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 117
1/18 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 118
1/19 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 119
1/20 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 120
1/21 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 121
1/22 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 122
1/23 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 123
1/24 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 124
1/25 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 125
```

Table 207 describes significant fields shown in this output.

Table 207 show voice permanent-call Field Descriptions

Field	Description
state	Current status of the call on this voice port.
coding	Codec type used for this call.
payload size	Size in bytes of the voice payload.
vad	Whether voice activity detection is turned on or off.
ec	Echo canceler length, in milliseconds.
cng	Whether comfort noise generation is used.
fax	Whether fax-relay is enabled.
digit_relay	Whether FRF.11 Annex A DTMF digit-relay is enabled.
Seq num	Whether sequence numbers are turned on or off.
VOFR	Interface used for this call.
dlci	DLCI for this call.
cid	DLCI subchannel for this call.
TX INFO:slow-mode	FRF.11 Annex B packets are being sent at the slow rate defined by the signal timing keepalive period.
TX INFO:seq#	Sequence number of the last packet sent.
TX INFO:sig pkt cnt	Number of signaling packets sent by this dial peer.
TX INFO:last-ABCD	Last ABCD signaling state sent by this dial peer to the network.
hardware-state	On-hook/off-hook state of the call when the signaling protocol in use is a supported protocol. Not valid when the signal type is “transparent.”
signal type	Type of call-control signaling used by this dial peer.
voice-gate	Whether voice packets are being sent (OPEN) or not sent (CLOSED).
network-path	Whether any type of packet is being sent (OPEN) or not sent (CLOSED) to the network. This field indicates CLOSED only if the port is configured as a slave using the **connection trunk answer-mode** command.
RX INFO:slow-mode	FRF.11 Annex B packets are being received at the slow rate. Successive packets have the same sequence number.
RX INFO:sig pkt cnt	Number of slow-mode signaling packets received by this dial peer.
RX INFO:under-run	Valid for fast-mode only. Counts the number of times the signaling playout buffer became empty during FRF.11 Annex B fast-mode. In this mode, signaling packets are expected to be received every 20 milliseconds.
RX INFO:over-run	Valid for fast-mode only. Counts the number of times the signaling playout buffer became full during FRF.11 Annex B fast-mode. In this mode, signaling packets are expected to be received every 20 milliseconds.
RX INFO:missing	Number of FRF.11 Annex B packets that were counted as missing based on checking Annex B sequence numbers.

Table 207 show voice permanent-call Field Descriptions (continued)

Field	Description
RX INFO:out of seq	Number of FRF.11 Annex B packets that were counted as received in the wrong order based on checking Annex B sequence numbers.
RX INFO:very late	Number of FRF.11 Annex B packets that were received with a sequence number significantly different from the expected sequence number.
RX INFO:playout depth	Valid for fast-mode only. Shows the current FRF.11 Annex B signaling buffer playout depth in milliseconds.
RX INFO:refill count	Indicates the number of times the FRF.11 Annex B signaling playout buffer was refilled as a result of a slow-mode to fast-mode transition.
RX INFO:prev-seq#	Sequence number of the last FRF.11 Annex B signaling packet received.
RX INFO:last-ABCD	Last ABCD signaling bit pattern sent to the attached PBX (telephone network side). In the out-of-service condition, this shows the OOS pattern being sent to the PBX.
RX INFO:slave standby timeout	Value configured using the **signal timing oos standby** command for the applicable voice class permanent entry.
max inter-arrival time	Maximum interval between the arrival of fast-mode FRF.11 Annex B packets since the last time this parameter was displayed.
current timer	Time, in milliseconds, since the last signaling packet was received.
max timeout timer	Maximum value of the "current timer" parameter since the last time it was displayed.
restart timeout	Connection restart timeout value.
signaling packet fast-mode inter-arrival time	Last several values of the fast-mode FRF.11 Annex B signaling packet inter-arrival time.
signaling playout history	Recent ABCD signaling bits received from the data network.

Related Commands

Command	Description
show frame-relay fragment	Displays Frame Relay fragmentation details.
show frame-relay pvc	Displays statistics about PVCs for Frame Relay interfaces.
show frame-relay vofr	Displays details about FRF.11 subchannels being used on Voice over Frame Relay DLCIs.

show voice port

To display configuration information about a specific voice port, use the **show voice port** command in privileged EXEC mode.

Cisco 1750 Router

show voice port *slot***/***port*

Cisco 2600 and Cisco 3600 Series Router with Analog Voice Ports

show voice port [*slot***/***subunit***/***port* | **summary**]

Cisco 2600 and Cisco 3600 Series Router with Digital Voice Ports (with T1 Packet Voice Trunk Network Modules)

show voice port [*slot***/***port***:***ds0-group* | **summary**]

Cisco AS5300 Universal Access Server

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

Cisco 7200 Series Router

show voice port {*slot***/***port***:***ds0-group-number* | *slot***/***subunit***/***port*}

Syntax Description

Cisco 1750 Router	
slot	Slot number in the router in which the VIC is installed. Valid entries are 0, 1, and 2, depending on the slot in which it is installed.
/port	Voice port. Valid entries are 0 and 1. The slash mark is required.

Cisco 2600 and Cisco 3600 Series Router with Analog Voice Ports	
*slot***/***subunit***/***port*	(Optional) The analog voice port designation: • *slot*—Router slot in which a voice network module (VNM) is installed. Valid entries are router slot numbers for the particular platform. • *subunit*—Voice Interface Card (VIC) in which the voice port is located. Valid entries are 0 and 1. (The VIC fits into the voice network module.) The slash mark is required. • *port*—Analog voice port number. Valid entries are 0 and 1. The slash mark is required.
summary	(Optional) Displays a summary of all voice ports.

Cisco 2600 and Cisco 3600 Series Router with Digital Voice Ports	
*slot***/***port***:***ds0-group*	(Optional) Specifies the digital voice port designation: • *slot*—Router slot in which the packet voice trunk network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • **/***port*—T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. (One VWIC fits in an NM.) The slash mark is required. • **:***ds0-group*—T1 or E1 logical port number. T1 range is 0 to 23. E1 range is 0 to 30. The colon is required.
summary	(Optional) Displays a summary of all voice ports.

Cisco AS5300 Universal Access Server	
controller-number	T1 or E1 controller.
:D	D channel that is associated with the ISDN PRI. The colon is required.

Cisco 7200 Series Router	
slot	Router location where the voice port adapter is installed. Range is 0 to 3.
/*port*	Voice interface card location. Valid entries are 0 and 1. The slash mark is required.
:*ds0-group-number*	Defined DS0 group number. Because each defined DS0 group number is represented on a separate voice port, you can define individual DS0s on the digital T1/E1 card. The colon is required.
slot	Slot number in the Cisco router where the VIC is installed. Range is 0 to 3, depending on the slot where it is installed.
/*subunit*	Subunit on the VIC where the voice port is located. Valid entries are 0 and 1. The slash mark is required.
/*port*	Voice port number. Valid entries are 0 and 1. The slash mark is required.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
11.3(1)MA	This command was modified. Port-specific values for the Cisco MC3810 were added.
12.0(3)T	This command was modified. Port-specific values for the Cisco MC3810 were added.
12.0(5)XK	This command was modified. The *ds0-group* argument was added for the Cisco 2600 series and Cisco 3600 series.

Release	Modification
12.0(5)XE	This command was modified. Additional syntax was created for digital voice to allow specification of the DS0 group. This command applies to VoIP on the Cisco 7200 series.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0(7)T.
12.0(7)XK	This command was modified. The **summary** keyword was added for the Cisco 2600 series and Cisco 3600 series. The *ds0-group* argument was added for the Cisco MC3810.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.2(8)T	This command was modified. This command was implemented for direct inward dial (DID) on the Cisco IAD2420 series.
12.2(2)XN	This command was modified. Support for enhanced Media Gateway Control Protocol (MGCP) voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series, Cisco 3600 series, and Cisco Gateway 200 (Cisco VG200).
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and Cisco CallManager Version 3.2. It was implemented on the Cisco IAD2420 series.
12.4(11)T	This command was modified. This command was enhanced to display voice class called-number-pool configuration information for the voice port.
12.4(12)	This command was modified. This command was integrated into Cisco IOS Release 12.4(12) and output was modified to display the parameter set by the **timing sup-disconnect** command.
15.0(1)XA	This command was modified. The output was enhanced to display the logical partitioning class of restriction (LPCOR) policy for incoming and outgoing calls.
15.1(1)T	This command was integrated into Cisco IOS Release 15.1(1)T.
15.1(3)T	This command was modified. The output of this command was enhanced to display the connection status of foreign exchange office (FXO) ports.

Usage Guidelines

Use this command to display configuration and VIC-specific information about a specific port.

This command works on Voice over IP, Voice over Frame Relay, and Voice over ATM.

The **ds0-group** command automatically creates a logical voice port that is numbered as follows on Cisco 2600, Cisco 3600 series, and Cisco 7200 series routers: *slot/port***:***ds0-group-number*. Although only one voice port is created for each group, applicable calls are routed to any channel in the group.

Note This command is not supported on Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 platforms for Non-Facility Associated Signaling (NFAS) configuration.

Examples

The following is sample output from the **show voice port** command for an E&M analog voice port:

```
Router# show voice port 1/0/0

E&M Slot is 1, Sub-unit is 0, Port is 0
 Type of VoicePort is E&M
 Operation State is unknown
 Administrative State is unknown
```

```
 The Interface Down Failure Cause is 0
 Alias is NULL
 Noise Regeneration is disabled
 Non Linear Processing is disabled
 Music On Hold Threshold is Set to 0 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is disabled
 Echo Cancel Coverage is set to 16ms
 Connection Mode is Normal
 Connection Number is not set
 Initial Time Out is set to 0 s
 Interdigit Time Out is set to 0 s
 Analog Info Follows:
 Region Tone is set for northamerica
 Currently processing none
 Maintenance Mode Set to None (not in mtc mode)
 Number of signaling protocol errors are 0

 Voice card specific Info Follows:
 Signal Type is wink-start
 Operation Type is 2-wire
 Impedance is set to 600r Ohm
 E&M Type is unknown
 Dial Type is dtmf
 In Seizure is inactive
 Out Seizure is inactive
 Digit Duration Timing is set to 0 ms
 InterDigit Duration Timing is set to 0 ms
 Pulse Rate Timing is set to 0 pulses/second
 InterDigit Pulse Duration Timing is set to 0 ms
 Clear Wait Duration Timing is set to 0 ms
 Wink Wait Duration Timing is set to 0 ms
 Wink Duration Timing is set to 0 ms
 Delay Start Timing is set to 0 ms
 Delay Duration Timing is set to 0 ms
```

The following is sample output from the **show voice port** command for an E&M digital voice port:

```
Router# show voice port 1/0/1

receEive and transMit Slot is 1, Sub-unit is 0, Port is 1
 Type of VoicePort is E&M
 Operation State is DORMANT
 Administrative State is UP
 No Interface Down Failure
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Music On Hold Threshold is Set to -38 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancel Coverage is set to 8 ms
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Region Tone is set for US
```

The following is sample output from the **show voice port** command for a foreign exchange station (FXS) analog voice port:

```
Router# show voice port 1/1/1
```

```
Foreign Exchange Station 1/1/1 Slot is 1, Sub-unit is 1, Port is 1
 Type of VoicePort is FXS  VIC2-2FXS
 Operation State is DORMANT
 Administrative State is UP
 The Last Interface Down Failure Cause is Administrative Shutdown
 Description is I am a FXS LoopStart port
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Non Linear Mute is disabled
 Non Linear Threshold is -21 dB
 Music On Hold Threshold is Set to -38 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 3 dB
 Echo Cancellation is enabled
 Echo Cancellation NLP mute is disabled
 Echo Cancellation NLP threshold is -21 dB
 Echo Cancel Coverage is set to 64 ms
 Echo Cancel worst case ERL is set to 6 dB
 Playout-delay Mode is set to adaptive
 Playout-delay Nominal is set to 60 ms
 Playout-delay Maximum is set to 250 ms
 Playout-delay Minimum mode is set to default, value 40 ms
 Playout-delay Fax is set to 300 ms
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Call Disconnect Time Out is set to 60 s
 Supervisory Disconnect Time Out is set to 750 ms
 Ringing Time Out is set to 180 s
 Wait Release Time Out is set to 30 s
 Companding Type is u-law
 Region Tone is set for US

 Analog Info Follows:
 Currently processing none
 Maintenance Mode Set to None (not in mtc mode)
 Number of signaling protocol errors are 0
 Impedance is set to 600r Ohm
 Station name None, Station number None
 Translation profile (Incoming):
 Translation profile (Outgoing):
 lpcor (Incoming): local_group
 lpcor (Outgoing): local_group

 Voice card specific Info Follows:
 Signal Type is loopStart
 Ring Frequency is 25 Hz
 Hook Status is On Hook
 Ring Active Status is inactive
 Ring Ground Status is inactive
 Tip Ground Status is active
 Digit Duration Timing is set to 100 ms
 InterDigit Duration Timing is set to 100 ms
 Hookflash-in Timing is set to max=1000 ms, min=150 ms
 Hookflash-out Timing is set to 400 ms
 No disconnect acknowledge
 Ring Cadence is defined by CPTone Selection
 Ring Cadence are [20 40] * 100 msec
 Ringer Equivalence Number is set to 1
```

The following is sample output from the **show voice port** command for an FXO analog voice port:

```
Router# show voice port 1/0/1

Foreign Exchange Office 1/0/1 Slot is 1, Sub-unit is 0, Port is 1
 Type of VoicePort is FXO
 Operation State is DORMANT
 Administrative State is UP
 The Last Interface Down Failure Cause is Administrative Shutdown
 Description is I am an FXO LoopStart port
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Non Linear Mute is disabled
 Non Linear Threshold is -21 dB
 Music On Hold Threshold is Set to -38 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 3 dB
 Echo Cancellation is enabled
 Echo Cancellation NLP mute is disabled
 Echo Cancellation NLP threshold is -21 dB
 Echo Cancel Coverage is set to 64 ms
 Echo Cancel worst case ERL is set to 6 dB
 Playout-delay Mode is set to adaptive
 Playout-delay Nominal is set to 60 ms
 Playout-delay Maximum is set to 250 ms
 Playout-delay Minimum mode is set to default, value 40 ms
 Playout-delay Fax is set to 300 ms
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Call Disconnect Time Out is set to 60 s
 Ringing Time Out is set to 180 s
 Wait Release Time Out is set to 30 s
 Companding Type is u-law
 Region Tone is set for US

 Analog Info Follows:
 Currently processing none
 Maintenance Mode Set to None (not in mtc mode)
 Number of signaling protocol errors are 0
 Impedance is set to 600r Ohm
 Station name None, Station number None
 Translation profile (Incoming):
 Translation profile (Outgoing):

 Voice card specific Info Follows:
 Signal Type is loopStart
 Battery-Reversal is enabled
 Number Of Rings is set to 1
 Supervisory Disconnect is signal
 Answer Supervision is inactive
 Hook Status is On Hook
 Ring Detect Status is inactive
 Ring Ground Status is inactive
 Tip Ground Status is inactive
 Dial Out Type is dtmf
 Digit Duration Timing is set to 100 ms
 InterDigit Duration Timing is set to 100 ms
 Pulse Rate Timing is set to 10 pulses/second
 InterDigit Pulse Duration Timing is set to 750 ms
 Percent Break of Pulse is 60 percent
 GuardOut timer is 2000 ms
 Minimum ring duration timer is 125 ms
 Hookflash-in Timing is set to 600 ms
 Hookflash-out Timing is set to 400 ms
```

```
 Supervisory Disconnect Timing (loopStart only) is set to 750 ms
 OPX Ring Wait Timing is set to 6000 ms
```

The following is sample output from the **show voice port summary** command. Note that for the connected FXO analog voice port 0/2/0, which has the ADMIN state of "up" and the OPER state of "dorm," this output shows that the IN STATUS is "idle" and the OUT STATUS is "on-hook":

```
Router# show voice port summary
                                               IN       OUT
PORT            CH   SIG-TYPE   ADMIN OPER STATUS   STATUS   EC
=============== == ============ ===== ==== ======== ======== ==
0/0/0           --  fxs-ls      up    dorm on-hook  idle     y
0/0/1           --  fxs-ls      up    dorm on-hook  idle     y
0/3/0:23        01  isdn-voice  up    dorm none     none     y
0/3/0:23        02  isdn-voice  up    dorm none     none     y
.
.
.
0/1/0           --  did-in-wnk  up    dorm idle     idle     y
0/1/1           --  did-in-wnk  up    dorm idle     idle     y
0/2/0           --  fxo-ls      up    dorm idle     on-hook  y
0/2/1           --  fxo-ls      up    down idle     off-hook y
2/0/0           --  fxs-ls      up    dorm on-hook  idle     y
2/0/1           --  fxs-ls      up    dorm on-hook  idle     y
2/0/2           --  fxs-ls      up    dorm on-hook  idle     y
2/0/3           --  fxs-ls      up    dorm on-hook  idle     y
2/0/4           --  fxs-ls      up    dorm on-hook  idle     y
2/0/5           --  fxs-ls      up    dorm on-hook  idle     y
2/0/6           --  fxs-ls      up    dorm on-hook  idle     y
2/0/7           --  fxs-ls      up    dorm on-hook  idle     y
```

Note If the FXO port 0/2/0 is disconnected, the output of the **show voice port summary** command changes so that the OUT STATUS is reported as "off-hook," and the OPER state changes to "down."

The following is sample output from the **show voice port** command for an ISDN voice port:

```
Router# show voice port

ISDN 2/0:23 Slot is 2, Sub-unit is 0, Port is 23
 Type of VoicePort is ISDN-VOICE
 Operation State is DORMANT
 Administrative State is UP
 No Interface Down Failure
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Non Linear Mute is disabled
 Non Linear Threshold is -21 dB
 Music On Hold Threshold is Set to -38 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancellation NLP mute is disabled
 Echo Cancellation NLP threshold is -21 dB
 Echo Cancel Coverage is set to 64 ms
 Echo Cancel worst case ERL is set to 6 dB
 Playout-delay Mode is set to adaptive
 Playout-delay Nominal is set to 60 ms
 Playout-delay Maximum is set to 250 ms
 Playout-delay Minimum mode is set to default, value 40 ms
 Playout-delay Fax is set to 300 ms
```

```
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Call Disconnect Time Out is set to 60 s
 Ringing Time Out is set to 180 s
 Wait Release Time Out is set to 30 s
 Companding Type is u-law
 Region Tone is set for US
 Station name None, Station number None
 Translation profile (Incoming):
 Translation profile (Outgoing):
 Voice class called number pool:

DS0 channel specific status info:
                                      IN      OUT
    PORT       CH  SIG-TYPE    OPER STATUS   STATUS    TIP     RING
    2/0:23     01  isdn-voice  up   none     none
    2/0:23     02  isdn-voice  up   none     none
    2/0:23     03  isdn-voice  up   none     none
    2/0:23     04  isdn-voice  up   none     none
    2/0:23     05  isdn-voice  up   none     none
    2/0:23     06  isdn-voice  up   none     none
    2/0:23     07  isdn-voice  dorm none     none
    2/0:23     08  isdn-voice  dorm none     none
    2/0:23     09  isdn-voice  dorm none     none
    2/0:23     10  isdn-voice  dorm none     none
    2/0:23     11  isdn-voice  dorm none     none
    2/0:23     12  isdn-voice  dorm none     none
    2/0:23     13  isdn-voice  dorm none     none
    2/0:23     14  isdn-voice  dorm none     none
    2/0:23     15  isdn-voice  dorm none     none
    2/0:23     16  isdn-voice  dorm none     none
    2/0:23     17  isdn-voice  dorm none     none
    2/0:23     18  isdn-voice  dorm none     none
    2/0:23     19  isdn-voice  dorm none     none
    2/0:23     20  isdn-voice  dorm none     none
    2/0:23     21  isdn-voice  dorm none     none
    2/0:23     22  isdn-voice  dorm none     none
    2/0:23     23  isdn-voice  dorm none     none
```

The following is sample output from the **show voice port** command for the connected FXO analog voice port 0/2/0, which has the Administrative State of “UP” and the Operation State of “DORMANT”:

```
Router# show voice port 0/2/0

Foreign Exchange Office 0/2/0 Slot is 0, Sub-unit is 2, Port is 0
 Type of VoicePort is FXO
 Operation State is DORMANT
 Administrative State is UP
 No Interface Down Failure
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Non Linear Mute is disabled
 Non Linear Threshold is -21 dB
 Music On Hold Threshold is Set to -38 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 3 dB
 Echo Cancellation is enabled
 Echo Cancellation NLP mute is disabled
 Echo Cancellation NLP threshold is -21 dB
 Echo Cancel Coverage is set to 128 ms
 Echo Cancel worst case ERL is set to 6 dB
```

```
Playout-delay Mode is set to adaptive
Playout-delay Nominal is set to 60 ms
Playout-delay Maximum is set to 1000 ms
Playout-delay Minimum mode is set to default, value 40 ms
Playout-delay Fax is set to 300 ms
Connection Mode is normal
Connection Number is not set
Initial Time Out is set to 15 s
Interdigit Time Out is set to 10 s
Call Disconnect Time Out is set to 60 s
Power Denial Disconnect Time Out is set to 1000 ms
Ringing Time Out is set to 180 s
Wait Release Time Out is set to 30 s
Companding Type is u-law
Region Tone is set for US

Analog Info Follows:
Currently processing none
Maintenance Mode Set to None (not in mtc mode)
Number of signaling protocol errors are 0
Impedance is set to 600r Ohm
Station name None, Station number None
Translation profile (Incoming):
Translation profile (Outgoing):
lpcor (Incoming):
lpcor (Outgoing):

Voice card specific Info Follows:
Signal Type is loopStart
Battery-Reversal is enabled
Number Of Rings is set to 1
Supervisory Disconnect is signal
Answer Supervision is inactive
Hook Status is On Hook
Ring Detect Status is inactive
Ring Ground Status is inactive
Tip Ground Status is inactive
Dial Out Type is dtmf
Digit Duration Timing is set to 100 ms
InterDigit Duration Timing is set to 100 ms
Pulse Rate Timing is set to 10 pulses/second
InterDigit Pulse Duration Timing is set to 750 ms
Percent Break of Pulse is 60 percent
GuardOut timer is 2000 ms
Minimum ring duration timer is 125 ms
Hookflash-in Timing is set to 600 ms
Hookflash-out Timing is set to 400 ms
Supervisory Disconnect Timing (loopStart only) is set to 350 ms
OPX Ring Wait Timing is set to 6000 ms
Secondary dialtone is disabled
```

Note If the FXO port 0/2/0 is disconnected, the output of the **show voice port** command changes so that the Administrative State remains "UP" but the Operation State is "DOWN."

Beginning in Cisco IOS Release 15.1(3)T, there is improved status monitoring of FXO ports—any time an FXO port is connected or disconnected, a message is displayed to indicate the status change. For example, the following message is displayed to report that a cable has been connected, and the status is changed to "up" for FXO port 0/2/0:

```
000118: Jul 14 18:06:05.122 EST: %LINK-3-UPDOWN: Interface Foreign Exchange Office 0/2/0,
changed state to operational status up due to cable reconnection
```

Table 208 describes significant fields shown in these outputs, in alphabetical order.

Table 208 show voice port Field Descriptions

Field	Description
Administrative State	Administrative state of the voice port.
Alias	User-supplied alias for the voice port.
Clear Wait Duration Timing	Time (in milliseconds [ms]) of inactive seizure signal to declare call cleared.
Companding Type	Companding standard used to convert between analog and digital signals in pulse code modulation (PCM) systems.
Connection Mode	Connection mode of the interface.
Connection Number	Full E.164 telephone number used to establish a connection with the trunk or private line automatic ringdown (PLAR) mode.
Currently Processing	Type of call currently being processed: none, voice, or fax.
Delay Duration Timing	Maximum delay signal duration (in ms) for delay dial signaling.
Delay Start Timing	Timing (in ms) of generation of delayed start signal from detection of incoming seizure.
Dial Type	Out-dialing type of the voice port.
Digit Duration Timing	Dual-tone multifrequency (DTMF) digit duration (in ms).
E&M Type	Type of E&M interface.
Echo Cancel Coverage	Echo cancel coverage for this port.
Echo Cancellation	Whether echo cancellation is enabled for this port.
Impedance	Configured terminating impedance for the E&M interface.
In Gain	Amount of gain (in decibels [dB]) inserted at the receiver side of the interface.
In Seizure	Incoming seizure state of the E&M interface.
Initial Time Out	Amount of time (in seconds) the system waits for an initial input digit from the caller.
Interdigit Duration Timing	DTMF interdigit duration (in seconds).
InterDigit Pulse Duration Timing	Pulse dialing interdigit timing (in ms).

Table 208 show voice port Field Descriptions (continued)

Field	Description
Interdigit Time Out	Amount of time (in seconds) the system waits for a subsequent input digit from the caller.
Lpcor (Incoming)	Setting of the **lpcor incoming** command.
Lpcor (Outgoing)	Setting of the **lpcor outgoing** command.
Maintenance Mode	Maintenance mode of the voice port.
Music On Hold Threshold	Configured music-on-hold threshold value for this interface.
Noise Regeneration	Whether background noise should be played to fill silent gaps if voice activity detection (VAD) is activated.
Non Linear Processing	Whether nonlinear processing is enabled for this port.
Number of signaling protocol errors	Number of signaling protocol errors.
Operation State	Operational state of the voice port.
Operation Type	Operation type of the E&M signal: 2-wire or 4-wire.
Out Attenuation	Amount of attenuation (in dB) inserted at the transmit side of the interface.
Out Seizure	Outgoing seizure state of the E&M interface.
Port	Port number for the interface associated with the voice interface card.
Pulse Rate Timing	Pulse dialing rate, in pulses per second (pps).
Region Tone	Configured regional tone for this interface.
Ring Active Status	Ring active indication.
Ring Cadence	Configured ring cadence for this interface.
Ring Frequency	Configured ring frequency (in hertz) for this interface.
Ring Ground Status	Ring ground indication.
Ringing Time Out	Ringing timeout duration (in seconds).
Signal Type	Type of signaling for a voice port: delay-dial, ground-start, immediate, loop-start, and wink-start.
Slot	Slot used in the voice interface card for this port.
Sub-unit	Subunit used in the voice interface card for this port.
Tip Ground Status	Tip ground indication.
Type of VoicePort	Type of voice port: FXO, FXS, or E&M.
The Interface Down Failure Cause	Text string describing why the interface is down,
Wait Release Time Out	Length of time (in seconds) that a voice port stays in call-failure state while a busy tone, reorder tone, or out-of-service tone is sent to the port.
Wink Duration Timing	Maximum wink duration (in ms) for wink-start signaling.
Wink Wait Duration Timing	Maximum wink wait duration (in ms) for wink-start signaling.

Related Commands

Command	Description
ds0 group	Specifies the DS0 time slots that make up a logical voice port on a T1 or E1 controller and specifies the signaling type by which the router communicates with the PBX or PSTN.
timing sup-disconnect	Defines the minimum time to ensure that an on-hook indication is intentional and not an electrical transient on the line before a supervisory disconnect occurs (based on power denial signaled by the PSTN or PBX).

show voice source-group

To display the details of one or more voice source IP groups, use the **show voice source-group** command in privileged EXEC mode.

show voice source-group [*name* | **sort** [**ascending** | **descending**]]

Syntax Description

name	(Optional) Name of the source IP group to display.
sort [**ascending** | **descending**]	(Optional) Displays the source IP groups in either ascending or descending alphanumerical order.

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 shows an invalid configuration.

```
Router# show voice source-group abc

Source Group: abc
    description="",
    carrier-id source="sj_area",
    carrier-id target="",
    trunk-group-label source="",
    trunk-group-label target="ny_main",
    h323zone-id="",
    access-list=,
    disconnect-cause="no-service",
    translation-profile="",
```

The following sample output shows a valid configuration for carrier-ID routing:

```
Router# show voice source-group abc

Source Group: abc
    description="",
    carrier-id source="",
    carrier-id target="",
    trunk-group-label source="texas_backup",
    trunk-group-label target="ny_main",
    h323zone-id="",
    access-list=,
    disconnect-cause="no-service",
    translation-profile="",
```

If you are using carrier-ID routing, both carrier-ID fields are filled in and the "trunk-group-label" fields are blank.

The following sample output displays the source groups in ascending order. Both source IP groups use carrier-ID routing.

```
Router# show voice source-group sort ascending

Source Group:1
        description="routec calls from 1311 to 1411",
        carrier-id source="1311",
        carrier-id target="1411",
        trunk-group-label source="",
        trunk-group-label target="",
        h323zone-id="fr1311",
        access-list= ,
        disconnect-cause="user-busy",
        destination-pattern="",
        incoming called-number="",
        translation-profile="10",

Source Group:2
        description="",
        carrier-id source="abcd",
        carrier-id target="xyz",
        trunk-group-label source="",
        trunk-group-label target="",
        h323zone-id="",
        access-list= ,
        disconnect-cause="no-service",
        destination-pattern="",
        incoming called-number="",
        translation-profile="",
```

Table 209 describes significant fields shown in this output.

Table 209 show voice source-group Field Descriptions

Field	Description
Source Group	Name of the voice source IP group.
description	Description of the voice source IP group.
carrier-id source	Name of the source carrier ID used by the terminating gateway to select a target carrier.
carrier-id target	Name of the target carrier ID used by the terminating gateway to select a dial peer for routing the call over a POTS line.
trunk-group-label source	Name of the source trunk group used by the originating gateway to route the call over an inbound dial peer.
trunk-group-label target	Name of the target trunk group used by the terminating gateway to select a dial peer for routing an outbound call over a POTS line.
h323zone-id	Name of the zone associated with incoming H.323 calls to the voice source IP group.
access-list	Number of the access list used by the voice source IP group to block calls.
disconnect-cause	Phrase returned by the voice source IP group when a call is blocked.
translation-profile	Name of the translation profile used by the voice source IP group to translate calls.

Related Commands

Command	Description
voice source-group	Initiates a voice source IP group definition.

show voice statistics csr interval accounting

To display accounting statistics by configured intervals, use the **show voice statistics csr interval accounting** command in privileged EXEC mode.

show voice statistics csr interval tag-number accounting {**all** | **method-list** *method-list-name*} [**push** {**all** | **ftp** | **syslog**}]

Syntax Description

tag-number	Interval that represents a specified time range. The valid range is from 1 to 36655. **Note** You must first enter the **show voice statistics interval-tag** command to obtain the valid tag numbers that you can enter for this command.
all	Displays all voice accounting statistics.
method-list-name *method-list-name*	Displays accounting statistics by method list. You must specify a method-list name.
push	(Optional) Statistics are downloaded to an FTP or syslog server, or to both servers. The keywords are as follows: • **all**—Pushes statistics to both the FTP and syslog servers. • **ftp**—Pushes statistics to the FTP server. • **syslog**—Pushes statistics to the syslog server.

Command Modes

Privileged EXEC

Command History

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

Examples

The following sample output shows all of the statistics that were collected for interval tag 102 for method list h323-1:

```
Router# show voice statistics csr interval 102 accounting method-list h323-1

Client Type: Voice ACCT Stats
        Start Time: 2002-05-01T19:35:17Z        End Time: 2002-05-01T19:36:29Z
methodlist=h323-1,acc_pass_criteria=1,pstn_in_pass=0,pstn_in_fail=0,pstn_out_pass=0,
pstn_out_fail=0,ip_in_pass=0,ip_in_fail=0,ip_out_pass=0,ip_out_fail=0
```

Table 210 lists and describes the significant output fields.

Table 210 show voice statistics csr interval accounting Field Descriptions

Field	Description
Client Type	The type of statistics collected.
Start Time	The start time of the statistics collection.
End Time	The ending time of the statistics collection.
method-list	The method list name.
acc_pass_criteria	Accounting pass criteria: • 1: all start/interim/stop messages passed. • 2: all start/stop messages passed. • 3: stop-only message passed.
pstn_in_pass	Number of incoming calls on the PSTN leg that meet acc_pass_criteria.
pstn_in_fail	Number of incoming calls on the PSTN leg that fail acc_pass_criteria.
pstn_out_pass	Number of outgoing calls on the PSTN leg that meet acc_pass_criteria.
pstn_out_fail	Number of outgoing calls on the PSTN leg that fail acc_pass_criteria.
ip_in_pass	Number of incoming calls on the IP leg that meet acc_pass_criteria.
ip_in_fail	Number of incoming calls on the IP leg that fail acc_pass_criteria.
ip_out_pass	Number of outgoing calls on the IP leg that meet acc_pass_criteria.
ip_out_fail	Number of outgoing calls on the IP leg that fail acc_pass_criteria.

Related Commands

Command	Description
show event-manager consumers	Displays event-manager statistics.
show voice statistics csr interval aggregation	Displays statistical information by configured intervals for signaling statistics.
show voice statistics csr since-reset accounting	Displays all accounting CSRs since the last reset.
show voice statistics csr since-reset aggregation-level	Displays all signaling CSRs since the last reset.
show voice statistics csr since-reset all	Displays all CSRs since the last reset.
show voice statistics interval-tag	Displays the configured interval numbers.
show voice statistics memory-usage	Displays current memory usage.

show voice statistics csr interval aggregation

To display signaling statistics by configured intervals, use the **show voice statistics csr interval aggregation** command in privileged EXEC mode.

show voice statistics csr interval tag-number aggregation {**all** | **gateway** | **ip** | **pstn** | **trunk-group** {*trunk-group-label* | **all** } | **voice-port** {*voice-port-label* | **all**}} [**mode** {**concise** | **verbose**}] [**push** {**all** | **ftp** | **syslog**}]

Syntax Description

tag-number	Interval that represents a specified time range. The valid range is from 1 to 36655. **Note** You must first enter the **show voice statistics interval-tag** command to obtain the valid tag numbers that you can enter for this command.
all	Displays all levels of signaling statistics.
gateway	Displays gateway-wide level statistics.
ip	Displays VoIP interface level statistics.
pstn	Displays telephone interface level statistics.
trunk-group	Displays trunk-group level statistics. • *trunk-group-label*—displays statistics for a specific trunk group • **all**—Displays statistics for all trunk groups.
voice-port	Displays voice-port level statistics: • *voice-port-label*—displays statistics for a specific voice port • **all**—Displays statistics for all voice ports.
mode	(Optional) Statistics are displayed in a specified mode. The keywords are as follows: • **concise**—Displays output that contains total calls, answered calls, and answered call duration. • **verbose**—Displays all fields contained in call statistic records (CSRs). This is the default setting.
push	(Optional) Statistics are downloaded to an FTP or syslog server, or to both servers. The keywords are as follows: • **all**—Pushes statistics to both the FTP and syslog servers. • **ftp**—Pushes statistics to the FTP server. • **syslog**—Pushes statistics to the syslog server.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

This command is valid only if the **voice statistics time-range** command is configured to either the **periodic** or **start-stop** value. If you enter the **show voice statistics csr interval aggregation** command but the gateway has been configured to collect statistics only since the last reset, the gateway displays an error message.

You must first enter the **show voice statistics interval-tag** to obtain the valid tag numbers that you can enter for this command.

Examples

The following sample output shows signaling statistics for all aggregation levels for interval tag 200:

```
Router# show voice statistics csr interval 200 aggregation all

Client Type: VCSR
        Start Time: 2002-04-28T01:48:24Z        End Time: 2002-04-28T01:50:01Z

record_type=gw,trunk_group_id=,voice_port_id=,in_call=0,in_ans=0,in_fail=0,out_call=0,
out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,orig_disconn=0,
in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,in_setup_delay=0,
out_setup_delay=0,lost_pkt=0,latency=0,jitter=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=ip,trunk_group_id=,voice_port_id=,in_call=0,in_ans=0,in_fail=0,out_call=0,
out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,orig_disconn=0,
in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,in_setup_delay=0,
out_setup_delay=0,lost_pkt=0,latency=0,jitter=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=pstn,trunk_group_id=,voice_port_id=,in_call=0,in_ans=0,in_fail=0,out_call=0,
out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,orig_disconn=0,
in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,in_setup_delay=0,
out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=vp,trunk_group_id=,voice_port_id=4/0/0,in_call=0,in_ans=0,in_fail=0,
out_call=0,out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,
orig_disconn=0,in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,
in_setup_delay=0,out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=vp,trunk_group_id=,voice_port_id=4/0/1,in_call=0,in_ans=0,in_fail=0,
out_call=0,out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,
orig_disconn=0,in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,
in_setup_delay=0,out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=vp,trunk_group_id=,voice_port_id=4/1/0,in_call=0,in_ans=0,in_fail=0,
out_call=0,out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,
orig_disconn=0,in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,
in_setup_delay=0,out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=vp,trunk_group_id=,voice_port_id=4/1/1,in_call=0,in_ans=0,in_fail=0,
out_call=0,out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,
orig_disconn=0,in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,
in_setup_delay=0,out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=vp,trunk_group_id=,voice_port_id=2/0:23,in_call=0,in_ans=0,in_fail=0
,out_call=0,out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,
orig_disconn=0,in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,
in_setup_delay=0,out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=vp,trunk_group_id=,voice_port_id=2/1:23,in_call=0,in_ans=0,in_fail=0
out_call=0,out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,
orig_disconn=0,in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,
in_setup_delay=0,out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
```

Table 211 lists and describes the significant output fields.

Table 211 show voice statistics csr interval aggregation Field Descriptions

Field	Description
Client Type	The type of statistics collected.
Start Time	The start time of the statistics collection.
End Time	The ending time of the statistics collection.
record_type	Call statistics record type. Symbols are gw, ip, pstn, tg, and vp.
trunk_group_id	Trunk group ID. **Note** For the symbols gw, ip, pstn, and some vp records, this field is empty.
voice_port_id	Voice port ID. **Note** For the symbols gw, ip, pstn, and some vp records, this field is empty.
in_call	Number of incoming calls.
in_ans	Number of incoming calls answered by the gateway.
in_fail	Number of incoming calls that failed.
out_call	Number of outgoing calls attempted.
out_ans	Number of outgoing calls that received answers.
out_fail	Number of outgoing calls that failed.
in_szre_d	Incoming seizure duration (in seconds).
out_szre_d	Outgoing seizure duration (in seconds).
in_conn_d	Incoming connected duration (in seconds).
out_conn_d	Outgoing connected duration (in seconds).
orig_disconn	Number of calls encountering the originating side having been disconnected before the outgoing calls were connected.
in_ans_abnorm	Number of incoming answered calls terminated with any cause code other than "normal".
out_ans_abnorm	Number of outgoing answered calls terminated with any cause code other than "normal".
in_mcd	Number of incoming calls lasting less than the configured minimum call duration (MCD).
out_mcd	Number of outgoing calls lasting less than the configured MCD.
in_pdd	Total post dial delay duration on incoming calls (in ms).
out_pdd	Total post dial delay duration on outgoing calls (in ms).
in_setup_delay	Total inbound setup delay duration (in ms).
out_setup_delay	Total outbound setup delay duration (in ms).
lost_pkt	Number of calls losing more than the configured number of packets. **Note** This field will exist only in IP records. In other types of records, this field will be empty and extra commas are expected.

Table 211 show voice statistics csr interval aggregation Field Descriptions (continued)

Field	Description
latency	Number of calls encountering more than the configured amount of latency. **Note** This field will exist only in IP records. In other types of records, this field will be empty and extra commas are expected.
jitter	Number of calls encountering more than configured amount of jitter. **Note** This field will exist only in IP records. In other types of records, this field will be empty and extra commas are expected.
in_cc_no	Number of the following disconnect cause code counters as per incoming calls (expected to be fewer than 5).
in_disc_cc	Incoming disconnect cause code. For example, in_disc_cc_16=3 indicates that 3 calls were disconnected or finished with a disconnect cause code of 16 (normal).
out_disc_cc	Outgoing disconnect cause code.
out_cc_no	Number of the following disconnect cause code counters as per outgoing calls (expected to be fewer than 5).
in_cc_id	Disconnect cause code ID for the following field for incoming calls.
in_cc_cntr	Disconnect cause code counter for incoming calls (any incoming cause code counter pairs).
out_cc_id	Disconnect cause code ID for the following field for outgoing calls.
out_cc_cntr	Disconnect cause code counter for outgoing calls (any outgoing cause code counter pairs).

Related Commands

Command	Description
show event-manager consumers	Displays event statistics.
show voice statistics csr interval accounting	Displays statistical information by configured intervals for accounting statistics.
show voice statistics csr since-reset accounting	Displays all accounting CSRs since the last reset.
show voice statistics csr since-reset aggregation-level	Displays all signaling CSRs since the last reset.
show voice statistics csr since-reset all	Displays all CSRs since the last reset.
show voice statistics interval-tag	Displays the configured interval numbers.
show voice statistics memory-usage	Displays current memory usage.
voice statistics time-range	Specifies the time range to collect CSRs.

show voice statistics csr since-reset accounting

To display VoIP AAA accounting statistics since the last reset, use the **show voice statistics csr since-reset accounting** command in privileged EXEC mode.

show voice statistics csr since-reset accounting {**all** | **method-list** *method-list-name*} [**push** {**all** | **ftp** | **syslog**}]

Syntax Description

all	All collected statistics since the last reset are displayed.
method-list *method-list-name*	Collected statistics by method list since the last reset are displayed. The *method-list-name* argument specifies the name of the method list.
push	(Optional) Statistics are downloaded to an FTP or syslog server, or to both servers. The keywords are as follows: • **all**—Pushes statistics to both the FTP and syslog servers. • **ftp**—Pushes statistics to the FTP server. • **syslog**—Pushes statistics to the syslog server.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

This command only applies if the **voice statistics time-range** command is configured to the **since-reset** value. Voice statistics collection on the gateway is reset using the **clear voice statistics csr** command.

If you enter the **show voice statistics csr since-reset accounting** command but the gateway has been configured for periodic collection or to a specific interval, the gateway will display an error message.

Examples

The following sample output shows the accounting statistics for method list h323-1 since the last reset:

```
Router# show voice statistics csr since-reset accounting method-list h323-1

Client Type: Voice ACCT Stats
        Start Time: 2002-05-05T17:39:17Z        End Time: 2002-05-09T19:00:16Z
methodlist=h323-1,acc_pass_criteria=1,pstn_in_pass=0,pstn_in_fail=1,pstn_out_pass=0,
pstn_out_fail=0,ip_in_pass=0,ip_in_fail=0,ip_out_pass=0,ip_out_fail=1
```

Table 212 lists and describes the significant output fields.

Table 212 show voice statistics csr since-reset accounting Field Descriptions

Field	Description
Client Type	The type of statistics collected.
Start Time	The start time of the statistics collection.

Table 212 show voice statistics csr since-reset accounting Field Descriptions (continued)

Field	Description
End Time	The ending time of the statistics collection.
method-list	The method list name.
acc_pass_criteria	Accounting pass criteria: • 1: all start/interim/stop messages passed. • 2: all start/stop messages passed. • 3: stop-only message passed.
pstn_in_pass	Number of incoming calls on the PSTN leg that meet acc_pass_criteria.
pstn_in_fail	Number of incoming calls on the PSTN leg that fail acc_pass_criteria.
pstn_out_pass	Number of outgoing calls on the PSTN leg that meet acc_pass_criteria.
pstn_out_fail	Number of outgoing calls on the PSTN leg that fail acc_pass_criteria.
ip_in_pass	Number of incoming calls on the IP leg that meet acc_pass_criteria.
ip_in_fail	Number of incoming calls on the IP leg that fail acc_pass_criteria.
ip_out_pass	Number of outgoing calls on the IP leg that meet acc_pass_criteria.
ip_out_fail	Number of outgoing calls on the IP leg that fail acc_pass_criteria.

Related Commands

Command	Description
clear voice statistics	Clears voice statistics, resetting the statistics collection.
show event-manager consumers	Displays event statistics.
show voice statistics csr interval accounting	Displays statistical information by configured intervals for accounting statistics.
show voice statistics csr interval aggregation	Displays statistical information by configured intervals for signaling statistics.
show voice statistics csr since-reset aggregation-level	Displays all signaling CSRs since the last reset.
show voice statistics interval-tag	Displays the configured interval numbers
show voice statistics memory-usage	Displays current memory usage.
voice statistics time-range	Specifies a time range to collect statistics from the gateway on a periodic basis, since the last reset, or for a specific time duration.

show voice statistics csr since-reset aggregation-level

To display signaling statistics since the last reset, use the **show voice statistics csr since-reset aggregation-level** command in privileged EXEC mode.

show voice statistics csr since-reset aggregation-level {**all** | **gateway** | **ip** | **pstn** | **trunk-group** {**all** | *trunk-group-label*} | **voice-port** {**all** | *voice-port-label*}} [**mode** {**concise** | **verbose**}] [**push** {**all** | **ftp** | **syslog**}]

Syntax Description

all	All signaling statistics.
gateway	Gateway-wide level statistics.
ip	VoIP-interface-level statistics.
pstn	PSTN-level statistics.
trunk-group	Trunk-group-level statistics. Keywords and arguments are as follows. • **all**—Statistics for all trunk groups. • *trunk-group-label*—Statistics for a specific trunk group.
voice-port	Voice-port-level statistics. Keywords and arguments are as follows: • **all**—Statistics for all voice ports. • *voice-port-label*—Statistics for a specific voice port.
mode	(Optional) Statistics in a specified mode. Keywords are as follows: • **concise—**Output contains total calls, answered calls, and answered call duration. • **verbose**—All fields contained in call statistic records (CSRs). This is the default.
push	(Optional) Statistics are downloaded to an FTP or syslog server, or to both servers. Keywords are as follows: • **all**—Pushes statistics to both the FTP and syslog servers. • **ftp**—Pushes statistics to the FTP server. • **syslog**—Pushes statistics to the syslog server.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

This command applies only if the **voice statistics time-range** command is configured to the **since-reset** value. Voice statistics collection on the gateway is reset using the **clear voice statistics csr** command.

If you enter the **show voice statistics csr since-reset aggregation-level** command but the gateway has been configured for periodic collection or to a specific interval, the gateway will display an error message.

Examples

The following sample output shows signaling statistics for all aggregation levels since the last reset:

```
Router# show voice statistics csr since-reset aggregation-level all

Client Type: VCSR
        Start Time: 2002-04-25T01:48:12Z        End Time: 2002-04-25T01:50:01Z

record_type=gw,trunk_group_id=,voice_port_id=,in_call=0,in_ans=0,in_fail=0,out_call=0,
out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,orig_disconn=0,
in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,in_setup_delay=0,
out_setup_delay=0,lost_pkt=0,latency=0,jitter=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=ip,trunk_group_id=,voice_port_id=,in_call=0,in_ans=0,in_fail=0,out_call=0,
out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,orig_disconn=0,
in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,in_setup_delay=0,
out_setup_delay=0,lost_pkt=0,latency=0,jitter=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=pstn,trunk_group_id=,voice_port_id=,in_call=0,in_ans=0,in_fail=0,out_call=0,
out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,orig_disconn=0,
in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,in_setup_delay=0,
out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=vp,trunk_group_id=,voice_port_id=4/0/0,in_call=0,in_ans=0,in_fail=0,
out_call=0,out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,
orig_disconn=0,in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,
in_setup_delay=0,out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=vp,trunk_group_id=,voice_port_id=4/0/1,in_call=0,in_ans=0,in_fail=0,
out_call=0,out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,
orig_disconn=0,in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,
in_setup_delay=0,out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=vp,trunk_group_id=,voice_port_id=4/1/0,in_call=0,in_ans=0,in_fail=0,
out_call=0,out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,
orig_disconn=0,in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,
in_setup_delay=0,out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=vp,trunk_group_id=,voice_port_id=4/1/1,in_call=0,in_ans=0,in_fail=0,
out_call=0,out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,
orig_disconn=0,in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,
in_setup_delay=0,out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=vp,trunk_group_id=,voice_port_id=2/0:23,in_call=0,in_ans=0,in_fail=0
,out_call=0,out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,
orig_disconn=0,in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,
in_setup_delay=0,out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=vp,trunk_group_id=,voice_port_id=2/1:23,in_call=0,in_ans=0,in_fail=0
out_call=0,out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,
orig_disconn=0,in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,
in_setup_delay=0,out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
```

The following sample output shows signaling statistics for the IP aggregation level since the last reset:

```
Router# show voice statistics csr since-reset aggregation-level ip

Client Type: VCSR
        Start Time: 2002-04-25T01:48:12Z        End Time: 2002-05-02T21:21:27Z

record_type=ip,trunk_group_id=10,voice_port_id=2,in_call=15,in_ans=15,in_fail=0,out_call=0
out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,orig_disconn=0,
in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,in_setup_delay=0,
out_setup_delay=0,lost_pkt=0,latency=0,jitter=0,in_disc_cc_16=0,out_disc_cc_16=0
```

The following sample output shows signaling statistics for the PSTN aggregation level since the last reset:

```
Router# show voice statistics csr since-reset aggregation-level pstn

Client Type: VCSR
        Start Time: 2002-04-25T01:48:12Z        End Time: 2002-05-02T21:21:42Z

record_type=pstn,trunk_group_id=25,voice_port_id=2,in_call=100,in_ans=10,in_fail=90,
out_call=0,out_ans=0,out_fail=0,in_szre_d=100,out_szre_d=0,in_conn_d=0,out_conn_d=0,
orig_disconn=0,in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,
in_setup_delay=0,out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
```

Table 213 lists and describes the significant output fields.

Table 213 show voice statistics csr since-reset aggregation-level Field Descriptions

Field	Description
Client Type	The type of statistics collected.
Start Time	The start time of the statistics collection.
End Time	The ending time of the statistics collection.
record_type	Call statistics record type. Symbols are gw, ip, pstn, tg, and vp.
trunk_group_id	Trunk group ID. **Note** For the symbols gw, ip, pstn, and some vp records, this field is empty.
voice_port_id	Voice port ID. **Note** For the symbols gw, ip, pstn, and some vp records, this field is empty.
in_call	Number of incoming calls.
in_ans	Number of incoming calls answered by the gateway.
in_fail	Number of incoming calls that failed.
out_call	Number of outgoing calls attempted.
out_ans	Number of outgoing calls that received answers.
out_fail	Number of outgoing calls that failed.
in_szre_d	Incoming seizure duration (in seconds).
out_szre_d	Outgoing seizure duration (in seconds).
in_conn_d	Incoming connected duration (in seconds).
out_conn_d	Outgoing connected duration (in seconds).
orig_disconn	Number of calls encountering the originating side having been disconnected before the outgoing calls were connected.
in_ans_abnorm	Number of incoming answered calls terminated with any cause code other than “normal”.
out_ans_abnorm	Number of outgoing answered calls terminated with any cause code other than “normal”.
in_mcd	Number of incoming calls lasting less than the configured minimum call duration (MCD).
out_mcd	Number of outgoing calls lasting less than the configured MCD.
in_pdd	Total post dial delay duration on incoming calls (in ms).

Table 213 show voice statistics csr since-reset aggregation-level Field Descriptions (continued)

Field	Description
out_pdd	Total post dial delay duration on outgoing calls (in ms).
in_setup_delay	Total inbound setup delay duration (in ms).
out_setup_delay	Total outbound setup delay duration (in ms).
lost_pkt	Number of calls losing more than the configured number of packets. **Note** This field will exist only in IP records. In other types of records, this field will be empty and extra commas are expected.
latency	Number of calls encountering more than the configured amount of latency. **Note** This field will exist only in "IP" records. In other types of records, this field will be empty and extra commas are expected.
jitter	Number of calls encountering more than configured amount of jitter. **Note** This field will exist only in IP records. In other types of records, this field will be empty and extra commas are expected.
in_disc_cc	Incoming disconnect cause code. For example, in_disc_cc_16=3 indicates that 3 calls were disconnected or finished with a disconnect cause code of 16 (normal).
out_disc_cc	Outgoing disconnect cause code.
in_cc_no	Number of the following disconnect cause code counters as per incoming calls (expected to be fewer than 5).
out_cc_no	Number of the following disconnect cause code counters as per outgoing calls (expected to be fewer than 5).
in_cc_id	Disconnect cause code ID for the following field for incoming calls.
in_cc_cntr	Disconnect cause code counter for incoming calls (any incoming cause code counter pairs).
out_cc_id	Disconnect cause code ID for the following field for outgoing calls.
out_cc_cntr	Disconnect cause code counter for outgoing calls (any outgoing cause code counter pairs).

Related Commands

Command	Description
clear voice statistics	Clears voice statistics, resetting the statistics collection.
clear voice statistics csr	Clears voice-statistic collection settings on the gateway.
show event-manager consumers	Displays event statistics.
show voice statistics csr interval accounting	Displays statistical information by configured intervals for accounting statistics.
show voice statistics csr interval aggregation	Displays statistical information by configured intervals for signaling statistics.
show voice statistics csr since-reset accounting	Displays all accounting CSRs since the last reset.

Command	Description
show voice statistics interval-tag	Displays voice statistics within a specified interval.
show voice statistics memory-usage	Displays current memory usage.
voice statistics time-range	Specifies the time range to collect CSRs.

show voice statistics csr since-reset all

To display all voice call statistical information since a reset occurred, use the **show voice statistics csr since-reset all** command in privileged EXEC mode.

show voice statistics csr since-reset all [**mode** {**concise** | **verbose**}] [**push** {**all** | **ftp** | **syslog**}]

Syntax Description

mode	(Optional) Statistics are displayed in a specified mode. The keywords are as follows: • **concise**—Displays output that contains total calls, answered calls, and answered call duration. • **verbose**—Displays all fields contained in call statistic records (CSRs). This is the default setting.
push	(Optional) Statistics are downloaded to an FTP or syslog server, or to both servers. The keywords are as follows: • **all**—Pushes statistics to both the FTP and syslog servers. • **ftp**—Pushes statistics to the FTP server. • **syslog**—Pushes statistics to the syslog server.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

This command can also be used to display and push VoIP internal error codes (IECs).

Examples

The following example shows all of the statistics that were collected since the last reset:

```
Router# show voice statistics csr since-reset all

Client Type: VCSR
        Start Time: 2002-05-01T19:35:17Z        End Time: 2002-05-01T19:36:26Z

record_type=gw,trunk_group_id=,voice_port_id=,in_call=0,in_ans=0,in_fail=0,out_call=0,
out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,orig_disconn=0,
in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,in_setup_delay=0,
out_setup_delay=0,lost_pkt=0,latency=0,jitter=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=ip,trunk_group_id=,voice_port_id=,in_call=0,in_ans=0,in_fail=0,out_call=0,
out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,orig_disconn=0,
in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,in_setup_delay=0,
out_setup_delay=0,lost_pkt=0,latency=0,jitter=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=pstn,trunk_group_id=,voice_port_id=,in_call=0,in_ans=0,in_fail=0,out_call=0,
out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,orig_disconn=0,
in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,in_setup_delay=0,
```

```
out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=vp,trunk_group_id=,voice_port_id=4/0/0,in_call=0,in_ans=0,in_fail=0,
out_call=0,out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,
orig_disconn=0,in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,
in_setup_delay=0,out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=vp,trunk_group_id=,voice_port_id=4/0/1,in_call=0,in_ans=0,in_fail=0,
out_call=0,out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,
orig_disconn=0,in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,
in_setup_delay=0,out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=vp,trunk_group_id=,voice_port_id=4/1/0,in_call=0,in_ans=0,in_fail=0,
out_call=0,out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,
orig_disconn=0,in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,
in_setup_delay=0,out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=vp,trunk_group_id=,voice_port_id=4/1/1,in_call=0,in_ans=0,in_fail=0,
out_call=0,out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,
orig_disconn=0,in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,
in_setup_delay=0,out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=vp,trunk_group_id=,voice_port_id=2/0:23,in_call=0,in_ans=0,in_fail=0
out_call=0,out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,
orig_disconn=0,in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,
in_setup_delay=0,out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0
!
record_type=vp,trunk_group_id=,voice_port_id=2/1:23,in_call=0,in_ans=0,in_fail=0
out_call=0,out_ans=0,out_fail=0,in_szre_d=0,out_szre_d=0,in_conn_d=0,out_conn_d=0,
orig_disconn=0,in_ans_abnorm=0,out_ans_abnorm=0,in_mcd=0,out_mcd=0,in_pdd=0,out_pdd=0,
in_setup_delay=0,out_setup_delay=0,in_disc_cc_16=0,out_disc_cc_16=0

Client Type: Voice ACCT Stats
        Start Time: 2002-05-01T19:35:17Z        End Time: 2002-05-01T19:36:29Z
methodlist=h323-1,acc_pass_criteria=1,pstn_in_pass=0,pstn_in_fail=0,pstn_out_pass=0,
pstn_out_fail=0,ip_in_pass=0,ip_in_fail=0,ip_out_pass=0,ip_out_fail=0
```

Table 214 lists and describes the significant output fields.

Table 214 show voice statistics csr since-reset all Field Descriptions

Field	Description
Client Type	The type of statistics collected.
Start Time	The start time of the statistics collection.
End Time	The ending time of the statistics collection.
record_type	Call statistics record type. Symbols are gw, ip, pstn, tg, and vp.
trunk_group_id	Trunk group ID. **Note** For the symbols gw, ip, pstn, and some vp records, this field is empty.
voice_port_id	Voice port ID. **Note** For the symbols gw, ip, pstn, and some vp records, this field is empty.
in_call	Number of incoming calls.
in_ans	Number of incoming calls answered by the gateway.

Table 214 show voice statistics csr since-reset all Field Descriptions (continued)

Field	Description
in_fail	Number of incoming calls that failed.
out_call	Number of outgoing calls attempted.
out_ans	Number of outgoing calls that received answers.
out_fail	Number of outgoing calls that failed.
in_szre_d	Incoming seizure duration (in seconds).
out_szre_d	Outgoing seizure duration (in seconds).
in_conn_d	Incoming connected duration (in seconds).
out_conn_d	Outgoing connected duration (in seconds).
orig_disconn	Number of calls encountering the originating side having been disconnected before the outgoing calls were connected.
in_ans_abnorm	Number of incoming answered calls terminated with any cause code other than "normal".
out_ans_abnorm	Number of outgoing answered calls terminated with any cause code other than "normal".
in_mcd	Number of incoming calls lasting less than the configured minimum call duration (MCD).
out_mcd	Number of outgoing calls lasting less than the configured MCD.
in_pdd	Total post dial delay duration on incoming calls (in ms).
out_pdd	Total post dial delay duration on outgoing calls (in ms).
in_setup_delay	Total inbound setup delay duration (in ms).
out_setup_delay	Total outbound setup delay duration (in ms).
lost_pkt	Number of calls losing more than the configured number of packets. **Note** This field will exist only in IP records. In other types of records, this field will be empty and extra commas are expected.
latency	Number of calls encountering more than the configured amount of latency. **Note** This field will exist only in IP records. In other types of records, this field will be empty and extra commas are expected.
jitter	Number of calls encountering more than the configured amount of jitter. **Note** This field will exist only in IP records. In other types of records, this field will be empty and extra commas are expected.
in_disc_cc	Incoming disconnect cause code. For example, in_disc_cc_16=3 indicates that 3 calls were disconnected or finished with a disconnect cause code of 16 (normal).
out_disc_cc	Outgoing disconnect cause code.

Table 214 show voice statistics csr since-reset all Field Descriptions (continued)

Field	Description
in_cc_no	Number of the following disconnect cause code counters as per incoming calls (expected to be fewer than 5).
out_cc_no	Number of the following disconnect cause code counters as per outgoing calls (expected to be fewer than 5).
in_cc_id	Disconnect cause code ID for the following field for incoming calls.
in_cc_cntr	Disconnect cause code counter for incoming calls (any incoming cause code counter pairs).
out_cc_id	Disconnect cause code ID for the following field for outgoing calls.
out_cc_cntr	Disconnect cause code counter for outgoing calls (any outgoing cause code counter pairs).

Related Commands

Command	Description
clear voice statistics	Clears voice statistics, resetting the statistics collection.
show event-manager consumers	Displays event statistics.
show voice statistics csr interval accounting	Displays statistical information by configured intervals for accounting statistics.
show voice statistics csr interval aggregation	Displays statistical information by configured intervals for signaling statistics.
show voice statistics csr since-reset accounting	Displays all accounting CSRs since the last reset.
show voice statistics csr since-reset aggregation-level	Displays all signaling CSRs since the last reset.
show voice statistics interval-tag	Displays voice statistics within a specified interval.
show voice statistics memory-usage	Displays current memory usage.

show voice statistics iec

To display Internal Error Code (IEC) statistics, use the **show voice statistics iec** command in user EXEC or privileged EXEC mode.

show voice statistics iec {**interval** *number* | **since-reboot** | **since-reset**} [**push** [**all** | **ftp** | **syslog**]]

Syntax Description

interval	Displays statistics for the specified interval.
number	The interval tag number. The range is from 1 to 36655.
since-reboot	Displays IEC statistics since the last reboot.
since-reset	Displays IEC statistics since the last reset.
push	Specifies the off-load pushing interface.
all	Indicates that IEC statistics will be off-loaded to all push interfaces.
ftp	Indicates that IEC statistics will be off-loaded to the FTP server.
syslog	Indicates that IEC statistics will be off-loaded to the syslog server.

Command Modes

User EXEC (#)
Privileged EXEC(#)

Command History

Release	Modification
12.3(4)T	This command was introduced.
12.4(24)T	This command was modified in a release earlier than Cisco IOS Release 12.4(24)T. The **push all**, **ftp** and **syslog** keywords were added.

Usage Guidelines

Before you can display IEC statistics for a specific interval, use the **show voice statistics interval-tag** command to display available interval options. Before you display view IEC statistics since reboot, you must configure the **voice statistics type iec** command. Before you can display IEC statistics since the last reset, you must configure the **voice statistics type iec** command and the **voice statistics time-range since-reset** command.

Examples

The following is sample output from the **show voice statistics iec since-reset** command, which displays statistics since the last instance when IEC counters were cleared:

```
Router# show voice statistics iec since-reset

Internal Error Code counters
----------------------------
Counters since last reset (2002-11-28T01:55:31Z):
  SUBSYSTEM CCAPI [subsystem code 1]
      [errcode   6] No DSP resource                           5

  SUBSYSTEM SSAPP [subsystem code 4]
      [errcode   5] No dial peer match                         2
      [errcode   3] CPU high                                 96
```

```
  SUBSYSTEM H323 [subsystem code 5]
      [errcode  22] No Usr Responding, H225 timeout            1
      [errcode 27] H225 invalid msg                           1
      [errcode 79] H225 chn, sock fail                       27

  SUBSYSTEM VTSP [subsystem code 9]
      [errcode   6] No DSP resource                           83
```

Table 215 describes the significant fields shown in the display.

Table 215 show voice statistics iec Field Descriptions

Field	Description
SUBSYSTEM	Indicates the specific subsystem within the physical entity where the IEC was generated.
errcode	Identifies the error code within the subsystem.

The following is sample output from the **show voice statistics iec since-reset push all** command, which displays statistics since the last instance when IEC counters were cleared and off-loaded to all push interfaces.

```
Router# show voice statistics iec since-reset push all

Internal Error Code counters
----------------------------
Counters since last reset (2009-07-16T01:40:59Z):
No errors.


Router#
*Jul 16 01:43:39.530: %VSTATS-6-IEC: SEQ=1:
stats_type,version,entity_id,start_time,end_time,record_count
IEC,1,7206-2,2009-07-16T01:40:59Z,2009-07-16T01:43:39Z,0
```

Related Commands

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

show voice statistics interval-tag

To display the interval numbers assigned by the gateway, use the **show voice statistics interval-tag** command in privileged EXEC mode.

show voice statistics interval-tag

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

This is used to obtain the interval tag number required for the **show voice statistics csr interval accounting** and **show voice statistics csr interval aggregation** commands.

Examples

The following example shows the start and end times for specific interval tags:

```
Router# show voice statistics interval-tag

Current System Time is: 2002-4-1T010:10:00Z

    Interval-Tag   Intervals Start Time       End Time
    101            2002-3-31T010:00:00Z       2002-3-31T010:55:00Z
    105            2002-3-31T012:15:00Z       2002-3-31T012:30:00Z
```

Table 216 lists and describes the significant output fields.

Table 216 show voice statistics interval-tag Field Descriptions

Field	Description
Current System Time	Current system time of the gateway.
Interval-Tag	Interval number.
Intervals Start Time	Interval start time.
End Time	Interval end time.

Related Commands

Command	Description
show event-manager consumers	Displays event statistics.
show voice statistics csr interval accounting	Displays statistical information by configured intervals for accounting statistics.

Command	Description
show voice statistics csr interval aggregation	Displays statistical information by configured intervals for signaling statistics.
show voice statistics csr since-reset accounting	Displays all accounting CSRs since the last reset.
show voice statistics csr since-reset aggregation-level	Displays all signaling CSRs since the last reset.
show voice statistics csr since-reset all	Displays all CSRs since the last reset.
show voice statistics memory-usage	Displays current memory usage.

show voice statistics memory-usage

To display the memory used for collecting call statistics and to estimate the future use of memory, use the **show voice statistics memory-usage** command in privileged EXEC mode.

show voice statistics memory-usage {**all** | **csr** | **iec**}

Syntax Description

all	Memory used to collect both signaling and accounting call statistics records (CSRs).
csr	Memory used to collect signaling CSRs only.
iec	Memory used to collect Cisco internal error codes (IECs) only.

Command Modes

Privileged EXEC

Command History

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

Examples

The following example shows all of the memory used at a fixed interval and since the last reset for signaling and accounting; it also shows the estimated future memory to be used.

```
Router# show voice statistics memory-usage all

*** Voice Call Statistics Record Memory Usage ***
        Fixed Interval Option -
                CSR size: 136 bytes
                Number of CSR per interval: 9
                Used memory size (proximate): 0
                Estimated future claimed memory size (proximate): 0
        Since Reset Option -
                CSR size: 136 bytes
                Total count of CSR: 9
                Used memory size (proximate): 1224

*** Voice Call Statistics Accounting Record Memory Usage ***
        Fixed Interval Option -
                ACCT REC size: 80 bytes
                Number of ACCT REC per interval: 1
                Used memory size (proximate): 0
                Estimated future claimed memory size (proximate): 0

        Since Reset Option -
                ACCT REC size: 80 bytes
                Total count of ACCT REC: 1
                Used memory size (proximate): 80
```

Table 217 lists and describes the significant output fields.

Table 217 show voice statistics memory-usage Field Descriptions

Field	Description
Voice Call Statistics Record Memory Usage	
Fixed Interval Option:	Statistics gathered for a fixed interval.
CSR size	Size of the CSR for the fixed interval.
Number of CSR per interval	Number of CSRs collected for the fixed interval.
Used memory size (proximate)	Amount of memory currently being used to store statistics.
Estimated future claimed memory size (proximate)	Amount of remaining memory available to store statistics.
Since Reset Option:	Statistics gathered since the last reset or reboot of the gateway.
CSR size	Size of the CSR since the last reset.
Total count of CSR	Total number of CSRs gathered since the last reset.
Used memory size (proximate)	Amount of memory currently being used to store statistics.
Voice Call Statistics Accounting Record Memory Usage	
Fixed Interval Option:	Statistics gathered for a fixed interval.
ACCT REC size	Accounting record size.
Number of ACCT REC per interval	Number of accounting records per interval.
Used memory size (proximate)	Amount of memory currently being used to store statistics.
Estimated future claimed memory size (proximate)	Amount of remaining memory available to store statistics.
Since Reset Option:	Statistics gathered since the last reset or reboot of the gateway.
ACCT REC size	Accounting record size.
Total count of ACCT REC	Total number of accounting records since the last reset or reboot of the gateway.
Used memory size (proximate)	Amount of memory currently being used to store statistics.

Related Commands

Command	Description
show event-manager consumers	Displays event statistics.
show voice statistics csr interval accounting	Displays statistical information by configured intervals for accounting statistics.
show voice statistics csr interval aggregation	Displays statistical information by configured intervals for signaling statistics.
show voice statistics csr since-reset accounting	Displays all accounting CSRs since the last reset.
show voice statistics csr since-reset aggregation-level	Displays all signaling CSRs since the last reset.

Command	Description
show voice statistics csr since-reset all	Displays all CSRs since the last reset.
show voice statistics interval-tag	Displays the configured interval numbers.