Troubleshooting Analog Voice Interfaces to the IP Network

If you are troubleshooting an analog connection, you must understand what type of circuit and interface your voice port is using. Analog voice port interfaces connect routers in packet-based networks to analog two-wire or four-wire analog circuits in telephony networks. Two-wire circuits connect to analog telephone or fax devices, and four-wire circuits connect to PBXs. Analog voice telephony interfaces include foreign exchange office (FXO), foreign exchange station (FXS), and receive and transmit (E&M). Direct Inward Dialing (DID) is a service offered by telephone companies that enables callers to dial directly to an extension on a PBX without the assistance of an operator or automated call attendant.

To troubleshoot analog voice interfaces, see the following sections:

- FXS Interfaces, page 1
- FXO Interfaces, page 7
- E&M Interfaces, page 14
- Analog DID Interfaces, page 41
- Voice Port Testing Commands, page 44

If you are troubleshooting a connection to a PBX, you might find the PBX interoperability notes useful. These notes contain configuration information for Cisco gateways and several types of PBXs. To access these notes, use the following website:


FXS Interfaces

An FXS interface connects the router or access server to end-user equipment such as telephones, fax machines, and modems. The FXS interface supplies ring, voltage, and dial tone to the station and includes an RJ-11 connector for basic telephone equipment, keysets, and PBXs. In Figure 10, FXS signaling is used for end-user telephony equipment, such as a telephone or fax machine.
If you are having trouble with an FXS port, check the following sections:

- **FXS Hardware Troubleshooting**, page 2
- **Ring Voltage Problems**, page 4
- **Unbreakable Dial Tone**, page 6

## FXS Hardware Troubleshooting

An FXS interface connects directly to a standard telephone, fax machine, or similar device and supplies ring, voltage, and dial tone.

Troubleshoot FXS hardware by checking the following sections:

- **Software Compatibility**, page 2
- **Cabling**, page 2
- **Shutdown Port**, page 3
- **Disabling a Port on a Multiple Port Card**, page 4

## Software Compatibility

To ensure that your FXS card is compatible with your software, check the following:

- For network modules inserted into Cisco 2600 series, Cisco 3600 series, and Cisco 3700 series, check the compatibility tables in the “Overview of Cisco Network Modules” chapter in the *Cisco Network Modules Hardware Installation Guide*.
- For interface cards inserted into Cisco 1600 series, Cisco 1700 series, Cisco 2600 series, Cisco 3600 series, Cisco 3700 series, and Cisco ICS 7750 platforms, check the compatibility tables in the “Overview of Cisco Interface Cards” chapter in the *Cisco Interface Cards Installation Guide*.

## Cabling

Two types of cabling are supported for Cisco FXS interfaces. They are described in the following sections:

- **RJ-11 Connectors**, page 3
- **RJ-21 Connectors on the High-Density Analog Telephony Network Module**, page 3

### Note

For FXS connections, use a 2-wire (RJ-11) cable. A 4-wire cable can cause the second port to busy out.
RJ-11 Connectors

The two-port and four-port FXS interface cards support the RJ-11 connector. Illustrations of the connector ports are shown in Figure 11 and Figure 12. Information about LEDs can be found in the “Connecting Voice Interface Cards to a Network” chapter of the Cisco Interface Card Hardware Installation Guide.

Figure 11 Two-Port FXS Card Front Panel

Figure 12 Four-Port FXS/DID Card Front Panel

For information about the VIC-2FXS interface card, refer to Understanding Foreign Exchange Station (FXS) Voice Interface Cards, document ID 7938.

RJ-21 Connectors on the High-Density Analog Telephony Network Module

The High-Density Analog Telephony network module supports an RJ-21 connector. This network module supports both FXS and FXO traffic. An illustration of the connector port is shown in Figure 13. Information about LEDs and pinouts can be found in the “Connecting High-Density Analog Telephony Network Modules to a Network” chapter of the Cisco Network Modules Hardware Installation Guide.

Figure 13 High-Density Analog Telephony Network Module

Shutdown Port

If the port is not working, be sure the port is not shut down. Enter the show voice port command with the voice port number that you are troubleshooting. The output will tell you:

- If the voice port is up. If it is not, use the no shutdown command to make it active.
- What parameter values have been set for the voice port, including default values (which do not appear in the output from the show running-config command). If these values do not match those of the telephony connection you are making, reconfigure the voice port.
Disabling a Port on a Multiple Port Card

If you shut down a port on a multiple-port card, you can disable all of the ports on that card. If only one port is bad and the others are working, in many cases you can disable the bad port and then use the working ports until a replacement arrives. To disable a bad port, use one of the following methods:

- On a Cisco universal gateway, such as the Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850, busy out the port using the `busyout` command. This setting allows the port to be taken out of service without disrupting the Cisco IOS configuration. See the product documentation for details:
  - Cisco AS5350 product documentation
  - Cisco AS5400 product documentation
  - Cisco AS5800 product documentation
  - Cisco AS5850 product documentation
- On other Cisco gateways, remove the port from the dial peer. Refer to “Dial Peer Features and Configuration” in the Dial Peer Configuration on Voice Gateway Routers document to configure the dial peer.

Ring Voltage Problems

Telephone exchanges and FXSs need to supply DC battery and AC ringing to enable the connected telephone equipment to transmit speech energy and to power the telephone equipment’s ringing device. This section discusses what voltages are supplied by various Cisco FXS interfaces and how to overcome some known issues regarding voltage levels.

Ringing Voltages

The industry standard for PBX and key systems requires that the ring detection circuit be able to detect a ringing signal as low as 40 Vrms. This voltage takes into account the effects of load and cabling voltage drop on a ringing signal generated from a central office (CO). Conversely, the CO (exchange) must supply ringing with enough power to drive the maximum load over the maximum cable length. In order to meet this requirement, a CO-based unit must present a ringing signal with an amplitude of approximately 85 to 100 Vrms. Cisco voice gateways are intended for use as on premise services (ONS) equipment that is colocated or fairly close to equipment that detects ringing, so it can therefore use a lower ringing voltage and still meet the 40 Vrms 5 Ringer Equivalence Number (REN) requirement.

Idle Battery Voltage

Cisco voice gateways were designed for ONS connections and by default the FXS interface supplies either –24 Vdc or –36 Vdc idle battery, whereas off premise services, such as a CO, would require voltages of –48 V because it might have to interconnect over much greater cable lengths. Certain Cisco FXS interfaces can be configured to supply higher voltages.
Idle Line Voltages

Table 24 shows idle line voltages supplied by various Cisco gateway FXS interfaces.

<table>
<thead>
<tr>
<th>FXS Interface</th>
<th>Idle Voltage</th>
</tr>
</thead>
<tbody>
<tr>
<td>VG248</td>
<td>–36V</td>
</tr>
<tr>
<td>VIC-2FXS</td>
<td>–26V</td>
</tr>
<tr>
<td>VIC-2DID</td>
<td>–24V (low) –48V (high)</td>
</tr>
<tr>
<td>ASI 81 and ASI 160</td>
<td>–24V (low) –48V (high)</td>
</tr>
<tr>
<td>IAD 24xx-FXS</td>
<td>–24V (low) –48V (high)</td>
</tr>
<tr>
<td>1730 IAD</td>
<td>–24V (low) –48V (high)</td>
</tr>
<tr>
<td>VIC-4FXS/DID</td>
<td>–24V (low) –48V (high)</td>
</tr>
</tbody>
</table>

Ring Voltage Problems

Voltage problems can cause three types of problems:
- Answering and Call Initiation Problems with Automated Telephony Devices, page 5
- Ringing Problems, page 5
- FXS Ring Failure in the United Kingdom, page 6

Certain automated devices, such as fax machines, answer machines, multiline phones and voice mail systems, look at the line voltage in order to deduce if the line is busy or idle. If another device is off hook, then the line voltage drops, and the automated system does not answer or initiate a call. If the threshold being used is close to –24 V or higher, this can cause the device not to work as expected.

Certain phones might not ring when the default ring voltage and ring frequency are applied from the Cisco FXS interface.

Answering and Call Initiation Problems with Automated Telephony Devices

In voice port configuration mode, configure the idle-voltage command on the voice port of the FXS to increase idle battery voltage from –24 V to –48 V. The idle-voltage low setting designates -24 V and the idle-voltage high setting designates -48 V.

Note

This option is not available on VG248, VIC-2FXS, and WS-x6624 FXS interfaces.

Ringing Problems

Phone manufacturers sometimes use frequency filters known as antitinkle circuits to prevent ringer devices from sounding while the user is dialing. Sometimes it is necessary to adjust the frequency of the ring to suit the connected device.

Configure the ring frequency for Cisco modular access routers by issuing the following command:

```
Router(config-voiceport)# ring frequency ?
25 ring frequency 25 Hertz
50 ring frequency 50 Hertz
```
Configure the ring frequency for the Cisco IAD2400 platform by issuing the following command:

```
Router(config-voiceport)# ring frequency ?
20 ring frequency 20 Hertz
30 ring frequency 30 Hertz
```

To prevent ringer devices from sounding, you can also provide a voltage threshold so that the lower voltages, which can be produced during dialing, are ignored. Increasing the voltage can overcome this.

Configure the DC offset voltage on Cisco IAD2400 series routers by issuing the following command:

```
Router(config-voiceport)# ring dc-offset ?
10-volts Ring DC offset 10 volts
20-volts Ring DC offset 20 volts
24-volts Ring DC offset 24 volts
```

**Note**

This command sequence can be used only for Cisco IAD2400 series routers. The 24-V ring DC offset setting is available for Cisco IOS 12.2(11)T and later releases.

---

**FXS Ring Failure in the United Kingdom**

A telephone approved for the United Kingdom might fail to ring when connected to a Cisco FXS port. The failure results from a physical interoperability issue and is independent of Cisco hardware or software. British Telecom did not implement RJ-11 type connectors when it adopted plug-and-socket connection methodology. RJ-11 connectors allow parallel connectivity for the transmission path and the ringer circuit. They were not used because older telephones needed to have their ringer circuits connected in series due to a requirement for high current.

Outside the United Kingdom, ringer circuitry is self-contained in each phone. The U.K. implementation puts the capacitor, which provides the AC ring path, and the antitinkle feature (prevents the bell or ringer from sounding when pulse dialing is used) externally in the first socket, connected to the local loop.

In the United Kingdom, certain British Approval Board for Telecommunications (BABT) telephones fail to ring when they are connected to FXS ports on Cisco voice-enabled routers and switches. Outgoing calls can be made and voice communication in both directions can be established. However, incoming calls do not ring the telephone. These telephones functioned correctly before they were connected to the FXS ports.

Because a proprietary connection system is implemented, you must use an adapter to connect the telephone to an FXS port. The adapter must be a *master* that contains the capacitor, or the telephone fails to ring.

For a schematic and more information, refer to *Understanding Why Telephones in the United Kingdom Connected to Cisco FXS Interfaces May Fail to Ring*, document ID 25800.

---

**Unbreakable Dial Tone**

A common problem encountered in a VoIP network is being unable to break dial tone. The router seizes a line on the local PBX but when digits are dialed, the dial tone stays. The calling party is unable to pass the dual-tone multifrequency (DTMF) tones or digits to the terminating device, resulting in callers being unable to dial the desired extension or interact with a device that needs DTMF tones such as a voice mail or an interactive voice response (IVR) application. This problem can result from a number of sources, for example:

- DTMF tones are not passed.
- DTMF tones are not understood.
DTMF tones are too distorted to be understood.

Other signaling and cabling issues occur.

Make sure the dial type is set as DTMF on both the router and the PBX. The FXS port does not pass on the digits; therefore, this setting is not available on an FXS port. However, this setting can be changed on FXO and E&M ports:

```
Router(config-voiceport)# dial-type ?
    dtmf   touch-tone dialer
    mf     mf-tone dialer
    pulse  pulse dialer
```

For more information, refer to *Inability To Break Dialtone in a Voice over IP Network*, document ID 22376.

### No LED When Phone Off the Hook

Verify if you have an analog or digital card. If you have an analog card like the VIC-2FXS or the VIC2-4FXS, you might have one of the following problems:

- The port is in a shutdown state.
- The port is in a park state.
- The port is bad.

If you have a digital card like the NM-2V, you might have bad DSPs.

Use the following procedure if there is no LED when your phone is off hook:

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Check the cable to make sure that it is RJ-11 with two pins for the FXS port.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Test the LED using a different phone.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Check your Cisco IOS version to make sure that the feature set is either IP Plus or Enterprise Plus.</td>
</tr>
<tr>
<td>Step 4</td>
<td>If Steps 1 to 3 do not work, replace the voice interface card (VIC).</td>
</tr>
</tbody>
</table>

### FXO Interfaces

An FXO interface is used for trunk, or tie line, connections to a PSTN CO or to a PBX that does not support E&M signaling (when local telecommunications authority permits). This interface is of value for off-premises station applications. *Figure 14* shows an FXS connection to a telephone and an FXO connection to the PSTN at the far side of a WAN.
If you are having trouble with an FXO port, check the following sections:

- FXO Hardware Troubleshooting, page 8
- FXO Disconnect Failure, page 10
- Troubleshooting FXO Answer and Disconnect Supervision, page 10
- Unbreakable Dial Tone, page 12
- Troubleshooting Caller ID Problems, page 12

**FXO Hardware Troubleshooting**

An FXO interface is used for trunk connections. Troubleshoot FXO hardware by checking the following sections:

- Software Compatibility, page 8
- Cabling, page 8
- Shutdown Port, page 9
- Disabling a Port on a Multiple Port Card, page 10

**Software Compatibility**

To ensure that your card is compatible with your software, check the following:

- For network modules inserted into Cisco 2600 series, Cisco 3600 series, and Cisco 3700 series routers, refer to the compatibility tables in the “Overview of Cisco Network Modules” chapter in the Cisco Network Modules Hardware Installation Guide.
- For interface cards inserted into Cisco 1600 series, Cisco 1700 series, Cisco 2600 series, Cisco 3600 series, Cisco 3700 series, and Cisco ICS 7750 platforms, refer to the compatibility tables in the “Overview of Cisco Interface Cards” chapter in the Cisco Interface Cards Installation Guide.

**Cabling**

Two types of cabling are supported for Cisco FXO interfaces. They are described in the following sections:

- RJ-11 Connectors, page 3
- RJ-21 Connectors on the High-Density Analog Telephony Network Module, page 3
For FXO connections, use a 2-wire (RJ-11) cable. A 4-wire cable can cause the second port to busy out.

**RJ-11 Connectors**

The two-port and four-port FXO interface cards support the RJ-11 connector. Illustrations of the connector ports are shown in Figure 15 and Figure 16. Information about LEDs can be found in the “Connecting Voice Interface Cards to a Network” chapter of the Cisco Interface Card Hardware Installation Guide.

**Figure 15 Two-Port FXO Card Front Panel**

![Two-Port FXO Card Front Panel](image)

**Figure 16 Four-Port FXO Card Front Panel**

![Four-Port FXO Card Front Panel](image)

**RJ-21 Connectors on the High-Density Analog Telephony Network Module**

The High-Density Analog Telephony network module supports an RJ-21 connector. This network module supports both FXS and FXO traffic. An illustration of the connector port is shown in Figure 17. Information about LEDs and pinouts can be found in the “Connecting High-Density Analog Telephony Network Modules to a Network” chapter of the Cisco Network Modules Hardware Installation Guide.

**Figure 17 High-Density Analog Telephony Network Module**

![High-Density Analog Telephony Network Module](image)

**Shutdown Port**

If the port is not working, be sure the port is not shut down. Enter the `show voice port` command with the voice port number that you are troubleshooting. The output will tell you:

- If the voice port is up. If it is not, use the `no shutdown` command to make it active.
• What parameter values have been set for the voice port, including default values (these values do not appear in the output from the `show running-config` command). If these values do not match those of the telephony connection you are making, reconfigure the voice port.

Disabling a Port on a Multiple Port Card

If you shut down a port on a multiple-port card, you can disable all of the ports on that card. If only one port is bad and the others are working, in many cases you can disable the bad port and use the working ports until a replacement arrives. To disable a bad port, use one of the following methods:

• On a Cisco universal gateway, such as the Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850, busy out the port using the `busyout` command. This allows the port to be taken out of service without disrupting the Cisco IOS configuration. Refer to the product documentation for details:

  – For Cisco AS5350 and Cisco AS5400 universal gateways, refer to the “Managing and Troubleshooting the Universal Port Card” chapter in the *Cisco AS5350 and Cisco AS5400 Universal Gateway Software Configuration Guide*.

  – For Cisco AS5800 access servers, refer to the *Managing and Troubleshooting NextPort Services on the AS5800* feature document.

  – For Cisco AS5850 universal gateways, refer to the *Managing Port Services on the Cisco AS5850 Universal Gateway* feature document.

• On other Cisco gateways, remove the port from the dial peer. Refer to “Dial Peer Features and Configuration” in the *Dial Peer Configuration on Voice Gateway Routers* document to configure the dial peer.

FXO Disconnect Failure

When loop-starting signaling is used, an FXO interface looks like a phone to the switch that it is connecting to. The FXO interface closes the loop to indicate off hook. The switch always provides a battery so there is no disconnect supervision from the switch side. Because a switch expects a phone user or modem to hang up the phone when the call is terminated on either side, it also expects the FXO port on the router to hang up. However, the FXO port expects the switch to tell it when to hang up. Because the port relies on the switch, there is no guarantee that a near- or far-end FXO port will disconnect the call once either end of the call has hung up.

The most common symptoms of this problem are phones that continue to ring when the caller has cleared, or FXO ports that remain busy after the previous call should have been cleared.

To troubleshoot this problem, refer to *Understanding FXO Disconnect Problem*, document ID 8120.

Troubleshooting FXO Answer and Disconnect Supervision

This section describes troubleshooting the FXO Answer and Disconnect Supervision feature for analog FXO voice ports. This feature applies to analog FXO voice ports with loop-start signaling connected to PSTNs, PBXs, or key systems.

The FXO Answer and Disconnect Supervision feature enables analog FXO ports to monitor call-progress tones and to monitor voice and fax transmissions returned from a PBX or from the PSTN.
Answer supervision can be accomplished in two ways: by detecting battery reversal, or by detecting voice, fax, or modem tones. If an FXO voice port is connected to the PSTN and battery reversal is supported, use the battery reversal method. Voice ports that do not support battery reversal must use the answer supervision method, in which answer supervision is triggered when the DSP detects voice, modem, or fax transmissions. Configuring answer supervision automatically enables disconnect supervision; however, you can configure disconnect supervision separately if answer supervision is not configured.

Disconnect supervision can be configured to detect call-progress tones sent by the PBX or PSTN (for example, busy, reorder, out-of-service, number-unavailable), or to detect any tone received (for example, busy tone or dial tone). When an incoming call ends, the DSP detects the associated call-progress tone, causing the analog FXO voice port to go on-hook.

This section provides solutions to problems that you might encounter when implementing the FXO Answer and Disconnect Supervision feature.

Typical problems with the answer supervision feature are as follows:

- Call-progress tones such as ringback are not heard by the calling party.
  - If any call legs have IVR configured, ensure that the IVR version is 2.0.
- Ringback timer is not initiated or ringback is not detected.
  - The wrong call-progress tone (\texttt{cptone}) command is configured on the voice port.
  - The wrong DTMF detection parameters are configured.
  - Custom call-progress tones are assigned to the voice port but ringback tone has not been configured; in this case, the default behavior is not to detect any ringback tones.
- Answer supervision is not triggered. Answer supervision—either by battery-reversal detection or by call-progress tone detection—is not configured on the voice port in use.
- Excessive delay before answer supervision is activated.
  - The level on the sensitivity parameter in the \texttt{supervisory answer dualtone} command is set too low. Configure the sensitivity for \texttt{high}.

If incorrect disconnect cause codes are reported, check the following:

- The values configured for custom call-progress tones could be incorrect.
- Overlapping detection frequencies might have been incorrectly specified in the voice class created by the \texttt{voice class dualtone-detect-params} command. For example if the \texttt{freq-max-deviation} parameter is configured to be 20 Hz, and the busy and reorder parameters are set for frequencies 350 and 370 respectively, the voice port cannot detect the reorder tone, resulting in an incorrect disconnect cause code.

\textbf{Note} If the frequencies and cadences (including error deviations as defined in the \texttt{voice class dualtone-detect-params} command) are the same for multiple call-progress tones, the order of detection is as follows: \texttt{busy}, \texttt{reorder}, \texttt{number-unobtainable}, \texttt{out-of-service}, \texttt{disconnect}.

If calls are not billed correctly, it might be that answer supervision is not being triggered. For answer supervision to be triggered, voice, fax, or data tones originating at the called-party end must be detected.

To configure the FXO Supervisory Disconnect Tone, refer to the \textit{Voice Port Configuration} document.
Monitoring and Maintaining FXO Answer and Disconnect Supervision

To monitor the status of the FXO Answer and Disconnect Supervision feature, use the `show voice port` command, which causes the FXO voice port to be monitored. The following table illustrates the use of the `show voice port` command for monitoring voice port 1/1/0.

<table>
<thead>
<tr>
<th>Command</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Router# show voice port 1/1/0</td>
<td>Shows a detailed status of the voice port. Under the heading “Voice card specific Info Follows:”, the status of the FXO Answer and Disconnect Supervision feature is indicated by one of the following messages: “Answer Supervision is active” or “Answer Supervision is inactive”.</td>
</tr>
</tbody>
</table>

Unbreakable Dial Tone

A common problem encountered in a VoIP network is being unable to break dial tone. The router puts a seizure on the local PBX but when digits are dialed, the dial tone stays. The calling party is unable to pass the DTMF tones or digits to the terminating device, resulting in callers being unable to dial the desired extension or interact with the device that needs DTMF tones, such as a voice mail or IVR application. Here are some possible causes of the problem:

- DTMF tones not being sent.
- DTMF tones not being understood.
- DTMF tones too distorted to be understood.
- Other signaling and cabling issues.

Make sure the dial type is set as DTMF on both the router and the PBX. The FXS port does not pass on the digits, therefore this setting is not available on an FXS port. However, this setting can be changed on FXO and E&M ports:

```
Router(config-voiceport)# dial-type ?
dtmf    touch-tone dialer
        mf        mf-tone dialer
        pulse   pulse dialer
```

For more information, refer to *Inability To Break Dialtone in a Voice over IP Network*, document ID 22376.

Troubleshooting Caller ID Problems

Several debugs can be used to troubleshoot the Caller ID feature on the routers. The voice port module (VPM) signaling debugs, such as the `debug vpm signal` command, track the standard debugs with Caller ID feature turned on. These debugs are analyzed from the perspective of the terminating router and its FXO port; the caller ID is sent from this end. The following example shows an FXO port receiving caller ID. In this example, the phone sends the caller ID to the FXO port.

```
Nov 20 10:40:15.861 EST: [1/0/0] htsp_start_caller_id_rx
Nov 20 10:40:15.861 EST: [1/0/0] htsp_set_caller_id_rx:BELLCORE
Nov 20 10:40:15.861 EST: htsp_timer - 10000 msec
Nov 20 10:40:17.757 EST: [1/0/0, FXOLS_RINGING, E_DSP_SIG_0100]
Nov 20 10:40:17.757 EST: fxols_ringing_not
Nov 20 10:40:17.761 EST: htsp_timer_stop
```
In this example, everything was working fine and both Name and Number Display were properly delivered to the phone. In the two scenarios below, the calling number is missing in one case, and the name display is missing in the other.

**Calling Number Lost, Name Delivered**

In the following example, the calling number is lost, but the name is delivered:

```
Nov 17 17:39:34.164 EST: [1/1/0] htsp_set_caller_id_tx
calling num=display_info=Outside called num=9913050
```

In the Caller ID String, looking at “04 01 4F” translates to:

```
04 : Reason for Absence of DN
01 : Length of message
4F : "Out of Area"
```

**Calling Number Delivered, Name Lost**

In the following example, the calling number is delivered, but the name is lost:

```
Nov 17 17:53:24.034 EST: [1/1/0] htsp_set_caller_id_tx
calling num=5550109display_info= called num=5550011
Nov 17 17:53:24.034 EST: [1/1/0] Caller ID String 80 16
01 08 31 31 31 37 32 32 33 33 39 04 01 4F
73 69 64 65 88
```

In the Caller ID String, looking at “04 01 4F” translates to:

```
04 : Reason for Absence of DN
01 : Length of message
4F : "Out of Area"
In the Caller ID String, looking at “**08 01 4F**” translates to:

- **08**: Reason for Absence of Display
- **01**: Length
- **4F**: Out of Area

For more information, refer to *Caller ID Name Delivery Issues on Cisco IOS Gateways*, document ID 23444.

**E&M Interfaces**

The difference between a conventional two-wire telephone interface such as FXS or FXO and an E&M interface is that the E&M interface has wires that pass the audio signals plus wires to act as an input (to sense an incoming call) or an output (to indicate an outgoing call). These control leads are normally called the E lead (input) and the M lead (output). Depending on the type of E&M interface, the signaling leads could be controlled by connecting them to the ground, switching a –48-Vdc source, or completing a current loop between the two devices.

E&M interfaces can normally be two- or four-wire operation, which does not refer to the total number of physical connections on the port but rather to the way that audio is passed between the devices. Two-wire operation means the transmitting and receiving audio signals are passed through a single pair of wires (one pair equals two wires). Four-wire operation uses one pair for transmitting and another pair for receiving audio.

In Figure 18, two PBXs are connected across a WAN by E&M interfaces. This topology illustrates the path over a WAN between two geographically separated offices in the same company.

**Figure 18 E&M Signaling Interfaces**

This section contains the following topics:

- E&M Hardware Troubleshooting, page 14
- E&M Interface Types, page 17
- Troubleshooting E&M Interfaces at the Physical Level, page 26
- Confirming E&M Configuration, page 33
- Unbreakable Dial Tone, page 41

**E&M Hardware Troubleshooting**

The E&M interface typically connects remote calls from an IP network to a PBX. Troubleshoot Cisco E&M hardware by checking the following sections:

- Software Compatibility, page 15
Software Compatibility

For interface cards inserted into Cisco modular access routers, check the compatibility tables in the “Overview of Cisco Interface Cards” chapter in the Cisco Interface Cards Installation Guide.

Cabling

E&M is a signaling technique for two-wire and four-wire telephone and trunk interfaces. The E&M interface typically connects remote calls from an IP network to a PBX. The card is connected to the PSTN or PBX through a telephone wall outlet by a straight-through RJ-48C cable.

Note Refer to the appropriate platform product documentation for specific interface information about your E&M card.

The connector port for the E&M voice interface card is shown in Figure 19. Information about LEDs can be found in the “Connecting Voice Interface Cards to a Network” chapter of the Cisco Interface Cards Installation Guide.

Note Ports on the E&M voice interface card are color-coded brown.

![Two-Port E&M Card Front Panel](image)

To verify that the analog E&M hardware is being recognized by the Cisco IOS platform, use the following commands:

- `show version`—This command displays the configuration of the system hardware, the software version, the names of configuration files, and the boot images. See the following sample output.

- `show running-config`—This command shows the configuration of the Cisco platform. The voice ports should appear in the configuration automatically. See the following sample output.

**show version Command on a Cisco 3640 Platform**

```plaintext
Router# show version
Cisco Internetwork Operating System Software
IOS (tm) 3600 Software (C3640-1S-32M), Version 12.1(2), RELEASE SOFTWARE (fc1)
Copyright (c) 1986-2000 by cisco Systems, Inc.
Compiled Wed 10-May-00 07:20 by linda
Image text-base: 0x600088F0, data-base: 0x60E38000
```
ROM: System Bootstrap, Version 11.1(20)AA2, EARLY DEPLOYMENT RELEASE SOFTWARE(fc1)
Router uptime is 0 minutes
System returned to ROM by power-on at 11:16:21 cst Mon Mar 12 2001
System image file is "flash:c3640-is-mz.121-2.bin"
cisco 3640 (R4700) processor (revision 0x00) with 126976K/4096K bytes of memory.
Processor board ID 16187704
R4700 CPU at 100Mhz, Implementation 33, Rev 1.0
Bridging software.
X.25 software, Version 3.0.0.
SuperLAT software (copyright 1990 by Meridian Technology Corp).
2 Ethernet/IEEE 802.3 interface(s)
2 Voice FXS interface(s)
2 Voice E&M interface(s)
DRAM configuration is 64 bits wide with parity disabled.
125K bytes of non-volatile configuration memory.
32768K bytes of processor board System flash (Read/Write)
20480K bytes of processor board PCMCIA Slot0 flash (Read/Write)
Configuration register is 0x2102

show running-config Command on a Cisco 3640 Platform

Router# show running-config

Building configuration...
Current configuration:
!
! --- Some output omitted.
version 12.1
service timestamps debug uptime
service timestamps log uptime
!
hostname Router
!
voice-port 3/0/0
!
voice-port 3/0/1
!
voice-port 3/1/0
!
voice-port 3/1/1
!
end

Shutdown Port

Check to make sure the port is not shut down. Enter the show voice port command with the voice port number that you are troubleshooting. The output will tell you:

- If the voice port is up. If it is not, use the no shutdown command to make it active.
- What parameter values have been set for the voice port, including default values (default values do not appear in the output from the show running-config command). If these values do not match those of the telephony connection you are making, reconfigure the voice port.
E&M Interface Types

This section describes the standard analog E&M interface types I, II, III, and V (IV is not supported by Cisco platforms). The following topics are covered:

- E&M Signaling Unit Side and Trunk Circuit Side Compatibility Issues, page 17
- E&M Type I Interface Model, page 18
- E&M Type II Interface Model, page 20
- E&M Type III Interface Model, page 22
- E&M Type V Interface Model, page 24

E&M Signaling Unit Side and Trunk Circuit Side Compatibility Issues

E&M signaling defines a trunk circuit side and a signaling unit side for each connection. Cisco's analog E&M interface functions as the signaling unit side, so it expects the other side to be a trunk circuit. When you use E&M interface model Type II or Type V, you can connect two signaling unit sides back to back by appropriate crossing of the signaling leads. When using the E&M Type I or Type III interface, you cannot connect two signaling unit sides back to back.

Many PBX brands have E&M analog trunk cards that can operate as either the trunk circuit side or the signaling unit side. Because the Cisco E&M interfaces are fixed as the signaling unit side of the interface, it may be necessary to change the E&M trunk settings on the PBX to operate as the trunk circuit side. If Type I or III E&M is being used, this is the only way the PBX can work with the Cisco E&M interface.

Some PBX products (and many key systems) can operate only as the signaling unit side of the E&M interface. They cannot interoperate with the Cisco E&M interface if Type I or Type III is chosen. If Type II or Type V E&M is being used, PBX products fixed as "signaling unit" side can still be used with the Cisco E&M interface via Type II or Type V.

Each E&M signaling type has a unique circuit model and connection diagram. The following sections describe the different types. Table 25 shows the E&M supervisory signal description.

<table>
<thead>
<tr>
<th>Signal</th>
<th>Meaning</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>E</td>
<td>Ear or earth</td>
<td>Signal wire from trunking (CO) side to signaling side.</td>
</tr>
<tr>
<td>M</td>
<td>Mouth or magnet</td>
<td>Signal wire from signaling side to trunking (CO) side.</td>
</tr>
<tr>
<td>SG</td>
<td>Signal ground</td>
<td>Used on E&amp;M Types II, III, and IV. (Type IV is not supported on Cisco gateways.)</td>
</tr>
<tr>
<td>SB</td>
<td>Signal battery</td>
<td>Used on E&amp;M Types II, III, and IV. (Type IV is not supported on Cisco gateways.)</td>
</tr>
<tr>
<td>T/R</td>
<td>Tip/Ring</td>
<td>Tip and ring leads carry audio between the signaling unit and the trunking circuit. On a two-wire audio operation circuit, this pair carries the full-duplex audio path.</td>
</tr>
<tr>
<td>T1/R1</td>
<td>Tip-1/Ring-1</td>
<td>Used on four-wire audio operation circuits only. The four-wire implementation provides separate paths for receiving and sending audio signals.</td>
</tr>
</tbody>
</table>
E&M Type I Interface Model

E&M Type I is the original E&M lead signaling arrangement, and it is the most common interface type in North America. Table 26 shows the sent signal states for on- and off-hook signaling.

<table>
<thead>
<tr>
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<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>M</td>
<td>Ground</td>
<td>Battery</td>
<td>E</td>
<td>Open</td>
<td>Ground</td>
</tr>
</tbody>
</table>

The gateway grounds its E-lead to signal a trunk seizure. The PBX applies battery to its M-lead to signal a seizure. Cisco gateways expect to see off-hook conditions on the M-lead, and they signal off-hook to a remote device on the E-lead. E&M Type I 2-wire operation is shown in Figure 20. E&M Type I 4-wire operation is shown in Figure 21.

Figure 20  E&M Type I 2-Wire Audio Operation

![E&M Type I 2-Wire Audio Operation Diagram]
For the four-wire audio setup, Pin 6 (Tip) and Pin 3 (Ring) on the router transport the audio path from the PBX to the router. Pin 5 (Tip1) and 4 (Ring1) on the router transport the audio path from the router to the PBX. Pins for the cable are shown in Figure 22.
Considerations for Type I interfaces include:

- Two signaling units cannot be connected back to back.
- A Type I signaling unit and a trunk circuit share a common ground.
- Type I does not provide isolation between trunk circuits and signaling units, can produce noise in audio circuits, and might be susceptible to electrical transients.
- It is critical to provide and ground connection directly between the Cisco product and the PBX. Otherwise, E&M signaling might be intermittent.
- Four wires are used for Type I, two-wire audio operation.
- Six wires are used for Type I, four-wire audio operation.

**E&M Type II Interface Model**

E&M Type II provides a 4-wire fully looped arrangement that provides full isolation between the trunks and signaling units. Type II is usually used on Centrex lines and Nortel PBX systems. Table 27 shows the sent signal states for on- and off-hook signaling.

<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>M</td>
<td>Open</td>
<td>Battery</td>
<td>E</td>
<td>Open</td>
<td>Ground</td>
</tr>
</tbody>
</table>

The gateway grounds its E-lead to signal a trunk seizure. The PBX applies battery to its M-lead to signal a seizure. Cisco gateways expect to see off-hook conditions on the M-lead, and they signal off-hook to a remote device on the E-lead. E&M Type II 2-wire operation is shown in Figure 23. E&M Type II 4-wire operation is shown in Figure 24.
Figure 23  E&M Type II 2-Wire Audio Operation

-48V

PBX

Analog E&M 2-wire audio operation

Tie-line Cisco equipment

Detect

E

SG

M

SB

2-wire audio

Trunk circuit

Signaling unit

-48V

On-hook

Off-hook

Detect

ptc

2-wire audio

Trunk circuit

Signaling unit
**E&M Type II Interface Model**

E&M Type II interfaces include:
- Two signaling unit sides can be connected back-to-back if the appropriate signaling leads are swapped.
- Six wires are used for Type II, two-wire audio operation.
- Eight wires are used for Type II, four-wire audio operation.

**E&M Type III Interface Model**

E&M Type III is a partially looped four-wire E&M arrangement with ground isolation. The signaling unit provides both the battery and the ground. Table 28 shows the sent signal states for on- and off-hook signaling.
Troubleshooting Analog Voice Interfaces to the IP Network

E&M Interfaces

The router senses loop current on the M-lead for an inbound seizure and grounds its E-lead for an outbound seizure. Cisco routers/gateways expect to see off-hook conditions on the M-lead, and they signal off-hook to a remote device on the E-lead. E&M Type III 2-wire operation is shown in Figure 25. E&M Type III 4-wire operation is shown in Figure 26.

**Table 28**  
**E&M Type III Signal States**

<table>
<thead>
<tr>
<th>PBX to Cisco Gateway</th>
<th>Cisco Gateway to PBX</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Lead</strong></td>
<td><strong>On-Hook</strong></td>
</tr>
<tr>
<td>M</td>
<td>Ground</td>
</tr>
</tbody>
</table>

The router senses loop current on the M-lead for an inbound seizure and grounds its E-lead for an outbound seizure. Cisco routers/gateways expect to see off-hook conditions on the M-lead, and they signal off-hook to a remote device on the E-lead. E&M Type III 2-wire operation is shown in Figure 25. E&M Type III 4-wire operation is shown in Figure 26.

**Figure 25**  
**E&M Type III 2-Wire Audio Operation**
For the four-wire audio setup, Pin 6 (Tip) and Pin 3 (Ring) on the router transport the audio path from the PBX to the router. Pin 5 (Tip1) and Pin 4 (Ring1) on the router transport the audio path from the router to the PBX.

Considerations for Type III interfaces include:

- Two signaling units cannot be connected back to back.
- Six wires are used for Type III, two-wire audio operation.
- Eight wires are used for Type III, four-wire audio operation.

**E&M Type V Interface Model**

E&M Type V is widely used outside North America (nearly a worldwide standard.) Type V is a symmetrical two-wire lead arrangement that signals in both directions (open for on-hook and ground for off-hook.) Table 29 shows the sent signal states for on- and off-hook signaling.
The gateway grounds its E-lead to signal a trunk seizure. The PBX grounds its M-lead to signal a seizure. Cisco gateways expect to see off-hook conditions on the M-lead, and they signal off-hook to remote device on the E-lead. E&M Type V 2-wire operation is shown in Figure 27. E&M Type V 4-wire operation is shown in Figure 28.

**Table 29**  
**E&M Type V Signal States**

<table>
<thead>
<tr>
<th>PBX to Cisco Gateway</th>
<th>Cisco Gateway to PBX</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Lead</strong></td>
<td><strong>On-Hook</strong></td>
</tr>
<tr>
<td>M</td>
<td>Open</td>
</tr>
</tbody>
</table>

**Figure 27**  
**E&M Type V 2-Wire Audio Operation**

![Diagram of E&M Type V 2-Wire Audio Operation]
Figure 28  E&M Type V 4-Wire Audio Operation

Note

For the four-wire audio setup, Pin 6 (Tip) and Pin 3 (Ring) on the router transport the audio path from the PBX to the router. Pin 5 (Tip1) and Pin 4 (Ring1) on the router transport the audio path from the router to the PBX.

Considerations for Type V interfaces include:

- Type V does not provide ground isolation.
- Two signaling unit sides can be connected back-to-back if the appropriate signaling leads are swapped.
- Four wires are used for Type V, two-wire audio operation.
- Six wires are used for Type V, four-wire audio operation.

Troubleshooting E&M Interfaces at the Physical Level

E&M provides the highest quality analog interface available, but it also is the most difficult to administer due to the number of leads, configurations, and protocol issues. Usually it is helpful to have the appropriate reference diagram available when verifying the connections.
Preparing to Troubleshoot E&M Physical Problems

Use the information in the following sections to prepare to troubleshoot E&M physical problems:

- Hardware Troubleshooting Tools, page 27
- Precautions, page 27
- PBX Interconnection, page 28
- Use Rollover Cable for E&M Port-to-Port Testing, page 28

Hardware Troubleshooting Tools

Test equipment is not required for every installation, but sometimes you need to use it to isolate problems with analog E&M ports. The most useful equipment is a digital multimeter and a technician's line test set. These tools allow measurement of signaling states and voltages, and monitoring of audio signals. A digital multimeter is used to measure the DC loop voltage and AC ringing voltage on FXS ports, E- or M-lead signaling transitions, voltages on E or M leads, and DC resistance of E&M signaling leads.

In the terminating mode of operation, the technician's line test set acts like a normal telephone handset when connected to a loopstart trunk, allowing telephone numbers to be dialed on the built-in keypad. When switched to the monitoring mode (bridging mode), the unit presents a high impedance to the TX or RX audio pairs of the E&M port, allowing the audio signals and tones to be heard on the built-in loudspeaker. This mode helps you find problems with one-way audio, incorrect digits being sent or received, distortion and level problems, and possible sources of noise and echo.

For an effective troubleshooting kit, have the following items available:

- Digital volt ohm meter (VOM) with sharp-tipped probes. Those with the analog bar graph and a beeper with pitch proportional to the display are particularly useful.
- Lineman's test set.
- RJ-45 breakout adapter. This adapter has an RJ-45 socket on each end, with terminals for each of the lines distributed about each side.
- RJ-45 straight-through cable (verify that it is straight through).
- Alligator-clip patch cables.

Precautions

<table>
<thead>
<tr>
<th>Warning</th>
</tr>
</thead>
</table>

Equipment closets where telecommunication devices exist, while usually not hazardous, can have some potentially harmful situations, including, but not limited to:

- **Lead acid battery stacks** able to supply large amounts of current, and possibly flammable hydrogen fumes. Ventilation and insulation are the keys to avoiding damage. Wear long-sleeved shirts, long pants, and steel-toed work boots. Keep electrically insulated work gloves and OSHA-approved eye protection available. Avoid wearing metal objects such as chains, bracelets, rings, and watches unless under cover and away from making any connection. Voltage does not injure; current does.

- **Many wires** for voice, data, power, and so on. Watch for potentially damaging outages caused by pulling a wire that is snagged on another wire. RJ plugs have a tendency to snag on other wires and loosen equipment.

- **Sharp edges**. Equipment deployed before there were safety requirements regarding snag or cut hazards often have protruding bolts and screws. Full clothing protection helps protect you in these cases.
• **Loose, heavy equipment.** Objects in the equipment room may be less than secure. These objects can fall and hurt the equipment, you, or others. If moving heavy objects is involved, leave it to the facility staff or other professional movers; otherwise, use a back protector belt and follow proper OSHA-approved lifting and moving guidelines.

## PBX Interconnection

The majority of PBXs interface with peripheral equipment using cable distribution frames (DFs). Multipair cables are run from the PBX equipment cabinet to the distribution frame, where they are jumpered (cross-connected) to the external devices. These DFs have various names, but the most common terms for them are 110 block, 66 block, and Krone frame. The DF is generally the place where all connections are made between the router voice port and the PBX, so it is where most wiring errors are made and would obviously be the best place to perform testing and troubleshooting.

## Use Rollover Cable for E&M Port-to-Port Testing

Past experience of Cisco Technical Assistance Center (TAC) engineers has shown that most E&M-related faults are due to incorrect wiring or PBX port programming. To assist in determining if the fault is external to the router, you can use the standard rollover console cable that is supplied with every Cisco router as an E&M cross-over cable. This cross-over cable connects the signaling output of one port to the input of the other port and maintains an audio path between the two ports. You can configure a dial peer so that a test call is sent out one port and looped back into the second port, proving the operation of the router.

The rollover console cable has the following RJ-45 connector wiring:

```
1-------8
2-------7
3-------6
4-------5
5-------4
6-------3
7-------2
8-------1
```

The signaling cross-over occurs as pins 2 (M-lead) and 7 (E-lead) on one port are connected to pins 7 (E-lead) and 2 (M-lead) on the other port. The two ports share a common internal ground. The cross-over on pins 4 and 5 (audio pair) has no effect on the audio signal. By setting both voice ports to 2-wire, type 5 operation, the E&M ports become symmetrical and an outward seizure on one port is seen as an incoming seizure on the second port. Any DTMF digits sent out immediately come back in and are then matched on another dial peer. If the test calls are successful, there is little doubt about the operation of the router voice ports. In the following example, the assumption is made that there are working devices on the IP network that can originate and accept VoIP calls.

The voice ports and dial peers are configured like this:

```
voice-port 1/0/0
!--- First port under test.
operation 2-wire
signal-type wink
type 5
!
voice-port 1/0/1
```
Troubleshooting Analog Voice Interfaces to the IP Network

E&M Interfaces

!--- Second port under test.
operation 2-wire
signal-type wink
type 5
Cisco - Analog E&M Troubleshooting Guidelines (Cisco IOS Platforms)
!
dial-peer voice 100 pots
!--- Send call out to port 1/0/0, strip the 100 and prefix with a called
!--- number 200.
destination-pattern 100
port 1/0/0
prefix 200
!
dial-peer voice 200 voip
!--- Incoming test call for 200 comes
!--- in on port 1/0/1 and is sent to 10.1.1.1 as VoIP call.
destination-pattern 200
session-target ipv4:10.1.1.1
!

When a VoIP call comes in to the router with a called number of 100, it is sent out port 1/0/0. By default, any explicitly matched digits on a POTS dial peer are assumed to be an access code and stripped off before the call is made. To route the call correctly, these digits need to be replaced. In this case the prefix command prepends the digits 200 as the called number. This call is immediately looped back in on port 1/0/1. The digits match on dial-peer 200 and make the new call to the designated IP address. The devices originating and accepting the VoIP calls should then have an audio connection that is across the IP network and goes out and back through the E&M ports. This connection proves the router is working properly and indicates that the fault is external to the router. The majority of faults are due to incorrect cabling or PBX port programming issues.

Troubleshooting Type I Interfaces

The four-wire Type I interface from the PBX (set up for the trunk circuit side) has the following characteristics:

- E detector “floats” at –48 V below ground.
- M contact has low ohms to ground on-hook, and is –48 V below ground when off-hook.
- Resistance is approximately 30 to 150 ohms between tip and ring, sometimes in series with 2.2 uF of capacitance.
- Resistance is approximately 30 to 150 ohms between tip-1 and ring-1, sometimes in series with 2.2 uF of capacitance.

Confirm the Cable Interface from the PBX

If you think the cable is bad, pull the suspect voice cable from the router and leave the other side connected to the PBX. Then do the following:

- With a VOM, measure DC voltage between pin 7 of the cable and the chassis ground. The meter should read between –24 V and –56 V. If it does not, pin 7 is likely not the E lead on the PBX.
- Measure the other pins, looking for –24 to –56 V to ground. Some devices, like an AT&T, Lucent or Avaya PBX, bias the tip/ring leads to –48 V to aid debugging. On pins that had no conclusive energy, measure the ohms to ground with a VOM. If one shows less than 500 ohms, it is likely the M lead. It should be pin 2 on the cable. If pin 2 shows between –24 v and –48 V to ground, it is possible that the PBX is off hook; sometimes a PBX busies out what seems to be a bad port.
With a VOM, measure the resistance (ohms) between tip and ring. It should read from 30 to 120 ohms if the PBX has no DC blocking capacitor. If there is a capacitor, you will see the meter jump to around 100 ohms, then climb to infinity as the capacitor charges. With either signature, there is an audio pair—you just need to figure out which direction it is.

Do the same for tip-1/ring-1. It should behave like tip/ring.

Attach a test to tip/ring. While listening, ground E (pin 7 on the cable). If the PBX is configured to provide a dial tone, you should hear it in the earpiece. If you hear nothing, try the other audio pair in case it is cross-wired. If you still hear nothing, the PBX might not give a dial tone on a trunk line.

It is acceptable to cross tip with ring or tip-1 with ring-1.

Additional Troubleshooting Tips

On either the router or the PBX, try a similar port that is known to work.

Listen in on both sides of the audio path (one at a time) with the test set to hear the call progress.

Try to spoof the signaling of one end or the other by clipping one of the active signals to see if the equipment reacts as expected. Grounding E should simulate an inbound call coming over the trunk to the PBX, and the PBX might respond with a dial tone (if provisioned to do so).

Using an extension off of the PBX, try to seize the trunk and see if M connects to ground.

Troubleshooting Type II Interfaces

The four-wire Type-II interface from the PBX (setup for trunk circuit side) has the following characteristics:

- E lead detector “floats” at -48 v below ground.
- SG lead has a low ohms to ground.
- M lead contact between M and SB is open when on-hook and closed when off-hook.
- M lead floats.
- SB lead floats.
- Approximately 30 to 150 ohms between tip and ring, sometimes in series with 2.2 uF of capacitance.
- Approximately 30 to 150 ohms between tip-1 and ring-1, sometimes in series with 2.2 uF of capacitance.

Confirm the Cable Interface from the PBX

Pull the suspect voice cable from the router and leave the other side connected to the PBX. Then do the following:

- With a VOM, measure the DC voltage between E (pin 7 of the cable) and the chassis ground. The meter should read between –24 V and –56 V. If it does not, pin 7 on the cable is likely not the E lead.
- Measure the other pins, looking for –24 to –56 V to ground. Some devices, like an AT&T, Lucent, or Avaya PBX, bias the tip/ring leads to –48 V to aid debugging. On pins that have no conclusive energy, measure the ohms to ground with a VOM. If one shows less than 500 ohms, it is likely the SG lead. It should be pin 8 on the cable.
- With a VOM, measure the resistance (ohms) between tip and ring. It should read from 30 to 120 ohms if the PBX has no DC blocking capacitor. If there is a capacitor, you will see the meter jump to around 100 ohms, then climb to infinity as the capacitor charges. With either signature, there is an audio pair—you just need to figure out which direction it is.
• Do the same for tip-1/ring-1. It should behave like tip/ring.
• Attach a test set to tip/ring. While listening, ground E (pin 7 on the cable). If the PBX is configured to provide a dial tone, you should hear it in the earpiece. If you hear nothing, try the other audio pair in case it is cross-wired. If you still hear nothing, the PBX might not give a dial tone on a trunk line.
• It is acceptable to cross tip with ring or tip-1 with ring-1.
• In most cases, you can get M/SB backwards and E/SG backwards and still have no problems.

Additional Troubleshooting Tips

• On either the router or the PBX, try a similar port that is known to work.
• Listen in on both sides of the audio path (one at a time) with the test set to hear the call progress.
• Try to spoof the signaling of one end or the other by clipping one of the active signals to see if the equipment reacts as expected. Grounding E should simulate an inbound call coming over the trunk to the PBX, and the PBX might respond with a dial tone (if provisioned to do so).
• Using an extension off of the PBX, try to seize the trunk and see if M connects to ground.

Troubleshooting Type III Interfaces

The four-wire Type-III interface from the PBX has the following characteristics:
• E lead detector “floats” at –48 V below ground.
• M lead contact between M and SG when on-hook, and between M and SB when off-hook.
• SG lead floats.
• M lead floats.
• SB lead floats.
• Approximately 30 to 150 ohms between tip and ring, sometimes in series with 2.2 uF of capacitance.
• Approximately 30 to 150 ohms between tip-1 and ring-1, sometimes in series with 2.2 uF of capacitance.

Confirm the Cable Interface from the PBX

Pull the suspect voice cable from the router and leave the other side connected to the PBX. Then do the following:
• With a VOM, measure DC voltage between E (pin 7 of the cable) and the chassis ground. The meter should read somewhere between –24 V and –56 V. If it does not, pin 7 is likely not the E lead.
• Measure the other pins, looking for –24 to –56 V to ground. Some PBXs bias (apply a DC voltage to control the operation of a device) the tip/ring leads to –48 V to aid debugging. On pins that have no conclusive energy:
  – Look for a contact closure (low ohms) between M and SG (if the PBX is on-hook).
  – Look for a contact closure (low ohms) between M and SB (if the PBX is off-hook).
• With a VOM, measure the resistance (ohms) between tip and ring. It should read from 30 to 120 ohms if the PBX has no DC blocking capacitor. If there is a capacitor, you’ll see the meter jump to around 100 ohms, then climb to infinity as the capacitor charges. With either signature, there is an audio pair—you just need to figure out which direction it is.
• Do the same for tip-1/ring-1. It should behave like tip/ring.
• Attach a test set to tip/ring. While listening, ground E (pin 7 on the cable). If the PBX is configured to provide a dial tone, you should hear it in the earpiece. If you hear nothing, try the other audio pair in case it is cross-wired. If you still hear nothing, the PBX might not give a dial tone on a trunk line.
• It is acceptable to cross tip with ring or tip-1 with ring-1.

Additional Troubleshooting Tips

• On either the router or the PBX, try a similar port that is known to work.
• Listen in on both sides of the audio path (one at a time) with the test set to hear the call progress.
• Try to spoof the signaling of one end or the other by clipping one of the active signals to see if the equipment reacts as expected. Grounding E should simulate an inbound call coming over the trunk to the PBX, and the PBX might respond with a dial tone (if provisioned to do so).
• Using an extension off of the PBX, try to seize the trunk and see if M (pin 2 on the cable) connects to SB (pin 1 on the cable).

Troubleshooting Type V Interfaces

The four-wire Type-V interface from the PBX has the following characteristics:
• E lead detector “floats” at –48 V below ground.
• M lead contact ground is open when on-hook, and closed when off-hook.
• Approximately 30 to 150 ohms between tip and ring, sometimes in series with 2.2 uF of capacitance.
• Approximately 30 to 150 ohms between tip-1 and ring-1, sometimes in series with 2.2 uF of capacitance.

Confirm the Cable Interface from the PBX

Pull the suspect voice cable from the router and leave the other side connected to the PBX. Then do the following:
• With a VOM, measure DC voltage between E (pin 7 of the cable) and the chassis ground. The meter should read between –24 V and –56 V. If it does not, pin 7 on the cable is likely not the E lead.
• With a VOM, measure the resistance (ohms) between tip and ring. It should read from 30 to 120 ohms if the PBX has no DC blocking capacitor. If there is a capacitor, you will see the meter jump to around 100 ohms, then climb to infinity as the capacitor charges. With either signature, there is an audio pair—you just need to figure out which direction it is.
• Do the same for tip-1/ring-1. It should behave like tip/ring.
• Attach a test set to tip/ring. While listening, ground E (pin 7 on the cable). If the PBX is configured to provide a dial tone, you should hear it in the earpiece. If you hear nothing, try the other audio pair in case it is cross-wired. If you still hear nothing, the PBX might not give a dial tone on a trunk line.
• It is acceptable to cross tip with ring or tip-1 with ring-1.

Additional Troubleshooting Tips

• On either the router or the PBX, try a similar port that is known to work.
• Listen in on both sides of the audio path (one at a time) with the test set to hear the call progress.
Troubleshooting Analog Voice Interfaces to the IP Network

E&M Interfaces

• Try to spoof the signaling of one end or the other by clipping one of the active signals to see if the equipment reacts as expected. Grounding E should simulate an inbound call coming over the trunk to the PBX, and the PBX might respond with a dial tone (if provisioned to do so).
• Using an extension off of the PBX, try to seize the trunk and see if M (pin 2 on the cable) connects to ground.

Confirming E&M Configuration

The following items should be checked to confirm the E&M configuration:

• Confirming the PBX E&M Configuration Parameters, page 33
• Confirming the Cisco IOS Gateway Configuration, page 33
• Verifying the Wiring Arrangement Between the PBX and the Cisco Gateway, page 34
• Verifying Supervision Signaling, page 35
• Verifying That the Cisco Equipment and PBX Are Sending and Receiving Digits, page 36
• Verifying That the Gateway Sends the Expected Digits to the PBX, page 39
• Verify That the Gateway Receives the Expected Digits from the PBX, page 40

Confirming the PBX E&M Configuration Parameters

The Cisco gateway needs to match the PBX configuration. One of the challenges of configuring and troubleshooting analog E&M circuits are the amount of configuration variables.

• E&M signaling type (I, II, III, V)
• Audio implementation (2-wire / 4-wire)
• Start dial supervision (wink-start, immediate, delay-dial)
• Dial method (DTMF, pulse)
• Call progress tones (standardized within geographic regions)
• PBX port impedance

For information about how specific PBX types interoperate with your gateway, go to the PBX interoperability portal, which is located here:


Note

E&M Type IV is not supported by Cisco gateways. E&M Type V is the most common interface type used outside of North America, but the term Type V is not commonly used outside of North America. From the viewpoint of many PBX operators, there is only one E&M type, what is called Type V in North America.

Confirming the Cisco IOS Gateway Configuration

The Cisco gateway configuration should match the connected PBX configuration. Use the following commands to verify the Cisco IOS platform configuration:

• show running-config—This command displays the running configuration of the router/ gateway.
Troubleshooting Analog Voice Interfaces to the IP Network

Note

The default configuration on E&M voice ports is Type I, wink-start, 2-wire operation, DTMF dialing. Default E&M voice port parameters are not displayed with the `show running-config` command.

- **show voice-port**— For E&M voice ports, this command displays specific configuration data such as E&M voice port, interface type, impedance, dial-supervision signal, audio operation, and dial method. For detailed information see the sample output below.

**Sample Output of show voice port Command**

```
Router# show voice port 1/0/0
receiver And transmitter slot is 1, Sub-unit is 0, Port is 0
Type of VoicePort is E&M
Operation State is DORMANT
Administrative State is UP
The Last Interface Down Failure Cause is Administrative Shutdown
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
Call-Disconnect Time Out is set to 60 s
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
Voice card specific Info Follows:
Signal Type is immediate
Operation Type is 2-wire
E&M Type is 5
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
```

Verifying the Wiring Arrangement Between the PBX and the Cisco Gateway

Physical wiring is often the primary source for analog E&M problems. It is imperative that you verify that the cable/wiring you are using is appropriate for the E&M setup in place. A few things to consider:
• **E&M Type I and Type V** use two leads for supervisory signaling (on/off hook signaling)—E (ear, earth) and M (mouth, magnet). Cisco routers/gateways expect to see off-hook conditions on the M-lead and signal off-hook to the remote device on the E-lead.

• **E&M Type II and Type III** use four leads for supervisory signaling (on/off hook signaling)—E (ear, earth), M (mouth, magnet), SG (signal ground), SB (signal battery). Cisco routers/gateways expect to see off-hook conditions on the M-lead and signal off-hook to a remote device on the E-lead.

• **Audio operation**—The 2-wire/4-wire operation is independent of the signaling type. For example, a 4-wire audio operation E&M circuit has 6 physical wires if configured for Type I or Type V and 8 physical wires if configured for Type II or Type III.

• **Audio path wiring**—In 4-wire audio mode, some PBXs and key systems reverse the normal usage of the tip and ring and tip-1 and ring-1 pairs. To match up the audio pairs with the Cisco E&M audio pairs, connect tip and ring on the PBX side to tip-1 and ring-1 on the Cisco side, and tip-1 and ring-1 on the PBX side to tip and ring on the Cisco side.

See the “Troubleshooting E&M Interfaces at the Physical Level” section on page 26 for more information on the wiring arrangement.

### Verifying Supervision Signaling

In this step, verify that on-hook/off-hook signals are being transmitted between the PBX and the gateway. If you are accessing the router through the console port, enter the command `terminal monitor`, otherwise no debug output is displayed.

Follow these steps to verify supervision signaling:

**SUMMARY STEPS**

1. enable
2. debug vpm signal
3. Place a call from the PBX to the gateway.

**DETAILED STEPS**

Step 1  At the Router> prompt, enter **enable** to enter privileged EXEC mode. Enter your password if prompted.

Step 2  Turn on the command **debug vpm signal** on the Cisco gateway. This command is used to collect debug information for signaling events (on-hook/ off-hook transitions).

Step 3  Place a call from the PBX to the gateway. The PBX should seize the E&M trunk and send the on-hook -> off-hook signal transition to the gateway. The following output displays a successful reception of these signals.

In this example, the PBX is seizing the router trunk. The router E&M voice port transitions from on-hook to off-hook. This shows that on-hook, off-hook signaling is being received from the PBX.

```
Router# debug vpm signal
Voice Port Module signaling debugging is enabled
*Mar 2 05:54:43.996: htsp_process_event: [1/0/0, 1.4 , 34] em_onhook_offhookhtsp_setup_ind
*Mar 2 05:4:44.000: htsp_process_event: [1/0/0, 1.7 , 8]
*Mar 2 05:4:44.784: htsp_process_event: [1/0/0, 1.7 , 10]
*Mar 2 05:4:44.784: htsp_process_event: [1/0/0, 1.2 , 5]
fxs1s_onhook_setuphtsp_alerhtsp_alert_notify
*Mar 2 05:4:44.788: htsp_process_event: [1/0/0, 1.7 , 11]
```
If no output is displayed, there is probably a problem with the E&M supervision signaling.

Table 30 describes some possible problems and the corresponding solutions.

### Table 30: E&M Supervisory Signaling Troubleshooting Table

<table>
<thead>
<tr>
<th>Symptom</th>
<th>Problem</th>
<th>Solution</th>
</tr>
</thead>
<tbody>
<tr>
<td>No dial tone from the Cisco port. No port seizure activity seen on the Cisco gateway.</td>
<td>The PBX is not configured to seize the E&amp;M port connected to the Cisco equipment.</td>
<td>Configure the PBX to seize the trunk.</td>
</tr>
<tr>
<td>The port is seized but the call does not go through.</td>
<td>There is an E&amp;M Type (I, II, III or V) mismatch between the PBX and the gateway.</td>
<td>Verify (and change if necessary) the E&amp;M type configured on the Cisco equipment. See the “Confirming the Cisco IOS Gateway Configuration” section on page 33.</td>
</tr>
<tr>
<td>The port has unbreakable dial tone. The Cisco gateway is unable to send digits when a port is seized.</td>
<td>Incorrect wiring arrangement (cabling) for the supervisory signaling leads (E and M leads for Type I and V; E, M, SB, and SG leads for Types II and III).</td>
<td>Wiring issues are usually the primary source of analog E&amp;M problems. Make sure the cable used corresponds to the required PBX and Cisco gateway pinout, interface type, and audio operation setup. For more information see the “Troubleshooting E&amp;M Interfaces at the Physical Level” section on page 26.</td>
</tr>
<tr>
<td>The port on the Cisco gateway cannot be seized. The Cisco gateway is unable to send digits. Calls cannot be made in two directions.</td>
<td>The Cisco gateway configuration changes are not enabled.</td>
<td>Issue the shutdown/no shutdown command sequence on the E&amp;M voice port after the configuration changes.</td>
</tr>
</tbody>
</table>

### Verifying That the Cisco Equipment and PBX Are Sending and Receiving Digits

After confirming successful supervisory (on-hook/off-hook) signaling between the PBX and the gateway, you need to verify that address information (DTMF digits or pulse dial) is being passed between both ends.

**Note**

DTMF digits are sent on the audio path. Pulse-dial address information is sent by pulsing on the E or M lead.
There are three start dial supervision line protocols that analog E&M uses to define how the equipment passes address information:

- Immediate start
- Wink start
- Delay dial

Make sure both the Cisco gateway and the PBX are configured with the same start dial supervision protocol. Verify that information is being passed by performing the following steps:

**SUMMARY STEPS**

1. **enable**
2. **debug vpm signal, debug vtsp dsp**
3. Place a call from the PBX to the gateway.
4. Place a call from the gateway to the PBX.

**DETAILED STEPS**

**Step 1**
At the Router> prompt, enter **enable** to enable privileged EXEC mode. Enter your password if prompted.

**Step 2**
Turn on the commands **debug vpm signal** and **debug vtsp dsp** on the Cisco gateway. The command **debug vtsp dsp** is useful for displaying the digits received and sent by the voice DSPs.

**Step 3**
Place a call from the PBX to the gateway. The following output displays a successful reception of the expected digits. In this example, the router receives a call from the PBX to extension 2000.

```
Router# show debugging
Voice Port Module signaling debugging is on
Voice Telephony dsp debugging is on
Router# *
```

```
*Mar 1 03:16:19.207: htsp_process_event: [1/0/0, 1.4 , 34]
em_onhook_offhookhtsp_setup.*Mar 1 03:16:19.207: htsp_process_event: [1/0/0, 1.7 , 8]
*Mar 1 03:16:19.339: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT_BEGIN:
digit=2,rtp_=0x9961CF03
*Mar 1 03:16:19.399: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT_OFF:
digit=2,duration=*Mar 1 03:16:19.539: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT_BEGIN:
digit=0,rtp_=0x9961CF03
*Mar 1 03:16:19.599: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT_OFF:
digit=0,duration=*Mar 1 03:16:19.739: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT_BEGIN:
digit=0,rtp_=0x9961CF03
*Mar 1 03:16:19.939: htsp_process_event: [1/0/0, 1.7 , 6]
fxsls_onhook_setuphtsp_alerthtsp.*Mar 1 03:16:20.003: htsp_process_event: [1/0/0, 1.7 , 11]
```

**Step 4**
Place a call from the gateway to the PBX. The following output displays the digits the Cisco equipment is sending. In this example, the PBX receives a call from the router to extension 1000. If digits are not parsed properly, the wink start timers being triggered.
Table 31 shows digit send and receive problems and the corresponding solutions. These problems can be diagnosed if you notice that the wink timers are being triggered.
In the 4-wire audio mode, some PBX and key system products reverse the normal usage of the tip and ring and tip-1 and ring-1 pairs. In that case, to match up the audio pairs with the Cisco E&M audio pairs, you might need to connect tip and ring on the PBX side to tip-1 and ring-1 on the Cisco side, and tip-1 and ring-1 on the PBX side to tip and ring on the Cisco side. If the audio pairs are not correctly matched up in 4-wire mode, there is no end-to-end audio path in either direction. If the E&M interface is configured to send dial strings as dial pulse (which works by pulsing on the E or M lead), it is possible to establish a call even with the 4-wire audio pairs reversed, but there will be little or no audio path in either direction after the call is established (there might be low-level transmission of audio, but the sound levels will be far too low for comfort). If you are using DTMF to send dial strings, the E&M interface goes off hook at the start of the call, but the call does not complete, because one end sends the DTMF tones on the wrong audio pair, and the other end does not receive these DTMF tones.

Verifying That the Gateway Sends the Expected Digits to the PBX

Once the two end devices are able to successfully send supervision and address signaling (on-hook, off-hook, digits), we can assume that the troubleshooting process is complete for analog E&M signaling, and it is now in the dial plan domain. For more information about dial plan design, refer to the Voice Design and Implementation Guide, document ID 5756.

If incomplete or incorrect digits are sent by the Cisco equipment, then the Telco switch (CO or PBX), cannot ring the correct station.

On POTS dial peers, the only digits that are sent to the other end are the ones specified with the command destination-pattern and the wild card character ("."). The POTS dial peer command prefix can be used to include a dial-out prefix that the system enters automatically instead of people dialing it. Refer to the following output example for a sample configuration.

```plaintext
!--- Some output omitted.
!
!--- E&M Voice Port
!
voice-port 1/0/0
type 2
signal immediate
```

Table 31  Digit Send and Receive Troubleshooting Table

<table>
<thead>
<tr>
<th>Problem</th>
<th>Solution</th>
</tr>
</thead>
<tbody>
<tr>
<td>Start dial supervision mismatch or timing issues between the PBX and gateway.</td>
<td>Make sure both end systems are configured with the same start dial protocol.</td>
</tr>
<tr>
<td>Audio operation mismatch (for example, one side configured for 2-wire, the other for 4-wire) or wiring problems on the audio path.</td>
<td>Verify the gateway configuration and PBX configuration and the wiring arrangement. For more information see the “Troubleshooting E&amp;M Interfaces at the Physical Level” section on page 26.</td>
</tr>
<tr>
<td>Note DTMF digits are passed on the audio path. Even if the line supervision signaling is operating correctly, DTMF digits are not passed if the audio path is broken.</td>
<td></td>
</tr>
<tr>
<td>Wiring problems in the audio path.</td>
<td>Verify the wiring arrangement. See the “Troubleshooting E&amp;M Interfaces at the Physical Level” section on page 26.</td>
</tr>
</tbody>
</table>
Troubleshooting Analog Voice Interfaces to the IP Network

E&M Interfaces

--- FXS Voice Port
voice-port 1/1/0

--- Dial peer 1 is in charge of forwarding calls to the E&M voiceport 1/0/0.
--- In this case the digit "1" in the destination pattern will be dropped and the system will transmit the 3 digits matched by the "." wildcard.
--- Notice that since the PBX is expecting the "1000" string, the prefix command is used.

--- FXS Voice Port

Verify That the Gateway Receives the Expected Digits from the PBX

Verify that the digits received from the PBX match a dial peer in the gateway. If incomplete or incorrect digits are sent by the PBX, a dial peer cannot be matched. Use the command `debug vtsp dsp` to view the digits received by the analog E&M voice port.

To verify which dial peers match a specific string use the command `show dialplan number`. Refer to the following sample output example.

```plaintext
Router# show dialplan number 1000
Macro Exp.: 1000
VoiceEncapPeer2
information type = voice,
tag = 2, destination-pattern = "1...",
answer-address = ", preference=0,
group = 2. Admin state is up, Operation state is up,
incoming called-number = ", connections/maximum = 0/unlimited,
application associated:
type = pots, prefix = "1",
session-target = ", voice-port = "1/0/0",
direct-inward-dial = disabled,
register E.164 number with GK = TRUE
Connect Time = 19644, Charged Units = 0,
Successful Calls = 63, Failed Calls = 2,
Accepted Calls = 65, Refused Calls = 0,
Last Disconnect Cause is ",
Last Disconnect Text is "normal call clearing.",
Last Setup Time = 28424467.
Matched: 1000 Digits: 1
Target:
```

Router# show dialplan number 2000
Macro Exp.: 2000
VoiceEncapPeer1
information type = voice,
tag = 1, destination-pattern = "2000",
answer-address = ", preference=0,
group = 1. Admin state is up, Operation state is up,
incoming called-number = ", connections/maximum = 0/unlimited,
application associated:
type = pots, prefix =",
session-target = ", voice-port = "1/1/1",
direct-inward-dial = disabled,
```
register E.164 number with GK = TRUE
Connect Time = 19357, Charged Units = 0,
Successful Calls = 68, Failed Calls = 8,
Accepted Calls = 76, Refused Calls = 0,
Last Disconnect Cause is "10 ",
Last Disconnect Text is "normal call clearing.",
Last Setup Time = 28424186.
Matched: 2000 Digits: 4
Target:

Unbreakable Dial Tone

A common problem occurs when the router seizes the local PBX but as digits are dialed, the dial tone stays. The calling party is unable to pass the DTMF tones or digits to the terminating device, resulting in callers being unable to dial the desired extension or interact with the device that needs DTMF tones, such as a voice mail or interactive voice response (IVR) application. This problem can result from a number of reasons such as:

- DTMF tones not sent
- DTMF tones not understood
- DTMF tones too distorted to be understood
- Other signaling and cabling issues

For more information, refer to *Inability To Break Dialtone in a Voice over IP Network*, document ID 22376.

Analog DID Interfaces

Direct inward dialing (DID) is a service offered by telephone companies that enables callers to dial directly to an extension on a PBX without the assistance of an operator or automated call attendant. This service makes use of DID trunks, which forward only the last three to five digits of a phone number to the PBX. The DID state machine is identical to the E&M state machine.

*Figure 29* shows a hypothetical topology in which a user connected to the PSTN (User A) dials various numbers and is connected to the appropriate extensions on a PBX.

*Figure 29*  DID Support for Cisco 2600 and Cisco 3600 Series Routers
DID Hardware Troubleshooting

A DID voice interface connects directly to a standard telephone, fax machine, or similar device and supplies ring, voltage, and dial tone.

Troubleshoot DID hardware by checking the following sections:

- Software Compatibility, page 42
- Cabling, page 42
- Shutdown Port, page 43

Software Compatibility

For interface cards inserted into Cisco 1600 series, Cisco 1700 series, Cisco 2600 series, Cisco 3600 series, Cisco 3700 series, and Cisco ICS 7750 platforms, refer to the compatibility tables in the “Overview of Cisco Interface Cards” chapter in the Cisco Interface Cards Installation Guide.

Cabling

The two-port and four-port DID interface cards support the RJ-11 connector. Illustrations of the connector ports are shown in Figure 30 and Figure 31. Information about LEDs can be found in the “Connecting Voice Interface Cards to a Network” chapter of the Cisco Interface Card Hardware Installation Guide.

<table>
<thead>
<tr>
<th>Number Dialed by User A</th>
<th>Number Received by Router</th>
<th>Extension Receiving Call</th>
</tr>
</thead>
<tbody>
<tr>
<td>555-1234</td>
<td>234</td>
<td>User C</td>
</tr>
<tr>
<td>555-1345</td>
<td>345</td>
<td>User D</td>
</tr>
<tr>
<td>555-1456</td>
<td>456</td>
<td>User B</td>
</tr>
<tr>
<td>555-1678</td>
<td>678</td>
<td>No dial-peer match found; fast busy tone is played</td>
</tr>
</tbody>
</table>
ShUTDOWN PORT

Check to make sure that the port is not shut down. Enter the `show voice port` command with the voice port number that you are troubleshooting, which will tell you:

- If the voice port is up. If it is not, use the `no shutdown` command to make it active.
- What parameter values have been set for the voice port, including default values (these do not appear in the output from `show running-config` command). If these values do not match those of the telephony connection you are making, reconfigure the voice port.

Verifying Direct Inward Dialing Voice-Port Configuration

To verify voice-port configuration, enter the `show voice port` command. You can specify a voice port or view the status of all configured voice ports. In the following example, the specified port is configured for DID.

```
Router# show voice port 1/1/0
Foreign Exchange Station with Direct Inward Dialing (FXS-DID) 1/1/0 Slot is 1, Sub-unit is 1, Port is 0
   Type of VoicePort is DID-IN
   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
   Playout-delay Minimum mode is set to default, value 4 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 3 s
   Ringing Time Out is set to 180 s
   Wait Release Time Out is set to 3 s
   Compressing 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 Chalil Mohanan, Station number 1234567

Voice card specific Info Follows:
Signal Type is wink-start
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 750 ms
Clear Wait Duration Timing is set to 400 ms
Wink Wait Duration Timing is set to 200 ms
Wait Wink Duration Timing is set to 550 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
Percent Break of Pulse is 60 percent
Auto Cut-through is disabled
Dialout Delay for immediate start is 300 ms

Voice Port Testing Commands

Voice port testing commands allow you to force voice ports into specific states for testing. The following types of voice-port tests are covered:

- **Detector-Related Function Tests**, page 44
- **Loopback Function Tests**, page 45
- **Tone Injection Tests**, page 46
- **Relay-Related Function Tests**, page 46
- **Fax/Voice Mode Tests**, page 47

Detector-Related Function Tests

Using the `test voice port detector` command, you are able to force a particular detector into an on or off state, perform tests on the detector, and then return the detector to its original state.
To configure this feature, enter these commands beginning in privileged EXEC mode:

<table>
<thead>
<tr>
<th>Command</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 1** Router# test voice port slot/subunit/port detector {m-lead | battery-reversal | loop-current | ring | tip-ground | ring-ground | ring-trip} {on | off} | Identifies the voice port you want to test.  
- Enter a keyword for the detector under test and specify whether to force it to the on or off state.  
**Note** For each signaling type (E&M, FXO, FXS), only the applicable keywords are displayed. The **disable** keyword is displayed only when a detector is in the forced state. |
| **Step 2** Router# test voice port slot/subunit/port detector {m-lead | battery-reversal | loop-current | ring | tip-ground | ring-ground | ring-trip} disable | Identifies the voice port on which you want to end the test.  
- Enter a keyword for the detector under test and the keyword **disable** to end the forced state.  
**Note** For each signaling type (E&M, FXO, FXS), only the applicable keywords are displayed. The **disable** keyword is displayed only when a detector is in the forced state. |

### Loopback Function Tests

To establish loopbacks on a voice port, enter the following commands beginning in privileged EXEC mode:

<table>
<thead>
<tr>
<th>Command</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 1** Router# test voice port slot/subunit/port loopback {local | network} | Identifies the voice port you want to test and enters a keyword for the loopback direction.  
**Note** A call must be established on the voice port under test. |
| **Step 2** Router# test voice port slot/subunit/port loopback disable | Identifies the voice port on which you want to end the test and enters the keyword disable to end the loopback. |
## Tone Injection Tests

To inject a test tone into a voice port, enter the following commands beginning in privileged EXEC mode:

<table>
<thead>
<tr>
<th>Command</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 1** | **Router# test voice port slot/subunit/port inject-tone** *(local | network) {1000hz | 2000hz | 200hz | 3000hz | 300hz | 3200hz | 3400hz | 500hz | quiet}** | Identifies the voice port you want to test and enter keywords for the direction to send the test tone and for the frequency of the test tone.  
**Note** A call must be established on the voice port under test. |
| **Step 2** | **Router# test voice port slot/subunit/port inject-tone disable** | Identifies the voice port on which you want to end the test and enter the keyword **disable** to end the test tone.  
**Note** The **disable** keyword is available only if a test condition is already activated. |

## Relay-Related Function Tests

To test relay-related functions on a voice port, enter the following commands beginning in privileged EXEC mode:

<table>
<thead>
<tr>
<th>Command</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 1** | **Router# test voice port slot/subunit/port relay** *(e-lead | loop | ring-ground | battery-reversal | power-denial | ring | tip-ground) (on | off)* | Identifies the voice port you want to test.  
• Enter a keyword for the relay under test and specify whether to force it to the on or off state.  
**Note** For each signaling type (E&M, FXO, FXS), only the applicable keywords are displayed. The **disable** keyword is displayed only when a relay is in the forced state. |
| **Step 2** | **Router# test voice port slot/subunit/port relay** *(e-lead | loop | ring-ground | battery-reversal | power-denial | ring | tip-ground) disable** | Identifies the voice port on which you want to end the test.  
• Enter a keyword for the relay under test, and the keyword **disable** to end the forced state.  
**Note** For each signaling type (E&M, FXO, FXS), only the applicable keywords are displayed. The **disable** keyword is displayed only when a relay is in the forced state. |
Fax/Voice Mode Tests

The test voice port switch fax command forces a voice port into fax mode for testing. After you enter this command, you can use the show voice call or show voice call summary command to check whether the voice port is able to operate in fax mode. If no fax data is detected by the voice port, the voice port remains in fax mode for 30 seconds and then reverts automatically to voice mode.

The disable keyword ends the forced mode switch; however, the fax mode ends automatically after 30 seconds. The disable keyword is available only while the voice port is in fax mode.

To force a voice port into fax mode and return it to voice mode, enter the following commands, beginning in privileged EXEC mode:

<table>
<thead>
<tr>
<th>Command</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td></td>
</tr>
<tr>
<td>Router# test voice port slot/subunit/port switch fax</td>
<td>Identifies the voice port you want to test.</td>
</tr>
<tr>
<td></td>
<td>• Enter the keyword fax to force the voice port into fax mode.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td></td>
</tr>
<tr>
<td>Router# test voice port slot/subunit/port switch disable</td>
<td>Identifies the voice port on which you want to end the test.</td>
</tr>
<tr>
<td></td>
<td>• Enter the keyword disable to return the voice port to voice mode.</td>
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</tbody>
</table>