



# Device Issues

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This section addresses the following common problems that you may experience with Cisco IP Phones, gateways, and related devices.

- [Voice Quality](#)
- [Codec and Region Mismatches](#)
- [Location and Bandwidth](#)
- [Phone Issues](#)
- [Gateway Issues](#)
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- [B-Channel Remains Locked When Restart\\_Ack Does Not Contain Channel IE](#)

## Voice Quality

You may experience voice quality issues including lost or distorted audio signal during phone calls.

Common problems include audio breaks (like broken words) or the presence of odd noises and audio distortion, such as echo, and watery or robotic voice quality. One-way audio, that is, a conversation between two people where only one person can hear anything, does not actually represent a voice quality issue, but this section covers this issue.

You may experience audio problems with one or more of the following items:

- Gateways
- Phones
- Networks

This section covers the following common voice quality problems:

- [Lost or Distorted Audio](#)
- [Correcting Audio Problems from the Cisco IP Phone](#)
- [Echo](#)
- [One-Way Audio or No Audio](#)

## Lost or Distorted Audio

### Symptom

One of the most common problems that you may encounter involves broken audio signal (often described as garbled speech or lost syllables within a word or sentence). Two common causes for this exist: packet loss and/or jitter. Packet loss means that audio packets do not arrive at their destination because they were dropped or arrived too late to be useful. Jitter describes the variation in the arrival times of packets. In the ideal situation, all Voice over IP (VoIP) packets would arrive exactly at a rate of 1 every 20 microseconds (ms). Notice that this is not the time that it takes for a packet to get from point A to point B but is simply the variation in packet arrival times.

### Possible Cause

Many sources of variable delay exist in a network. You can control some of these but not others. You cannot entirely eliminate variable delay in a packetized voice network. Digital Signal Processors (DSP) on phones and other voice-capable devices by design buffer some of the audio in anticipation of variable delay. This dejittering occurs only when the audio packet has reached its destination and is ready to be put into a conventional audio stream.

The Cisco Unified IP Phone 7960 can buffer as much as 1 second of voice samples. The jitter buffer is adaptive, meaning if a burst of packets is received, the Cisco IP Phone 7960 can play them out in an attempt to control the jitter. The network administrator needs to minimize the variation between packet arrival times by applying quality-of-service (QoS) and other measures in advance (especially if calls cross a WAN).

Some video endpoints may not support G.728 and using G.728 may result in noise. Use another codec, such as G.729.

### Recommended Action

1. When you are faced with a lost or distorted audio problem, first try to isolate the path of the audio. Try to identify each network device (switches and routers) in the path of the call audio stream. Keep in mind that the audio may be between two phones, or between a phone and a gateway, or it could have multiple legs (from a phone to a transcoding device and from there to another phone). Try to identify whether the problem occurs only between two sites, only through a certain gateway, on a certain subnet, and so on. This will help narrow the number of devices that you need to look at more carefully.
2. Next, disable silence suppression (also known as Voice Activation Detection or VAD). This mechanism does save bandwidth by not transmitting any audio when there is silence but may cause noticeable or unacceptable clipping at the beginning of words.

Disable the service in Cisco Unified CallManager Administration, and choose **Service > Service Parameters**. From there, choose the server and the Cisco CallManager service.

3. Set SilenceSuppression to **False to disable for all devices in a Cisco CallManager cluster**; alternatively, you can set SilenceSuppressionForGateways to **False**. When in doubt, turn both off by choosing the value **False** for each.
4. Using a network analyzer, if a network analyzer is available, check whether a monitored call between two phones has 50 packets per second (or 1 packet every 20 ms) when silence suppression is disabled. With proper filtering, you can identify whether an excessive number of packets are lost or delayed.

Remember that delay by itself will not cause clipping, only variable delay. Notice in the table below, which represents a perfect trace, the arrival times between the audio packets (which will have an RTP header) that will be 20 ms. In a poor quality call (such as a call with a lot of jitter), the arrival times would vary greatly.

The following table illustrates a perfect trace.

Packet Number	Time - absolute (sec)	Time - delta (ms)
1	0	
2	0.02	20
3	0.04	20
4	0.06	20
5	0.08	20

Placing the packet analyzer into various points in the network will help narrow the number of places from which the delay is coming. If no analyzer is available, you will need to use other methods. Examine interface statistics of each device in the path of the audio.

Diagnostic Call Detail Records (CDR) specifies another tool for tracking calls with poor voice quality. Refer to the *Cisco Unified CallManager Serviceability Administration Guide* for more information about CDRs.

## Correcting Audio Problems from the Cisco IP Phone

### Symptom

Audio problems occur while a call is in progress.

### Possible Cause

Devices, where a higher speed interface feeds into a lower speed interface, provide the most common sources for delay and packet loss. For example, a router may have a 100 Megabyte (MB) fast Ethernet interface connected to the LAN and a slow frame-relay interface, connected to the WAN. If the poor audio quality occurs only when communicating to the remote site, the most likely causes of the problem include

- The router has not been properly configured to give voice traffic priority over data traffic.
- Too many active calls exist for the WAN to support (that is, no call admission control restricts the number of calls that can be placed).
- Physical port errors occur.
- Congestion in the WAN itself occurs.

On the LAN, the most common problems represent physical-level errors (such as CRC errors) that are caused by faulty cables, interfaces, or by incorrectly configured devices (such as a port speed or duplex mismatch). Make sure that the traffic is not crossing any shared-media device, such as a hub.

**Recommended Action**

The Cisco Unified IP Phone 7960 provides another tool for diagnosing possible audio problems.

1. On an active call, you can press the *i* button twice rapidly and the phone will display an information screen that contains packet receive and transmit statistics, as well as average and maximum jitter counters.



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**Note** On this screen, jitter represents the average of the last five packets that arrived; the maximum jitter designates the maximum for the average jitter.

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2. Situations could also occur where the traffic is taking a slower path through the network than expected. If QoS has been configured correctly, the possibility exists that there is no call admission control. Depending on your topology, you can accomplish this through the use of **Locations** in Cisco Unified CallManager Administration configuration, or by using a Cisco IOS router as a gatekeeper. In any case, you should always know the maximum calls supported across your WAN.

**Diagnosing Crackling Sounds**

3. Crackling is another poor quality symptom, which is sometimes caused by a defective power supply or some kind of strong electrical interference close to the phone. Try swapping the power supply and moving the phone.

**Checking Your Loads**

4. Verify gateway and phone loads. Check Cisco Connection Online (CCO) at [www.cisco.com](http://www.cisco.com) for the latest software loads, new patches, or release notes relating to the problem.

**Verification**

1. Test by disabling silence suppression as described in [Lost or Distorted Audio](#); then, place calls between the two sites. Do not place the calls on hold or on mute because this will stop packets from being transmitted.
2. With the maximum number of calls across the WAN, the calls should all have acceptable quality.
3. Test to make sure that a fast busy is returned when you try to make one more call.

# Echo

## Symptom

Echo occurs when the speech energy being generated and transmitted down the primary signal path is coupled into the receive path from the far end. The speaker then hears his or her own voice, delayed by the total echo path delay time.

Voice can reflect back. This can happen but goes unnoticed in a traditional voice network because the delay is so low. To the user, it sounds more like a side-tone than an echo. In a VoIP network, it will always be noticeable because packetization and compression contribute to the delay.

## Possible Cause

Remember that the cause of the echo is always with analog components and wiring. For instance, IP packets cannot simply turn around and go back to the source at a lower audio level or on digital T1/E1 circuits. The only exception may occur if one party is using a speakerphone that has the volume set too high or other situations where an audio loop is created.

## Recommended Action

1. Make sure that the problem phones are not using the speakerphone and that they have the headset volume set to reasonable levels (start with 50 percent of the maximum audio level). Most of the time, the problems occur when you attach to the PSTN by way of a digital or analog gateway.

## Testing the Gateway

2. Determine which gateway is being used. If a digital gateway is in use, you may be able to add additional padding in the transmit direction (towards the PSTN). Because lower signal strength will yield less reflected energy, this should clear the problem.

Additionally, you can adjust the receive level, so any reflected audio is reduced even further. Remember to make small adjustments at a time. Too much attenuation of the signal will make the audio impossible to hear on both sides.

3. Alternatively, you can contact the carrier and request to have the lines checked. On a typical T1/PRI circuit in North America, the input signal should be -15 dB. If the signal level is much higher (-5 dB, for example), echo likely will result.

## Keeping an Echo Log

4. You should keep a log of all calls that experience echo.

Record the time of the problem, the source phone number, and the number called. Gateways have a fixed time of 16 ms of echo cancellation.

If the delay in the reflected audio is longer than this, the echo canceller cannot work properly. This should not be an issue for local calls, and long distance calls should have external echo cancellers built into the network at the Central Office. This fact provides one of the reasons why it is important to note the external phone number of a call that experiences echo.

## Checking Your Loads

5. Verify your gateway and phone loads. Check Cisco Connection Online at [www.cisco.com](http://www.cisco.com) for the latest software loads, new patches, or release notes that may relate to the problem.

## One-Way Audio or No Audio

### Symptom

When establishing a phone call from an IP station through a Cisco IOS voice gateway/router, only one of the parties receives audio (one-way communication).

When establishing a toll-bypass call between two Cisco gateways, only one of the parties receives audio (one-way communication).

### Possible Cause

An improperly configured Cisco IOS Gateway, a firewall, or a routing or default gateway problem, among other things, can cause this problem.

### Recommended Action

#### Make Sure IP Routing is Enabled on Cisco IOS Gateway/Routers

Some Cisco IOS gateways, such as the VG200, have IP routing disabled by default. This will lead to one-way voice problems.



**Note** Before going any further, make sure your router has IP routing enabled (in other words, does not have the global configuration command **no ip routing**).

To enable IP routing, simply type the following global configuration command in your Cisco IOS gateway:

**voice-ios-gwy(config)#ip routing**

Check Basic IP Routing

Basic IP reachability should always be checked first. As RTP streams are connectionless (transported over UDP), traffic may travel successfully in one direction, but get lost in the opposite direction.

Check the following:

- Default gateways configured at the end stations
- IP routes on the default gateways, mentioned above, leading to the destination networks



**Note** The following list explains how to verify the default router/gateway configuration on various Cisco IP phones:

- Cisco Unified IP Phone 7910—Press **Settings**, select option 6, push volume down until the Default Router field shows up.
- Cisco Unified IP Phone 7960/40—Press **Settings**, select option 3, scroll down until the Default Router field shows up.
- Cisco Unified IP Phone 2sp+/30vip—Press **\*\*#**, then press **#** until **gtwy=** shows up.

**Note**

When using the Cisco IP SoftPhone application and more than one NIC (network interface card) is installed in the box, make sure the box sources the correct NIC. This issue is commonly present in Cisco IP SoftPhone software version 1.1.x (version 1.2 should resolve this issues).

**Note**

For Cisco DT24+ Gateways, check the DHCP Scope and make sure there is a Default Gateway (003 router) option in the scope. The 003 router parameter is what populates the Default Gateway field in the devices and PCs. Scope option 3 should have the IP address of the router interface that will be doing routing for the gateway.

### Bind the H.323 Signaling to a Specific IP Address on Cisco IOS Gateway/Routers

When the Cisco IOS gateway has multiple active IP interfaces, some of the H.323 signaling may be sourced from one IP address and other parts of it may reference a different source addresses. This can generate various kinds of problems, one of them being one-way audio.

To avoid the problem, the H.323 signaling can be bound to a specific source address, which can belong to a physical or virtual interface (loopback). The command syntax to use under the interface configuration mode is

**h323-gateway voip bind srcaddr<ip address>** . Configure this command under the interface with the IP address to which the Cisco Unified CallManager points.

This command was introduced in Cisco IOS Release 12.1.2T and is documented in *Configuring H.323 Support for Virtual Interfaces*.

**Note**

A bug exists in version 12.2(6) where this solution can actually cause a one-way audio problem. For more information, refer to bug ID CSCdw69681 ( registered customers only) in Cisco Software Bug Toolkit ( registered customers only) .

### Check that Answer Supervision is being Sent and Received Correctly from the Telco or Switch

In an implementation that has a Cisco IOS gateway connected to a Telco or switch, verify that answer supervision is being sent correctly when the called device behind the Telco or switch answers the call. Failure to receive the answer supervision will cause the Cisco IOS gateway not to cut-through (open) the audio path in a forward direction causing one-way voice. A workaround is to configure **voice rtp send-recv on**.

### Cut-through Two Way Audio Early Using voice rtp send-recv on Cisco IOS Gateway/Routers

The voice path is established in the backward direction as soon as the RTP stream is started. The forward audio path will not be cut-through until the Cisco IOS gateway receives a Connect message from the remote end.

In some cases it is necessary to establish a two-way audio path as soon as the RTP channel is opened—before the connect message is received. To achieve this, use the **voice rtp send-recv** global configuration command.

### Check cRTP Settings on a Link-by-Link Basis on Cisco IOS Gateway/Routers

This issue applies to scenarios, such as toll-bypass, where more than one Cisco IOS router/gateway is involved in the voice path and Compressed RTP (cRTP) is used. cRTP, or RTP Header Compression, is a method for making the VoIP packet headers smaller to regain bandwidth. cRTP takes the 40-byte IP/UDP/RTP header on a VoIP packet and compresses it to 2-4 bytes per packet, yielding approximately 12Kb of bandwidth for a G.729 encoded call with cRTP.

cRTP is done on a hop-by-hop basis with decompression and recompression on every hop. Each packet header needs to be examined for routing, therefore cRTP needs to be enabled on both sides of an IP link.

It is also important to verify that cRTP is working as expected on both ends of the link. Cisco IOS levels vary in terms of switching paths and concurrent cRTP support.

In summary, the history is:

- Until Cisco IOS Software Release 12.0.5T, cRTP gets process-switched.
- In Cisco IOS Software Release 12.0.7T, and continuing in 12.1.1T, fast- and Cisco express forwarding (CEF)-switching support for cRTP is introduced.
- In Cisco IOS Software Release 12.1.2T, algorithmic performance improvements are introduced.

If you are running cRTP on Cisco IOS platforms (IOS Release 12.1), verify that bug CSCds08210 (registered customers only) (VoIP and FAX not working with RTP header compression ON) does not affect your IOS version.

Verify Minimum Software Level for NAT on Cisco IOS Gateway/Routers

If using Network Address Translation (NAT), the minimum software level requirements must be met. Earlier versions of NAT do not support skinny protocol translation and will lead to one-way voice issues.

The minimum software levels required for using NAT and skinny simultaneously are Cisco IOS® Software 12.1(5)T for IOS gateways to support skinny and H.323v2 with NAT.



**Note** If your Cisco Unified CallManager is using a TCP port for skinny signaling different from the default 2000, you need to adjust the NAT router with the **ip nat service skinny tcp port<number>** global configuration command.

The minimum software level required for using NAT and skinny simultaneously on a PIX firewall is 6.0.



**Note** These levels of software will not necessarily support all of the RAS messages necessary for full gatekeeper support. Gatekeeper support is outside the scope of this document.

Disable voice-fastpath on AS5350 and AS5400

The Cisco IOS command **voice-fastpath enable** gets a hidden global configuration command for the AS5350 and AS5400, which is enabled by default. To disable it, use the **no voice-fastpath enable** global configuration command.

When enabled, this command caches the IP address and UDP port number information for the logical channel opened for a specific call and prevents the RTP stream from getting to the application layer, but rather forwards the packets at a lower layer. This helps marginally reduce CPU utilization in high call volume scenarios.

When supplementary services, such as hold or transfer are used, the voice-fastpath command causes the router to stream the audio to the cached IP address and UDP port, disregarding the new logical channel information generated after resuming a call on hold or completing a transfer. To avoid this problem, traffic should go to the application layer constantly so that redefinition of the logical channel is taken into account and audio is streamed to the new IP address/UDP port pair. That is why you should disable voice-fastpath to support supplementary services.

Configure the VPN IP Address with SoftPhone



Cisco IP SoftPhone offers the ability to make a PC work like a Cisco Unified IP Phone 7900 Series phone. Remote users who connect back to their company network through VPN need to configure some additional settings in order to avoid a one-way voice problem.

The solution is to configure the VPN IP address, instead of the IP address of the network adapter under the Network Audio Settings.

## Verification

A useful command to verify packet flow is **debug cch323 rtp**. This command displays packets that the router transmits (X) and receives (R). An uppercase character indicates successful transmission/reception whereas a lowercase character indicates a dropped packet. See the following example:

[illegible]

## Codec and Region Mismatches

If a user gets a reorder tone (busy signal) when going off hook, it could be the result of codec disagreement between regions. Verify that both call ends support at least one common codec (for example, G.711). If they do not, you will need to use transcoders.

A region specifies the range of supported codecs that can be used with each of the other regions. Every device belongs to a region.



### Note

Codec negotiation with a Cisco IOS router is not supported.

For example, Region1<->Region2 = G.711, means that a call between a device in Region1 and a device in Region2 can use G.711 or any other supported codec that requires the same or less bandwidth as G.711 (any supported codecs within G.711, G.729, G.723, and so on).



### Note

The following list gives codecs that are supported for each device:

Cisco IP Phone 7960—G.711A-law/μ-law, G.729, G.729A, G.729Annex-B

Cisco IP Phone SP12 series and VIP 30—G.711a-law/mu-law, G.723.1

Cisco Access Gateway DE30 and DT-24+—G.711a-law/mu-law, G.723.1

## Location and Bandwidth

If a user receives a reorder tone after dialing a number, the cause may be that the Cisco Unified CallManager bandwidth allocation for the location of one of the call end devices has been exceeded. Cisco Unified CallManager checks for the available bandwidth for each device before making a call. If no bandwidth is available, Cisco Unified CallManager will not set up the call, and the user receives a reorder tone.

```
12:42:09.017 Cisco CallManager|Locations:Orig=1 BW=12Dest=0 BW=-1(-1 implies infinite bw
available)
12:42:09.017 Cisco CallManager|StationD - stationOutputCallState tcpHandle=0x4f1ad98
12:42:09.017 Cisco CallManager|StationD - stationOutputCallInfo CallingPartyName=,
CallingParty=5003, CalledPartyName=, CalledParty=5005, tcpHandle=0x4f1ad98
12:42:09.017 Cisco CallManager|StationD - stationOutputStartTone: 37=ReorderTone
tcpHandle=0x4f1ad98
```

Once the call is established, the Cisco Unified CallManager will subtract bandwidth from the locations, depending on the codec used in that call.

- If the call is using G.711, Cisco Unified CallManager subtracts 80k.
- If the call is using G.723, Cisco Unified CallManager subtracts 24k.
- If the call is using G.729, Cisco Unified CallManager subtracts 24k.

## Phone Issues

This section addresses the following phone issues:

- [Phone Resets](#)
- [Dropped Calls](#)

## Phone Resets

### Symptom

Phone resets.

### Possible Cause

Phones will power cycle or reset for two reasons:

- TCP failure connecting to Cisco Unified CallManager
- Failure to receive an acknowledgment to the phone KeepAlive messages.

### Recommended Action

1. Check the phones and gateways to ensure that you are using the latest software loads.
  2. Check Cisco Connection Online at [www.cisco.com](http://www.cisco.com) for the latest software loads, new patches, or release notes that may relate to the problem.
  3. Check the Event Viewer for instances of phone(s) resetting. Phone resets represent Information events.
  4. Look for any errors that may have occurred around the time that the phone(s) reset.
  5. Start an SDI trace and try to isolate the problem by identifying any common characteristics in the phones that are resetting. For example, check whether they are all located on the same subnet, same VLAN, and so on. Look at the trace and determine:
    - If the resets occur during a call or happen intermittently
    - If there any similarities of phone model (such as Cisco Unified IP Phone 7960 or Cisco Unified IP Phone 30VIP)
  6. Start a Sniffer trace on a phone that frequently resets. After it has reset, look at the trace to determine if there are any TCP retries occurring. If so, this indicates a network problem. The trace may show some consistencies in the resets, such as the phone resetting every seven days. This might indicate DHCP lease expiration every seven days (this value is user-configurable; for example, it could be every two minutes).
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## Dropped Calls

### Symptom

Premature termination of dropped calls.

### Possible Cause

Premature termination of dropped calls can be the result of a phone or gateway resetting (see [Phone Resets](#)) or a circuit problem, such as incorrect PRI configuration.

### Recommended Action

1. Determine whether this problem is isolated to one phone or to a group of phones. Perhaps you will find that the affected phones are all on a particular subnet or location.

2. Check the Event Viewer for phone or gateway resets.

You will see one Warning and one Error message for each phone that resets. This indicates that the phone cannot keep its TCP connection to the Cisco Unified CallManager alive, so the Cisco Unified CallManager resets the connection. This may occur because a phone was turned off or a problem may exist in the network. If this is an intermittent problem, you may find it useful to use Microsoft Performance to record phone registrations.

3. If the problem seems to be occurring only through a certain gateway, such as a Cisco Access DT-24+, enable tracing and/or view the Call Detail Records (CDR). The CDR files will give a cause of termination (CoT) that may help determine the cause of the problem. Refer to the *Cisco Unified CallManager Serviceability Administration Guide* for detailed information on CDRs.
4. Find the disconnect cause values (origCause\_value and destCause\_value—depending on which side hung up the call, that map to Q.931 disconnect cause codes (in decimal) at the following location:  
  
<http://www.cisco.com/univercd/cc/td/doc/product/software/ios113ed/dbook/disdn.htm>.
5. If the call is going out of a gateway to the PSTN, you can use the CDR to determine which side is hanging up the call. Obtain much of the same information by enabling tracing on the Cisco Unified CallManager. Because the trace tool can affect Cisco Unified CallManager performance, you will want to use this option only as a last resort or if your network is not yet in production.

## Gateway Issues

This section addresses the following gateway issues:

- [Gateway Reorder Tone](#)
- [Gateway Registration Failure](#)

## Gateway Reorder Tone

### Symptom

Reorder tone occurs.

### Possible Cause

Users placing a call through the gateway might get a reorder tone if they are attempting to make a restricted call or to call a number that has been blocked. A reorder tone may occur if the dialed number is out of service or if the PSTN has an equipment or service problem.

Check to be sure that the device giving the reorder tone has registered. Also, check your dial plan configuration to ensure that the call can be successfully routed.

### Recommended Action

The following procedure shows the steps for troubleshooting reorder tones through gateways.

1. Check the gateways to ensure that you are using the latest software loads.

2. Check Cisco Connection Online at [www.cisco.com](http://www.cisco.com) for the latest software loads, new patches, or release notes relating to the problem.
3. Start an SDI trace and re-create the problem. Reorder tones could be the result of a configuration issue with location-based admission control or gatekeeper-based admission control where the Cisco Unified CallManager might limit the number of allowable calls. In the SDI trace, locate the call to determine if it was blocked intentionally by a route pattern or the calling search space or by any other configuration setting.
4. Reorder tones can also occur when calling occurs through the PSTN. Check the SDI trace for Q.931 messages, in particular for disconnect messages. If a Q.931 disconnect message is present, it means that the other party caused the disconnect, and you cannot correct for that.

## Gateway Registration Failure

This section describes two similar but different categories of gateways. The Cisco Access AS-X, AT-X and Cisco Access DT-24+ and DE-30+ belong to one category. These gateways identify standalone units that are not directly connected to a Network Management Processor (NMP). The second category includes the Analog Access WS-X6624 and Digital Access WS-X6608. These gateways, as blades installed in a Catalyst 6000 chassis, provide direct connectivity to the NMP for control and statusing.

### Symptom

A registration problem represents one of the most common issues that are encountered with gateways on a Cisco Unified CallManager.

### Possible Cause

Registration can fail for a variety of reasons.

### Recommended Action

1. First, check that the gateway is up and running. All gateways have a heartbeat LED that blinks 1-second-on, 1-second-off when the gateway software is running normally.

If this LED is not blinking at all, or blinking very rapidly, the gateway software is not running. Normally, this results in an automatic reset of the gateway. Also, it is normal for the gateway to reset itself if it cannot complete the registration process after about 2 to 3 minutes. So, you may happen to look at the heartbeat LED while the device is resetting, but if the normal blinking pattern does not appear in 10 to 15 seconds, the gateway has suffered a serious failure.

On the Cisco Access Analog gateways, find the green heartbeat LED on the far right of the front panel. On the Cisco Access Digital gateways, find the red LED on the far left on the top edge of the card. On the Cisco Analog Access WS-X6624, a green LED appears inside the blade (not visible from the front panel) on the far right card edge near the front. Finally, on the Digital Access WS-X6608, a separate heartbeat LED exists for each of the eight spans on the blade. Eight red LEDs appear across the card (not visible from the front panel) about two thirds of the way towards the back.

2. Check that the gateway received its IP address. A standalone gateway must receive its IP address via DHCP or BOOTP. A Catalyst gateway may receive its IP address by DHCP, BOOTP or by manual configuration through the NMP.

- The IP address uses this format:

For example, for port 1 on module 7, you would enter

```
Console (enable) ping 127.1.7.1
127.1.7.1 is alive
```

- tracy\_start**
- mod port*

The following output actually comes from the 860-console port on the gateway board itself; however, the output of the `tracy` command represents nothing more than a remote copy of the 860-console port.

## Troubleshooting Guide for Cisco Unified CallManager Release 5.0(2)

```

00:00:06.780 (CFG) IP Configuration Change! Restarting now...
00:00:10.480 (CFG) DHCP Request or Discovery Sent, DHCPState = INIT
00:00:14.480 (CFG) DHCP Timeout Waiting on Server, DHCPState = INIT
00:00:22.480 (CFG) DHCP Timeout Waiting on Server, DHCPState = INIT
00:00:38.480 (CFG) DHCP Timeout Waiting on Server, DHCPState = INIT

```

If this timeout message continues to scroll by, a problem exists with contacting the DHCP server.

11. First, check that the Catalyst 6000 gateway port is in the correct VLAN.  
You will find this information in the information that you retrieved by using the **show port** command.
12. If the DHCP server is not on the same VLAN as the Catalyst 6000 gateway, then make sure that the appropriate IP helper addresses have been configured to forward the DHCP requests to the DHCP server. The gateway can get stuck in the INIT state after a VLAN number change until the gateway resets.
13. When in the INIT state, try resetting the gateway. Every time that the 860 gets reset, your tracy session will be lost, so you must close your existing session and reestablish a new one by issuing the following commands:  
**tracy\_close mod port**  
**tracy\_start mod port**
14. If you are still seeing the **DHCPState = INIT** messages, check whether the DHCP server is functioning correctly.
15. If so, start a sniffer trace to see whether the requests are being sent and the server is responding.  
Once DHCP is working correctly, the gateway will have an IP address that allows the use of the tracy debugging utility. This utility includes a built in feature of the NMP command set for the Catalyst gateways and is available as a helper application that runs on Windows 98/NT/2000 for the standalone gateways.
16. To use the helper application tracy utility, connect to the gateway by using the IP address to which it is assigned. This tracy application works on all the gateways, provides a separate trace window for each gateway (up to eight may be traced at once), and allows traces to be logged directly to a file that you specify.
17. Verify that the TFTP server IP address was correctly provided to the gateway. DHCP normally provides DHCP in Option 66 (by name or IP address), Option 150 (IP address only), or si\_addr (IP address only). If your server has multiple Options configured, si\_addr will take precedence over Option 150, which will take precedence over Option 66.

If Option 66 provides the DNS\_NAME of the TFTP server, then the DNS server(s) IP address(es) must have been specified by DHCP, and the name entered in Option 66 must resolve to the correct TFTP server IP address. The NMP could configure a Catalyst gateway could be configured by the NMP to disable DHCP, and the NMP operator must then manually enter all configuration parameters at the console, including the TFTP server address.

Additionally, the gateways will always attempt to resolve the name CiscoCM1 via DNS. If successful, the CiscoCM1 IP address will take precedence over anything that the DHCP server or NMP tells it for the TFTP server address, even if the NMP has DHCP disabled.

18. You can check the current TFTP server IP address in a gateway by using the tracy utility. Enter the following command to get the configuration task number:

```

TaskID: 0
Cmd:    show tl

```

Look for a line with config or CFG and use the corresponding number as the taskID for the next line, such as for the Cisco Access Digital gateway. In the examples that follow, bold lines of text make it easier for you to see the messages being explained. In the actual display output, text does not appear bolded. The examples come from an WS-X6624 model; the command to dump the DHCP information is

```
TaskID: 6
Cmd:      show dhcp
```

19. The TFTP server IP address then appears. If it is not correct, verify that your DHCP options and other information that it provides are correct.
20. Once the TFTP address is correct, ensure that the gateway is getting its configuration file from the TFTP server. If you see the following information in the Tracy output, your TFTP service may not be working correctly, or the gateway might not be configured on the Cisco Unified CallManager:

```
00:09:05.620 (CFG) Requesting SAA00107B0013DE.cnf File From TFTP Server
00:09:18.620 (CFG) TFTP Error: Timeout Awaiting Server Response for .cnf File!
```

The gateway attempts to connect to the same IP address as the TFTP server if it does not get a configuration file. This works fine unless you are in a clustered environment in which the gateway needs to receive its list of redundant Cisco Unified CallManagers.

21. If the card is not getting its TFTP information correctly, check the TFTP service on the Cisco Unified CallManager and make sure it is running.
22. Check the TFTP trace on the Cisco Unified CallManager.

Another common problem occurs if the gateway is not configured correctly on the Cisco Unified CallManager. A typical error involves entering an incorrect MAC address for the gateway. If this is the case, for a Catalyst 6000 gateway, you will probably get the following messages on the NMP console every 2 minutes:

```
2000 Apr 14 19:24:08 %SYS-4-MODHPRESET:Host process (860) 7/1 got reset
asynchronously
2000 Apr 14 19:26:05 %SYS-4-MODHPRESET:Host process (860) 7/1 got reset
asynchronously
2000 Apr 14 19:28:02 %SYS-4-MODHPRESET:Host process (860) 7/1 got reset
asynchronously
```

The following example shows what the Tracy output would look like if the gateway is not in the Cisco CallManager database:

```
00:00:01.670 (CFG) Booting DHCP for dynamic configuration.
00:00:05.370 (CFG) DHCP Request or Discovery Sent, DHCPState = INIT_REBOOT
00:00:05.370 (CFG) DHCP Server Response Processed, DHCPState = BOUND
00:00:05.370 (CFG) Requesting DNS Resolution of CiscoCM1
00:00:05.370 (CFG) DNS Error on Resolving TFTP Server Name.
00:00:05.370 (CFG) TFTP Server IP Set by DHCP Option 150 = 10.123.9.2
00:00:05.370 (CFG) Requesting SAA00107B0013DE.cnf File From TFTP Server
00:00:05.370 (CFG) TFTP Error: .cnf File Not Found!
00:00:05.370 (CFG) Requesting SAADefault.cnf File From TFTP Server
00:00:05.380 (CFG) .cnf File Received and Parsed Successfully.
00:00:05.380 (CFG) Updating Configuration ROM...
00:00:05.610 GMSG: GWEvent = CFG_DONE --> GWState = SrchActive
00:00:05.610 GMSG: CCM#0 CPEvent = CONNECT_REQ --> CPState = AttemptingSocket
00:00:05.610 GMSG: Attempting TCP socket with CCM 10.123.9.2
00:00:05.610 GMSG: CCM#0 CPEvent = SOCKET_ACK --> CPState = BackupUnified CM
00:00:05.610 GMSG: GWEvent = SOCKET_ACK --> GWState = RegActive
00:00:05.610 GMSG: CCM#0 CPEvent = REGISTER_REQ --> CPState = SentRegister
00:00:05.680 GMSG: CCM#0 CPEvent = CLOSED --> CPState = NotCPSocket
00:00:05.680 GMSG: GWEvent = DISCONNECT --> GWState = Rollover
00:00:20.600 GMSG: GWEvent = TIMEOUT --> GWState = SrchActive
```



```
00:00:20.600 MSG: CCM#0 CPEvent = CONNECT_REQ --> CPState = AttemptingSocket
00:00:20.600 MSG: Attempting TCP socket with CCM 10.123.9.2
00:00:20.600 MSG: CCM#0 CPEvent = SOCKET_ACK --> CPState = Backup CCM
```

Another possible registration problem could be that the load information is incorrect or the load file is corrupt. The problem could also occur if the TFTP server is not working. In this case, tracy shows that the TFTP server reported that the file is not found:

```
00:00:07.390 MSG: CCM#0 CPEvent = REGISTER_REQ --> CPState = SentRegister
00:00:08.010 MSG: TFTP Request for application load A0021300
00:00:08.010 MSG: CCM#0 CPEvent = LOADID --> CPState = AppLoadRequest
00:00:08.010 MSG: ***TFTP Error: File Not Found***
00:00:08.010 MSG: CCM#0 CPEvent = LOAD_UPDATE --> CPState = LoadResponse
```

In this case, the gateway requests application load A0021300, although the correct load name would be A0020300. For a Catalyst 6000 gateway, the same problem can occur when a new application load needs to get its corresponding DSP load as well. If the new DSP load is not found, a similar message will appear.

```
ELVIS>> 00:00:00.020 (XA) MAC Addr : 00-10-7B-00-13-DE
00:00:00.050 NMPTask:got message from XA Task
00:00:00.050 (NMP) Open TCP Connection ip:7f010101
00:00:00.050 NMPTask:Send Module Slot Info
00:00:00.060 NMPTask:get DIAGCMD
00:00:00.160 (DSP) Test Begin -> Mask<0x00FFFFFF>
00:00:01.260 (DSP) Test Complete -> Results<0x00FFFFFF/0x00FFFFFF>
00:00:01.260 NMPTask:get VLANCONFIG
00:00:02.030 (CFG) Starting DHCP
00:00:02.030 (CFG) Booting DHCP for dynamic configuration.
00:00:05.730 (CFG) DHCP Request or Discovery Sent, DHCPState = INIT_REBOOT
00:00:05.730 (CFG) DHCP Server Response Processed, DHCPState = BOUND
00:00:05.730 (CFG) Requesting DNS Resolution of CiscoCM1
00:00:05.730 (CFG) DNS Error on Resolving TFTP Server Name.
00:00:05.730 (CFG) TFTP Server IP Set by DHCP Option 150 = 10.123.9.2
00:00:05.730 (CFG) Requesting SAA00107B0013DE.cnf File From TFTP Server
00:00:05.730 (CFG) .cnf File Received and Parsed Successfully.
00:00:05.730 MSG: GWEvent = CFG_DONE --> GWState = SrchActive
00:00:05.730 MSG: CCM#0 CPEvent = CONNECT_REQ --> CPState = AttemptingSocket
00:00:05.730 MSG: Attempting TCP socket with CCM 10.123.9.2
00:00:05.730 MSG: CCM#0 CPEvent = SOCKET_ACK --> CPState = Backup CCM
00:00:05.730 MSG: GWEvent = SOCKET_ACK --> GWState = RegActive
00:00:05.730 MSG: CCM#0 CPEvent = REGISTER_REQ --> CPState = SentRegister
00:00:06.320 MSG: CCM#0 CPEvent = LOADID --> CPState = LoadResponse
00:01:36.300 MSG: CCM#0 CPEvent = TIMEOUT --> CPState = BadUnified CM
00:01:36.300 MSG: GWEvent = DISCONNECT --> GWState = Rollover
00:01:46.870 MSG: CCM#0 CPEvent = CLOSED --> CPState = NoTCPSocket
00:01:51.300 MSG: GWEvent = TIMEOUT --> GWState = SrchActive
00:01:51.300 MSG: CCM#0 CPEvent = CONNECT_REQ --> CPState = AttemptingSocket
00:01:51.300 MSG: Attempting TCP socket with CCM 10.123.9.2
00:01:51.300 MSG: CCM#0 CPEvent = SOCKET_ACK --> CPState = Backup CCM
00:01:51.300 MSG: GWEvent = SOCKET_ACK --> GWState = RegActive
00:01:51.300 MSG: CCM#0 CPEvent = REGISTER_REQ --> CPState = SentRegister
00:01:51.890 MSG: Unified CM#0 CPEvent = LOADID --> CPState = LoadResponse
```

The difference here is that the gateway gets stuck in the **LoadResponse** stage and eventually times out. You can resolve this problem by correcting the load file name in the Device Defaults area of Cisco Unified CallManager Administration.

# Gatekeeper Issues

Before starting any gatekeeper troubleshooting, verify that IP connectivity exists within the network. Assuming that there is IP connectivity, use the following information in this section to troubleshoot your gatekeeper calls:

- [Intercluster Trunks or H.225 Trunks](#)
- [Admission Rejects](#)
- [Registration Rejects](#)

## Intercluster Trunks or H.225 Trunks

Refer to the *Cisco Unified CallManager Administration Guide* and *Cisco Unified CallManager System Guide* at the following location:

[http://www.cisco.com/univercd/cc/td/doc/product/voice/c\\_callmg/4\\_1/index.htm](http://www.cisco.com/univercd/cc/td/doc/product/voice/c_callmg/4_1/index.htm)

## Admission Rejects

### Symptom

The system issues Admission Rejects (ARJ) when Cisco Unified CallManager has registered with the Gatekeeper but cannot send a phone call.

### Possible Cause

Configuration issues on the gatekeeper should be the primary focus when the gatekeeper issues an ARJ.

### Recommended Action

1. Verify IP connectivity from the Cisco Unified CallManager to the gatekeeper.
2. Show gatekeeper status and verify that the gatekeeper state is up.
3. Is there a zone subnet defined on the gatekeeper? If so, verify that the subnet of the Cisco Unified CallManager is in the allowed subnets.
4. Verify that the technology prefix matches between the Cisco Unified CallManager and the gatekeeper configuration.
5. Verify the bandwidth configuration.

## Registration Rejects

### Symptom

The system issues Registration Rejects (RRJ) when Cisco Unified CallManager cannot register with the gatekeeper.

**Possible Cause**

Configuration issues on the gatekeeper should be the primary focus when the gatekeeper is issuing a RRJ.

**Recommended Action**

1. Verify IP connectivity from the Cisco Unified CallManager to the gatekeeper.
  2. Show gatekeeper status and verify that the gatekeeper state is up.
  3. Is there a zone subnet defined on the gatekeeper? If so, verify that the subnet of the gateway is in the allowed subnets.
- 

## B-Channel Remains Locked When Restart\_Ack Does Not Contain Channel IE

**Symptom**

When the Cisco Unified CallManager system receives a Release Complete with cause ie=channel not available, the system sends out a Restart to bring this channel back to the idle state.

**Possible Cause**

In the Restart, you specify with the Channel IE which channel(s) must be restarted. If the network responds with Restart\_Ack without the Channel IE, the system keeps this channel in a locked state. While on network side, this same channel goes back to idle state.

Now you end up with the network requesting this channel for inbound calls.

Because the channel is locked on the Cisco Unified CallManager server, the Cisco Unified CallManager releases any call requests for this channel.

This behavior occurs on numerous sites in the UK and when the gateway is an E1 blade (most likely the same happens when using MGCP backhaul on the 2600/3600).

A glare condition provides the likely reason for the Release Complete.

You see this frequent happening on sites where a high call volume occurs.

If the B-channel selection on the network is top-down or bottom up, all inbound calls will fail until a B-channel in the higher/lower range is freed (if an active call gets cleared).

When B-channel selection is round-robin over a certain time, you will end up with an E1 blade with all locked B-channels.

**Recommended Action**

Reset the E1 port.

Verification

The B-channel(s) return to the idle state.

