After configuring the voice ports on your router, follow these steps to verify proper operation:

Guide Contents
Troubleshooting Cisco IOS Voice Overview Debug Command Output on Cisco IOS Voice Gateways Filtering Troubleshooting Output Cisco VoIP Internal Error Codes Troubleshooting Cisco IOS Voice Telephony Troubleshooting Cisco IOS Voice Protocols Troubleshooting Cisco IOS Telephony Applications Monitoring the Cisco IOS Voice Network Cause Codes and Debug Values

Contents

SUMMARY STEPS

1. Check for dial tone.
2. Check for DTMF detection.
3. **show voice port summary**
4. **show voice port**
5. **show running-config**
6. show controller {t1 | e1} *controller-number*
7. **show voice dsp**
8. **show voice call summary**
9. **show call active voice**
10. **show call history voice** {**last** | *number* | **brief**}

DETAILED STEPS

1. Pick up the handset of an attached telephony device and check for a dial tone.

2. If you have dial tone, check for DTMF detection. If the dial tone stops when you dial a digit, then the voice port is most likely configured properly for DTMF detection.

3. To identify port numbers of voice interfaces installed in your router, use the **show voice port summary** command.

4. To verify voice-port parameter settings, enter the **show voice port** command with the appropriate syntax for your voice interfaces.

5. For digital T1/E1 connections, to verify the codec complexity configuration, enter the **show running-config** command to display the current voice-card setting. If medium complexity is specified, the codec complexity setting is not displayed. If high complexity is specified, the setting codec complexity **high** is displayed. The following example shows an excerpt from the command output when high complexity has been specified:

```
Router# show running-config
.
.
.
hostname router-alpha

voice-card 0
 codec complexity high
.
.
.
```

6. For digital T1/E1 connections, to verify that the controller is up and that no alarms have been reported, and to display information about clock sources and other controller settings, use the **show controller** command. For output examples, see the show controller Samples.

```
Router# show controller {t1 | e1} controller-number
```

7. To display voice-channel configuration information for all DSP channels, enter the **show voice dsp'** *command'*. For output examples, see the show voice dsp Samples.

```
Router# show voice dsp
```

8. To verify the call status for all voice ports, enter the **show voice call summary'** *command'*. For output examples, see the show voice call summary Samples.

```
Router# show voice call summary
```

9. To display the contents of the active call table, which shows all of the calls currently connected through the router or concentrator, enter **the '***show call active voice'* **command**. For output examples, see the show call active voice Samples.

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

10. To display the contents of the call history table, enter the **show call history voice'** *command'*. To limit the display to the last calls connected through this router, enter the keyword **last** and define the number of calls to be displayed with the argument *number*. To limit the display to a shortened version of the call history table, use the keyword **brief**. For output examples, see the show call history voice Sample.

```
Router# show call history voice {last | number | brief}
```

show voice port Samples

In the following sections, output examples of the following types are shown:

- Cisco 3600 series digital E&M voice port
- Cisco AS5300 universal access server T1 CAS voice port
- Cisco 7200 series router digital E&M voice port

Cisco 3600 Digital E&M 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
```

Cisco AS5300 Universal Access Server T1 CAS Voice Port

```
Router# show voice port

DS0 Group 1:0 - 1:0
 Type of VoicePort is CAS
 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
 Playout-delay Mode is set to default
 Playout-delay Nominal is set to 60 ms
 Playout-delay Maximum is set to 200 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
 Companding Type is u-law
 Region Tone is set for US
 Wait Release Time Out is 30 s
 Station name None, Station number None

 Voice card specific Info Follows:

 DS0 channel specific status info:
                              IN       OUT
    PORT   CH SIG-TYPE   OPER STATUS   STATUS    TIP     RING
```

Cisco 7200 Series Router Digital E&M Voice Port

```
Router# show voice port 1/0:1
receEive and transMit Slot is 1, Sub-unit is 0, Port is 1  << voice-port 1/0:1
 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
```

show controller Samples

In the following sections, output examples of the following types are shown:

- Cisco 3600 series router T1 controller
- Cisco AS5800 universal access server T1 controller

Cisco 3600 Series Router T1 Controller

```
Router# show controller T1 1/1/0
T1 1/0/0 is up.
  Applique type is Channelized T1
  Cablelength is long gain36 0db
  No alarms detected.
  alarm-trigger is not set
  Framing is ESF, Line Code is B8ZS, Clock Source is Line.
  Data in current interval (180 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
```

Cisco AS5800 Universal Access Server T1 Controller

```
Router# show controller t1 2
 T1 2 is up.
   No alarms detected.
   Version info of slot 0:  HW: 2, Firmware: 16, PLD Rev: 0

 Manufacture Cookie Info:
  EEPROM Type 0x0001, EEPROM Version 0x01, Board ID 0x42,
  Board Hardware Version 1.0, Item Number 73-2217-4,
  Board Revision A0, Serial Number 06467665,
  PLD/ISP Version 0.0, Manufacture Date 14-Nov-1997.

   Framing is ESF, Line Code is B8ZS, Clock Source is Internal.
   Data in current interval (269 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
```

show voice dsp Samples

The following output is from a Cisco 3640 router when a digital voice port is configured.

```
Router# show voice dsp
TYPE DSP CH CODEC    VERS STATE STATE   RST AI PORT    TS ABORT   TX/RX-PAK-CNT
==== === == ======== ==== ===== ======= === == ======= == ===== ===============
C549 010 00 g729r8    3.3 busy  idle      0  0 1/015   1     0      67400/85384
         01 g729r8     .8 busy  idle      0  0 1/015   7     0      67566/83623
         02 g729r8        busy  idle      0  0 1/015  13     0      65675/81851
         03 g729r8        busy  idle      0  0 1/015  20     0      65530/83610
C549 011 00 g729r8    3.3 busy  idle      0  0 1/015   2     0      66820/84799
         01 g729r8     .8 busy  idle      0  0 1/015   8     0      59028/66946
         02 g729r8        busy  idle      0  0 1/015  14     0      65591/81084
         03 g729r8        busy  idle      0  0 1/015  21     0      66336/82739
C549 012 00 g729r8    3.3 busy  idle      0  0 1/015   3     0      59036/65245
         01 g729r8     .8 busy  idle      0  0 1/015   9     0      65826/81950
         02 g729r8        busy  idle      0  0 1/015  15     0      65606/80733
         03 g729r8        busy  idle      0  0 1/015  22     0      65577/83532
C549 013 00 g729r8    3.3 busy  idle      0  0 1/015   4     0      67655/82974
         01 g729r8     .8 busy  idle      0  0 1/015  10     0      65647/82088
         02 g729r8        busy  idle      0  0 1/015  17     0      66366/80894
         03 g729r8        busy  idle      0  0 1/015  23     0      66339/82628
C549 014 00 g729r8    3.3 busy  idle      0  0 1/015   5     0      68439/84677
         01 g729r8     .8 busy  idle      0  0 1/015  11     0      65664/81737
         02 g729r8        busy  idle      0  0 1/015  18     0      65607/81820
         03 g729r8        busy  idle      0  0 1/015  24     0      65589/83889
C549 015 00 g729r8    3.3 busy  idle      0  0 1/015   6     0      66889/83331
         01 g729r8     .8 busy  idle      0  0 1/015  12     0      65690/81700
         02 g729r8        busy  idle      0  0 1/015  19     0      66422/82099
         03 g729r8        busy  idle      0  0 1/015  25     0      65566/83852
Router# show voice dsp
TYPE DSP CH CODEC    VERS STATE STATE   RST AI PORT    TS ABORT   TX/RX-PAK-CNT
==== === == ======== ==== ===== ======= === == ======= == ===== ===============
C549 007 00 {medium}  3.3 IDLE  idle      0  0 1/0:1    4      0            0/0
                      .13
C549 008 00 {medium}  3.3 IDLE  idle      0  0 1/0:1    5      0            0/0
                      .13
C549 009 00 {medium}  3.3 IDLE  idle      0  0 1/0:1    6      0            0/0
                      .13
C549 010 00 {medium}  3.3 IDLE  idle      0  0 1/0:1    7      0            0/0
                      .13
C549 011 00 {medium}  3.3 IDLE  idle      0  0 1/0:1    8      0            0/0
```

```
                      .13
C549 012 00 {medium}  3.3 IDLE  idle      0  0 1/0:1    9     0            0/0
                      .13
C542 001 01 g711ulaw  3.3 IDLE  idle      0  0 2/0/0          0        512/519
                      .13
C542 002 01 g711ulaw  3.3 IDLE  idle      0  0 2/0/1          0        505/502
                      .13
C542 003 01 g711alaw  3.3 IDLE  idle      0  0 2/1/0          0    28756/28966
                      .13
C542 004 01 g711ulaw  3.3 IDLE  idle      0  0 2/1/1          0        834/838
                      .13
```

show voice call summary Samples

The following is an output example of show voice call summary on a Cisco 3600 series router digital voice port:

Cisco 3600 Series Router Digital Voice Port

```
Router# show voice call summary
PORT      CODEC    VAD VTSP STATE            VPM STATE
========= ======== === ==================== ========================
1/015.1  g729r8    y  S_CONNECT             S_TSP_CONNECT
1/015.2  g729r8    y  S_CONNECT             S_TSP_CONNECT
1/015.3  g729r8    y  S_CONNECT             S_TSP_CONNECT
1/015.4  g729r8    y  S_CONNECT             S_TSP_CONNECT
1/015.5  g729r8    y  S_CONNECT             S_TSP_CONNECT
1/015.6  g729r8    y  S_CONNECT             S_TSP_CONNECT
1/015.7  g729r8    y  S_CONNECT             S_TSP_CONNECT
1/015.8  g729r8    y  S_CONNECT             S_TSP_CONNECT
1/015.9  g729r8    y  S_CONNECT             S_TSP_CONNECT
1/015.10 g729r8    y  S_CONNECT             S_TSP_CONNECT
1/015.11 g729r8    y  S_CONNECT             S_TSP_CONNECT
1/015.12 g729r8    y  S_CONNECT             S_TSP_CONNECT
```

show call active voice Samples

The output from the Cisco 7200 series router is shown below.

```
Router# show call active voice
GENERIC:
SetupTime=94523746 ms
Index=448
PeerAddress=##73072
PeerSubAddress=
PeerId=70000
PeerIfIndex=37
LogicalIfIndex=0
ConnectTime=94524043
DisconectTime=94546241
CallOrigin=1
ChargedUnits=0
InfoType=2
TransmitPackets=6251
TransmitBytes=125020
ReceivePackets=3300
ReceiveBytes=66000
VOIP:
ConnectionId[0x142E62FB 0x5C6705AF 0x0 0x385722B0]
RemoteIPAddress=171.68.235.18
```

```
RemoteUDPPort=16580
RoundTripDelay=29 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
SessionProtocol=cisco
SessionTarget=ipv4:171.68.235.18
OnTimeRvPlayout=63690
GapFillWithSilence=0 ms
GapFillWithPrediction=180 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=70 ms
LoWaterPlayoutDelay=30 ms
ReceiveDelay=40 ms
LostPackets=0 ms
EarlyPackets=1 ms
LatePackets=18 ms
VAD = disabled
CoderTypeRate=g729r8
CodecBytes=20
cvVoIPCallHistoryIcpif=0
SignalingType=cas
```

show call history voice Sample

The following output is for the Cisco 7200 series router.

```
Router# show call history voice
GENERIC:
SetupTime=94893250 ms
Index=450
PeerAddress=##52258
PeerSubAddress=
PeerId=50000
PeerIfIndex=35
LogicalIfIndex=0
DisconnectCause=10
DisconnectText=normal call clearing.
ConnectTime=94893780
DisconectTime=95015500
CallOrigin=1
ChargedUnits=0
InfoType=2
TransmitPackets=32258
TransmitBytes=645160
ReceivePackets=20061
ReceiveBytes=401220
VOIP:
ConnectionId[0x142E62FB 0x5C6705B3 0x0 0x388F851C]
RemoteIPAddress=171.68.235.18
RemoteUDPPort=16552
RoundTripDelay=23 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
SessionProtocol=cisco
SessionTarget=ipv4:171.68.235.18
OnTimeRvPlayout=398000
GapFillWithSilence=0 ms
GapFillWithPrediction=1440 ms
GapFillWithInterpolation=0 ms
```

```
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=97 ms
LoWaterPlayoutDelay=30 ms
ReceiveDelay=49 ms
LostPackets=1 ms
EarlyPackets=1 ms
LatePackets=132 ms
VAD = disabled
CoderTypeRate=g729r8
CodecBytes=20
cvVoIPCallHistoryIcpif=0
```

Verifying Digits Received and Sent on the POTS Call Leg

Once the on-hook and off-hook signaling are verified to be working correctly, verify the correct digits are being received or sent on the digital voice port. A dial peer that does not matched or the switch (CO or PBX) cannot ring the correct station if incomplete or incorrect digits are being sent or received. Some commands that can be used to verify the digits received or 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**-This command displays the digits as they are received by the voice port.
- **debug vtsp all**-This command enables the following debug voice telephony service provider (VTSP) commands: **debug vtsp session**, **debug vtsp error**, and **debug vtsp dsp**.

show dialplan number

The **show dialplan number** 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. The output of this command looks like this:

```
Router# show dialplan number 5000
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 dsp

debug vtsp dsp shows the digits as they are received by the voice-port. The following output shows the collection of DTMF digits from the DSP:

```
Router# debug vtsp dsp
Voice telephony call control dsp debugging is on
!-- ACTION: Caller picked up handset and dialed
!-- digits 5000.
!-- The DSP detects DTMF digits. Digit 5 was
!-- detected with ON time of 130msec.
*Mar 10 17:57:08.505: vtsp_process_dsp_message:
MSG_TX_DTMF_DIGIT_BEGIN: digit=5,
*Mar 10 17:57:08.585: vtsp_process_dsp_message:
MSG_TX_DTMF_DIGIT_OFF: digit=5,
duration=130
*Mar 10 17:57:09.385: vtsp_process_dsp_message:
MSG_TX_DTMF_DIGIT_BEGIN: digit=0
*Mar 10 17:57:09.485: vtsp_process_dsp_message:
MSG_TX_DTMF_DIGIT_OFF: digit=0,
duration=150
*Mar 10 17:57:10.697: vtsp_process_dsp_message:
MSG_TX_DTMF_DIGIT_BEGIN: digit=0
*Mar 10 17:57:10.825: vtsp_process_dsp_message:
MSG_TX_DTMF_DIGIT_OFF: digit=0,
duration=180
*Mar 10 17:57:12.865: vtsp_process_dsp_message:
MSG_TX_DTMF_DIGIT_BEGIN: digit=0
*Mar 10 17:57:12.917: vtsp_process_dsp_message:
MSG_TX_DTMF_DIGIT_OFF: digit=0,
duration=100

Router# debug vtsp session
Voice telephony call control session debugging is on
!--- <some output have been omitted>
!-- 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=-38 act_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 the digits are not being sent or received properly, then it might be necessary to use either a digit-grabber (test tool) or T1 tester to verify that 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 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 may add or remove digits. Refer to your switch documentation to check the translation tables on your switch.