This chapter provides troubleshooting tips about the following topics for problems that can occur after installing and configuring Cisco Unified MeetingPlace H.323/SIP IP Gateway:

- Troubleshooting Network Connectivity
- Troubleshooting Caller Connectivity
- Troubleshooting Audio Problems

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Troubleshooting Network Connectivity

If you experience a network connectivity problem, perform the following steps to make sure that the IP-gateway server has not lost its connection to the Cisco Unified MeetingPlace Audio Server system.

1. To verify that Cisco Unified MeetingPlace H.323/SIP IP Gateway services are running, choose **Start > Settings > Control Panel > Services** from the IP-gateway server.
2. Make sure the following services are started:
    - Cisco Unified MeetingPlace Gateway SIM
    - Cisco Unified MeetingPlace IP Gateway
3. To verify that the IP-gateway server is logging in, telnet to the Cisco Unified MeetingPlace Audio Server system.
4. To verify that the IP-gateway server status is OK, enter gwstatus.
5. Check the Cisco Unified MeetingPlace Audio Server System eventlog for any errors relating to the IP-gateway server.
6. Make sure that all cards are seated properly in the chassis.
7. Check all cables and connections.
8. Verify card configuration by entering the **blade**, **dcard**, and **span** commands.
9. Verify port configuration by entering the **port** command.
10. Check the error log by entering the **errorlog** command.

Troubleshooting Caller Connectivity

- Unable to Make Calls From a Cisco IP Phone
- Unable to Call a PSTN Telephone From a Cisco IP Phone or Vice Versa
- Dead Air Heard When Using an H.323 Device
- Dead Air Heard When Using a Cisco IP Phone
- Fast Busy Signal Heard When Using a Cisco IP Phone
- Unable to Make Dial-Pad Key Selections When Using an H.323 Device
- IP Ports Do Not Answer
- IP Calls Connect But No Audio Is Heard
- Unable to Dial Out on IP Ports

Unable to Make Calls From a Cisco IP Phone

Possible Cause-The network may not be functioning properly.

Corrective Action-Verify your network access.

Possible Cause-Cisco Unified Communications Manager may not be configured correctly.

Corrective Action-Verify your Cisco Unified Communications Manager configuration.

Unable to Call a PSTN Telephone From a Cisco IP Phone or Vice Versa

Possible Cause-The voice gateway may not be functioning or configured properly.

Corrective Action-Verify your configuration settings.

Dead Air Heard When Using an H.323 Device

Possible Cause-Data packets transmitted across IP are at times inconsistently sized.

Corrective Action-Ensure that Cisco Unified Communications Manager, the IP-gateway server, and the Cisco Unified MeetingPlace Audio Server system are all be set to handle the same size data packet.

Dead Air Heard When Using a Cisco IP Phone

Possible Cause-There may be a poor connection between the Cisco IP phone and the Cisco Unified MeetingPlace Audio Server system.

Corrective Action-Verify that all associated connections are secure.

Fast Busy Signal Heard When Using a Cisco IP Phone

Possible Cause-The route pattern to IP-gateway server may not be configured properly in Cisco Unified Communications Manager.

Corrective Action -To resolve a fast busy-signal problem, verify that the configuration information that you entered in the Assigning a Cisco Unified Communications Manager Route Pattern to Point to the Cisco Unified MeetingPlace H.323/SIP IP Gateway Server page is correct.

To verify the configuration, perform the following steps:

1. In the Cisco Unified Communications Manager Administration page, choose **Route Plan > Route Pattern**.
2. Verify that the settings are correct and make changes if necessary.
3. When finished, click **Insert**.

Possible Cause-All IP ports on the Cisco Unified MeetingPlace Audio Server system are in use.

Corrective Action-Confirm that Cisco Unified Communications Manager and the voice gateway have been configured to handle IP call overflow.

Unable to Make Dial-Pad Key Selections When Using an H.323 Device

Possible Cause-The audio compression setting of the H.323 device may be incorrect.

Corrective Action-Use CCITT u-Law, 8.000 kHz, 8 Bit Mono for a Cisco Unified MeetingPlace Audio Server system with T1; use CCITT A-Law, 8.000 kHz, 8 Bit Mono for a Cisco Unified MeetingPlace Audio Server system with E1.

IP Ports Do Not Answer

- Checking the Cisco Unified MeetingPlace Audio Server System
- Checking the Cisco Unified MeetingPlace H.323/SIP IP Gateway Server
- Checking Cisco Unified Communications Manager

Checking the Cisco Unified MeetingPlace Audio Server System

When IP ports do not answer:

1. Check that the Ethernet switch port or any other network devices to which the MA-16 blade connects directly is set to fixed 100 Base-TX Full Duplex.
2. Make sure that the IP ports on the server are configured and active by using the **blade** and **portstat** commands.
3. Check the port status by performing the following steps:
    1. Log in to the CLI.
    2. At the tech$ prompt, enter the **tvportstat -all** command and monitor the output.
    3. Make a test call.
    4. Verify that the incoming call is seen by the server.
4. Trace a test call by performing the following steps:
    1. At the tech$ prompt, enter the **cptrace -T 5** command and monitor the output.
    2. Make another test call.
    3. Verify that the incoming call is seen by the server.
5. Check for warnings and alarms, especially those that occur in "cpiphandler.cc" by performing the following steps:
    1. At the tech$ prompt, enter the **viewexlog -s info -l | more** command.
    2. Scroll through the log by entering **f**.
6. At the tech$ prompt, enter **gwstatus** to verify that both the Cisco Unified MeetingPlace Gateway SIM and IP-gateway server have a status of OK.

Checking the Cisco Unified MeetingPlace H.323/SIP IP Gateway Server

When IP ports do not answer:

1. To verify that both the Gateway SIM and IP-gateway server have a status of OK, enter **gwstatus** at the tech$ prompt.
2. Verify that the Cisco Unified MeetingPlace H.323/SIP IP Gateway configuration has the appropriate call control enabled-either H.323 or SIP.
3. Open the Cisco Unified MeetingPlace Gateway SIM eventlog.
4. Make a test call.
5. From the Cisco Unified MeetingPlace Gateway SIM eventlog, verify that the test call is received by the IP-gateway server and that the call-processing server is returning a response code of 0, as shown the following example:
   ```
   MP Resp. Msg=3 CPerr=0 SeqNum=0x16
   ```
6. Verify that soft phones are not running on the gateway.
7. If Cisco Unified MeetingPlace Web Conferencing is on the same server as Cisco Unified MeetingPlace H.323/SIP IP Gateway, make sure that they are each assigned different IP addresses.

Checking Cisco Unified Communications Manager

When IP ports do not answer:

1. Verify that an H.323 gateway has been created for the IP-gateway server and that a route pattern has been assigned to it.
2. Verify that the Cisco Unified Communications Manager server can ping the IP-gateway server and vice versa.

IP Calls Connect But No Audio Is Heard

- Checking the Cisco Unified MeetingPlace Audio Server System
- Checking the Cisco Unified MeetingPlace H.323/SIP IP Gateway
- Checking the Cisco IP Phone

Checking the Cisco Unified MeetingPlace Audio Server System

When IP calls connect but no audio is heard:

1. Check that the Ethernet switch port or any other network devices to which the MA-16 connects directly is set to fixed 100Base-TX Full Duplex.

2. Verify that the subnet mask address is correct by entering the **blade** command. If it is not correct, Cisco Unified MeetingPlace Audio Server system will not be able to send voice packets to the phone. Restart the Cisco Unified MeetingPlace Audio Server system for any changes to take effect.
3. At the tech$ prompt, enter **tvportstat -all**.
4. While monitoring the output, make a test call to verify that the IP call is seen by the Cisco Unified MeetingPlace Audio Server system.
5. At the tech$ prompt, enter **cptrace -T 5**.
6. While monitoring the output of the trace command, make a test call to verify that the IP call is seen by the Cisco Unified MeetingPlace Audio Server system.
7. At the tech$ prompt, enter **tvportstat** *number*, where *number* is the port number that you used in Step 6.
8. Look for the **`RTCP packets sent by far end`** message to verify that the phone is transmitting voice data to the Cisco Unified MeetingPlace Audio Server system.

If the message is present, there is a one-way connection.

Checking the Cisco Unified MeetingPlace H.323/SIP IP Gateway

When IP calls connect but no audio is heard:

1. Open the Cisco Unified MeetingPlace Gateway SIM eventlog and verify that the following log entries have the correct IP address of the IP MA-16 blade:
   ```
   MP RTP info. IP=10.10.10.1 Port=5010
   MP RTCP info. IP=10.10.10.2 Port=5011
   ```
2. From the Cisco Unified MeetingPlace Gateway SIM eventlog, verify that the following log entries have the correct IP address of the IP phone:
   ```
   Remote RTP info. IP=10.10.10.3 Port=6510
   Remote RTCP info. IP=10.10.10.4 Port=6511
   ```
3. Ping the IP addresses of all MA-16 blades and of the IP phone.

Checking the Cisco IP Phone

When IP calls connect but no audio is heard:

1. Press the blue **i** button quickly twice.
2. Verify that the phone is receiving and sending packets.
3. Verify that the expected codec has been negotiated.

Unable to Dial Out on IP Ports

* Checking the Translation Table
* Checking the Cisco Unified MeetingPlace Audio Server System

* Checking the Cisco Unified MeetingPlace H.323/SIP IP Gateway Server
* Checking Cisco Unified Communications Manager

Checking the Translation Table

Possible Cause-Dialing out may be prevented because of information that is in the translation table.

Corrective Action-Verify that the table contains the necessary numbering plans to allow for dialing out.

Note: In a mixed IP-PSTN environment, the translation table must contain numbering plans for each type of call.

Checking the Cisco Unified MeetingPlace Audio Server System

When unable to dial out on IP ports:

1. Verify that incoming calls to the server are connecting. If not, perform the following procedures:
    - Checking the Cisco Unified MeetingPlace Audio Server System
    - Checking the Cisco Unified MeetingPlace H.323/SIP IP Gateway Server
    - Checking Cisco Unified Communications Manager
2. Verify that the port group is enabled for outgoing calls by using the **port** command.
3. Check the translation table to verify IP calls are being directed to a port group that is configured for IP.

    Tip: You can use the **xltest** utility to check which port group will be used for the dialed number. This is especially important for mixed PSTN and IP systems.
4. At the tech$ prompt, enter **cptrace -T 5**.
5. While monitoring the output of the trace command, make a test call.
6. At the tech$ prompt, enter **viewexlog -s info -l | more**.

    Tip: Enter **f** to move forward in the log.
7. Check for warnings and alarms, especially those that occur in "cpiphandler.cc" and "cpplacecall.cc".
8. At the tech$ prompt, enter **activity**.
9. Choose option **4** to make a test call.
10. Test internal extensions and outside numbers to isolate the problem.

Checking the Cisco Unified MeetingPlace H.323/SIP IP Gateway Server

When unable to dial out on IP ports:

1. Open the Cisco Unified MeetingPlace Gateway SIM eventlog and verify that the IP-gateway server receives the **outdial** command from the Cisco Unified MeetingPlace Audio Server system.

2. In the Cisco Unified MeetingPlace Gateway SIM eventlog, verify that the correct phone number was received by the IP-gateway server, as shown in the following example:
   ```
   MeetingPlace IP outdial. Phone=651515 IRC=0 PSTN=46 Unit=0
   ```
3. In the Cisco Unified MeetingPlace H.323/SIP IP Gateway configuration, verify that the outdial is sent by using the appropriate protocol.
4. Verify that the gateway, gatekeeper, and proxy server addresses and ports are correct according to the desired protocol.
5. Verify that the E.164 Address and H.323 ID fields are correct for H.323 outdials.
6. Verify that the Display Name, User Name, and Session Name fields are correct for SIP outdials.

Checking Cisco Unified Communications Manager

When unable to dial out on IP ports:

1. If Cisco Unified MeetingPlace H.323/SIP IP Gateway is installed on a gateway with multiple IP addresses, verify that Cisco Unified Communications Manager has an H.323 gateway configuration for each address.
2. Verify that the gateway settings created for Cisco Unified MeetingPlace H.323/SIP IP Gateway allow dialing out.

Troubleshooting Audio Problems

See the following sections for information about troubleshooting audio problems:

- Poor or Low-Audio Quality
- Echo

Poor or Low-Audio Quality

Possible Cause-The caller is using a low-quality headset with the Cisco IP phone.

Corrective Action-Reduce the speaker volume to a volume that is comfortable but not loud enough to cause feedback from the microphone back to the other end of the call.

Corrective Action-Use a headset that is approved by Cisco Systems.

Possible Cause-Cisco IP phone audio settings need adjustment.

Corrective Action-During a meeting, on a Cisco 7960, press the blue **i** button twice to obtain network settings. The information that you receive provides statistics needed to optimize your network for VoIP.

Corrective Action-Lower the volume. Voice quality degrades if the volume on a Cisco IP phone is set to maximum.

Possible Cause-Network settings may need to be modified.

Corrective Action-Consider the CoS/QoS setting on your network. If the CoS setting is IP Precedence 5, you should hear considerable improvement in audio quality.

Corrective Action-Establish locations on your network. Locations enable you to regulate voice quality by limiting the amount of bandwidth that is available for calls.

For more information, refer to the Location Configuration section in the appropriate Cisco Unified Communications Manager Administration Guide for your release.

Echo

Possible Cause-The caller is using a low-quality headset with Cisco IP phone.

Corrective Action-Reduce the speaker volume to a volume that is comfortable but not loud enough to cause feedback from the microphone back to the other end of the call.

Corrective Action-Use a headset that is approved by Cisco Systems.

Possible Cause-Cisco IP phone audio settings need adjustment.

Corrective Action-Lower the volume. Voice quality degrades if the volume on a Cisco IP phone is set to maximum.