Using 'show call active voice brief' to Track and Troubleshoot Voice Calls

The `show call active voice brief` command is one of the most useful show commands for troubleshooting voice issues. It can be used for tracking calls, voice-quality problems, one-way audio issues, and collecting PCM captures for TAC. There are many different voice components in the command, and using it effectively is a matter of knowing what they are, where they are, and understanding how they relate to the problem.

Voice components gathered from this command:

- Packet loss and jitter for IP legs
- Number of active calls
- Connected IP addresses in RTP streams
- UDP ports used in RTP streams
- Port/Controller and Interface in use
- Dial peer matched for each call leg
- Codec in use
- Duration of current call
- Fax switchover method

This command is the concise version of `show call active voice`. Shown below is the template information printed when the command is used, and is helpful in checking unknown fields. This document only covers the most useful fields for troubleshooting common voice problems.

```
Router#show call active voice brief

<ID>: <CallID> <start>ms.<index> +<connect> pid:<peer_id> <dir> <addr> <state>
  dur hh:mm:ss tx:<packets>/<bytes> rx:<packets>/<bytes>
 IP <ip>:<udp> rtt:<time>ms pl:<play>/<gap>ms lost:<lost>/<early>/<late>
  delay:<last>/<min>/<max>ms <codec>

 media inactive detected:<y/n> media cntrl rcvd:<y/n> timestamp:<time>

 long duration call detected:<y/n> long duration call duration :<sec> timestamp:<time>
  MODEMPASS <method> buf:<fills>/<drains> loss <overall%> <multipkt>/<corrected>
    last <buf event time>s dur:<Min>/<Max>s
 FR <protocol> [int dlci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
  <codec> (payload size)
 ATM <protocol> [int vpi/vci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
  <codec> (payload size)
 Tele <int> (callID) [channel_id] tx:<tot>/<v>/<fax>ms <codec> noise:<l> acom:<l>
i/o:<l>/<l> dBm
  MODEMRELAY info:<rcvd>/<sent>/<resent> xid:<rcvd>/<sent> total:<rcvd>/<sent>/<drops>
        speeds(bps): local <rx>/<tx> remote <rx>/<tx>
 Proxy <ip>:<audio udp>,<video udp>,<tcp0>,<tcp1>,<tcp2>,<tcp3> endpt: <type>/<manf>
 bw: <req>/<act> codec: <audio>/<video>
  tx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>
 rx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>
```

For more information on the components, see the command reference:
http://www.cisco.com/en/US/docs/ios/voice/command/reference/vr_s03.html#wp1315853

Tracking a Call

```
Router#show call active voice brief
```

Call ID
2FEF : 3917822 -2046659508ms.1 +50 pid:5 Answer   active
  dur 00 **Controller / Port** 6/3771 **Channel / Voice Port**
  Tele **0/0/1:15** (3917822) **[0/0/1.13]** tx:43735/8185/0ms g711ulaw
noise:-61 acom:54 i/0:-61/-6 dBm
Call ID   **Dial Peer**   **Calling Number**
2FEF : 39 **Duration** 046659508ms.2 +50 **pid:1** Originate **5554071** connected
  dur **00:00:45** t **IP Address** 040 rx:2246/359184
  IP **10.128.136.58** 8828 SRTP: off rtt:0ms pl:24360/220ms lost:15/0/0
delay:60/60/65ms g711ulaw
  media inactive detected:n media contrl rcvd:n/a timestamp:n/a

2FFA : 3917844 -2046617638ms.1 +30 pid:5 Answer 5553236 active
  dur 00:00:04 tx:200/33600 rx:189/30240
  Tele 0/0/1:15 (3917844) [0/0/1.25] tx:4065/3785/0ms g711ulaw
noise:-84 acom:7 i/0:-24/-17 dBm

2FFA : 3917845 -2046617628ms.1 +10 pid:1 Originate 5554039 active
  dur 00:00:04 tx:189/30240 rx:200/32000
  IP 10.128.136.58:8864 SRTP: off rtt:0ms pl:3555/100ms lost:5/1/0
delay:65/65/65ms g711ulaw
  media inactive detected:n media contrl rcvd:n/a timestamp:n/a
```

Note: The entry with Call ID of 2FFA, has duration of 4 seconds, is connected to 10.128.136.58 on port 8864, is a call between 5554039 and 5553236, and is on PRI 0/0/1:15 channel 25

| | |
|---|---|
| Call ID: | A 16 bit hex ID that identifies the call. Standard calls will have two legs, and you can match the two Call IDs together for a call. This will show whether it is IP-to-IP or IP-to-TDM. The value next to this is the CCAPI call leg ID. |
| Controller/Port: | This field exists in telephony legs, and identifies which Serial interface, controller, or voice port the call is active on. This format of this field will be different based on the configuration of the controller. Some examples: |

**T1 PRI**: 0/0/1:23
**E1 PRI**: 0/0/1:15
**T1 CAS**: 0/0/1:0          *note: the 0 here is the ds0-group number*
**FXO/FXS**: 0/0/1
**CME**: 50/0/*<ephone dn>*

| | |
|---|---|
| Channel: | For telephony legs this will show the particular channel of the PRI or CAS circuit in use, or the channel of the ephone-dn if it is a CME/SRST ephone. |
| Duration: | The duration of the call – this can be used to check against the time on the endpoints to track the call more accurately. Calls are ordered by duration. |
| Dial Peer: | The pid: lists the dial peer matched for both legs of the call. |
| Calling Number: | This is available for call flows where the gateway has the calling or called number information available. MGCP calls and calls without CLID will not list this. This can be a very simple way to find a call, but may not be available. |
| IP Address: | For IP legs this can be used to identify an MTP, an IP phone, or another gateway. By checking this address against an IP phone you can find where a particular call exists on a gateway. |

- To track a call: start with the duration, IP address, or calling/called number, and narrow the call down with the duration to find the IP Call ID. Then match the Call ID to find which analog port and channel the call is using.

# Troubleshooting Voice Quality

```
Router#show call active voice brief

2FEF : 3917822 -2046659508ms.1 +50 pid:5 Answer  active
 dur 00:00:45 tx:2246/3771 Channel / Voice Port
  Tele 0/0/1:15 (3917822) [0/0/1.13] tx:43735/8185/0ms g711ulaw
noise:-61 acom:54  i/0:-61/-6 dBm

2FEF : 3917823 -2046659508ms.2 +50 pid:1 Originate 5554071 connected
 dur IP Address  tx:472/63040 rx:2246/359184          Packet Loss / Jitter
  IP 10.128.136.58 8828 SRTP: off rtt:0ms pl:24360/220ms lost:15/0/0
delay:60/60/65ms g711ulaw
  media  Latency   d Codec  d:n media contrl rcvd:n/a timestamp:n/a

2FFA : 3917844 -2046617638ms.1 +30 pid:5 Answer 5553236 active
 dur 00:00:04 tx:200/33600 rx:189/30240
  Tele 0/0/1:15 (3917844) [0/0/1.25] tx:4065/3785/0ms g711ulaw
noise:-84 acom:7  i/0:-24/-17 dBm

2FFA : 3917845 -2046617628ms.1 +10 pid:1 Originate 5554039 active
 dur 00:00:04 tx:189/30240 rx:200/32000
  IP 10.128.136.58:8864 SRTP: off rtt:0ms pl:3555/100ms lost:5/1/0
delay:65/65/65ms g711ulaw
  media inactive detected:n media contrl rcvd:n/a timestamp:n/a
```

Channel: Some voice quality problems are related to specific controllers or specific channels on a circuit. This field can be used to track whether voice quality is following a particular channel or interface.

Packet Loss/Jitter: In this field, it is in the format of <lost packets/early packets/late packets>. For troubleshooting purposes, both the early packets and late packets simply indicate jitter. In this example, there is a packet loss of 15 packets. This is enough to detect a problem, and a good call should not have any packet loss or jitter. Jitter and packet less can cause static, pops, crackling, choppiness, and clipping of voice.

IP Address: The IP address can be used to find where the RTP path is going, as well as track down which gateway or IP phone is in use. For Cisco phones, there is a great deal of stream information that can be retrieved from the webpage of the phone. If there is jitter or packet loss, inspect the path between the gateway and this IP address.

Latency: The minimum latency shown is seen in the example with 60/60/65ms. According to ITU G.114 standards, the delay for a voice call should be less than 150 ms, but Cisco voice will perform better with a delay of less than 100 ms. High latency can cause echo, static, and clipping of voice.

Codec: This is an easy way to confirm which codec is in use. If bandwidth is a factor, switching to a low bandwidth codec such as G.729 or G722 may improve voice quality. Alternatively, G.711 will be able to handle packet loss more gracefully. Note that low-bandwidth codecs are more sensitive to packet loss.

- If there are no lost or jittered packets, the delay is lower than 100-150ms, and the stream is directly to the phone, the quality is not caused in the IP network.

Troubleshooting One-way Audio

```
Router#show call active voice brief


2FEF : 3917822   Transmit / Receive Counters   5 Answer  active
 dur 00:00:45 tx:2246/377152 rx:472/63040
 Tele 0/0/1:15 (3917822) [0/0/1.13] tx:43735/8185/0ms g711ulaw
noise:-61 acom:54  i/0:-61/-6 dBm

2FEF : 3917823   Transmit / Receive Counters   Originate 5554071 connected
 dur IP Address  tx:472/63040 rx:2246/359184
 IP 10.128.136.58 8828 SRTP: off rtt:0ms pl:24360/220ms lost:15/0/0
delay:60/60/65ms g711ulaw
 media inactive detected:n media contrl rcvd:n/a timestamp:n/a


2FFA : 3917844 -2046617638ms.1 +30 pid:5 Answer 5553236 active
 dur 00:00:04 tx:200/33600 rx:189/30240
 Tele 0/0/1:15 (3917844) [0/0/1.25] tx:4065/3785/0ms g711ulaw
noise:-84 acom:7  i/0:-24/-17 dBm


2FFA : 3917845 -2046617628ms.1 +10 pid:1 Originate 5554039 active
 dur 00:00:04 tx:189/30240 rx:200/32000
 IP 10.128.136.58:8864 SRTP: off rtt:0ms pl:3555/100ms lost:5/1/0
delay:65/65/65ms g711ulaw
 media inactive detected:n media contrl rcvd:n/a timestamp:n/a
```

IP Address: The IP address can be used to find where the RTP path is going, as well as track down which gateway or IP phone is in use. For Cisco phones, there is a great deal of stream information that can be retrieved from the webpage of the phone. The stream page of the phone will offer the IP address the phone is connected to, which is helpful in determining routing issues. Confirm reachability from one address to the other.   Check if the phone has a default gateway, and if the default gateway can reach the gateway address. Use a bind statement on the gateway as needed.

Transmit / Receive
Counters: The transmit and receive counters are very helpful in determining if a leg is currently receiving or transmitting packets. If the incoming call leg is not receiving packets, it will not transmit any out the outgoing call leg. Troubleshooting one-way audio is simplified when it is determined whether the gateway is sending or receiving packets on a certain leg. Having a rx: counter of zero typically means the far end does not have a route to the gateway. If the receive counter on a telephony leg is 0, this points towards a problem on the analog circuit, or a hardware problem with the DSP or voice card.

- The webpage of the IP phone can also be used to check the default gateway as well as the stream statistics. You can see lost packets, jitter statistics, and if it is sending and receiving packets. Clicking the '?' twice quickly while on a call with a Cisco IP phone will bring up a similar stream statistic menu.

Collecting a PCM Capture

A PCM capture is a procedure used by TAC to capture audio directly off of the DSPs for troubleshooting various audio issues, and should only be captured if required by TAC.

1. First, enable the capture buffer on the gateway:

```
Router(config)#voice hpi capture buffer 1000000
```

Next, check for free memory in the flash:

```
Router#dir flash:

    1  -rw-     45252276   Nov 3 2008 22:37:06 -05:00   c2800nm-ipvoice_ivs-mz.124-22.T
    2  -rw-       496521   Aug 27 2008 21:36:30 -05:00   music-on-hold.au

63983616 bytes total (18231296 bytes free)
```

In this example there is approximately 18 MB free. If there is not at least 15 MB or more of free flash, the PCM capture should be sent to a TFTP server.

2. Enable the destination for the capture

```
Router(config)#voice hpi capture destination flash:pcm.dat
```
or
```
Router(config)#voice hpi capture destination tftp://192.168.0.1/pcm.dat
```
Note: These commands continuously write to the file destination once they are enabled. It is important to remove these when the capture is completed.

It is important to understand that a PCM capture pulls the data directly from the audio stream, so the problem that you're troubleshooting needs to be happening after the capture has started.

3. Place the call, and keep the call up. Go to 'Tracking a call' and find the interface/controller for the call. If this is a T1/E1, record the channel that the call is on as well. The call can be found by combining the approximate call duration, the address of the IP phone (or other endpoint/gateway), or the calling/called numbers if available. The Call ID will match an IP leg with a 'Tele' leg, and the controller and channel will be found in the Telephony leg.

3. Once the call is up, and you have found the port and channel, you are ready to capture. This command starts the PCM capture:

```
Router#test voice port <port.channel> pcm-dump caplog 7 duration 255
```

Note that if the circuit is on a channel, the *<port.channel>* will be similar to: 0/0/0:23.3.
The format will be the voice-port in use, followed by a period and the channel.
The command **show voice call stat** will also display the active calls in this format.

These are guidelines for the *<port.channel>* format:
T1 PRI: 0/0/0:23.channel
E1 PRI: 0/0/0:15.channel
T1 CAS: 0/0/0:0.channel
FXO/FXS: 0/0/0

While the capture is running, reproduce the issue you're troubleshooting. To confirm that the PCM capture is running correctly, use **show voice hpi capture**.

If the capture is running correctly, you should see "messages sent to URL":

```
Router#show voice hpi capture
HPI Capture is on and is logging to URL tftp://172.18.251.73/pcm.dat
5801 mesages sent to URL, 0 messages dropped
Message Buffer (total:inuse:free)  2049:0005:2044
Buffer Memory: 999912 bytes, Message size: 488 bytes
```

4. Now the PCM capture is running, and you should try to reproduce the problem, or wait for up to a minute for the problem to happen.

5. Disable the PCM capture:

```
Router#test voice port <port.channel> pcm-dump disable
```

6. If the test was successful, remove the buffer and destination to prevent further writing to the file:

```
Router(config)#no voice hpi capture buffer 1000000
Router(config)#no voice hpi capture destination flash:pcm.dat
  OR
Router(config)#no voice hpi capture buffer 1000000
Router(config)#no voice hpi capture destination tftp://1.1.1.1/pcm.dat
```

If the test was not successful, change the capture destination:

```
Router(config)#voice hpi capture destination tftp://1.1.1.1/pcm2.dat
```

Then, delete the old file, and continue at Step 3:

```
Router#delete flash:pcm.dat
```

7. When completed, copy the pcm.dat flash to a tftp server (if it is on flash), and email to TAC.