



CHAPTER 5

Operate

Introduction to Operating the System

To ensure that your network operates efficiently and reliably, you should follow system and performance management practices as part of daily operations. These practices include performing scheduled routine maintenance; keeping maintenance records; and maintaining up-to-date upgrade, troubleshooting, and recovery strategies.

Before You Begin

User acceptance testing is completed and any problems that surfaced have been resolved. Users have been trained in using the new system.

Output of This Process

The Operations phase produces data and performance criteria that provide crucial information for optimizing your system.

Major Tasks in This Process

- [Managing Your System](#)
- [Backing Up and Restoring Components](#)
- [Using Network Monitoring Tools](#)
- [Operating Contact Center Systems](#)
- [Troubleshooting Daily Operations](#)

Managing Your System

This topic provides a high-level summary of the ongoing tasks that are required for managing your system and the options for how these tasks can be performed. For detailed maintenance and operation guides for each component in your Cisco Unified Communications system, see the product documentation listed in [Component Resources Documentation](#) in the Resource Library.

System Management Tasks

Managing a Cisco Unified Communications system consists of performing the following activities:

- Integrating monitoring and management tools—Select, order, configure, integrate, and test a set of tools for monitoring and managing the Cisco Unified Communications system.
- Monitoring—Set thresholds, monitor events, and generate notifications when service-impacting events occur.
- Ticketing—Generate and track system trouble tickets for each event.
- Diagnosing incidents—Analyze and troubleshoot incidents to determine the cause.
- Resolving incidents—Define and execute an action plan which can include performing break and fix activities, applying software updates and patches, managing hardware replacements, and executing change management processes.
- Managing changes in the network—Define a change management process for performing moves, adds, changes, and disconnects (MACDs) for your Cisco Unified Communications system including network devices, phones/endpoints, software upgrades, voice-mail boxes, dial plan updates, security patches, OS applications, and voice applications.
- Archiving configurations—Back up device configurations daily and restore device configurations when necessary.
- Managing voice as a network service—Track, measure, and resolve quality of service (QoS) issues such as jitter, delay, and dropped packets, and monitor service level agreements (SLAs) with service providers.
- Managing security posture—Detect, analyze, and address security events.
- Reporting—Define, develop, and generate performance, availability, event, and inventory reports.
- Backing up and restoring system components—Define backup methodologies and schedules, define a verification process for backups, secure storage of backups, and document backup processes.

System Management Options

There are two options for managing a Cisco Unified Communications System:

- Do It Yourself—In this model, you are responsible for managing the entire Cisco Unified Communications System. This approach requires developing business processes; integrating, provisioning and maintaining network management tools; and developing data and voice management skills and knowledge. Cisco offers tools as a means for monitoring your network; see [Using Network Monitoring Tools](#) for more information.
- Outtasking Hybrid Model—Using the [Cisco Lifecycle Services](#) approach, Cisco and its partners provide a broad portfolio of end-to-end services and support that can help increase your Cisco Unified Communications system's business value and return on investment. This approach includes two services that provide different levels of management:
 - [Cisco Unified Communications Essential Operate Service](#) combines Cisco award-winning maintenance support with basic voice applications monitoring and reporting.
 - [Cisco Unified Communications Remote Management Service](#) includes monitoring and reporting plus managing day-to-day system issues such logical moves, adds, changes, and disconnects; resolving incidents; performing configuration backups; and reporting.

For more information about the Cisco Unified Communications Essential Operate Service, Cisco Unified Communications Remote Management Service or other Cisco Unified Communications services, see <http://www.cisco.com/go/ipcservices> or contact your Cisco service account manager.

Backing Up and Restoring Components

This topic provides details on backup and restore for Cisco Unified Communications components. First and foremost, the backup of Cisco Unified Communications components needs to be incorporated into your corporate-wide backup operations. It is an important aspect of disaster recovery and is also essential before doing component upgrades. If you do not have a process in place, you must develop and document a backup and recovery management process. Some items to consider for this process are the following:

- Provide proper storage of Operating System and Cisco Unified Communications Application CDs.
- Define incremental and full backup methodologies and schedules, assign an owner for each Unified Communications component and database server.
- Define a verification process for backups:
 - Monitor backup logs on a daily basis for errors.
 - Periodically restore backup images to ensure validity.
- Secure onsite and offsite storage of backups.
- Develop well documented processes for system and configuration restoration.
- Ideally, provide central location(s) (for example, SFTP servers) for backup of data from all the Cisco Unified Communications components.

The following topics provide backup and restore details on a component basis along with links to the appropriate component documentation:

- [Cisco Unified Communications Manager](#)
- [Cisco Unified IP Interactive Voice Response](#)
- [Cisco Unified Intelligent Contact Management Enterprise](#)
- [Cisco Unified Presence](#)
- [Cisco Unity Connection](#)

For additional information on backing up and restoring Cisco Unified Communications system components, as well as other system operations topics, see the documentation wiki (DocWiki) at http://docwiki.cisco.com/wiki/Unified_Communications_System_Operations.

Cisco Unified Communications Manager

Cisco Unified Communications Manager provides the Disaster Recovery System (DRS) for full backup and restore for all servers in a Unified Communications Manager cluster. The DRS performs a cluster-level backup, which means that it collects backups for all servers in a Unified Communications Manager cluster to a central location and archives the backup data to a physical storage device (tape or SFTP). For customers with multiple clusters, DRS must be configured per cluster.

DRS is invoked via the Unified Communications Manager Platform Administration. It allows you to perform scheduled (daily, weekly, monthly) automatic or user-invoked backups. DRS only supports a single backup schedule at a time. It provides a history (last 20 operations) of backup and restore operations.

With Cisco Unified Communications Manager Business Edition, DRS also provides backup and restore capabilities for Unity Connection.

**Note**

DRS does not support hostname or IP address change during restore. For more information about the Disaster Recovery System, see the [Disaster Recovery System Administration Guide](#).

Cisco Unified IP Interactive Voice Response

In Unified IP Interactive Voice Response (Unified IP IVR) Releases 4.0, 4.1, 6.0, the Backup and Restore System (BARS) utility is used for backing up and restoring data. Unified IP IVR system Releases 4.5 and 5.0 provide a Backup and Restore application that is embedded with the underlying Cisco Customer Response Solutions platform. For Unified IP IVR 5.0(1), this Backup and Restore application is described in Chapter 15 of the [Cisco Customer Response Solutions Administration Guide, Release 5.0\(1\)](#).

Cisco Unified Intelligent Contact Management Enterprise

For Cisco Unified Intelligent Contact Management Enterprise, the Microsoft backup strategies for SQL Server are recommended. In addition, the AW Database should not require a backup as it is populated by the Logger. When backing up the Logger, the Logger process should be stopped to prevent read/write conflicts. For more information, see the [Installation Guide for Cisco ICM/IPCC Enterprise & Hosted Editions](#).

Also, when upgrading the ICM Enterprise, see [Back Up the ICM Registry for Comparison](#) for backing up the Windows registry for comparison.

Cisco Unified Presence

Cisco Unified Presence uses the Disaster Recovery System (DRS) for full data backup and restore capabilities of all Unified Presence Administration. For more information, see the [Disaster Recovery System Administration Guide for Cisco Unified Presence](#).

Cisco Unity Connection

Cisco Unity Connection will make use of the Disaster Recovery System (DRS) instead of the previously used Disaster Recovery Backup and Restore Tool (DiRT). DRS will provide backup of Unity Connection database (system and users), all files containing the audio portion of greetings and voice names, and all voice-mail messages. It will provide restoration of this data onto a clean, running installation of the same version of Unity Connection that was backed up.

All the capabilities of DRS described in the [Cisco Unified Communications Manager](#) section apply. For more information, see the [Disaster Recovery System Administration Guide for Cisco Unity Connection](#).

Using Network Monitoring Tools

The Cisco Unified Communications Management Suite allows businesses to actively monitor their Cisco Unified Communications solution to discover potential problems, maintain quality and user satisfaction, and help minimize service downtime. The following network monitoring tools are available:

- [Cisco Unified Operations Manager](#)

Cisco Unified Operations Manager

Cisco Unified Operations Manager provides comprehensive monitoring with proactive and reactive diagnostics for the entire Cisco Unified Communications system, including the underlying transport infrastructure. Its built-in rules, which provide contextual diagnostics, enable rapid troubleshooting of key service-impacting outages.

Cisco Unified Operations Manager provides a real-time, service-level view of the entire Cisco Unified Communications system and presents contextual tools to look at the current alert status, historical information, and service impact of any outages. It continuously monitors the different elements such as Cisco Unified Communications Manager, Cisco Unity Connection, Cisco Unified Contact Center Enterprise, and Cisco Unified Presence, as well as Cisco gateways, routers, switches, and IP phones. For a complete list of devices that can be monitored, see the appropriate [Device Support Table for Cisco Unified Operations Manager](#).

Other Cisco Unified Operations Manager capabilities include:

- Synthetic tests that replicate end-user activity and verify gateway availability as well as other configuration aspects of the Cisco Unified Communications infrastructure. Tests may be run on synthetic phones or real IP phones (both SIP- and SCCP-based phones) deployed in the network.
- Cisco IOS IP Service Level Agreement (SLA)-based diagnostic tests that can be used to troubleshoot network-related issues, determine paths, and proactively monitor voice quality across WAN links.
- Tools to discover and report on the status of different video-enabled IP endpoints (for both SIP- and SCCP-based phones) in the Cisco Unified Communications system, as well as additional contextual information to locate and identify the IP phones. It can also track the status of these endpoints, such as when IP phones in your network that have become disconnected from the switch, are no longer registered to a Unified Communications Manager server, or have gone into SRST mode.
- Test probes to run dial-plan tests, acceptance tests, and phone-feature tests. Such phone-testing capabilities may be used to rapidly troubleshoot issues related to connectivity (signaling/media stream) and voice quality as well as call processing/dial-plan management issues.
- Visibility into key performance metrics of different Cisco Unified Communications elements, such as resource usage (CPU, memory, MTP resources, transcoder resources), call statistics (active calls), and trunk statistics (trunk usage, port usage, and gateway statistics) that aid in troubleshooting and capacity planning.
- Correlation and presentation of voice-quality alerts using the information available through Cisco Unified Service Monitor (when the latter is also deployed). Cisco Unified Operations Manager displays mean opinion scores associated with voice quality between pairs of endpoints (IP phones, Cisco Unity messaging systems, or voice gateways) at specified times involved in the monitored call segment and other associated details about the voice-quality problem. It can also trace a probable path between the two endpoints and report on any outages or problems on intermediate nodes in the path.

- Tracking of Cisco Unified Communications devices and IP phone inventory, including IP phone status changes, and creation of reports that document move, add, and change operations on IP phones in the network.

Because Cisco Unified Operations Manager does not deploy any agent software on the devices being monitored, it is completely nondisruptive to system operations. For more information on Cisco Unified Operations Manager, see the documentation available at:

http://www.cisco.com/en/US/products/ps6535/tsd_products_support_series_home.html

Operating Contact Center Systems

Sample call flows were tested and verified for the three test beds in the contact center system:

- [Test Bed 1: Unified IP IVR Test Sites](#)
- [Test Bed 2: Parent and Child Test Sites](#)
- [Test Bed 3: Unified CVP Test Sites](#)

Sites for the three test beds are defined in [Review Tested Deployment Models](#) on the Design tab.

Test Bed 1

Test Bed 1—Unified IP IVR test bed, which handles the following call flows:

- [Cisco Unified Communications Manager \(Unified Communications Manager\) Post-Routed call flow](#)
- [Cisco Unified Outbound Dialer \(Unified OUTD\) call flow](#)
- [Cisco Unified Mobile Agent \(Unified MA\) call flow](#)

Test Bed 2

Test Bed 2—Parent and Child test bed, which handles the following call flow:

- [Parent/Child call flow](#)

**Note**

Cisco Unified Customer Voice Portal (Unified CVP) implemented at Site1/Site4 provides initial call treatment for the Parent/Child call flow, while Customer Response Solutions (CRS) provides call queuing capabilities.

Test Bed 3

Test Bed 3—Unified CVP test bed, which handles the following call flows:

- [Unified Customer Voice Portal \(Unified CVP\) Post-Routed call flow](#)
- [Cisco Unified Outbound Dialer \(Unified OUTD\) call flow](#)
- [Cisco Unified Mobile Agent \(Unified MA\) call flow](#)

Failover and Redundancy

Failover testing was done to verify the redundancy and failover capabilities of specific components such as gatekeepers, WAN access routers, and the private connection between the Roggers in the data centers. Failover testing was done with:

- Contact center components that have redundancy capabilities in the event of a failure
- Contact center components that did not have redundancy capabilities in the event of a failure

For detailed information on the failover testing, see [Failure, Failover, and Recovery](#).

Troubleshooting Daily Operations

This topic describes how to diagnose and resolve system-level problems that occur during daily operations of a Cisco Unified Communications network. It contains the following sections:

- [Common Problems Reported by Users](#)

Common Problems Reported by Users


This section describes basic approaches to diagnose and resolve common problems reported by end users. Note that this section demonstrates various tools and diagnostic approaches available in the context of specific problems, but does not provide a comprehensive list of all possible problems that may occur. Problems described in this section include:

- [One-Way Audio](#)
- [Poor Voice Quality](#)

One-Way Audio

One-way audio and no audio at all (no-way audio) are problems that are fairly common during a newnetwork installation. The majority of these problems are caused by misconfigurations. For one-way audio problems, always pay attention the direction in which the one-way audio is occurring. For no audio in either direction, the troubleshooting methodology is the same. You might need to repeat the procedure for each direction of audio, but more likely you will find the source of the problem when trying to troubleshoot one direction. There are several steps you can take to troubleshoot a one-way/no-way audio problem:

1. [Verify Bidirectional IP Connectivity](#).
2. [Check Cisco IOS Software Gateway Configurations](#).
3. [Check for NAT or Firewall Restrictions](#).

For additional directions on troubleshooting one-way audio problems, refer to the [Troubleshooting One-Way Voice Issues Tech Note](#) .

Verify Bidirectional IP Connectivity

You should verify IP connectivity as the first step in troubleshooting a one-way or no-way audio problem because IP connectivity must be present for voice packets to be exchanged between two devices. A large number of one-way or no-way audio problems are caused by lack of IP connectivity. Check that:

- If the two endpoints involved in the call are on different IP subnets, each endpoint has the correct default gateway and subnet mask settings
- If one of the endpoints is a Unified IP phone, the DHCP scope has an incorrectly configured default gateway parameter.
- If one of the endpoints is a Cisco IOS software gateway, the default route is correct. Also, ping the other endpoint from the gateway. If the ping is successful, you know that you have IP connectivity. If the ping is unsuccessful, perform a **tracert** to determine where the problem lies.

**Note**

Remember that signaling packet traffic is always between Unified Communications Manager and the endpoint, whereas the RTP voice packet traffic is directly between the endpoints. So just because the endpoints are registered to Unified Communications Manager and can set up a call through Unified Communications Manager does not mean that the endpoints have proper IP connectivity between them.

Another useful tool for troubleshooting such a problem is the help (**i** or **?**) button on Cisco Unified IP phones. Press the help (**i** or **?**) button twice in quick succession during an active call. The display shows you receive and transmit statistics for the call. If you do not see the receive counter (RxCnt) incrementing, the packets are probably not arriving on that IP phone. If you go to the originating IP phone and the transmit count (TxCnt) is incrementing, the packets are probably being lost somewhere in the network. If a ping or traceroute does not provide enough information about where the packets are being lost, you may need to connect a sniffer to the network and perform the following steps:

1. Connect the sniffer to the back of the originating IP phone and make verify that the phone is actually transmitting packets.
2. On the originating phone, verify that the IP address and MAC address information is correct.
3. If the network settings on the originating phone are correct, go to the terminating IP phone to verify that the packets are not arriving.
4. If the voice packets are not arriving at the terminating phone, move the sniffer from network hop to network hop to isolate where the packets are being dropped. A common reason for a problem such as this is a missing or improperly configured IP route.

Check Cisco IOS Software Gateway Configurations

There are various reasons why you might encounter one-way audio on calls to a Cisco IOS software gateway. Most of these problems can be solved using simple configuration commands.

1. Check if IP routing is enabled on the gateway that you are using—You do not need to be running a routing protocol such as RIP, EIGRP, or OSPF, but IP routing must not be disabled. Make sure that the **no ip routing** command is not in your configuration. If it is, be sure to eliminate it by configuring the **ip routing** command. You can also issue the **show ip route** command to see if IP routing is enabled. If IP routing is disabled, there are no routes listed in the output, and the list of routing protocols is not present.
2. Determine if the VoIP subsystem is enabled—The VoIP subsystem in Cisco IOS software uses the IP routing code to aid in encapsulating and transmitting the VoIP packets, so the subsystem must be enabled to transmit and receive VoIP packets. It does not need the IP routing code to perform signaling such as H.323 or MGCP, so the signaling still works with IP routing disabled.
3. Check IP address configurations on gateway interfaces—Another common occurrence of one-way audio appears on Cisco IOS software H.323 voice gateways that have more than one data interface, such as a gateway that has both an Ethernet connection to the LAN and a serial connection to the WAN. When an H.323 gateway is configured in Cisco Unified Communications Manager Administration, you configure a specific IP address. Cisco Unified Communications Manager

always uses this IP address for all its signaling to the gateway; however, Cisco IOS software voice gateways by default use the IP address of the interface that is closest to the destination. This could be a problem if Unified Communications Manager is connected via one interface and the device to which the RTP audio stream is destined for is connected to a different interface. To force the voice gateway to always use the same IP address, configure the **h323-gateway voip bind srcaddr ip-address** command on the interface that you are using for signaling on the Cisco IOS software voice gateway. Make sure this is the same IP address configured in Cisco Unified Communications Manager Administration. Failure to do so could result in one-way audio when the gateway tries to use a different source interface than the one configured in Unified Communications Manager.

4. Configure voice rtp send-recv on the gateway—Sometimes you have one-way audio problems only when calling specific numbers, such as 411 or 911 in the North American numbering plan (NANP) or after you transfer a call or put it on hold. If you are having these problems when going through a Cisco IOS software voice gateway, be sure that the **voice rtp send-recv** command is configured on the gateway. Numbers such as 411 and 911 sometimes do not send back answer supervision (that is, an ISDN connect message) when the remote end answers. As a result, the Cisco IOS software voice gateway does not cut through audio in both directions to prevent toll fraud. Configuring the **voice rtp send-recv** command forces the voice gateway to cut through audio in both directions immediately.
5. If you are using a Cisco AS5350 or AS5400 as a gateway, configure the **no voice-fastpath enable** command in global configuration mode—When enabled, this command causes the voice gateway to cache the IP address and UDP port number information for the logical channel opened for a specific call and forwards the packets using the cached information. This helps marginally reduce CPU utilization in high-call-volume scenarios. Because of how Cisco Unified Communications Manager opens and closes logical channels to redirect RTP audio streams, such as in the case of a transfer or music on hold (MOH) server, the Cisco AS5350 and AS5400 cache the IP address information of the old IP address. Therefore, you end up with one-way audio when the call gets redirected to a new IP address because the voice gateway still uses the cached information instead of the newly negotiated information.

Check for NAT or Firewall Restrictions

One common cause of one-way or no-way audio is when Network Address Translation (NAT), Port Address Translation (PAT), or firewalls exist between two endpoints. The SCCP protocol embeds IP addresses in the IP packet's payload to signal which IP address to send RTP packets to. If the device performing NAT or PAT is unaware of this fact, the embedded IP addresses are not translated. Therefore, one-way or no-way audio results.

Firewalls can also be a problem if they are unaware of the voice traffic passing through them. Firewalls often are configured to block all UDP traffic going through them. Because voice traffic is carried over UDP, it might be blocked while the signaling carried over TCP is passed. A sniffer is the best tool for debugging such a scenario. If both devices appear to be transmitting audio but the audio is not reaching the opposite side, take a sniffer trace at each hop along the way until you find the hop where the audio is not passing through. If the firewall is blocking UDP packets, you might need to open a hole in it to allow the voice traffic to pass through.

Problems Occurring After the Call Connects Successfully

The scenarios discussed so far are cases in which you have one-way audio or no-way audio from the beginning of the call or after a hold/transfer. Occasionally, however, you might encounter scenarios in which a call is up and suddenly becomes one-way or audio disappears entirely. Network problems are largely to blame for failures of this sort. Ensure that network connectivity between the two endpoints still exists and that nothing on the network might be causing intermittent network connectivity. An example would be a *flapping* network connection—a network connection that is transitioning between

up and down states over and over again—or a routing protocol that cannot converge correctly. Again, a sniffer is the best tool for diagnosing this kind of problem. The best place to start is on the device that originates the RTP stream to ensure that the stream is still being generated when the loss of audio occurs. If you discover that the originating device stops sending packets for no reason, you might be dealing with a software or hardware problem on the originating device.

A common cause of such a failure is a Digital Signal Processor (DSP) crash. If the end device is a Cisco IOS software voice gateway, you see an error displayed on the console that looks similar to the following:

```
%VTSP-3-DSP_TIMEOUT: DSP timeout on event 6: DSP ID=0x2312: DSP error stats
```

This message is also sent to a Syslog server if the Cisco IOS software voice gateway is configured to send Syslog information to a Syslog server. On a Cisco VG200, 2600, or 3600, you can issue the following command to check the status of the DSPs:

```
test dsprm slot #
```

The **show voice dsp** command displays which port and time slot are allocated to each DSP. If the **test dsprm slot #** command detects a DSP that has crashed, you can compare this with the information obtained from a **show call active voice** command (or a **show call history voice** command if the call has been disconnected) to see if the time slot of the failed call is the same as the slot of the DSP that is no longer available. Unfortunately, the only way to recover from this condition is to reload the gateway.

Poor Voice Quality

Nearly all voice quality problems can be attributed to some kind of degradation on the IP network that the voice traffic traverses. Network problems that might not be noticeable for normal data traffic are very apparent in a voice conversation because of the need to minimize packet loss and variable delay in an IP telephony network.

A variety of issues can result in poor voice quality:

- [Packet Drops](#)
- [Queuing Problems](#)

In addition to the information in this section, refer to the [Troubleshooting QoS Choppy Voice Issues](#) document on Cisco.com for additional techniques on resolving voice quality issues.

Packet Drops

IP telephony demands that voice packets reach their destination within a predictable amount of time and without being dropped somewhere along the path from the source to the destination. In a properly designed network with appropriate QoS provisioning in place, packet loss should be near zero. All voice codecs can tolerate some degree of packet loss without dramatically affecting voice quality. Upon detecting a missing packet, the codec decoder on the receiving device makes a best guess as to what the waveform during the missing period of time should have been. Most codecs can tolerate up to five percent random packet loss without noticeable voice quality degradation. This assumes that the five percent of packets being lost are not being lost at the same time, but rather are randomly dropped in groups of one or two packets. Losing multiple simultaneous packets, even as a low percentage of total packets, can cause noticeable voice quality problems.

**Note**

You should design your network for zero packet loss for packets that are tagged as voice packets. A converged voice/data network should be engineered to ensure that only a specific number of calls are allowed over a limited-bandwidth link. You should guarantee the bandwidth for those calls by giving priority treatment to voice traffic over all other traffic. For more information on prioritizing voice over data, refer to the [Voice Quality](#) information available on Cisco.com.

There are various tools that you can use to determine whether you are experiencing packet loss in your network and where in the network the packets are getting dropped. The starting point to look for lost packets is the call statistics screen on Cisco Unified IP Phones.

1. Do one of the following:
 - If you are troubleshooting at the phone experiencing the problem, access these statistics by pressing the help (i or ?) button on the IP phone twice in quick succession during an active call.
 - If you are working with a remote user, open a web browser on your computer and enter the IP address of the user's phone. During an active call, choose the **Streaming Statistics > Stream 1** options from the display.
2. Examine the counters RxDisc and RxLost shown on the IP phone (or Rcvr Lost Packets if you are viewing the statistics remotely using a web browser).
 - RxLost measures the number of packets that were never received because they were dropped in the network somewhere. By detecting a missing RTP sequence number, the IP phone can determine that a packet has been lost.
 - RxDisc corresponds to packets that were received but were discarded because they could not be used at the time they arrived. RxDisc can come from an out-of-order packet or a packet that arrived too late.
3. If either of these two counters increments, you should investigate to learn why packets are being lost or discarded.

Regardless of how low your packet loss is, if it is not zero, you should investigate the root cause because it might be a sign of a bigger problem that will get worse with higher call volume. Also, although small packet loss might not be perceptible in a conversation between two people, it can be detrimental to fax and modem transmissions. The packet loss can be occurring at any layer of the OSI model, so be sure to check for all possibilities for each hop. For example, if there is a Frame Relay connection over a T1 between two sites, you should:

- Make certain that there are no errors at the physical layer on the T1.
- Determine if you are exceeding your committed information rate (CIR) on the Frame Relay connection.
- Verify that you are not dropping the packets at the IP layer because you are exceeding your buffer sizes.
- Check that you have your QoS improperly configured.
- Ensure that your service provider not only guarantees packet delivery but also guarantees a low-jitter link. Some service providers may tell you that they do not provide a CIR but guarantee that they will not drop any packets. In a voice environment, delay is as important as packet loss. Many service providers' switches can buffer a large amount of data, thereby causing a large amount of jitter.

One common cause of drops in an Ethernet environment is a duplex mismatch, when one side of a connection is set to full duplex and the other side is set to half duplex. To determine if this is the case, perform the following steps:

1. Check all the switch ports through which a given call must travel and ensure that there are no alignment or frame check sequence (FCS) errors. Poor cabling or connectors can also contribute to such errors; however, duplex mismatches are a far more common cause of this kind of problem.
2. Examine each link between the two endpoints that are experiencing packet loss and verify that the speed and duplex settings match on either side.

Although duplex mismatches are responsible for a large number of packet loss problems, there are many other opportunities for packet loss in other places in the network as well. When voice traffic must traverse a WAN, there are several places to look. First, check each interface between the two endpoints, and look for packet loss. On all Cisco IOS software platforms, you can find this information using the **show interface** command. If you are seeing dropped packets on any interface, there is a good chance that you are oversubscribing the link. This could also be indicative of some other traffic that you are not expecting on your network. The best solution in this case is to take a sniffer trace to examine which traffic is congesting the link.

Sniffers are invaluable in troubleshooting voice quality problems. With a sniffer, you can examine each packet in an RTP stream to see if packets are really being lost and where in the network they are being lost. To troubleshoot using a sniffer, perform the following steps:

1. Start at the endpoint that is experiencing the poor-quality audio where you suspect packet loss.
2. Take a sniffer trace of a poor-quality call and filter it so that it shows you only packets from the far end to the endpoint that is hearing the problem. The packets should be equally spaced, and the sequence numbers should be consecutive with no gaps.
3. If you are seeing all the packets in the sniffer trace, continue taking traces after each hop until you get a trace where packets are missing.
4. When you have isolated the point in the network where the packet loss is occurring, look for any counters on that device that might indicate where the packets are being lost.

Queuing Problems

Queuing delay can be a significant contributor to variable delay (*jitter*). When you have too much jitter end-to-end, you encounter voice quality problems. A voice sample that is delayed over the size of the receiving device's jitter buffer is no better than a packet that is dropped in the network because the delay still causes a noticeable break in the audio stream. In fact, high jitter is actually worse than a small amount of packet loss because most codecs can compensate for small amounts of packet loss. The only way to compensate for high jitter is to make the jitter buffer larger, but as the jitter buffer gets larger, the voice stream is delayed longer in the jitter buffer. If the jitter buffer gets large enough such that the end-to-end delay is more than 200 ms, the two parties on the conference feel like the conversation is not interactive and start talking over each other.

Remember that every network device between the two endpoints involved in a call (switches, routers, firewalls, and so on) is a potential source of queuing or buffering delays. The ideal way to troubleshoot a problem in which the symptoms point to delayed or jittered packets is to use a sniffer trace at each network hop to see where the delay or jitter is being introduced.


For more information on jitter, refer to the [Understanding Jitter in Packet Voice Networks](#) document on Cisco.com.

Features and Applications

This topic addresses various operational features and functions that can be employed in a Cisco Unified Communications system. The following features are discussed:

- [Silent Monitoring and Recording Using Unified Communications Manager](#)

Additional Sites and Services

Steps to Success is a Cisco methodology that outlines the tasks required to complete a successful customer engagement. Registered users can visit the [Steps to Success](#)  resource site for Cisco Unified Communications process flows.

Cisco Unified Communications Services is a Cisco service offering that provides engineering expertise and best practices.

- Registered users can visit the [Cisco Unified Communications Services](#)  partner site.
- Nonregistered users can visit the [Cisco Unified Communications Services](#) site.

