Contents

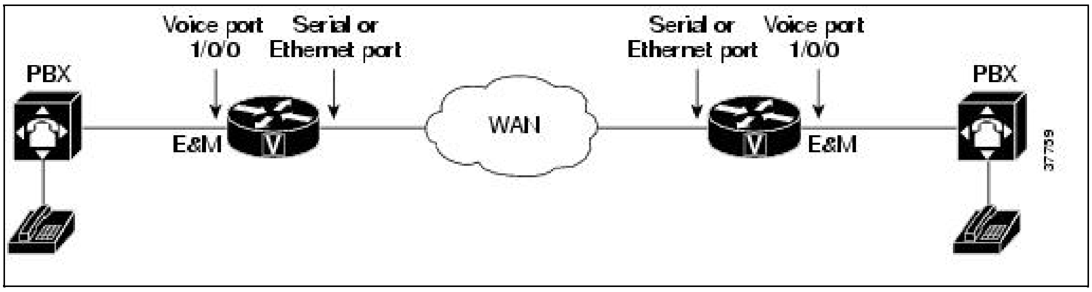
E&M Overview

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: E&M Signaling Interfaces, 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: E&M Signaling Interfaces



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
- Cabling
- Shutdown Port

Software Compatibility

For interface cards inserted into Cisco modular access routers, refer to Voice Interface Cards to check the software compatibility for your voice interface card.

Cabling

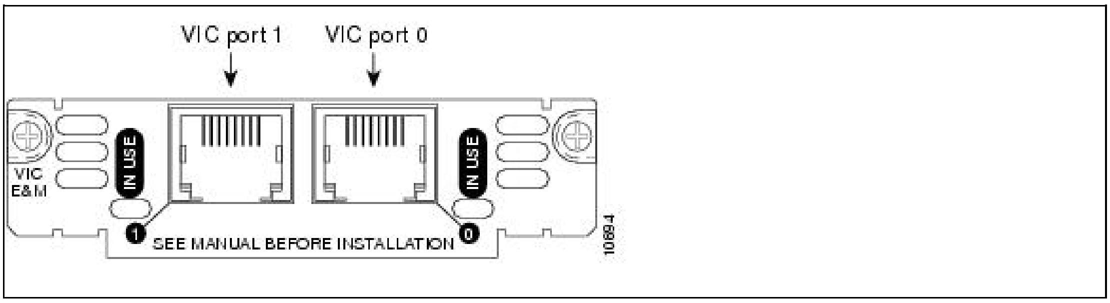
E&M is a signaling technique for two-wire and four-wire telephone and trunk interfaces. The E&M interface typically connects remote calls from an IP network to a PBX. The card is connected to the PSTN or PBX through a telephone wall outlet by a straight-through RJ-48C cable.

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: Two-Port E&M Card Front Panel. Information about LEDs can be found in the Voice Interface Cards document.

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

Figure: Two-Port E&M Card Front Panel



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

```
Router# show version
Cisco Internetwork Operating System Software
IOS (tm) 3600 Software (C3640-IS-M), 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

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: E&M Interface Supervision Signal Description shows the E&M supervisory signal description.

Table: E&M Interface Supervision Signal Description

Signal	Meaning	Description
E	Ear or earth	Signal wire from trunking (CO) side to signaling side.
M	Mouth or magnet	Signal wire from signaling side to trunking (CO) side.
SG	Signal ground	Used on E&M Types II, III, and IV. (Type IV is not supported on Cisco gateways.)
SB	Signal battery	Used on E&M Types II, III, and IV. (Type IV is not supported on Cisco gateways.)
T/R	Tip/Ring	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.
T1/R1	Tip-1/Ring-1	Used on four-wire audio operation circuits only. The four-wire implementation provides separate paths for receiving and sending audio signals.

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: E&M Type I Signal States shows the sent signal states for on- and off-hook signaling.

Table: E&M Type I Signal States

PBX to Cisco Gateway			Cisco Gateway to PBX		
Lead	**On-Hook**	**Off-Hook**	**Lead**	**On-Hook**	**Off-Hook**
M	Ground	Battery	E	Open	Ground

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: E&M Type I 2-Wire Audio Operation. E&M Type I 4-wire operation is shown in Figure: E&M Type I 4-Wire Audio Operation.

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

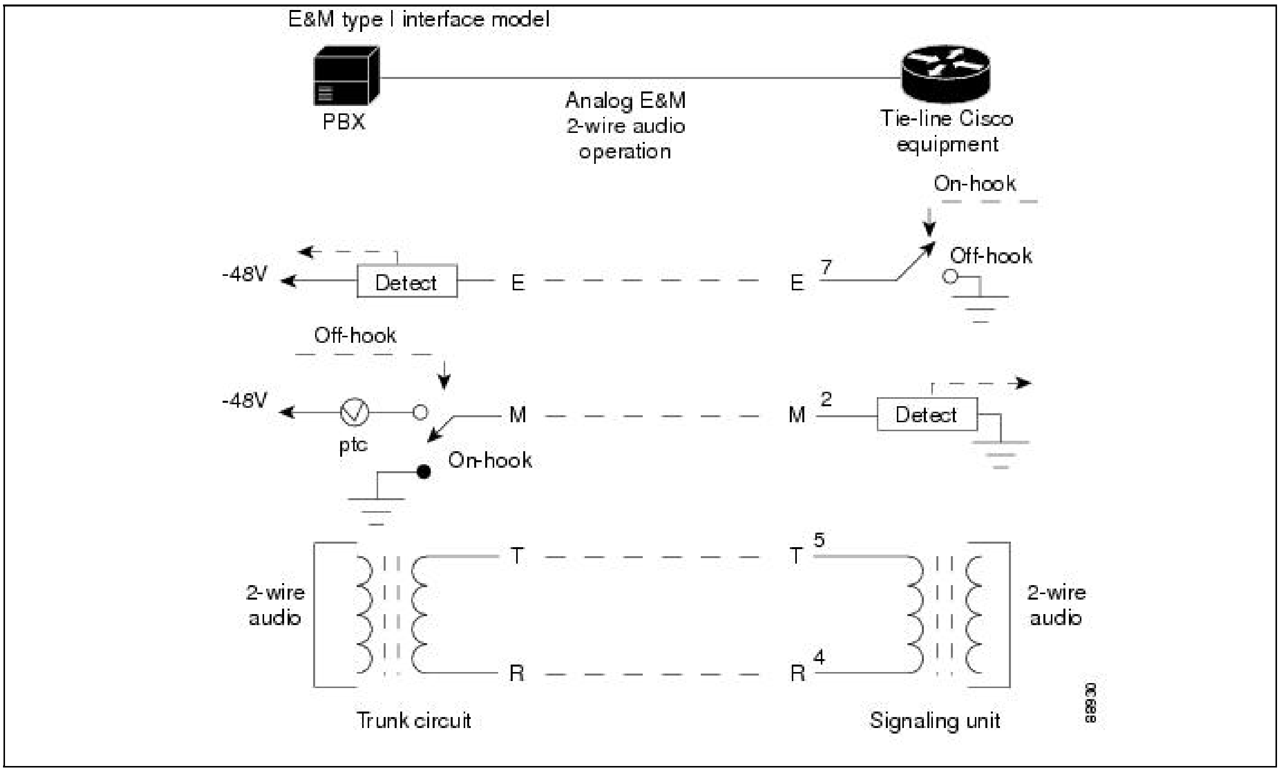
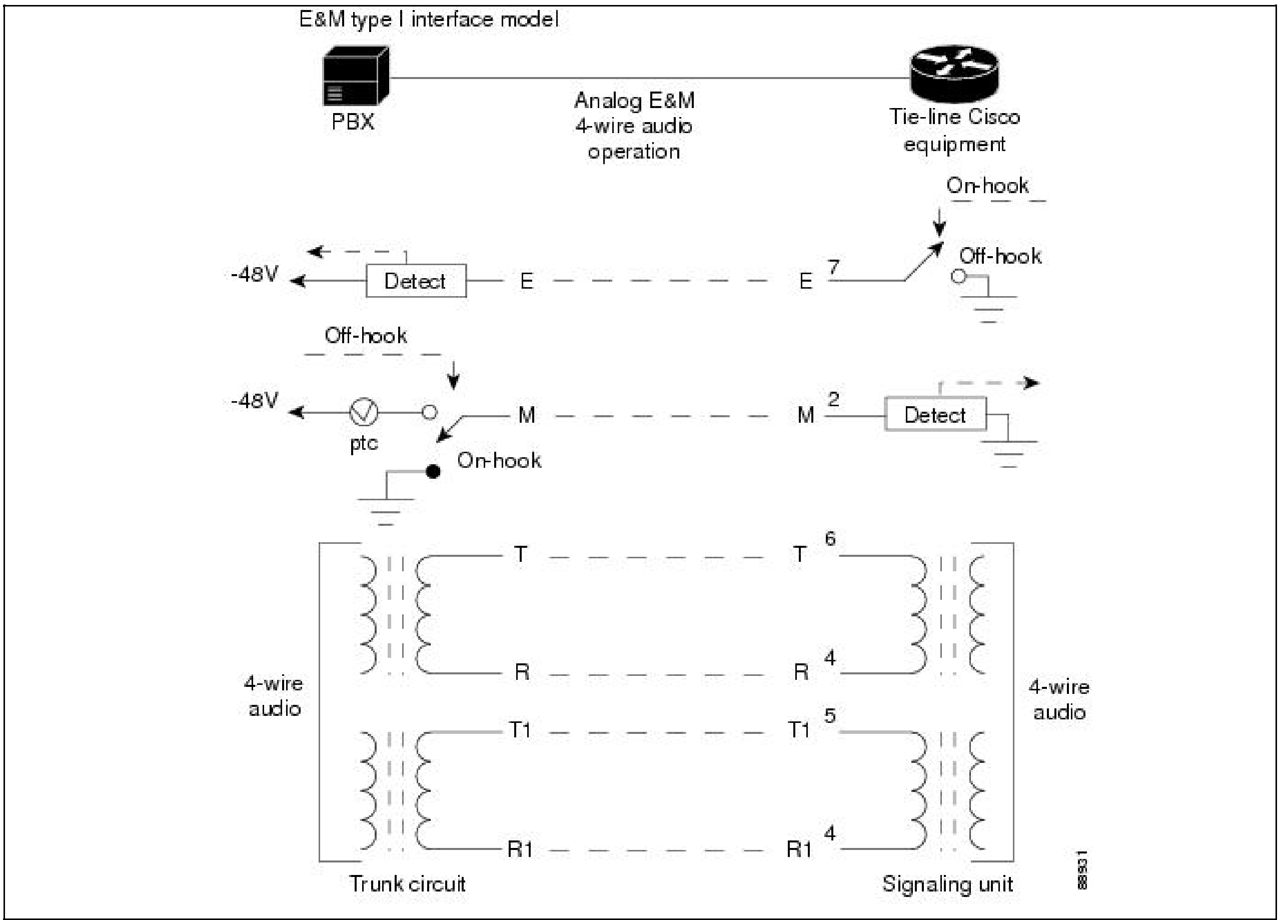
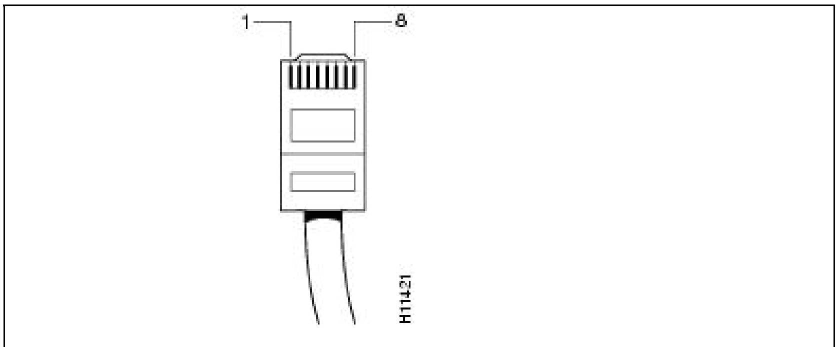


Figure: E&M Type I 4-Wire Audio Operation



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 4 (Ring1) on the router transport the audio path from the router to the PBX. Pins for the cable are shown in Figure: E&M Cabling Pins.

Figure: Figure 19 E&M Cabling Pins



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: E&M Type II Signal States shows the sent signal states for on- and off-hook signaling.

Table: E&M Type II Signal States

PBX to Cisco Gateway			Cisco Gateway to PBX		
Lead	**On-Hook**	**Off-Hook**	**Lead**	**On-Hook**	**Off-Hook**
M	Open	Battery	E	Open	Ground

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: E&M Type II 2-Wire Audio Operation. E&M Type II 4-wire operation is shown in Figure: E&M Type II 4-Wire Audio Operation.

Figure: E&M Type II 2-Wire Audio Operation

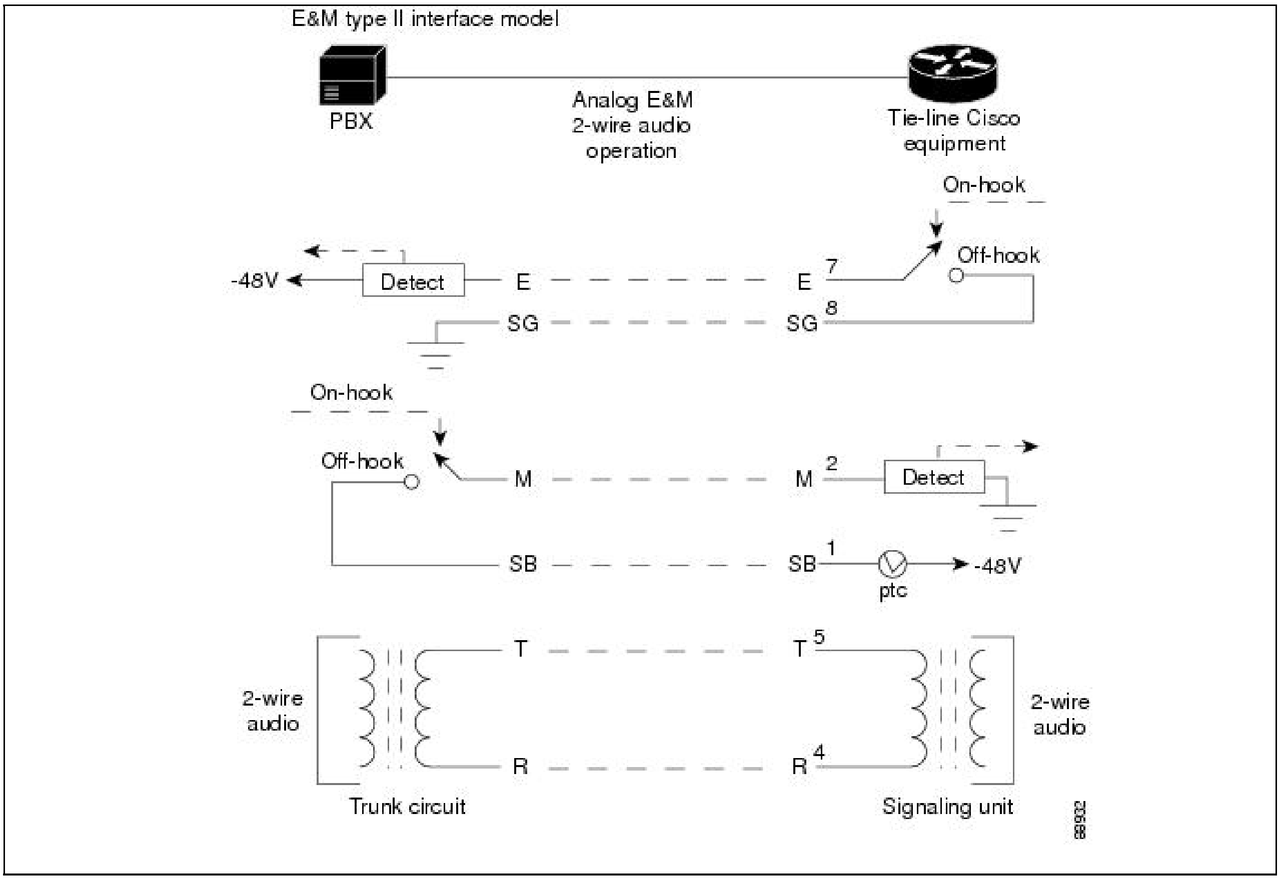
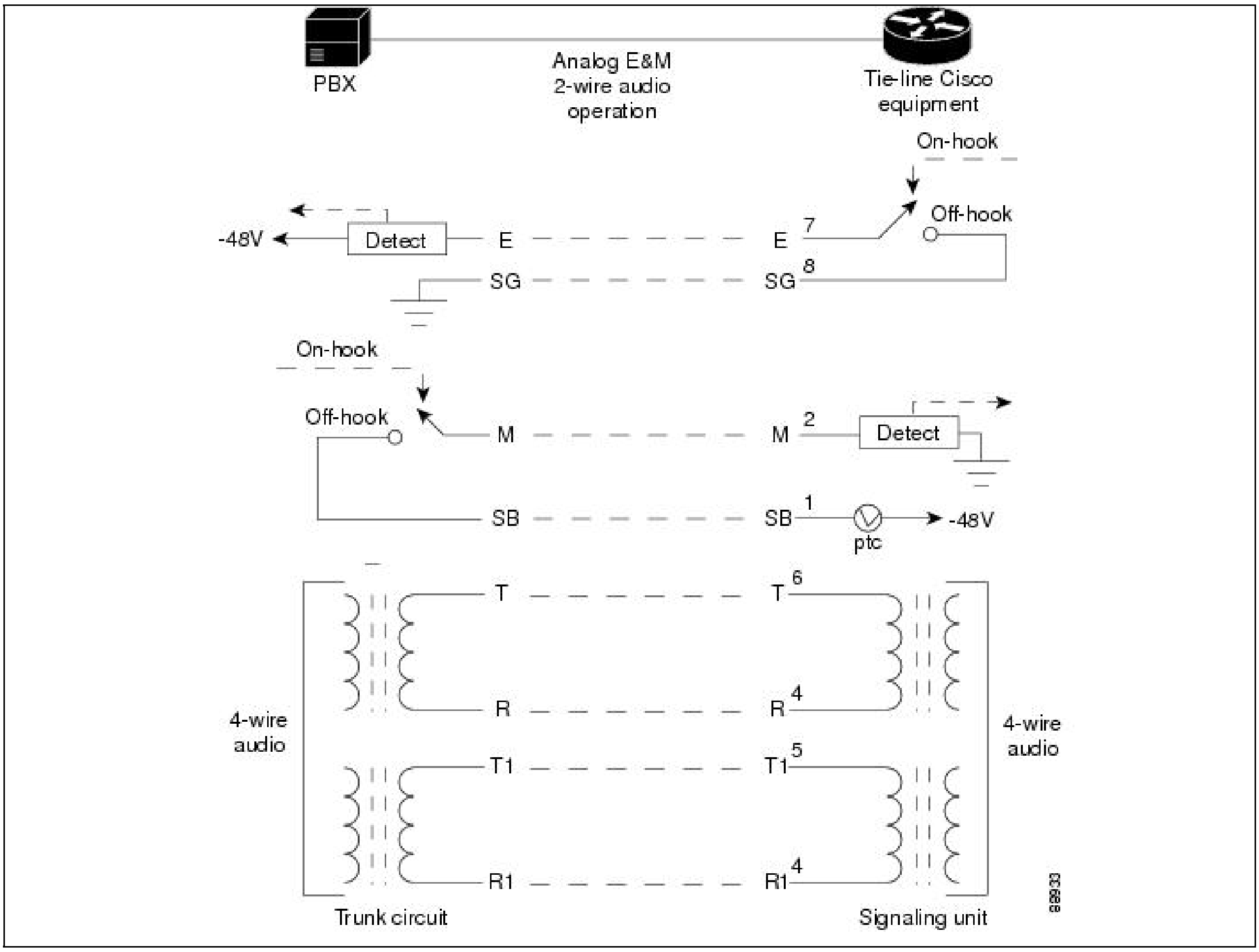


Figure: E&M Type II 4-Wire Audio Operation



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 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: E&M Type III Signal States shows the sent signal states for on- and off-hook signaling.

Table: E&M Type III Signal States

PBX to Cisco Gateway			Cisco Gateway to PBX		
Lead	**On-Hook**	**Off-Hook**	**Lead**	**On-Hook**	**Off-Hook**
M	Ground	Battery	E	Open	Ground

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: E&M Type III 2-Wire

Audio Operation. E&M Type III 4-wire operation is shown in Figure: E&M Type III 4-Wire Audio Operation.

Figure: E&M Type III 2-Wire Audio Operation

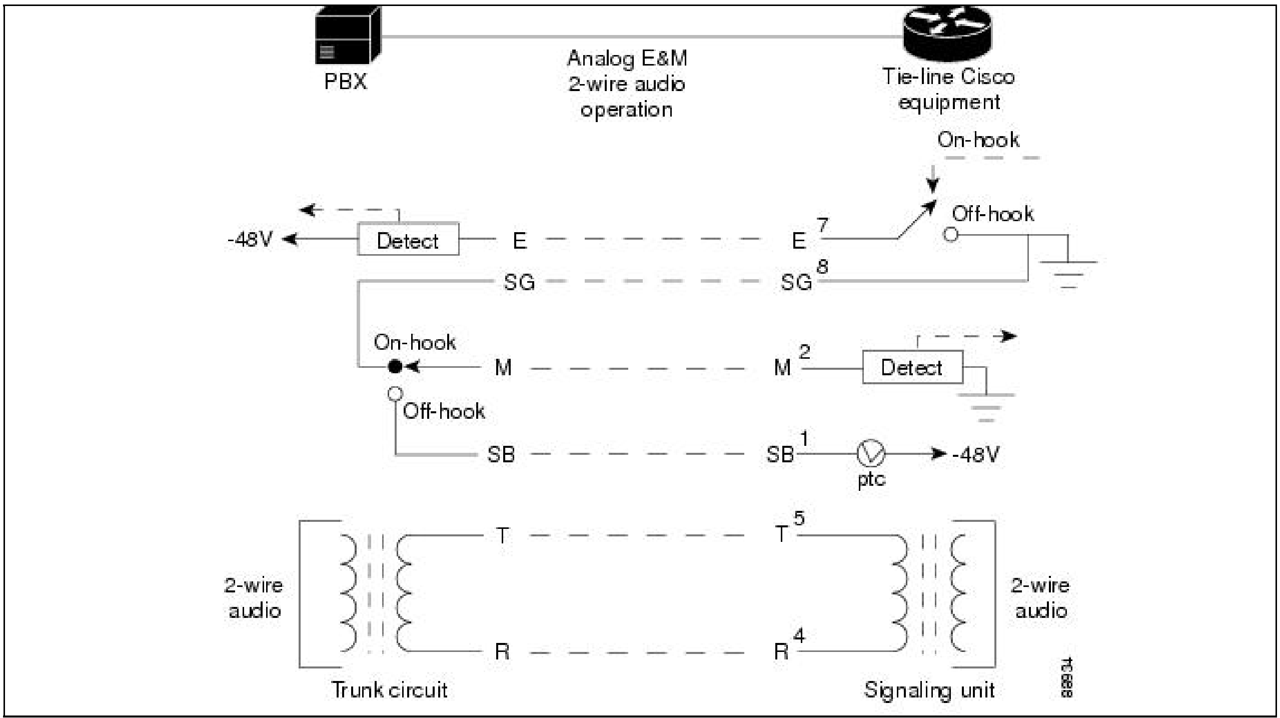


Figure: E&M Type III 4-Wire Audio Operation

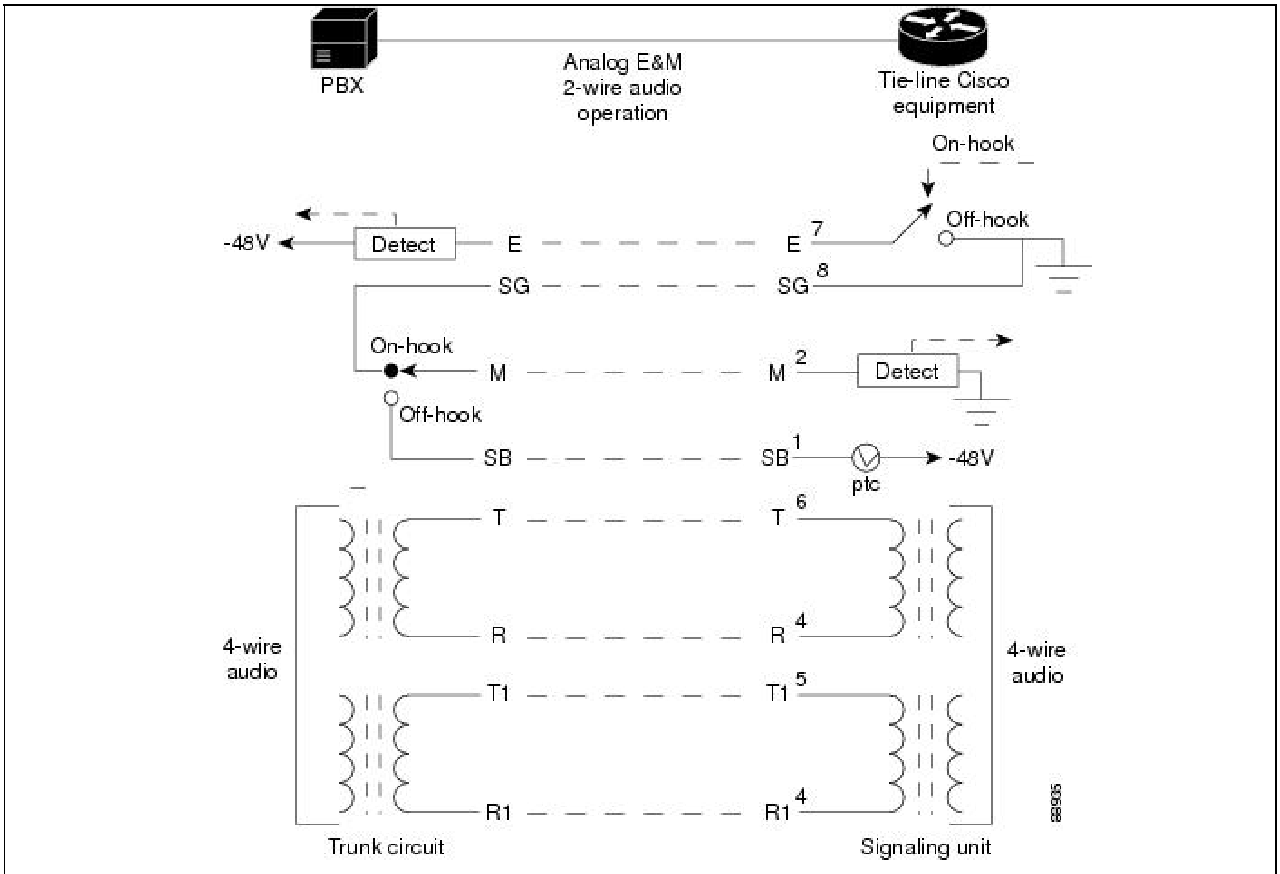


Table: E&M Type III Signal States

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 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: E&M Type V Signal States shows the sent signal states for on- and off-hook signaling.

Table: E&M Type V Signal States

PBX to Cisco Gateway			Cisco Gateway to PBX		
Lead	**On-Hook**	**Off-Hook**	**Lead**	**On-Hook**	**Off-Hook**
M	Open	Ground	E	Open	Ground

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: E&M Type V 2-Wire Audio Operation. E&M Type V 4-wire operation is shown in Figure: E&M Type V 4-Wire Audio Operation.

Figure: E&M Type V 2-Wire Audio Operation

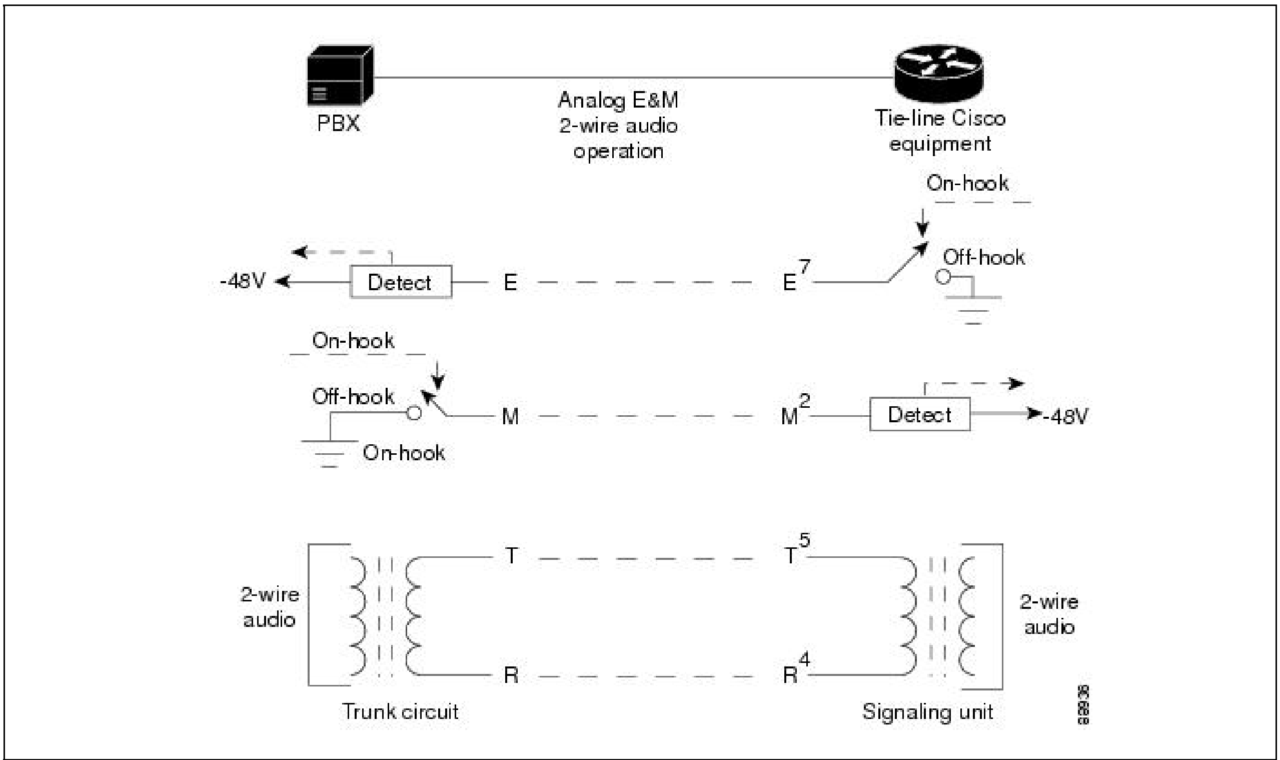
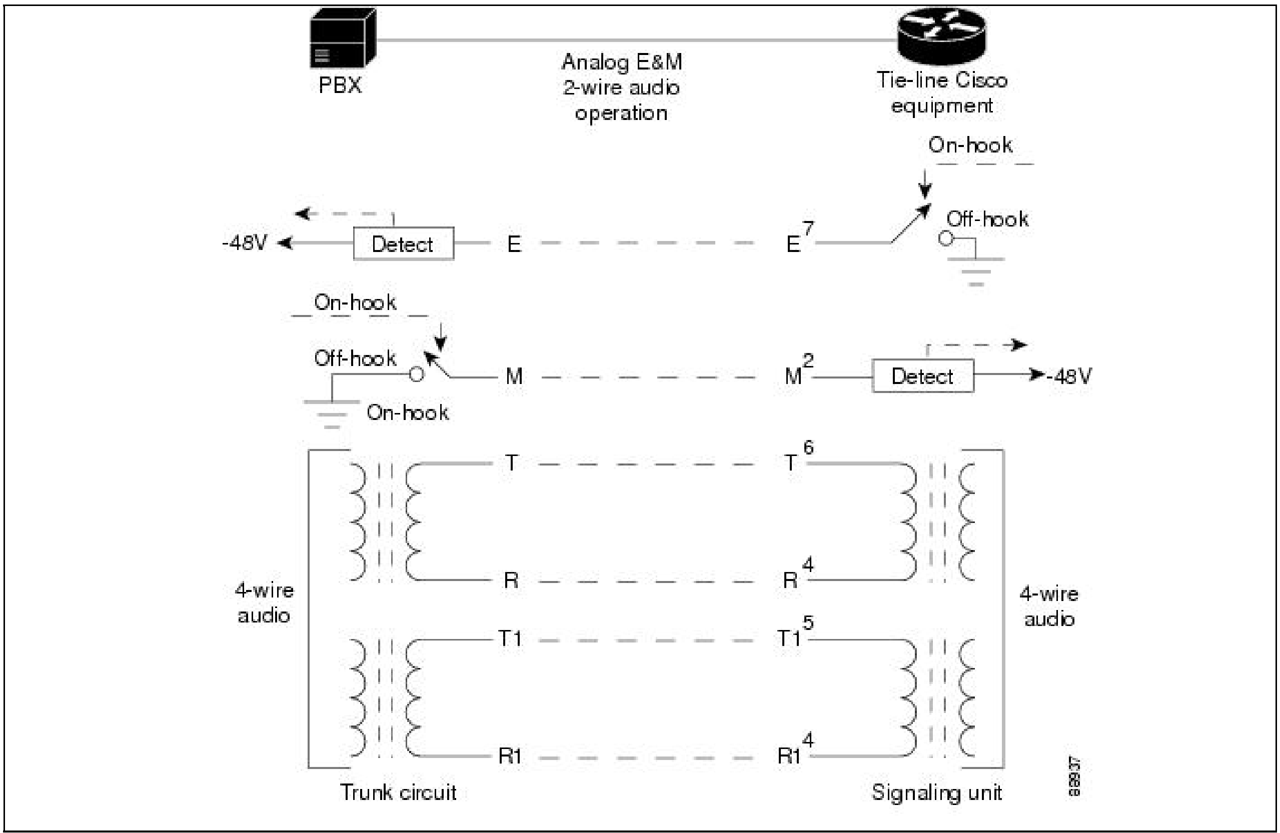


Figure: E&M Type V 4-Wire Audio Operation



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
- Precautions
- PBX Interconnection
- Use Rollover Cable for E&M Port-to-Port Testing

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

Warning! 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
!--- 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.
- 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
- Confirming the Cisco IOS Gateway Configuration
- Verifying the Wiring Arrangement Between the PBX and the Cisco Gateway
- Verifying Supervision Signaling
- Verifying That the Cisco Equipment and PBX Are Sending and Receiving Digits
- Verifyinng That the Gateway Sends the Expected Digits to the PBX
- Verify That the Gateway Receives the Expected Digits from the PBX

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 Cisco Interoperability Portal

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.

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
recEive And transMit 1/0/0 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 PBX es 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 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

1. At the Router> prompt, enter **enable** to enter privileged EXEC mode. Enter your password if prompted.
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).
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 05:54:44.000: htsp_process_event: [1/0/0, 1.7 , 8]
*Mar 2 05:54:44.784: htsp_process_event: [1/0/0, 1.7 , 10]
*Mar 2 05:54:44.784: htsp_process_event: [1/1/0, 1.2 , 5]
fxsls_onhook_setuphtsp_alerthtsp_alert_notify
*Mar 2 05:54:44.788: htsp_process_event: [1/0/0, 1.7 , 11]
*Mar 2 05:54:44.788: htsp_process_event: [1/1/0, 1.5 , 11]
fxsls_waitoff_voice
```

If no output is displayed, there is probably a problem with the E&M supervision signaling.

Table: E&M Supervisory Signaling Troubleshooting Table describes some possible problems and the corresponding solutions.

Table: E&M Supervisory Signaling Troubleshooting Table

Symptom	Problem	Solution
No dial tone from the Cisco port. No port seizure activity seen on the Cisco gateway.	The PBX is not configured to seize the E&M port connected to the Cisco equipment.	Configure the PBX to seize the trunk.
The port is seized but the call does not go through.	There is an E&M Type (I, II, III or V) mismatch between the PBX and the gateway.	Verify (and change if necessary) the E&M type configured on the Cisco equipment. See the Confirming the Cisco IOS Gateway Configuration.
The port has unbreakable dial tone. The Cisco gateway is unable to send	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).	Wiring issues are usually the primary source of analog E&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&M Interfaces at the Physical Level.

digits when a port is seized.		
The port on the Cisco gateway cannot be seized. The Cisco gateway is unable to send digits.	The Cisco gateway configuration changes are not enabled.	Issue the **shutdown/no shutdown** command sequence on the E&M voice port after the configuration changes.
Calls cannot be made in two directions.	On-hook or off-hook signals have been sent one way only.	This is probably an indication of a defective cable, where one path of the signaling leads is wired correctly and the other side is not.

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

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

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.

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 03:16:19.799: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT_OFF: digit=0,duration=
*Mar 1 03:16:19.939: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT_BEGIN: digit=0,=rtp_=0x9961CF03
*Mar 1 03:16:19.999: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT_OFF: digit=0,duration=
*Mar 1 03:16:19.999: htsp_process_event: [1/0/0, 1.7 , 10]
*Mar 1 03:16:19.999: htsp_process_event: [1/1/0, 1.2 , 5] fxsls_onhook_setuphtsp_alerthtsp_
*Mar 1 03:16:20.003: htsp_process_event: [1/0/0, 1.7 , 11]
*Mar 1 03:16:20.003: htsp_process_event: [1/1/0, 1.5 , 11] fxsls_waitoff_voice
*Mar 1 03:16:27.527: htsp_process_event: [1/1/0, 1.5 , 34] fxsls_waitoff_offhook
*Mar 1 03:16:27.531: htsp_process_event: [1/0/0, 1.7 , 6] em_offhook_connectem_stop_
```

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.

```
Log Buffer (1000000 bytes):
*Mar 1 03:45:31.287: htsp_process_event: [1/1/1, 1.2 , 34] fxsls_onhook_offhook
htsp_setup_ind
*Mar 1 03:45:31.291: htsp_process_event: [1/1/1, 1.3 , 8]
*Mar 1 03:45:33.123: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT_BEGIN: digit=1
, rtp_timestamp=0xCD4365D8
*Mar 1 03:45:33.283: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT_OFF: digit=1,
duration=205
*Mar 1 03:45:33.463: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT_BEGIN: digit=0
, rtp_timestamp=0xCD4365D8
*Mar 1 03:45:33.643: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT_OFF: digit=0,
duration=225
*Mar 1 03:45:33.823: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT_BEGIN: digit=0
, rtp_timestamp=0xCD4365F0
*Mar 1 03:45:34.003: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT_OFF: digit=0,
duration=222
*Mar 1 03:45:34.203: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT_BEGIN: digit=0
, rtp_timestamp=0xCD4365F0
*Mar 1 03:45:34.411: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT_OFF: digit=0,
duration=252
*Mar 1 03:45:34.415: htsp_process_event: [1/1/1, 1.3 , 10]
*Mar 1 03:45:34.415: htsp_process_event: [1/0/0, 1.4 , 5] em_onhook_setup em_of
fhook
*Mar 1 03:45:34.415: htsp_process_event: [1/0/0, 1.13 , 43] em_start_timer: 120
0 ms
*Mar 1 03:45:34.715: htsp_process_event: [1/0/0, 1.10 , 34] em_wink_offhookem_s
top_timers em_start_timer: 1200 ms
*Mar 1 03:45:34.923: htsp_process_event: [1/0/0, 1.11 , 22] em_wink_onhook em_s
top_timers em_send_digit htsp_dial
*Mar 1 03:45:34.923: digit=1, components=2, freq_of_first=697, freq_of_second
=1209, amp_of_first=16384, amp_of_second=16384
*Mar 1 03:45:34.923: digit=0, components=2, freq_of_first=941, freq_of_second
=1336, amp_of_first=16384, amp_of_second=16384
*Mar 1 03:45:34.923: digit=0, components=2, freq_of_first=941, freq_of_second
=1336, amp_of_first=16384, amp_of_second=16384
*Mar 1 03:45:34.923: digit=0, components=2, freq_of_first=941, freq_of_second
3.
=1336, amp_of_first=16384, amp_of_second=16384
```

```
*Mar 1 03:45:35.727: vtsp_process_dsp_message: MSG_TX_DIALING_DONE
*Mar 1 03:45:35.727: htsp_process_event: [1/0/0, 1.7 , 19] em_offhook_digit_don
ehtsp_alerthtsp_alert_notify
```

Table: Digit Send and Receive Troubleshooting Table 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.

Table: Digit Send and Receive Troubleshooting Table

Problem	Solution
Start dial supervision mismatch or timing issues between the PBX and gateway.	Make sure both end systems are configured with the same start dial protocol.
Audio operation mismatch (for example, one side configured for 2-wire, the other for 4-wire) or wiring problems on the audio path.	Verify the gateway configuration and PBX configuration and the wiring arrangement. For more information see the Troubleshooting E&M Interfaces at the Physical Level. **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.
Wiring problems in the audio path.	Verify the wiring arrangement. See the Troubleshooting E&M Interfaces at the Physical Level.

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.

```
!
!--- Some output ommited.
!
```

```
!--- E&M Voice Port
!
voice-port 1/0/0
type 2
signal immediate
!
!--- FXS Voice Port
voice-port 1/1/0
!
dial-peer voice 1 pots
destination-pattern 2000
port 1/1/0
!
!--- Dial peer 2 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 syste
!--- will transmit the 3 digits matched by the "." wildcard.
!--- Notice that since the PBX is expecting the "1000" string, the prefix command is used.
!
dial-peer voice 2 pots
destination-pattern 1...
port 1/0/0
prefix 1
!
```

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.

```
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 "10 ",
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.