Troubleshooting and Debugging VoIP Call Basics

TAC Notice: What's Changing on TAC Web

Contents

Introduction

This document demonstrates basic techniques and commands to troubleshoot and debug VoIP networks. An overview of the Voice Call Flow and Telephony Architecture in a Cisco Router is presented, followed by a step-by-step VoIP troubleshooting approach presented in these steps:

1. Verify digital and analog signaling.
2. Verify digits received and sent from the analog and digital voice ports.
3. Verify end-to-end VoIP signaling.
4. Understand VoIP Quality of Service (QoS) issues.

5. Understand details of cause codes and debug values for VoIP.

Note: This document does not explain every facet of the Cisco IOS® architecture used in Cisco VoIP gateways and gatekeepers. Instead, it is intended to show which commands can be used and which fields from the command outputs can be most valuable.

Caution: Debugging the Cisco IOS can be processor intensive. Exercise caution when you use the debugs listed in this document. For more information, refer to Important Information on Debug Commands.

Debugs need to be run with timestamps enabled in the log. Enable timestamping by adding the commands: **service timestamps** *debug datetime msec* , **service timestamps** *log datetime msec* in enable mode. The timestamps help determine the interval of time between state changes.

Prerequisites

Requirements

This document is intended for the networking personnel involved in the design and deployment of VoIP networks. Readers of this document should be knowledgeable of these topics:

- VoIP configuration
- Voice QoS

Components Used

This document is not restricted to specific software and hardware versions. However, the outputs shown are based on Cisco IOS® Software Release 12.3(8).

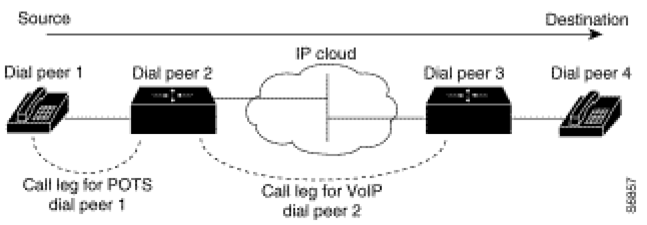
The information presented in this document was created from devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If you are working in a live network, ensure that you understand the potential impact of any command before using it.

Conventions

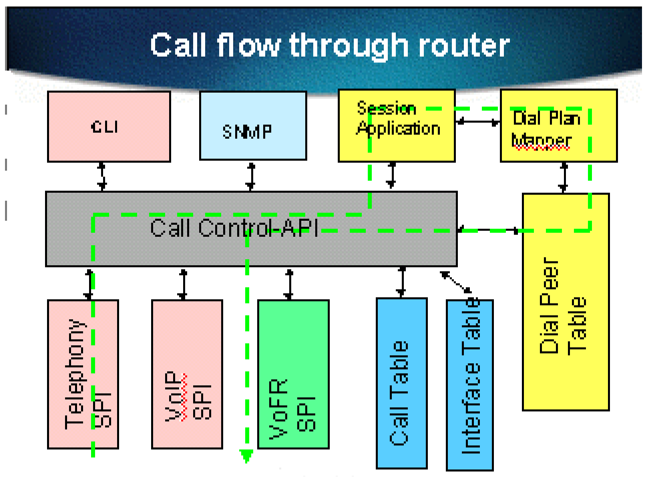
For more information on document conventions, refer to the Cisco Technical Tips Conventions.

Call Flow in the Network

An important factor to consider before you start any VoIP troubleshooting or debugging is that VoIP calls are made up of three call legs. These call legs are source Plain Old Telephone Systems (POTS), VoIP, and destination POTS. This is shown in this diagram. Troubleshooting and debugging needs to first focus on each leg independently and then on the VoIP call as a whole.



Router Call Flow



These definitions explain the function of the main components displayed in the router call flow diagram:

Call Control API (Application Programming Interface)—Three clients make use of the call control API. The three clients are CLI, simple network management protocol (SNMP) agent, and the Session Application. The Call Control API's (also referred to as CCAPI) main functions are to:

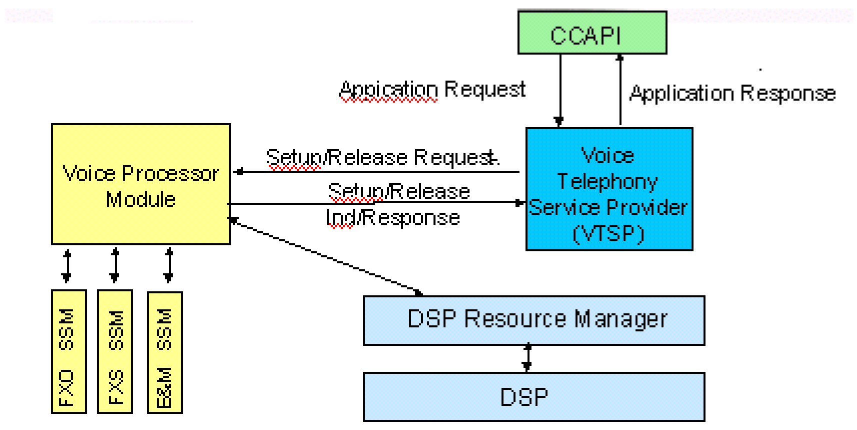
- Identify the call legs (for example, which dial peer is it? where did it come from?).
- Decide which session application takes the call (for example, who handles it?).
- Invoke the packet handler.
- Conference the call legs together.
- Start to record call statistics.

Session Application and Dial Plan Mapper—The Session Application uses the Dial Plan Mapper to map a number to a dial-peer (local POTS or remote VoIP). The Dial Plan Mapper uses the Dial Peer Table to find active dial-peers.

Telephony and VoIP Service Provider Interface (SPI)—The Telephony SPI communicates with the POTS (analog: fxs, fxo, e&m Digital: isdn, qsig, e&m, and so forth) dial-peers. The VoIP SPI is the specific interface to the VoIP peers. Telephony/DSP drivers deliver services to the Telephony SPI while the VoIP SPI relies on session protocols.

Telephony Interface Architecture

This diagram shows the architecture of Cisco router Telephony building blocks and how they interact with each other.



This list describes the functions and definitions of the main diagram components:

- **Call Control Application Programming Interface (CCAPI)**—Software entity that establishes, terminates and bridges call legs.
- **Voice Telephony Service Provider (VTSP)**—An IOS process that services requests from the Call Control API and formulates the appropriate requests to the digital signal processor (DSP) or the VPM.
- **Voice Processor Module (VPM)**—The VPM is in charge of bridging and coordinating signaling processes between the Telephony ports signaling state machine (SSM), the DSP Resource Manager, and the VTSP.
- **DSP Resource Manager**—The DSPRM provides interfaces by which the VTSP can send and receive messages to and from the DSPs.
- **Packet Handler**—The packet handler forwards packets between the DSPs and the Peer Call legs.
- **Call Peer**—The Call Peer is the opposite call leg. This can be another Telephony voice connection (POTS), VoFR, VoATM, or a VoIP connection.

Verify Digital and Analog Signaling (POTS Call-Leg)

The objectives for verifying digital and analog signaling are to:

- Determine that the proper on-hook and off-hook analog or digital signaling is received.
- Determine that proper E&M, FXO and FXS signaling is configured on both the router and switch (CO or PBX).
- Verify that the DSPs are in digit collection mode.

The commands outlined in these sections can be used to verify the signaling.

show controllers T1 / E1 (digital)

show controllers t1 [slot/port]—Use this command first. It shows if the digital T1 connection between the router and switch (CO or PBX) is up or down and if it functions properly. The output of this command looks like this:

```
router# show controllers T1 1/0
T1 1/0 is up.
Applique type is Channelized T1
Cablelength is short 133
No alarms detected.
Framing is ESF, Line Code is B8ZS, Clock Source is Line
Primary.
Data in current interval (6 seconds elapsed):

        0 Line Code Violations, 0 Path Code Violations
        0 Slip Secs, 0 Fr Loss Secs, 0 Line Err Secs,
  0 Degraded Mins
        0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs,
  0 Unavail Secs
```

If using E1 use the **show controllers e1** command. For more information, refer to:

- T1 Layer 1 Troubleshooting
- T1 Troubleshooting Flowchart
- Troubleshooting Serial Line Problems

show voice port

show voice port *slot-number/port* —Use this command to display the port state and the parameters configured on the voice-port of Cisco voice interface cards (VIC). Like all IOS commands, defaults do not display in **show running-config**, but they do display with this command.

This is sample output for an E&M voice port:

```

router# show voice port 1/0:1
recEive 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 16 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
Region Tone is set for US

Voice card specific Info Follows:
Out Attenuation is Set to 0 dB
```

```
Echo Cancellation is enabled
Echo Cancel Coverage is set to 16 ms
Connection Mode is normal (could be trunk or plar)
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
Region Tone is set for US

Voice card specific Info Follows:
Signal Type is wink-start
Operation Type is 2-wire
E&M Type is 1
Dial Type is dtmf
In Seizure is inactive
Out Seizure is inactive
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 500 ms
Clear Wait Duration Timing is set to 400 ms
Wink Wait Duration Timing is set to 200 ms
Wink Duration Timing is set to 200 ms
Delay Start Timing is set to 300 ms
Delay Duration Timing is set to 2000 ms
Dial Pulse Min. Delay is set to 140 ms
```

debug vpm (voice processor module)

These commands are used to debug the VPM Telephony interface:

- **debug vpm signal** —This command is used to collect debug information for signaling events and can be useful in resolving problems with signaling to a PBX.

- **debug vpm spi** —This command traces how the voice port module service provider interface (SPI) interfaces with the call control API. This **debug** command displays information about how each network indication and application request is handled.

- **debug vpm dsp** —This command displays messages from the DSP on the VPM to the router and can be useful if you suspect the VPM is not functional. It is a simple way to check if the VPM responds to off-hook indications and to evaluate timing for signaling messages from the interface.

- **debug vpm all** —This EXEC command enables all of the debug vpm commands: **debug vpm spi**, **debug vpm signal**, and **debug vpm dsp**.

- **debug vpm port** —Use this command to limit the debug output to a particular port. For example, this output shows **debug vpm dsp** messages only for port 1/0/0:

  ```
  debug vpm dsp

  debug vpm port 1/0/0
  ```

  For more information, refer to VoIP Debug Commands.

Sample Output for debug vpm signal Command

```
maui-voip-austin#debug vpm signal


!--- FXS port 1/0/0 goes from the "on-hook" to "off-hook"
!--- state.

htsp_process_event: [1/0/0, 1.2 , 36]
fxsls_onhook_offhook htsp_setup_ind
*Mar 10 16:08:55.958: htsp_process_event:
[1/0/0, 1.3 , 8]


!--- Sends ringing alert to the called phone.

*Mar 10 16:09:02.410: htsp_process_event:
[1/0/0, 1.3 , 10] htsp_alert_notify
*Mar 10 16:09:03.378: htsp_process_event:
[1/0/0, 1.3 , 11]


!--- End of phone call, port goes "on-hook".

*Mar 10 16:09:11.966: htsp_process_event:
[1/0/0, 1.3 , 6]
*Mar 10 16:09:17.218: htsp_process_event:
[1/0/0, 1.3 , 28] fxsls_offhook_onhook
*Mar 10 16:09:17.370: htsp_process_event:
[1/0/0, 1.3 , 41] fxsls_offhook_timer
*Mar 10 16:09:17.382: htsp_process_event:
[1/0/0, 1.2 , 7] fxsls_onhook_release
```

If the on-hook and off-hook are not signaling properly, check these items:

- Verify the cabling is correct.
- Verify that both the router and switch (CO or PBX) are properly grounded.
- Verify that both ends of the connection have matching signaling configurations. Mismatched configurations can cause incomplete or one-way signaling.

For more information on E&M troubleshooting, refer to Understanding and Troubleshooting Analog E & M Interface Types and Wiring Arrangements.

Sample Output for debug vpm spi Command

```
maui-voip-austin#debug vpm spi
Voice Port Module Session debugging is enabled


!--- The DSP is put into digit collection mode.

*Mar 10 16:48:55.710: dsp_digit_collect_on:
[1/0/0] packet_len=20 channel_id=128
 packet_id=35 min_inter_delay=290
 max_inter_delay=3200 mim_make_time=18 max_make
_time=75 min_brake_time=18 max_brake_time=75
```

Verify Digits Received and Sent (POTS Call-Leg)

Once the on-hook and off-hook signaling are verified and work correctly, verify the correct digits are received or sent on the voice-port (digital or analog). A dial-peer is not matched or the switch (CO or PBX) is not able to ring the correct station if incomplete or incorrect digits are sent or received. Some commands that can be used to verify the digits received/sent are:

- **show dialplan number** —This command is used to show which dial peer is reached when a particular telephone number is dialed.
- **debug vtsp session** —This command displays information on how each network indication and application request is processed, signaling indications, and DSP control messages.
- **debug vtsp dsp** —Earlier than Cisco IOS Software Release 12.3, this command displays the digits as they are received by the voice-port. However, in Cisco IOS Software Release 12.3 and later, the output of the **debug** command no longer displays the digits. The combination of **debug hpi** *detail* and **debug hpi** *notification* can be used to see the incoming digits.
- **debug vtsp all** —This command enables these debug voice telephony service provider (VTSP) commands: **debug vtsp session**, **debug vtsp error** , and **debug vtsp dsp**.

For more information, refer to VoIP Debug Commands.

show dialplan number

show dialplan number <*digit_string*>—This command displays the dial-peer that is matched by a string of digits. If multiple dial-peers can be matched, they are all shown in the order in which they are matched.

Note: You need to use the # sign at the end of phone numbers for dial-peers with variable length in order to match on destination-patterns that end with T.

The output of this command looks like this:

```
maui-voip-austin#show dialplan number 5000
Dial string terminator: #
Macro Exp.: 5000

VoiceOverIpPeer2
        information type = voice,
        tag = 2, destination-pattern = `5000',
        answer-address = `', preference=0,
        group = 2, Admin state is up, Operation
        state is up,
        incoming called-number = `',
        connections/maximum = 0/unlimited,
        application associated:
        type = voip, session-target =
        `ipv4:192.168.10.2',
        technology prefix:
        ip precedence = 5, UDP checksum =
        disabled, session-protocol = cisco,
        req-qos = best-effort,
        acc-qos = best-effort,
        dtmf-relay = cisco-rtp,
        fax-rate = voice,
        payload size =  20 bytes
        codec = g729r8,
        payload size =  20 bytes,
        Expect factor = 10, Icpif = 30,
        signaling-type = cas,
        VAD = enabled, Poor QOV Trap = disabled,
        Connect Time = 25630, Charged Units = 0,
        Successful Calls = 25, Failed Calls = 0,
        Accepted Calls = 25, Refused Calls = 0,
        Last Disconnect Cause is "10  ",
        Last Disconnect Text is "normal call
        clearing.",
        Last Setup Time = 84427934.
        Matched: 5000   Digits: 4
        Target: ipv4:192.168.10.2
```

debug vtsp session

The **debug vtsp session** command shows information on how the router interacts with the DSP based on the signaling indications from the signaling stack and requests from the application. This **debug** command displays information about how each network indication and application request is handled, signaling indications, and DSP control messages.

```
maui-voip-austin#debug vtsp session
Voice telephony call control session debugging is on


!--- Output is suppressed.
!--- ACTION: Caller picked up handset.
!--- The DSP is allocated, jitter buffers, VAD
!--- thresholds, and signal levels are set.


*Mar 10 18:14:22.865: dsp_set_playout: [1/0/0 (69)]
packet_len=18 channel_id=1 packet_id=76 mode=1
initial=60 min=4 max=200 fax_nom=300
*Mar 10 18:14:22.865: dsp_echo_canceller_control:
[1/0/0 (69)] packet_len=10 channel_id=1 packet_id=66
flags=0x0
*Mar 10 18:14:22.865: dsp_set_gains: [1/0/0 (69)]
packet_len=12 channel_id=1 packet_id=91
in_gain=0 out_gain=65506
*Mar 10 18:14:22.865: dsp_vad_enable: [1/0/0 (69)]
packet_len=10 channel_id=1 packet_id=78
thresh=-38act_setup_ind_ack
*Mar 10 18:14:22.869: dsp_voice_mode: [1/0/0 (69)]
packet_len=24 channel_id=1 packet_id=73 coding_type=1
 voice_field_size=80
VAD_flag=0 echo_length=64 comfort_noise=1
inband_detect=1 digit_relay=2
AGC_flag=0act_setup_ind_ack(): dsp_dtmf_mod
e()act_setup_ind_ack: passthru_mode = 0,
no_auto_switchover = 0dsp_dtmf_mode
(VTSP_TONE_DTMF_MODE)


!--- The DSP is put into "voice mode" and dial-tone is
!--- generated.


*Mar 10 18:14:22.873: dsp_cp_tone_on: [1/0/0 (69)]
packet_len=30 channel_id=1 packet_id=72 tone_id=4
n_freq=2 freq_of_first=350 freq_of_second=440 amp_of_first=
4000 amp_of_second=4000 direction=1 on_time_first=65535
```

```
off_time_first=0 on_time
_second=65535 off_time_second=0
```

If it is determined the digits are not being sent or received properly, then it might possibly be necessary to use either a digit-grabber (test tool) or T1 tester to verify the digits are being sent at the correct frequency and timing interval. If they are being sent "incorrectly" for the switch (CO or PBX), some values on the router or switch (CO or PBX) might possibly need to be adjusted so that they match and can interoperate. These are usually digit duration and inter-digit duration values. Another item to examine if the digits appear to be sent correctly are any number translation tables in the switch (CO or PBX) that can add or remove digits.

Verify End-to-End VoIP Signaling (VOIP Call-Leg)

After you verify that voice-port signaling works properly and the correct digits are received, move to the VoIP call control troubleshooting and debugging. These factors explain why call control debugging can become a complex job:

- Cisco VoIP gateways use H.323 signaling to complete calls. H.323 is made up of three layers of call-negotiation and call-establishment: H.225, H.245, and H.323. These protocols use a combination of TCP and UDP to set up and establish a call.
- End-to-End VoIP debugging shows a number of IOS state-machines. Problems with any state-machine can cause a call to fail.
- End-to-End VoIP debugging can be very verbose and create a lot of debug output.

debug voip ccapi inout

The primary command to debug end-to-end VoIP calls is **debug voip ccapi inout** . The output from a call debug is shown in this output.

```


!--- Action: A VoIP call is originated through the
!--- Telephony SPI (pots leg) to extension 5000.
!--- Some output is omitted.


maui-voip-austin#debug voip ccapi inout
voip ccAPI function enter/exit debugging is on


!--- Call leg identification, source peer: Call
!--- originated from dial-peer 1 pots
!--- (extension 4000).


*Mar 15 22:07:11.959: cc_api_call_setup_ind
(vdbPtr=0x81B09EFC, callInfo={called=,
calling=4000, fdest=0 peer_tag=1}, callID=0x81B628F0)

!--- CCAPI invokes the session application.


*Mar 15 22:07:11.963: cc_process_call_setup_ind
(event=0x81B67E44) handed call to app "SESSION"
*Mar 15 22:07:11.963: sess_appl:
```

```
ev(23=CC_EV_CALL_SETUP_IND), cid(88), disp(0)


!--- Allocate call leg identifiers "callid = 0x59"


*Mar 15 22:07:11.963: ccCallSetContext
(callID=0x58, context=0x81BAF154)
*Mar 15 22:07:11.963: ccCallSetupAck
(callID=0x58)


!--- Instruct VTSP to generate dialtone
.

*Mar 15 22:07:11.963: ccGenerateTone
(callID=0x58 tone=8)


!--- VTSP passes digits to CCAPI.


*Mar 15 22:07:20.275:cc_api_call_digit_begin
(vdbPtr=0x81B09EFC,callID=0x58,digit=5, flags=0x1, timestamp=0xC2E63BB7, expirat
*Mar 15 22:07:20.279: sess_appl:
ev(10=CC_EV_CALL_DIGIT_BEGIN), cid(88), disp(0)
*Mar 15 22:07:20.279: ssaTraceSct:
cid(88)st(0)oldst(0)cfid(-1)csize(0)in(1)fDest(0)
*Mar 15 22:07:20.279: ssaIgnore cid(88),
st(0),oldst(0), ev(10)
*Mar 15 22:07:20.327: cc_api_call_digit
(vdbPtr=0x81B09EFC, callID=0x58, digit=5
, duration=100)
*Mar 15 22:07:20.327: sess_appl:
ev(9=CC_EV_CALL_DIGIT), cid(88), disp(0)
*Mar 15 22:07:20.327: ssaTraceSct:
cid(88)st(0)oldst(0)cfid(-1)csize(0)in(1)fDes
t(0)
*Mar 15 22:07:21.975:cc_api_call_digit_begin
(vdbPtr=0x81B09EFC,callID=0x58,digit=0,
flags=0x1, timestamp=0xC2E63BB7, expiration=0x0)
*Mar 15 22:07:21.979: sess_appl:
ev(10=CC_EV_CALL_DIGIT_BEGIN), cid(88), disp(0)
*Mar 15 22:07:21.979: ssaTraceSct:
cid(88)st(0)oldst(0)cfid(-1)csize(0)in(1)fDes
t(0)
*Mar 15 22:07:21.979: ssaIgnore cid(88),
st(0),oldst(0), ev(10)
*Mar 15 22:07:22.075: cc_api_call_digit
(vdbPtr=0x81B09EFC, callID=0x58, digit=0
, duration=150)
*Mar 15 22:07:22.079: sess_appl:
ev(9=CC_EV_CALL_DIGIT), cid(88), disp(0)
*Mar 15 22:07:22.079: ssaTraceSct:
cid(88)st(0)oldst(0)cfid(-1)csize(0)in(1)fDest(0)
*Mar 15 22:07:23.235: cc_api_call_digit_begin
(vdbPtr=0x81B09EFC, callID=0x58, dgit=0,
flags=0x1, timestamp=0xC2E63BB7, expiration=0x0)
*Mar 15 22:07:23.239: sess_appl:
ev(10=CC_EV_CALL_DIGIT_BEGIN), cid(88), disp(0)
*Mar 15 22:07:23.239: ssaTraceSct:
cid(88)st(0)oldst(0)cfid(-1)csize(0)in(1)fDest(0)
*Mar 15 22:07:23.239: ssaIgnore cid(88),
st(0),oldst(0), ev(10)
*Mar 15 22:07:23.335: cc_api_call_digit
(vdbPtr=0x81B09EFC, callID=0x58, digit=0
, duration=150)
*Mar 15 22:07:23.339: sess_appl:
ev(9=CC_EV_CALL_DIGIT), cid(88), disp(0)
*Mar 15 22:07:23.339: ssaTraceSct:
cid(88)st(0)oldst(0)cfid(-1)csize(0)in(1)fDes
t(0)
*Mar 15 22:07:25.147: cc_api_call_digit_begin
(vdbPtr=0x81B09EFC, callID=0x58, d
igit=0, flags=0x1, timestamp=0xC2E63BB7,
expiration=0x0)
*Mar 15 22:07:25.147: sess_appl:
ev(10=CC_EV_CALL_DIGIT_BEGIN), cid(88), disp(0)
*Mar 15 22:07:25.147: ssaTraceSct:
cid(88)st(0)oldst(0)cfid(-1)csize(0)in(1)fDest(0)
*Mar 15 22:07:25.147: ssaIgnore cid(88),
st(0),oldst(0), ev(10)
*Mar 15 22:07:25.255: cc_api_call_digit
(vdbPtr=0x81B09EFC, callID=0x58, digit=0
, duration=160)
*Mar 15 22:07:25.259: sess_appl:
ev(9=CC_EV_CALL_DIGIT), cid(88), disp(0)
*Mar 15 22:07:25.259: ssaTraceSct:
cid(88)st(0)oldst(0)cfid(-1)csize(0)in(1)fDest(0)


!--- Matched dial-peer 2 voip. Destination number
!--- 5000


*Mar 15 22:07:25.259: ssaSetupPeer cid(88)
peer list:tag(2) called number(5000)
*Mar 15 22:07:25.259: ssaSetupPeer cid(88),
destPat(5000), matched(4), prefix(),
peer(81C04A10)


!--- Continue to call an interface and start the
!--- next call leg.


*Mar 15 22:07:25.259: ccCallProceeding
(callID=0x58, prog_ind=0x0)
*Mar 15 22:07:25.259: ccCallSetupRequest
(Inbound call = 0x58, outbound peer =2,
dest=, params=0x81BAF168 mode=0,
*callID=0x81B6DE58)
*Mar 15 22:07:25.259: callingNumber=4000,
calledNumber=5000, redirectNumber=


!--- VoIP call setup.


*Mar 15 22:07:25.263: ccIFCallSetupRequest:
(vdbPtr=0x81A75558, dest=,
callParams={called=5000, calling=4000,
fdest=0, voice_peer_tag=2}, mode=0x0)
*Mar 15 22:07:25.263: ccCallSetContext
(callID=0x59, context=0x81BAF3E4)
*Mar 15 22:07:25.375: ccCallAlert
```

```
(callID=0x58, prog_ind=0x8, sig_ind=0x1)


!--- POTS and VoIP call legs are tied together.


*Mar 15 22:07:25.375: ccConferenceCreate
(confID=0x81B6DEA0, callID1=0x58, callI
D2=0x59, tag=0x0)
*Mar 15 22:07:25.375: cc_api_bridge_done
(confID=0x1E, srcIF=0x81B09EFC, srcCall
ID=0x58, dstCallID=0x59, disposition=0,
tag=0x0)


!--- Exchange capability bitmasks with remote
!--- the VoIP gateway
!--- (Codec, VAD, VoIP or FAX, FAX-rate, and so forth).


*Mar 15 22:07:26.127: cc_api_caps_ind
(dstVdbPtr=0x81B09EFC, dstCallId=0x58, src
CallId=0x59,caps={codec=0x4, fax_rate=0x2,
vad=0x2, modem=0x1 codec_bytes=20,
signal_type=0})


!--- Both gateways agree on capabilities.

*Mar 15 22:07:26.127: cc_api_caps_ack
(dstVdbPtr=0x81B09EFC, dstCallId=0x58, src
CallId=0x59, caps={codec=0x4, fax_rate=0x2,
vad=0x2, modem=0x1 codec_bytes=20,
signal_type=0})
*Mar 15 22:07:26.139: cc_api_caps_ack
(dstVdbPtr=0x81A75558, dstCallId=0x59, src
CallId=0x58, caps={codec=0x4, fax_rate=0x2,
vad=0x2, modem=0x1 codec_bytes=20,
signal_type=0})
*Mar 15 22:07:35.259: cc_api_call_digit
(vdbPtr=0x81B09EFC, callID=0x58, digit=T
, duration=0)
*Mar 15 22:07:35.259: sess_appl:
ev(9=CC_EV_CALL_DIGIT), cid(88), disp(0)
*Mar 15 22:07:35.259: ssaTraceSct:
cid(88)st(4)oldst(3)cfid(30)csize(0)in(1)
fDest(0)-cid2(89)st2(4)oldst2(1)
*Mar 15 22:07:35.399: cc_api_call_connected
(vdbPtr=0x81A75558, callID=0x59)
*Mar 15 22:07:35.399: sess_appl:
ev(8=CC_EV_CALL_CONNECTED), cid(89), disp(0)
*Mar 15 22:07:35.399: ssaTraceSct:
cid(89)st(4)oldst(1)cfid(30)csize(0)in(0)
fDest(0)-cid2(88)st2(4)oldst2(4)


!--- VoIP call is connected.


*Mar 15 22:07:35.399: ccCallConnect
(callID=0x58)
```

```
!--- VoIP call is disconnected. Cause = 0x10


*Mar 15 23:29:39.530: ccCallDisconnect
(callID=0x5B, cause=0x10 tag=0x0)
```

If the call fails and the cause appears to be in the VoIP portion of the call setup, you might possibly need to look at the H.225 or H.245 TCP part of the call setup, as opposed to just the UDP portion of the H.323 setup. The commands that can be used to debug the H.225 or H.245 call setup are:

- **debug ip tcp transactions** and **debug ip tcp packet**—These commands examine the TCP portion of the H.225 and H.245 negotiation. They return the IP addresses, TCP ports, and states of the TCP connections.
- **debug cch323 h225** —This command examines the H.225 portion of the call negotiation and traces the state transition of the H.225 state machine based on the processed event. Think of this as the Layer 1 part of the three part H.323 call setup.
- **debug cch323 h245** —This command examines the H.245 portion of the call negotiation and traces the state transition of the H.245 state machine based on the processed events. Think of this as the Layer 2 part of the three part H.323 call setup.

Understand VoIP Quality of Service (QoS) Issues

When VoIP calls are properly established, the next step is to verify that the voice quality is good. Although QoS troubleshooting is not covered in this document, these guidelines need to be considered to achieve good voice quality:

- Understand how much bandwidth a VoIP call consumes with each codec. This includes Layer 2 and IP/UDP/RTP headers. For more information, refer to Voice over IP - Per Call Bandwidth Consumption.
- Understand the characteristics of the IP network the calls travels over. For example, the bandwidth of a frame-relay network at CIR is much different than that above-CIR (or burst), where packets can be dropped or queued in the Frame-Relay cloud. Ensure that delay and jitter are controlled and eliminated as much as possible. One-way transmit delay should not exceed 150 ms (per G.114 recommendation).
- Use a queuing technique that allows VoIP traffic to be identified and prioritized.
- When you transmit VoIP over low-speed links, consider using Layer 2 packet fragmentation techniques, such as MLPPP with Link Fragmentation and Interleaving (LFI) on point-to-point links, or FRF.12 on Frame-Relay links. Fragmentation of larger data packets allows less jitter and delay in transmitting VoIP traffic because the VoIP packets can be interleaved onto the link.
- Try to use a different codec and try the call with VAD enabled and disabled to possibly narrow down the issue to the DSP, as opposed to the IP network.

With VoIP, the main things to look for when troubleshooting QoS issues are dropped packets and network bottlenecks that can cause delay and jitter.

Look for:

- interface drops
- buffer drops
- interface congestion
- link congestion

Each interface in the path of the VoIP call needs to be examined. Also, eliminate drops and congestion. Also, round-trip delay needs to be reduced as much as possible. Pings between the VoIP end points give an indication of the round trip delay of a link. The round trip delay should not exceed 300 ms whenever possible. If the delay does have to exceed this value, efforts need to be taken to ensure this delay is constant, so as not to introduce jitter or variable delay.

Verification should also be made to ensure the IOS queuing mechanism is placing the VoIP packets within the proper queues. IOS commands, such as **show queue** *interface* or **show priority** can assist in verification of queueing.

Details of Cause Codes and Debug Values for VoIP

Use these tables when you read debugs and the associated values within the debugs.

Q.931 Call Disconnection Causes (cause_codes from debug voip ccapi inout)

For more information on Q.931 Cause Codes and Values, refer to ISDN Switch Types, Codes, and Values

Call Disconnection Cause Value (in Hex)	Meaning and Number (in Decimal)
CC_CAUSE_UANUM = 0x1	unassigned number. (1)
CC_CAUSE_NO_ROUTE = 0x3	no route to destination. (3)
CC_CAUSE_NORM = 0x10	normal call clearing. (16)
CC_CAUSE_BUSY = 0x11	user busy. (17)
CC_CAUSE_NORS = 0x12	no user response. (18)
CC_CAUSE_NOAN = 0x13	no user answer. (19)
CC_CAUSE_REJECT = 0x15	call rejected. (21)
CC_CAUSE_INVALID_NUMBER = 0x1C	invalid number. (28)
CC_CAUSE_UNSP = 0x1F	normal, unspecified. (31)
CC_CAUSE_NO_CIRCUIT = 0x22	no circuit. (34)

CC_CAUSE_NO_REQ_CIRCUIT = 0x2C	no requested circuit. (44)
CC_CAUSE_NO_RESOURCE = 0x2F	no resource. (47) [1]
CC_CAUSE_NOSV = 0x3F	service or option not available, or Unspecified. (63)

[1] This issue can occur due to a codec mismatch within the H323 setup, so the first troubleshooting step is to hardcode the VoIP dial-peers to use the correct codec.

Codec Negotiation Values (from debug voip ccapi inout)

For more information on CODECs, refer to VoIP - Understanding Codecs: Complexity, Support, MOS, and Negotiation.

Negotiation Value	Meaning
codec=0x00000001	G711 ULAW 64K PCM
codec=0x00000002	G711 ALAW 64K PCM
codec=0x00000004	G729
codec=0x00000004	G729IETF
codec=0x00000008	G729a
codec=0x00000010	G726r16
codec=0x00000020	G726r24
codec=0x00000040	G726r32
codec=0x00000080	G728
codec=0x00000100	G723r63
codec=0x00000200	G723r53
codec=0x00000400	GSMFR
codec=0x00000800	G729b
codec=0x00001000	G729ab
codec=0x00002000	G723ar63
codec=0x00004000	G723ar53
codec=0x00008000	CLEAR_CHANNEL

Tone Types

Tone Types	Meaning
CC_TONE_RINGBACK 0x1	Ring Tone
CC_TONE_FAX 0x2	Fax Tone
CC_TONE_BUSY 0x4	Busy Tone

CC_TONE_DIALTONE 0x8	Dial Tone
CC_TONE_OOS 0x10	Out of Service Tone
CC_TONE_ADDR_ACK 0x20	Address Acknowledgement Tone
CC_TONE_DISCONNECT 0x40	Disconnect Tone
CC_TONE_OFF_HOOK_NOTICE 0x80	Tone indicating the phone was left off hook
CC_TONE_OFF_HOOK_ALERT 0x100	A more urgent version of CC_TONE_OFF_HOOK_NOTICE
CC_TONE_CUSTOM 0x200	Custom Tone - used when specifying a custom tone
CC_TONE_NULL 0x0	Null Tone

FAX-Rate and VAD Capabilities Values

Values	Meaning
CC_CAP_FAX_NONE 0x1	Fax disables or not available
CC_CAP_FAX_VOICE 0x2	Voice Call
CC_CAP_FAX_144 0x4	14,400 baud
CC_CAP_FAX_96 0x8	9,600 baud
CC_CAP_FAX_72 0x10	7,200 baud
CC_CAP_FAX_48 0x20	4,800 baud
CC_CAP_FAX_24 0x40	2,400 baud
CC_CAP_VAD_OFF 0x1	VAD Disabled
CC_CAP_VAD_ON 0x2	VAD Enabled

Related Information

- **Configuring Dial Plans, Dial Peers, and Digit Manipulation**
- **VoIP Debug Commands**
- **T1 Layer 1 Troubleshooting**
- **T1 Troubleshooting Flowchart**
- **Troubleshooting Serial Line Problems**
- **Voice Technology Support**
- **Voice and Unified Communications Product Support**
- **Recommended Reading: Troubleshooting Cisco IP Telephony**
- **Technical Support - Cisco Systems**