Preface

This preface introduces the *QoS for Voice Over IP Solutions Guide*, which explains quality of service for Voice over IP (QoS for VoIP) and addresses how to deploy end-to-end QoS for voice traffic throughout the components of an internetwork. It also identifies who should use this guide and how it is intended to be used. Read this preface to determine if this guide is appropriate for your internetwork requirements.

About the QoS for Voice over IP Solutions Guide

Because they are real-time based, voice applications tolerate minimal packet delay and loss. Cisco IOS QoS features collectively embody techniques that offer the means to provide priority service that meets the stringent requirements of voice applications. In describing why and how you should deploy QoS for VoIP throughout your network, the guide does the following:

- Gives an overview of QoS for VoIP and describes applicable QoS features.
- Explains the optimum approaches to take in applying QoS for voice applications in the campus (enterprise) network using Frame Relay or PPP across 64 K or T1 lines.

A later version of this guide will include an overview of the internetwork topology used throughout this book to illustrate end-to-end QoS for VoIP. The later version will also include scenarios describing corporate use of an Internet service provider (ISP) for long-distance voice communication using ATM or using Packet over Sonet (POS).

Who Should Use This Guide?

You should use this guide if your network is configured to support VoIP applications concurrent with data applications or if you intend to configure your network as such, and you fit the following described audience profile. The audience for this publication should understand basic networking principles and terminology, and should have hands-on experience in administering a network.

The assumed target audience for this guide is broadly characterized as follows:

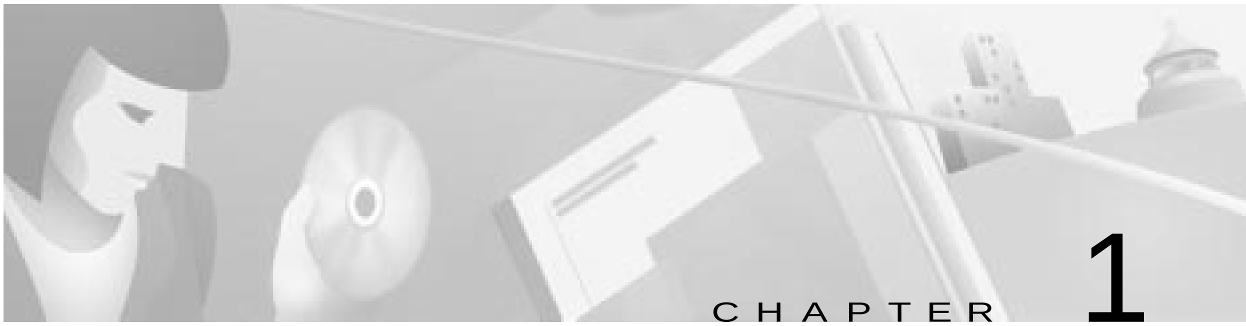
- System administrators responsible for installing and configuring networking equipment that are familiar with the fundamentals of router-based internetworking, and are familiar with Cisco IOS software and Cisco products.
- System administrators that have substantial background in configuring networks, but that might not have experience with Cisco products and Cisco-supported protocols.
- Customers with technical networking background and experience.

This guide does not require that users be familiar with QoS concepts or protocols or how they apply to VoIP. This guide gives an overview of the QoS features and protocols specific to VoIP. For those users new to Cisco products and Cisco-supported protocols, refer to the *Quality of Service Solutions Configuration Guide*, which belongs to the Cisco IOS Reference Library (http://www.cisco.com/univercd/cc/td/doc/product/software/ios120/12cgcr/qos_c/index.htm) for additional QoS concepts.

How to Use This Guide

The first two chapters of this guide provide conceptual information; the last chapter gives examples of how to apply this information to configure end-to-end QoS for VoIP paths throughout a campus (enterprise) network. Reading the entire guide will enable you to identify which QoS protocols are appropriate for your network. Use this guide in the following way:

- To gain understanding of the issues entailed in configuring a network for concurrent voice application and data application support using VoIP for voice traffic, read Chapter 1.
- For detailed information about the QoS features applicable to voice applications using VoIP, read Chapter 2.
- For illustration of how to apply these QoS features to paths throughout an enterprise network across various link types and speeds, read Chapter 3.



CHAPTER 1

QoS for Voice over IP Solutions Overview

This chapter briefly discusses how the rapid and widespread movement toward integrated transport of voice and data across IP has brought forth specific requirements and challenges best addressed by strategic deployment of quality of service (QoS) technologies. Many of these challenges exist because the requirements of real-time voice applications are so different from those of traditional data applications.

This chapter explains the fundamental requirements intrinsic to end-to-end internetwork transportation of packetized voice traffic and why deployment of QoS features is necessary to meet voice traffic requirements and adequately surmount the challenges inherent in integrating voice and data transport.

This chapter includes these sections:

- About Integration of Voice and Data in Internetworks
- About QoS for VoIP
- About the Basic Requirements for Voice Traffic

About Integration of Voice and Data in Internetworks

Corporations are integrating transport of voice and data communication across the same infrastructure for fiscal as well as technological advantage. Some companies are designing entirely new voice-and-data integrated internetworks. Other companies are overhauling their traditional data networks, redesigning them to include infrastructure to support packetized voice transmission.

Companies that carry data traffic that exceeds voice traffic in volume design their networks principally for data transport. These companies build into the design, as a secondary requirement, the ability to also carry voice traffic. Other companies give preference to voice traffic. Thus, companies take various approaches to integration of voice and data traffic in their networks.

Geographically dispersed enterprises with large WAN networks are migrating to Frame Relay (FR) routed networks and ATM switched networks because these networks support both voice and data traffic. Enterprises that depend on Systems Network Architecture (SNA) and other transaction-oriented protocols are migrating to IP networks to establish infrastructure for voice transmission.

The Cisco Voice over IP (VoIP) technology transcends the differences among these transport media and mechanisms because the lower-layer media used is transparent to an IP infrastructure. For VoIP, the underlying technology might be ATM, FR, point-to-point lines, POS, or a WAN link. In fact, many internetworks include all of these media types. Cisco IOS operates with all of these link layer technologies, creating interoperability at the both the IP and link layers, integrating them to produce end-to-end solutions.

The *Quality of Service for VoIP Solutions Guide* focuses exclusively on use of the Cisco VoIP to provide end-to-end QoS support for voice traffic. You should read this guide if your network carries voice traffic today or if you plan to implement support for voice traffic.

About VoIP

This section describes VoIP, and it suggests why you should use VoIP for voice transmission. It includes these subsections:

- What Is VoIP?
- Why Use VoIP for Packetized Voice?

What Is VoIP?

VoIP enables Cisco routers and switches to carry telephony-style voice traffic—that is, live, packetized voice traffic such as telephone calls—over IP-based data networks (intranetworks or internetworks) rather than Public Switched Telephone Networks (PSTN). Cisco routers and switches are equipped to handle origination, transport, and termination of VoIP traffic. VoIP enables toll bypass, remote PBX presence over WANs, unified voice and data trunking, and POTS-Internet telephony gateways.

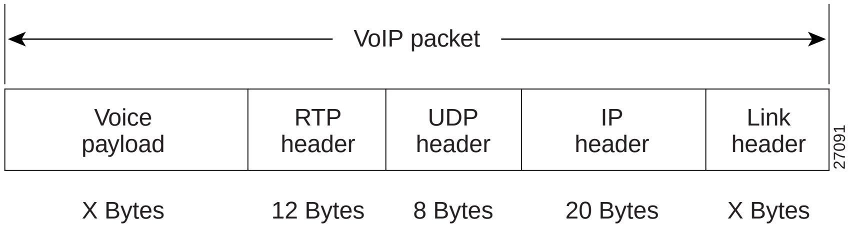
VoIP is IP-based. IP is considered a connectionless or best-effort transport when used in conjunction with the User Datagram Protocol (UDP). UDP, which is datagram-based (or connectionless), suits the specific requirements of voice traffic, so it is used as the transport for VoIP rather than TCP. UDP is preferable for voice despite the fact that TCP, which is connection-oriented, is considered a more ideal transport mechanism because of its built-in reliability.

Consider the underlying reasons for using UDP as the transport for voice traffic:

- Retransmission of dropped packets—the behavior in TCP—is far worse for delay-sensitive voice traffic than is packet loss.
- Because UDP is stateless, it removes from the CPU the burden of overhead entailed in maintaining state on connection-oriented protocols such as TCP.
- From an application perspective, VoIP uses small-sized packets that are sent out at consistent intervals depending on the digital signal processor (DSP) and codec (coder-decoder) used. The UDP header, which is 8 bytes long, is smaller in size than the TCP 20-byte header and thus costs less in bandwidth and overhead.

Figure 1-1 shows the VoIP packet.

Figure 1-1 VoIP Packet Structure



TCP offers reliability in that it guarantees retransmission of lost frames, but this reliable delivery is useless in the internetwork transportation of packetized voice because a frame that arrives late as a result of retransmission is as useful as no frame at all—that is, it has no effect. In other words, retransmission of packets is not meaningful. By the time the resent packet arrives at the end user endpoint, the required delivery time has long been transgressed.

Why Use VoIP for Packetized Voice?

The many reasons to use VoIP for voice traffic include the following:

- Because IP is ubiquitous, it provides contiguous connectivity independent of the media transport that carries it.

  Use of VoIP transcends the differences among various transport media and mechanisms because the media used is transparent to an IP infrastructure. The contiguous connectivity of IP offers an important benefit to real-time applications that is not available through direct use of other Cisco technologies, such as Voice over ATM (VoATM) or Voice over Frame Relay (VoFR).
- VoIP traffic is easily integrated with traffic from modern applications such as unified messaging or virtual call centers.
- As a technology for transporting voice calls, VoIP packet-switched traffic offers cost benefit over circuit-switched networks. One reason for this cost benefit is that Cisco IOS IP-based networks are less expensive to build and maintain than are circuit-switched networks.

About QoS for VoIP

This section briefly explains QoS and its purposes. Then, it explains why QoS is necessary for voice traffic.

Cisco IOS QoS features collectively embody techniques that you can employ to meet the stringent requirements of voice traffic delivery, including curtailment of packet loss and constancy of delay. They offer the means to provide priority service through service differentiation, a derived or secondary benefit of which is the ability to offer customers different classes of service with different cost structures.

This section includes these subsections:

- What Is QoS?
- Why Is QoS for VoIP Necessary?

What Is QoS?

QoS refers to the ability of a network to provide better service to selected network traffic over various underlying technologies. QoS is not inherent in a network infrastructure. Rather, you must institute QoS by strategically deploying features that implement it throughout the network.

Effective end-to-end QoS throughout an internetwork must serve disparate users, applications, organizations, and technologies, all at a reasonable cost and effort. QoS technologies for VoIP enable you to balance service levels for user satisfaction—granting priority service to voice, for instance, while servicing data transmission to the degree of fairness that you require—with efficient backbone and access utilization to minimize operations expenses.

QoS features for voice that implement reliability and predictability eliminate poor quality voice transmission, including crackles and missing syllables that render the call unsatisfactory (even incoherent) to the recipient. For a voice application, minimal QoS support consists of mechanisms that provide these assurances:

- Reliability, which ensures voice packet delivery without packet loss.
- Predictability, which promises voice packet delivery without an excessive amount of delay. (Delay is often expressed in distorted reconstruction of the transmitted conversation.)

QoS features offer other advantages for transmission of voice traffic. For instance, use of QoS for voice gives Internet Service Providers (ISPs) the means to offer their customers differentiated services with different associated costs. ISPs can offer a spectrum of new applications and additional paid-for services based on these levels of service. Without differentiated services, most ISPs offer a standard $20 a month service to residential subscribers. Use of a standard fee significantly reduces profit margins afforded the ISP, limiting any revenue gains the ISP might accrue to revenues from a small number of business clients.

Why Is QoS for VoIP Necessary?

With increasingly pervasive and heavy use of the Internet and intranets, deployment of QoS for voice becomes a fundamental necessity. In traditional voice and data terminal networks, data flow and throughput were predictable. Network usage today makes it hard to predict data flow and to time bursts of data.

Moreover, networking equipment and end stations that carry both data and voice cannot differentiate traffic that requires high-priority connections from traffic that does not require priority service. Without QoS, it is impossible to ensure that voice traffic (considered critical traffic) is expedited or that it will receive constant, predictable transmission performance across a backbone shared by data traffic.

The requirements and behaviors intrinsic to the transmission of voice versus data across an internetwork differ in a number of ways. Here is how they compare:

- Data is bursty by nature, while voice is deterministic (smooth).
- TCP-based data applications react to dropped packets, while UDP-based voice applications can only conceal dropped packets.

  Data applications respond to dropped packets with some degree of correction because often they are TCP-based (TCP resends dropped packets). Voice (which relies on the best-effort transmission of UDP) cannot truly respond to and recover from packet loss, although in some cases the complex algorithms underlying voice transmission can conceal packet loss.
- Data is delay-insensitive, while voice is delay-sensitive.

  Delay-insensitivity means that data applications can tolerate delay well because they are not real-time-based. Voice responds negatively to delay, creating so-called “holes” in the transmission as heard by the receiver.

These differences alone mandate use of QoS strategies for internetworks that carry both voice and data.

Effective transport of voice traffic over IP must entail reliable delivery of packets with low latency. Because VoIP appropriately uses UDP/RTP as its transport and UDP is not reliable, other mechanisms must be put in place to ensure reliable delivery of voice packets. QoS features offer strict priority service to voice traffic to ensure reliable delivery.

Transmission of voice packets, usually small in size, ranging from 80 to 256 bytes, can be unduly delayed between large data packets unless QoS techniques such as packet fragmentation and interleaving are used.

About the Basic Requirements for Voice Traffic

This section identifies packet delay and packet loss as the two most stringent requirements that characterize voice traffic transmission. To gain sufficient understanding of why these requirements must be met for acceptable transmission of voice traffic, see "Basic Requirements for Voice Traffic."

It is not necessary to read the details on delay and loss in order to understand the QoS features for voice, which are described in Chapter 2. However, understanding loss and delay in detail helps explain why and how certain QoS features are used under certain circumstances for integration of voice and data traffic.

This section includes these subsections:

- Basic Requirements for Voice Traffic
- About Delay
- About Loss

Basic Requirements for Voice Traffic

Voice traffic is intolerant of packet loss and delay primarily because these conditions degrade the quality of voice transmission delivered to the recipient end user. Delay must be constant for voice applications. The complete end-to-end absolute delay budget for voice traffic is 200 milliseconds (ms).

Here are some voice application requirements that address loss and delay:

- The network must provide strict policing of traffic.
- Bandwidth for voice traffic must meet minimal requirements.
- Voice traffic requires priority service over large data packets using the same link.

About Delay

Here are some causes of voice packet delay at the campus edge and egress switches and routers:

- Congestion
- Lack of traffic shaping
- Large packet serialization on slow links
- Variable size packets

Here are some causes of delay in the WAN:

- Global WAN congestion
- Central-to-remote site speed mismatches (that is, transmission of voice and data from a fast link to a slow one without adequate traffic shaping)
- Oversubscription of PVCs
- Bursting above committed rates

Two characteristic types of delay affect voice traffic: absolute delay and delay variation. Absolute delay is essentially the time it takes for voice packets, or speech, to travel from the source to the destination. Delay variation (jitter) is delay in which the absolute delay from the source to the destination varies from packet to packet. Variation in delay occurs because of variation of interpacket arrival time. Even though absolute delay might be minimal, a variation in this delay on a packet-by-packet basis can degrade voice quality.

Absolute delay can interfere with the standard rhythm or cadence of a phone call. Variation in delay can impact speech quality. If the wait between when signal elements are sent and when they arrive varies, voice traffic no longer will be synchronized or it may fail. (In other words, a slight time or phase movement in a transmission signal can introduce loss of synchronization.)

Two sources of delay are handling delay and propagation delay. If the amounts of these kinds of delay vary, they contribute to delay variation. Handling delay is incurred as the result of a process such as encoding (codec). Analog voice undergoes encoding during its conversion to digital information before it is packetized. As mentioned previously, handling delay can also occur when a voice packet is moved to the outbound queue for transmission. (This type of handling delay, which is called serialization delay, can occur on a hop-by-hop basis.) Propagation delay can also occur when a voip packet is moved to an I/O queue for transmission.

Another factor contributing to delay is latency. Latency refers to the time between when a device requests access to a network and when it is granted permission to send. End-to-end latency describes the overall delay associated with a network. Serialization delay is an aspect of latency that addresses the time it takes to send a packet out an interface—that is, the time it takes to move the actual packet to the output queue. The time it takes to put voice traffic onto a transmission line depends on the data volume and the speed of the line—for instance, it takes about 5 ms to send a 1024-byte packet on a 1.544–Mbps T1 line.

Note You should hold output queue delay to under 10 ms if possible through use of the most optimal QoS queueing feature for the node and network.

The effect of serialization delay can be such that a single link can cause enough delay to exceed the entire end-to-end 200–ms delay budget for voice traffic.

Here are two causes of serialization delay:

- The encoding process and the codec used. For instance, the G.729 codec, which is a type of compression that enables voice to be coded into 8-kbps streams, has an algorithmic delay of about 20 ms. (Different codec compression methods introduce different amounts of delay.)
- Packetization. VoIP supports a variable payload size, allowing you to specify how many bytes of payload should be included in each voice packet. In the Cisco IOS VoIP product, the DSP generates a frame every 10 ms. You can decide how many frames you want to send in one packet. Larger payloads reduce the packet-per-second load of each voice channel, which is traded off against delay of the voice connection.

Packet-switching is another underlying source of delay. Packet-switching delay refers to the latency accrued when bridges, switches, and routers forward data. The latency depends on the speed of the internal circuitry and CPU, and the switching architecture of the internetworking device.

About Loss

Networks can drop voice packets for a number of reasons under different circumstances, even under circumstances meant to provide benefits. Here are some examples of ways packet-drop problems can be introduced by strategies otherwise beneficial to data traffic:

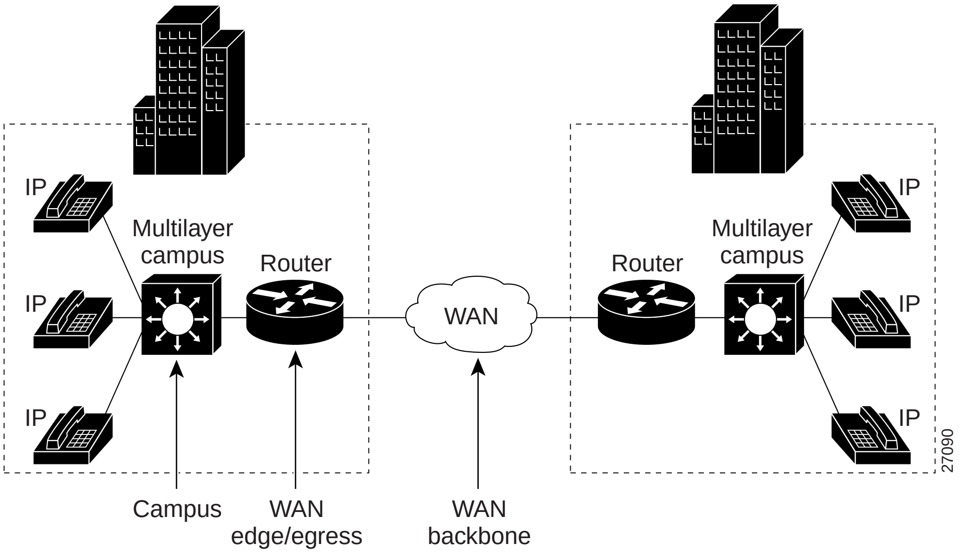
- On Frame Relay networks, the committed information rate (CIR) specifies the guaranteed amount of information carried during periods of congestion. During bursting over the CIR—a beneficial, common, and intentional practice in a data-only Frame Relay network—voice packets are sent out into the Frame Relay network essentially in a best-effort manner, subjecting them to packet drop. (Configuring traffic shaping—which is applicable to Frame Relay and ATM networks only, not leased lines—ensures that the CIR is not exceeded, thus avoiding occurrence of packet drop under these circumstances.)
- Oversubscription, another commonly used Frame Relay design implementation on data-only environments, makes it possible for many remote sites to feed into a central site. Depending on network conditions, oversubscription puts at risk the quality of voice traffic transmission. Under conditions of oversubscription, the aggregate line speeds and CIR of the remote sites can easily exceed the line speed (link bandwidth) of the central site circuit. Problems affecting voice traffic can also occur if the CIRs of the remote sites all equal less than the central site link speed but the bursting at the remote sites exceeds the central site link speed. If you run voice traffic over a network designed along these lines and the amount of traffic from the remote sites exceeds the circuit speed buffering at the central site, delay will result. Moreover, if the remote-to-central burst period is large enough, packet drop might also occur. (To eliminate congestion resulting from the oversubscription of the remote sites to the central site in order to avoid delay and packet drop, use traffic shaping from the remote sites.)

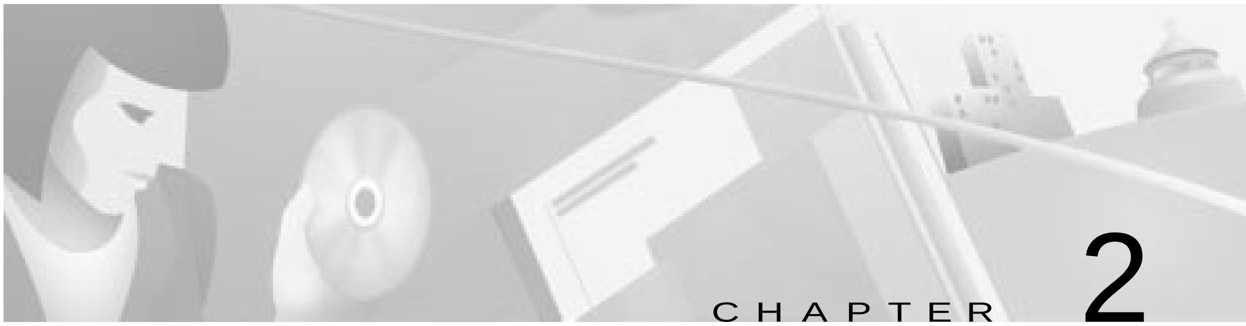
To avoid packet loss that can severely degrade voice quality, you should deploy mechanisms that inhibit its occurrence, such as strict priority service.

To configure guaranteed strict priority for voice traffic, you can use the IP RTP Priority feature on Cisco 2600, 3600, and 7200 series systems running release 12.0(7)T or later. This feature allows you to specify the exact amount of bandwidth allocated for the priority queue used to handle voice flows. IP RTP Priority closely polices use of bandwidth and if the configured amount is exceeded, IP RTP Priority drops voice packets. (Allocating more than the exact requisite bandwidth for the voice flow—taking into account the type of codec compression used and the interface characteristics—protects against occurrence of packet drop under these circumstances.)

Packet loss is most likely to occur in the part of the network referred to as the router egress into the WAN, although it can occur anywhere in the network. Figure 1-2 shows a basic representation of an internetwork composed of two campus networks communicating across a WAN. Notice that the router at the edge of the campus on the left is the egress into the WAN to its right. It is here that you would configure QoS features to inhibit packet loss.

Figure 1-2 Internetwork Domains and QoS Considerations





CHAPTER 2

QoS Features for Voice over IP

Cisco IOS QoS includes a rich set of features designed to enable you to deploy mechanisms that deliver quality of service throughout your network. Many of these features address the requirements of end-to-end QoS and service differentiation for voice packet delivery. The subset of QoS features for Voice over IP (VoIP) includes technologies that enable you to do the following:

- Classify traffic in order to differentiate it
- Police traffic and shape it
- Manage traffic congestion when it occurs
- Configure the system to avoid congestion where possible
- Fragment large data packets and interleave them with voice packets to meet the delivery requirements of voice
- Offer bandwidth guarantees and reservation to high-priority voice traffic

Thus, QoS for VoIP entails deploying features that ensure no loss, low and constant delay, no or minimal jitter, and guaranteed bandwidth—requirements for voice explained in Chapter 1, “QoS for Voice over IP Solutions Overview.”

Cisco IOS QoS for VoIP features have the following properties:

- They are best deployed at different points in the network and are designed to be used in conjunction with other QoS features to achieve goals such as control over jitter and delay.
- They are designed to support packet transmission over certain link types. (Not all QoS for VoIP features are supported on all platforms.)

Complete details for the QoS features introduced in this chapter, including configuration information, are provided in the appropriate Cisco IOS configuration and command reference documentation.

Cisco IOS QoS software includes these major feature categories applicable to VoIP traffic:

- Congestion Management
- Link Efficiency Mechanisms
- Congestion Avoidance
- Traffic Shaping and Policing
- Classification
- IP to ATM CoS
- Signalling

Note Cisco IOS software is enhanced and extended on a continuing basis to include new QoS features, many of which are being implemented or planned for near-future implementation. Consult with your support engineer (SE) to determine if releases of Cisco IOS software later than 12.0(5)T support additional QoS techniques applicable to voice exceeding those described in this guide.

Congestion Management

A network carrying voice traffic also carries data. Voice and data traffic sharing a path through the network can interact with one another in ways that affect the application performance of each, sometimes resulting in network congestion and packet loss. Congestion results from a sustained overload of traffic and it can manifest in performance degradation and packet loss unacceptable in the case of voice traffic delivery.

Congestion management features have the following properties:

- They operate to control congestion once it occurs.
- The embody queueing and scheduling disciplines that allow individual connections such as those used for voice to obtain guarantees on bandwidth, delay, and jitter, thus enabling guaranteed service that meets the performance requirements of a voice application.
- They support cooperative transmission of voice and data across a single path between routers.

To control congestion once it occurs, you can implement strategies using queueing and scheduling features. The use of queueing and scheduling mechanisms to meet specified bandwidth allocation or delay bounds applies to both the output of the edge devices and the core devices of the network.

Once you deploy congestion management features by configuring them, the techniques dynamically tailor themselves to existing network conditions as congestion arises. Deployment of congestion management features throughout your network allows you to ensure that time-critical voice traffic receives the priority transmission it requires.

About Queueing and Scheduling

When voice traffic is to be carried on packet networks, queueing generally functions to give voice priority over data traffic. Queueing is a mechanism that packet-based networks use to absorb bursts of traffic that are in excess of trunk bandwidth; packets awaiting transmission are buffered, or queued. Queueing is only necessary if congestion can occur in the network. When there is more trunk bandwidth available than traffic using it—say, up to 50 percent utilization—and trunk bandwidth allows several data frames to be queued before a voice frame without undue transmission delay to the voice frame, any configured queues would be empty or near-empty, and thus not needed.

A scheduling discipline determines which queue to service next. It decides the order in which the switch or router serves the buffered data. It allocates different delays to different users by its choice of service order, and it allocates different loss rates to different users by its choice of which requests to drop.

Apart from guaranteed strict priority service, most queueing and scheduling techniques enable you to grant voice traffic standard priority service. Standard priority treatment is not strict priority and the techniques that provide it must also address the needs of other enqueued data. For this reason, you must use other QoS features such as fragmentation and interleaving to fragment data packets so as to reduce their size and interleave the fragments with voice packets.

Cisco IOS QoS software offers many congestion management protocols for different platforms whose features address the requirements of voice traffic while ensuring that data transmission is not penalized. This section describes the following congestion management features:

- MDRR (Modified Deficit Round Robin)
- WRR (Weighted Round Robin)
- WFQ (Weighted Fair Queueing)

  There are two kinds of standard WFQ and three kinds of Distributed Weighted Fair Queueing (DWFQ):

  – Flow-Based WFQ
  – CBWFQ (Class-based Weighted Fair Queueing)

  There are three kinds of DWFQ:

  – Flow-Based DWFQ
  – QoS Group-Based DWFQ
  – ToS-Based DWFQ
- IP RTP Priority (Internet Protocol Real-Time Transfer Priority)
- Priority Queueing within CBWFQ

The last two features allow you to reserve a queue for voice. IP RTP Priority grants strict priority based on port range, and priority queueing within CBWFQ grants strict priority based on a wide range of criteria that you can use to determine what constitutes a class.

How Do WFQ, DWFQ, and CBWFQ Apply to Voice Traffic?

Availability of strict priority for voice traffic through use of IP RTP Priority in Cisco IOS Release 12.0(5)T renders use of WFQ as a queueing and scheduling mechanism far less essential and necessary than it was in prior releases. You can use IP RTP Priority to specify voice traffic to be enqueued in the strict priority queue. Also, you can use the priority queueing within CBWFQ feature to configure a class for strict priority and control the bandwidth allocated to that class. For instance in CBWFQ, you can give high bandwidth to a class, thereby giving it very low weight.

WFQ by design gives fair treatment to all classes. This aspect alone poses problems for voice traffic because voice traffic requires priority treatment.

However, WFQ and DWFQ are still useful for voice traffic on fast links that do not support the IP RTP Priority feature. Marking voice packets with a priority of 5 will still give them some degree of precedence over data packets marked with lower weights. If you use WFQ on a fast link, you should also configure the link for packet fragmentation. Because the link is fast, the fair queueing treatment afforded data packets, which can slow down voice packet transmission, is less perceptible than it would be on a slow link. You should avoid configuring WFQ (or DWFQ) on slow links if you have other choices for scheduling and queueing.

Congestion Management Features Supported in Versions of Cisco IOS Software

For each congestion management feature, Table 2-1 shows the versions of the Cisco IOS software that support the feature, the switching mode used, and the platforms the feature runs on.

Terms used in Table 2-1 are explained as follows:

- “All Cisco IOS platforms” refers to this set of platforms: 1000, 1600 series, 1720, 2500 series, 2600 series, 3600 series, 4500 series, 4700 series, 7200 series, and RSP in the 7500.

- "VIP distributed" refers to this set of platforms: 7000, 7500, and RSM.
- The following abbreviations are used to represent various switching modes in this table:
  - P = Process
  - F = Fast
  - N = NetFlow
  - O = Optimum
  - CEF = Cisco Express Forwarding
  - d = distributed (VIP)
  - dCEF = distributed CEF

Table 2-1 Cisco IOS Version, Switching Modes, and Platform Support for Congestion Management Features

Feature	Cisco IOS Version and Switching Mode							Platform Support
	10.3	11.0	11.1	11.2	11.1CC	11.3	12.0	
Congestion Management								
WFQ	—	P, F, N, O	P, F, N, O	P, F, N, O	P, F, N, O	P, F, N, O	P, F, N, O	All Cisco IOS platforms
CBWFQ	—	—	—	—	—	—	—	All Cisco 12.0(5)T IOS platforms
Distributed WFQ (Flow-based, ToS-based and Class-based)	—	—	—	—	dCEF	—	dCEF	VIP distributed
WRR	—	—	—	—	—	—	—	8500 series
MDRR	—	—	—	—	—	—	—	1200
IP RTP Priority	—	—	—	—	—	—	—	With serial and ISDN, IP RTP Priority in 12.0(5)T. For FR with FRTS in 12.0(7)T.
Priority Queueing within CBWFQ	—	—	—	—	—	—	—	12.0(7)T

MDRR

MDRR extends DRR to provide special support for delay sensitive traffic, such as VoIP, on the Cisco 12000 GSR series routers. MDRR includes a low-latency, high-priority (LLHP) queue that is treated differently from the other queues associated with service classes. This special queue is used to handle delay-sensitive traffic.You can configure MDRR for strict priority handling of the LLHP queue. If the queue contains packets, it is serviced first until all of its packets are sent. Within MDRR, IP packets are mapped to different class-of-service queues based on precedence bits. The queues are serviced in round-robin fashion except for one, the special queue used to handle voice traffic. You can configure WRED for each of the MDRR queues, specifying a discrete WRED profile in each case.

MDRR Overview

DRR is a packet queueing and scheduling protocol designed to provide features similar to those provided by WFQ such as class and flow differentiation, bandwidth allocation, and delay bounding, but for high-speed transport links operating at OC-3, OC-12, and higher. MDRR extends the DRR protocol to include a high-priority queue that is treated differently from the other queues associated with service classes.

For each set of CoS queues supported, MDRR includes an LLHP queue designed to handle special traffic such as voice in a manner that is different from the other queues associated with service classes. Except for the LLHP queue, MDRR services all queues in round-robin fashion.

Using the command-line interface, you can define MDRR to be used in either of the following two modes: strict priority or alternate priority.

Strict Priority Mode

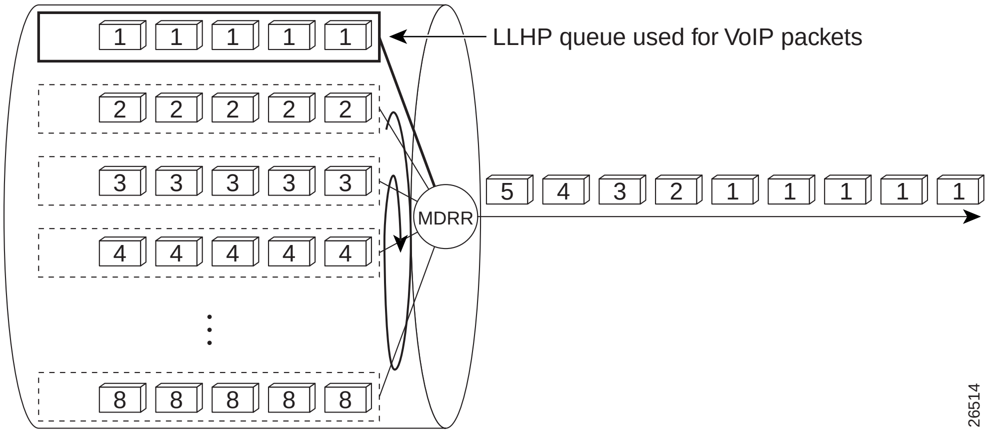
Also referred to as high priority mode, in this mode, if the LLHP queue contains packets, it is serviced first until all of its packets are sent and the queue is empty. Then the CoS queues are serviced in round-robin fashion according to the DRR algorithm.

Using the LLHP queue in high priority mode ensures the lowest possible delay for low-latency, high-priority packets. However, high priority mode incurs the risk that the CoS queues might not be serviced for extended periods of time, especially if the LLHP queue itself utilizes a large portion of the bandwidth. To avoid starvation of the CoS queues, when you use the LLHP queue in high priority mode, you should combine it with careful rate limiting of high-priority, low-latency packets.

Figure 2-1 shows MDRR configured for strict priority mode. All voice packets (labeled 1) in the strict priority LLHP queue used exclusively for VoIP are sent (exhaustively) before packets in the other eight queues are serviced in round-robin fashion. In other words, when the LLHP VoIP queue is empty, the other queues are serviced in round-robin fashion.

In Figure 2-1, all of the voice packets (labeled 1) in the LLHP queue will be serviced before any of the packets in the other queues.

Figure 2-1 MDRR Strict Priority Mode



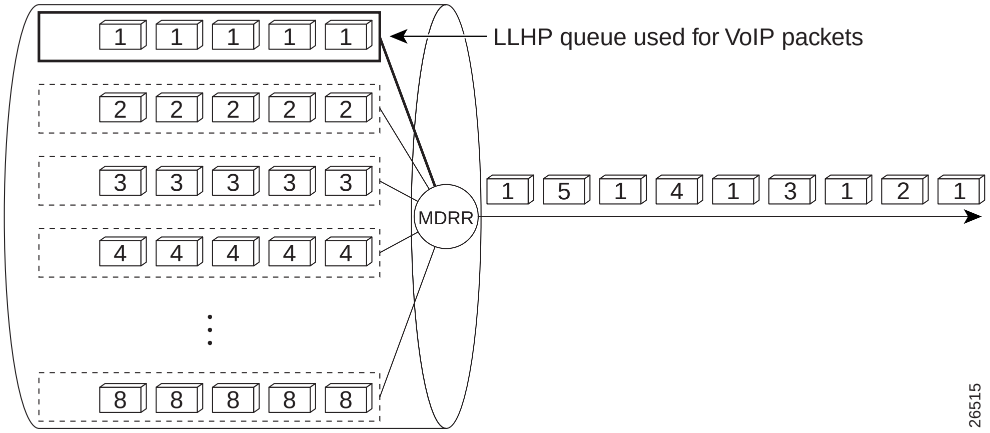
Alternate Priority Mode

Also referred to as fair priority, in this mode, service alternates between the LLHP queue and the CoS queues. Packets are serviced from the LLHP queue and then from an active CoS queue that is selected from among the CoS queues in round-robin fashion. This process of servicing the queues is repeated—service alternates between the LLHP queue and the next CoS queue, the LLHP queue and the next CoS queue, and so on—until the queues are empty. Alternate, or fair, priority mode does not delay indefinitely delivery of traffic in the CoS queues, hence its name. The CoS queues are serviced fairly in relation to one another, with the LLHP queue receiving alternating priority. This mode guarantees that all queues are serviced but at the expense of some latency on the LLHP queue.

Figure 2-2 shows MDRR configured in alternate priority mode: a single VoIP packet, labeled 1, enqueued in the LLHP queue, is serviced first, then a packet from queue 2, then another VoIP packet from the LLHP queue followed by a packet from queue 3, then another VoIP packet from the LLHP queue, then a packet from queue 4, and so on. Whenever the LLHP queue is not empty, its packets are serviced in this fashion.

When VoIP packets are enqueued, service is returned to the LLHP queue. As you can see in Figure 2-2, every other packet on the line is a voice packet.

Figure 2-2 MDRR Alternate Priority Mode



Advanced Concepts

Because it is difficult to scale up WFQ for high-speed transport links running at OC-3, OC-12, and higher rates, MDRR, a variant of DRR, is implemented for the Cisco 12000 GSR router to support delay-sensitive voice traffic.

This section gives explains how DDR works; then it explains how MDRR extends the functionality for delay-sensitive traffic, such as voice. For complete information on how to configure MDRR, see the Cisco IOS documentation.

For DRR to enact round robin fashion, each queue has assigned to it a configurable value called a service quantum. A service quantum provides a measure of how much traffic should be handled from the queue in each round. Packets from that queue are serviced until their cumulative length (byte count) exceeds the service quantum.

A deficit counter, which is a memory mechanism designed to enhance fairness and packet size independence, is used as a credit mechanism. The deficit counter value is added to the service quantum to determine the measure of service available for each queue during each round. For example, given the round-robin algorithm described later in this section, in a particular round a queue may not be able to get a full quantum worth of service because the next packet to be dequeued is larger than the remaining amount allowed to be serviced as specified by the remaining quantum. In this example, the deficit is used in addition to the quantum the next time the queue is serviced.

These are the basic steps that define how DRR works:

1. Packets are classified based on IP precedence and inserted in the appropriate queues.
2. Active queues are serviced in round-robin order:
    a. Deficit counter values for each queue are initialized to 0.
    b. The configured quantum size is added to the deficit counter of the first queue. The first packet in the first queue is dequeued and the deficit counter is decremented. This process repeats until the queue is empty or the deficit counter goes negative. A full packet is serviced even if the deficit counter runs out during the processing. If there is a remaining deficit, it is added to the quantum to be used to service the queue next round.
    c. The process described in Step b is repeated for this queue and so on, for each successive queue.

       If the receive (input) interface is an engine 0 card, for example, which supports up to 16 slots depending on the type of chassis used, there are 8 queues per slot. On the transmit (output) side, there are 8 queues per interface. For each set of 8 queues, you can configure whether the LLHP queue is used in strict priority mode or alternate priority mode. Data is sorted and enqueued from the receive queue to the appropriate transmit queue. MDRR maps IP traffic to different CoS. That is, it enqueues packets based on the IP precedence value of the packet.

These are the basic steps that define how MDRR works as a modification to DRR:

1. If MDRR is configured for high priority mode and the LLHP queue contains packets, MDRR services that queue first. If MDRR is configured for fair priority mode, a queue other than the LLHP queue was last serviced, and the LLHP queue contains packets, then the LLHP queue is serviced first; if the LLHP queue is empty, then the next active CoS queue in round-robin fashion is serviced.
2. The deficit counter for the queue is incremented for the queue to be serviced.
3. Packets from the queue are serviced until the until the queue is empty or the deficit counter goes negative. The remaining deficit, if any, is added to the quantum to be used to service the queue next round.
4. The process described in Step 3. is repeated for this queue and so on, for each successive queue.

WRR

WRR is a packet queueing and scheduling algorithm used on the Catalyst 8500 series switches. It provides class differentiation, bandwidth allocation, and delay bounding— features that make it possible to give voice packets premium service, although not strict priority. WRR interprets IP Precedence to assess how packets are classified. (You cannot use WRR to mark or change the IP Precedence of a packet.)

Overview

WRR queues and schedules packets at the campus backbone on Catalyst 8500 series switches. WRR interprets the ToS field of the IP header of the packet and enqueues packets based on the ToS value. Unlike WFQ, which recognizes seven IP Precedence categories, WRR classifies packets into four categories based on the first two (most significant, high-order) bits of the ToS field. WRR reserves four associated queues for enqueuing packets belonging to these four classes.

Packets sent to a core Catalyst 8500 switch have been previously marked for IP Precedence at the edge of the network. For instance, in the example topology used in this guide, the IP Precedence for voice packets delivered to a Catalyst 8500 at the core might be marked at the edge using Cisco IP phones.

Table 2-2 shows how WRR running on the Catalyst 8500 series switches uses the IP Precedence bits, mapping them into four categories, each of which has its own transmission queue.

Table 2-2 IP Precedence and Associated WRR Queues

IP Precedence Value	IP Precedence Bits	Delay Priority	Associated Queue
0	0 0 0	0 0	Q-0
1	0 0 1	0 0	Q-0
2	0 1 0	0 1	Q-1
3	0 1 1	0 1	Q-1
4	1 0 0	1 0	Q-2
5	1 0 1	1 0	Q-2
6	1 1 0	1 1	Q-3
7	1 1 1	1 1	Q-3

WRR does not explicitly reserve bandwidth for each of these four queues. Rather, each queue is assigned a different WRR scheduling weight, which determines the way the queues share the interface bandwidth.

Although you cannot mark the IP Precedence value of a packet, you can use WRR to configure the weight assigned to a specific class of traffic, such as the traffic class to which voice packets are assigned, in order to give priority treatment to that traffic. In assigning IP Precedence to packets, we recommend that an IP Precedence of 5 be assigned to voice packets. WRR would interpret an IP Precedence of 5 to mean that voice packets should be enqueued to the second WRR queue (queue 2). You can improve the service that WRR gives to voice traffic by using WRR to give voice packets a weight. Packets with a weight of 8 are enqueued to the third WRR queue (queue 3), which gets the highest priority service from WRR.

Table 2-3 shows the default weight assignments to the WRR queues.

Table 2-3 Default WRR Weight Assignments

WRR Queues	Weight
0	1
1	2
2	4
3	8

In configuring WRR, you can assign a different weight to each queue. For WRR, the higher the weight assigned to a queue, the higher the effective bandwidth attributed to that particular queue. This relationship of weight to bandwidth is different from that of WFQ in which a lower weight gets a higher precedence and, thus, more bandwidth. Considering the effect of weight, you can better ensure that voice traffic receives premium treatment by assigning a high weight to the queue used for voice packets. Given that voice packets marked with an IP Precedence of 5 are enqueued to WRR queue 2, you should assign a WRR weight of 8 to WRR queue 2.

You can use the following formula to determine the effective bandwidth in megabits per second (Mbps) for a particular queue:

```
(W / S) x B = number mbps
```

where:

- W = WRR weight of the specified queue.
- S = Sum of the weight of all active queues on the outgoing interface.
- B = Available bandwidth in Mbps.

The weight for any queue is 0 to 15. However, the sum of the WRR weight for all four queues on an interface cannot exceed 15. If the total weight exceeds 15, most the bandwidth of the interface is exceeded.

Advanced Concepts

The WRR scheduler has two main responsibilities within the Catalyst 8500 switch:

- To schedule frames into the switching fabric based on the priority queue being requested.
- To schedule frames out of the switching fabric based on the WRR scheduling algorithm.

As stated previously, you can assign different WRR-scheduling weights to the queues to determine the way bandwidth is to be shared among the queues. Use of weights allows you to provide bandwidth to higher priority applications such as voice based on the IP Precedence of the packet, while fairly granting access to lower priority queues. When queues are weighted and congestion occurs, the WRR schedule affords each queue the bandwidth allotted to it based on its weight.

You can configure the mapping between queues and weights at both the system and interface levels. As is customary, interface-level configuration takes precedence over system-level configuration.

When there is no network congestion, all queues between any pair of interfaces are granted the same weight, and bandwidth is not explicitly reserved for them. Under these circumstances, WRR and the weights provided do not strongly influence how packets are switched out the fabric because there is sufficient bandwidth available. However, WRR scheduling becomes increasingly important when an outgoing interface is congested.

When a link is congested, WRR services each queue per port based on the priority determined by the configured weight. Consider, for example, the weights assigned by a network manager in Table 2-4.

Table 2-4 Sample WRR Priority Weights and Bandwidth

Quality of Service Priority/Queue	Weight Given by Network Manager	Bandwidth Assignment
QoS-0	1	=(1/(8+4+2+1)) x 100
QoS-1	2	=((2/(8+4+2+1)) x 100
QoS-2	3	=((3/(8+4+2+1)) x 100
QoS-3	4	=((4/(8+4+2+1)) x 100

Based on these priorities and weights, WRR services QoS-3 more often.

WFQ

WFQ is a congestion management algorithm that provides priority management, but not strict prioritization for voice, during periods of traffic congestion. WFQ offers a solution that provides consistent, fair response time, based on weights, to heavy and light traffic alike without adding excessive bandwidth. WFQ provides features such as traffic isolation and delay bandwidth guarantees. Implicit within WFQ is a strict priority queue that is created when WFQ is enabled. However, this queue cannot be used until the IP RTP Priority feature is enabled.

Note Because they do not provide the strict priority required by voice traffic, WFQ and DWFQ largely are not useful for voice applications. For conditions under which you should use them for voice traffic, see the "How Do WFQ, DWFQ, and CBWFQ Apply to Voice Traffic?" section in this chapter.

There are two kinds of WFQ:

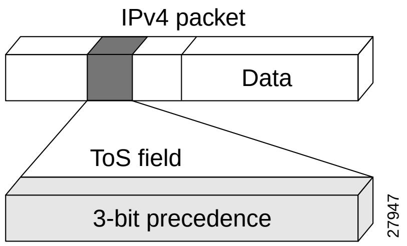
- Flow-Based WFQ
- CBWFQ

Before either of these WFQ types is discussed, this section explains IP Precedence and its use.

IP Precedence

Packet classification is pivotal to techniques that select packets traversing a network element or a particular interface for different types of QoS service. One method of classifying traffic is to mark it with IP Precedence values. IP Precedence allows you to specify the CoS for a packet using the three precedence bits in the ToS field of the IPv4 header. Figure 2-3 shows the ToS field within the structure of the IP packet.

Figure 2-3 IPv4 Packet ToS Field



How It Works

Using the ToS bits, you can define up to eight classes of service. Other features configured throughout the network can then use these bits to determine how to treat the packet in regard to the type of service to grant it. These other QoS features can assign appropriate traffic-handling policies including congestion management strategy and bandwidth allocation. For example, although IP Precedence is not a queueing method, queueing methods WFQ and WRED can use the IP Precedence setting of the packet to prioritize traffic.

By setting precedence levels on incoming traffic and using precedence values in combination with the Cisco IOS QoS queueing features for VoIP, you can create differentiated service. You can use BGP Policy Propagation and Cisco IP phones to set the precedence for VoIP traffic to 5, giving it the highest priority.

Note Even if you plan to use strict priority mode features such as IP RTP Priority and priority queueing within CBWFQ, you should set precedence on voice flows that will traverse an internetwork. This approach is recommended because IP Precedence values are persistent; that is, the voice packet carries the value throughout the internetwork. Some platforms such as the GSR, which is used extensively within the core of many networks, do not support strict priority and may relay on IP Precedence for classification and packet differentiation. For this reason, it is always beneficial to give voice traffic an IP Precedence of 5.

So that each subsequent network element can provide service based on the determined policy, IP Precedence is usually deployed as close to the edge of the network or the administrative domain as possible. You can think of IP Precedence as an edge function that allows core (or backbone) QoS features, such as WRED, to forward traffic based on CoS. IP Precedence can be used to control the dequeueing and scheduling behavior of WFQ.

IP Precedence can be mapped into adjacent technologies, such as ATM or Frame Relay. The Cisco IOS QoS IP to ATM CoS feature, described in the “IP to ATM CoS” section on page 2-44, relies on this capability.

Weights and Bandwidth

In considering class weights, it is helpful to think of them as inversely proportional to the class bandwidths. Therefore, the lower the weight, the greater the bandwidth and better the service given to the class.

Based on the way weights are assessed, each class receives at least 95 percent of the bandwidth configured for it, given the amount of bandwidth assigned to a class is within legal bounds not to exceed 75 percent of the interface bandwidth.

Flow-Based WFQ

Flow-based WFQ, referred to henceforth in this guide as WFQ, is enabled by default on links with speeds of 2 Mbps or less.Without requiring configuration or analysis or that you first define access lists, WFQ automatically sorts among individual traffic streams and categorizes traffic into two kinds of flows: high-bandwidth sessions and low-bandwidth sessions.

WFQ is IP Precedence-aware, meaning that it is able to detect higher priority packets designated by IP Precedence, give them superior response time, and schedule them for faster transmission than other packets. Voice packets are usually assigned a high precedence (Precedence 5) and thus WFQ gives voice traffic better service than data traffic.

WFQ dynamically adapts to changing network traffic conditions. It offers a solution that provides consistent, fair response time, based on weights, to heavy and light traffic alike without adding excessive bandwidth.

WFQ does not distinguish flows by their type, such as Telnet, voice, Systems Network Architecture (SNA), or File Transfer Protocol (FTP). Rather, it categorizes traffic by its flow characteristics, dynamically sorting traffic into messages that make up a conversation. WFQ classifies traffic into different flows based on packet header addressing, using such characteristics as source IP address, destination IP address, source or destination Transmission Control Protocol (TCP) or User Datagram Protocol (UDP) port, MAC address, Frame Relay data-link connection identifier (DLCI) value, and ToS value.

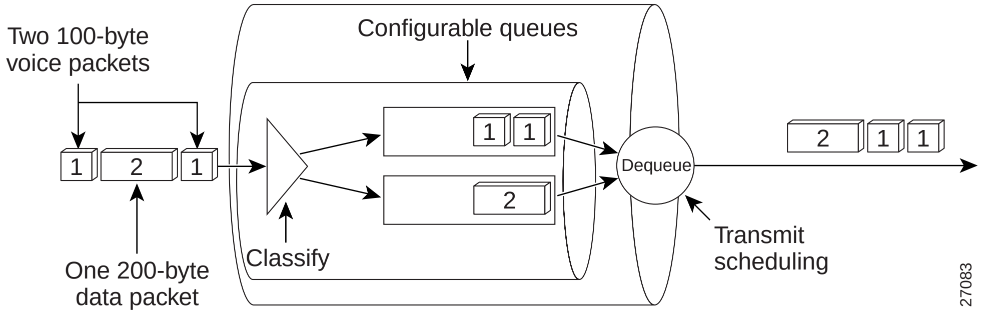
You can think of standard WFQ as fair queueing with the addition of weights:

- It supports per-flow queueing—each flow is relegated to a separate queue.
- It applies weights, or priorities, to identified traffic flows to determine how much bandwidth each conversation is allowed relative to other conversations. The weight that WFQ assigns to a flow determines the transmit order for packets queued in that flow.
- It breaks up the train of packets within a conversation to ensure that bandwidth is shared fairly between individual conversations and that low-volume traffic is transferred in a timely fashion.
- It grants low-bandwidth (low-volume) traffic effective priority over high-bandwidth (high-volume) traffic, and high-bandwidth traffic shares the transmission service proportionally according to assigned weights.

  WFQ apportions bandwidth based on weight. Thus, it does not allow you to guarantee bandwidth for voice applications. WFQ simultaneously schedules interactive traffic such as voice packets to the front of a queue to reduce response time and fairly shares the remaining bandwidth among high-bandwidth flows. Low-bandwidth voice traffic streams receive preferential service, sending their entire offered loads in a timely fashion.

Figure 2-4 shows two flows. In this figure, the voice packets are 100-byte packets; they are smaller than the data packets, which are 200-byte data packets. To fairly schedule the packets for transmission and give equal bandwidth, WFQ sends two voice packets for each single data packet.

Figure 2-4 WFQ Classification and Scheduling



WFQ addresses the problem of round-trip delay variability. If multiple high-bandwidth conversations are active, their transfer rates and interarrival periods are made much more predictable, resulting in more predictable throughput and response time for each active flow.

Advanced Concepts

WFQ manages simplex data streams, such as voice, and duplex data streams such as those between pairs of applications.WFQ segregates traffic into flows and then schedules traffic onto the outputs to meet specified bandwidth allocation or delay bounds.

WFQ cooperates with the RSVP and IP Precedence Cisco IOS software scheduling techniques in the following manner:

- RSVP uses WFQ to allocate buffer space, schedule packets, and guarantee bandwidth for reserved flows. WFQ works with RSVP to help provide differentiated and guaranteed QoS services.

- WFQ is IP Precedence-aware. The IP Precedence field has values from 0 to 7. As the precedence value increases, the WFQ allocates more bandwidth to that conversation to make sure that it gets served more quickly when congestion occurs.

In the weighting scheme for WFQ, lower weights are served first. IP Precedence serves as a divisor to this weighting factor. For instance, traffic with an IP Precedence field value of 7 gets a lower weight than traffic with an IP Precedence field value of 3, and thus has priority in the transmit order.

CBWFQ

CBWFQ extends the standard WFQ functionality to provide support for user-defined traffic classes. CBWFQ runs on Cisco 7200 series routers with NPE150 or higher for T1/E1/Ethernet rates or with NPE200 or higher for T3 rates.

Note Because they do not provide the strict priority required by voice traffic, WFQ and DWFQ are not useful largely for voice applications. See the "How Do WFQ, DWFQ, and CBWFQ Apply to Voice Traffic?" section.

Overview

CBWFQ allows you to define what constitutes a class based on criteria that exceed the confines of flow. Using CBWFQ, you can create a specific class for voice traffic. CBWFQ allows you to use standard and extended numbered access control lists and protocols or input interface names to define how traffic will be classified, thereby providing coarser granularity. You need not maintain traffic classification on a flow basis.

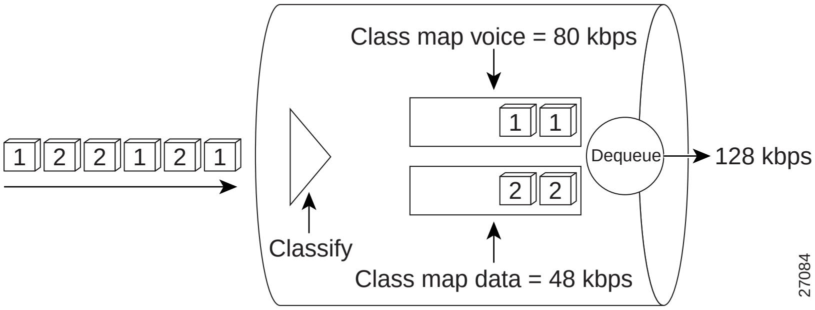
The classes you create determine how packets are grouped in queues. CBWFQ allows you to specify the exact amount of bandwidth to be allocated for a specific class of traffic. Taking into account available bandwidth on the interface, you can configure up to 64 classes and control distribution among them. (This is not the case with WFQ. For WFQ, weights determine how much bandwidth each conversation is allowed relative to other conversations, and the weights and traffic classification are dependent on and limited to the 7 IP Precedence levels.)

For CBWFQ, each class is associated with a separate queue. You can allocate a specific minimum amount of guaranteed bandwidth to the class as a percentage of the link or in kbps. Bandwidth not used can be shared by other classes in proportion to their assigned weights. When configuring CBWFQ, you should consider that bandwidth allocation does not necessarily mean that the traffic belonging to a class will experience low delay; however, you can skew weights to simulate priority queueing.

Packets satisfying the match criteria for a class constitute the traffic for that class. A queue is reserved for each class, and traffic belonging to a class is directed to the queue of that class.

Figure 2-5 shows two queues, the first of which is for voice traffic. Any packet with an IP Precedence of 5 is assigned to the voice class and gets a minimum of 80 kbps of bandwidth on the 128-kbps link.

Figure 2-5 Example Queues for CBWFQ Classes



For CBWFQ, the weight specified for the class becomes the weight of each packet that meets the match criteria of the class. Packets that arrive at the output interface are classified according to the match criteria filters you define, then each one is assigned the appropriate weight. The weight for a packet belonging to a specific class is derived from the bandwidth you assigned to the class when you configured it; in this sense the weight for a class is user-configurable.

After the weight of a packet is assigned, the packet is enqueued in the appropriate class queue. CBWFQ uses the weights assigned to the queued packets to ensure that the class queue is serviced fairly.

Configuring a class policy—and, thus, configuring CBWFQ—entails these processes:

- Defining traffic classes to specify the classification policy (class maps).

  This process determines how many types of packets are to be differentiated from one another.

- Associating policies—that is class characteristics—with each traffic class (policy maps).

  This process entails configuration of policies to be applied to packets belonging to one of the classes previously defined through a class map. For this process, you configure a policy map that specifies the policy for each traffic class.

- Attaching policies to interfaces (service policies).

  This process requires that you associate an existing policy map, or service policy, with an interface to apply the particular set of policies of the map to that interface.

Advanced Concepts

Once a class has been defined according to its match criteria, you can assign it characteristics. To characterize a class, you assign it bandwidth, weight, and maximum packet limit. The bandwidth assigned to a class is the minimum bandwidth delivered to the class during congestion.

To characterize a class, you also specify the queue limit for that class, which is the maximum number of packets allowed to accumulate in the queue of the class. Packets belonging to a class are subject to the bandwidth and queue limits that characterize the class.

After a queue has reached its configured queue limit, enqueuing of additional packets to the class causes tail drop or packet drop to take effect, depending on how class policy is configured.

Tail drop is used for CBWFQ classes unless you explicitly configure policy for a class to use WRED to drop packets as a means of avoiding congestion. You can apply WRED on a per-class basis. Note that if you use WRED packet drop instead of tail drop for one or more classes comprising a policy map, you must ensure that WRED is not configured for the interface to which you attach that service policy.

If a default class is configured, all unclassified traffic is treated as belonging to the default class. If no default class is configured, then by default the traffic that does not match any of the configured classes is flow classified and given best-effort treatment. Once a packet is classified, all of the standard mechanisms that can be used to differentiate service among the classes apply.

CBWFQ and WFQ are mutually exclusive, as are CBWFQ and WRED, and these rules apply:

- Attaching a service policy to an interface disables WFQ on that interface if WFQ is configured for the interface. For this reason, you should ensure that WFQ is not enabled on such an interface.
- Attaching a service policy containing classes configured to use WRED to an interface disables WRED on that interface. If any of the classes that you configure in a policy map use WRED for packet drop instead of tail drop, you must ensure that WRED is not configured on the interface to which you intend to attach that service policy.

DWFQ

DWFQ provides bandwidth allocations and delay bounds to specified IP traffic sources by segregating the traffic into flows or classes and then providing first-in, first-out (FIFO) service to the various queues according to their assigned weights.

Note Because they do not provide the strict priority required by voice traffic, WFQ and DWFQ largely are not useful for voice applications. For conditions under which you should use them for voice traffic, see the "How Do WFQ, DWFQ, and CBWFQ Apply to Voice Traffic?" section.

There are three kinds of DWFQ:

- Flow-Based DWFQ
- QoS Group-Based DWFQ
- ToS-Based DWFQ

To use any type of DWFQ, dCEF switching must be enabled on the interface. These features require a use of Versatile Interface Processor (VIP) 2-40 or later. DWFQ features cannot be configured on subinterfaces.

Note dCEF is an advanced Layer 3 IP forwarding technology designed to optimize network performance and scalability. It is a switching paradigm that is thought of as a full topology-driven architecture because it uses the first packet in a flow to build an IP destination cache to be used by packets subsequently delivered to the same network destination. The dCEF feature uses all available routing information to build an IP Forwarding Information Base (FIB) so that a deterministic switching decision can be made, even for the first packet to a new destination. This capability is significant, given the changing traffic dynamics on the Internet and within enterprise intranets, where flows are increasingly of shorter duration and more topologically dispersed, such as Web traffic.

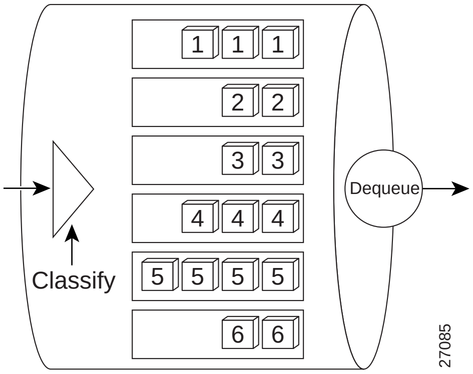
Flow-Based DWFQ

For flow-based DWFQ, packets are classified by flow. Packets with the same source IP address, destination IP address, source TCP or UDP port, destination TCP or UDP port, protocol, and ToS fields belong to the same flow.

Each flow corresponds to a separate output queue. When a packet is assigned to a flow, it is placed in the queue for that flow. During periods of congestion, WFQ allocates an equal share of the bandwidth to each active queue. All flows are equally weighted. Flow-based DWFQ uses fair queueing, therefore packets do not get priority treatment based on IP Precedence.

Figure 2-6 shows six flow-based queues. There is one queue per flow. Flow-based DWFQ is fair queueing; priority is not determined by IP Precedence values.

Figure 2-6 Flow-Based Distributed Weighted Fair Queueing



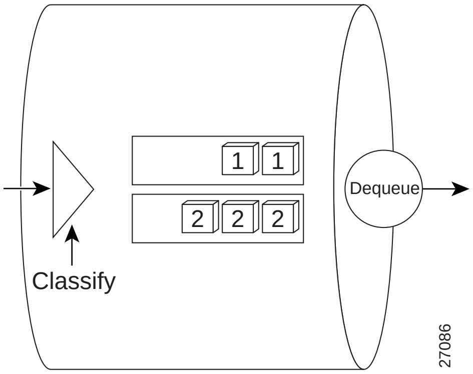
QoS Group-Based DWFQ

For QoS group-based DWFQ, packets are assigned to different queues by policy routing based on their QoS group or the IP Precedence in the ToS field. A QoS group is an internal classification of packets used by the router to determine how certain QoS features, such as WFQ and CAR, treat packets.You can use QoS policy propagation via BGP to assign packets to QoS groups.

For QoS Group-Based DWFQ, you can specify a weight for each class. In periods of congestion, each group is allocated a percentage of the output bandwidth equal to the weight of the class. For example, if a class is assigned a weight of 50, packets from this class will be allocated at least 50 percent of the outgoing bandwidth during periods of congestion. When the interface is not congested, queues can use any available bandwidth.

Figure 2-7 shows two QoS group-based queues.

Figure 2-7 QoS Group-Based DWFQ

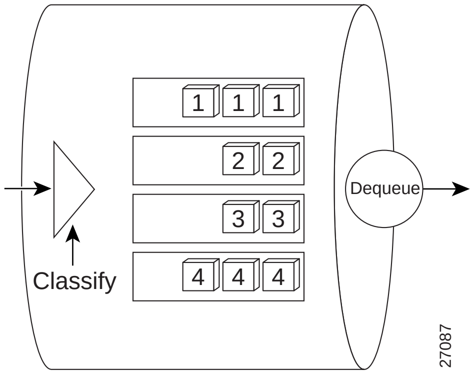


ToS-Based DWFQ

For ToS-based DWFQ, packets are classified based on only the two low-order IP Precedence bits, the last two bits. ToS-based DWFQ supports four queues. You assign weights to the four queues using percentages.

Figure 2-8 shows the four ToS-based DWFQ queues.

Figure 2-8 ToS-Based DWFQ



IP RTP Priority

When WFQ is enabled, it creates a strict priority queue whose use is not available until the IP RTP Priority feature is configured to enable it. The IP RTP Priority feature enables use of this queue.

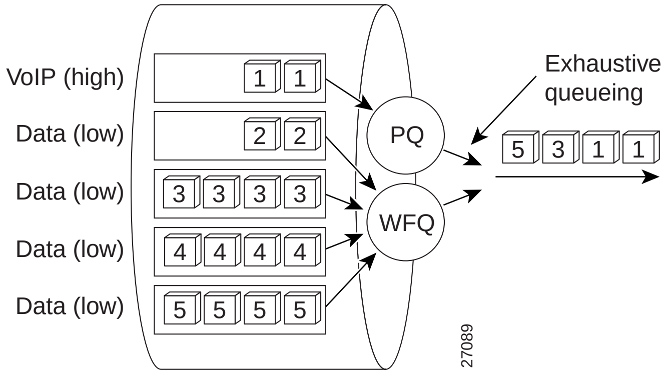
Introduced in Cisco IOS Release 12.0(5)T, the IP RTP Priority feature allows you to specify a range of UDP/RTP ports whose voice traffic is guaranteed *strict* priority service over any other queues or classes using the same output interface. Strict priority means that if packets exist in the priority queue, they are dequeued and sent first—that is, before packets in other queues are dequeued. The result is that voice is given strict priority service in preference to other nonvoice traffic.

For Release 12.0(7)T and later, you can use the priority queueing within CBWFQ feature, which allows you to configure the strict priority queueing for voice traffic belonging to a class. Without use of IP RTP Priority, CBWFQ provides true WFQ based on defined classes with no strict priority queue available for real-time traffic.

The IP RTP Priority feature does not require that you know the port of a voice call. Rather, the feature gives you the ability to identify a range of ports whose traffic is put into a single priority queue. Moreover, you can specify the entire voice port range—16384 to 32767—to ensure that all voice traffic is given strict priority service. IP RTP Priority is especially useful on slow-speed links whose speed is less than 1.544 Mbps.

Figure 2-9 shows five queues configured for IP RTP Priority. The priority queue, the high queue, is dedicated to voice traffic; the entire contents of the priority queue are scheduled for transmission before any other queues are serviced. (The strict priority process is also referred to as exhaustive queueing.) Once the priority queue is empty, the other queues are serviced using WFQ.

Figure 2-9 IP RTP Priority



You can use IP RTP Priority in conjunction with either WFQ or CBWFQ on the same outgoing interface. In either case, traffic matching the range of ports specified for the priority queue is guaranteed strict priority over other CBWFQ classes or WFQ flows; voice packets in the priority queue are all and always serviced first. Here is how the other classes and queues are handled after the priority queue is serviced:

- When used in conjunction with WFQ, the **ip rtp priority** command provides strict priority to voice, and WFQ scheduling is applied to the remaining queues.
- When used in conjunction with CBWFQ, the **ip rtp priority** command provides strict priority to voice. CBWFQ can be used to set up classes for other types of traffic (such as SNA) that need dedicated bandwidth and need to be treated better than best effort and not as strict priority; the nonvoice traffic is serviced fairly based on the weights assigned to the enqueued packets. CBWFQ can also support flow-based WFQ within the default CBWFQ class if so configured.

Because voice packets are small in size and can be detained between large data packets sent out the same interface, you should use Link Fragmentation and Interleaving (LFI) on slow-speed PPP links and FRF.12 on slow-speed Frame Relay links. (When you enable LFI or FRF.12, large data packets are fragmented and the small voice packets are interleaved between the data fragments. LFI and FRF.12 help to prevent voice packet delay.)

If you want to understand its behavior and properly use the IP RTP Priority feature, it is important to consider its admission control and policing characteristics. When you use the **ip rtp priority** command to configure the priority queue for voice, you specify a strict bandwidth limitation. The specified amount of bandwidth and no more is guaranteed to voice traffic enqueued in the priority queue. (This is the case whether you use the IP RTP Priority feature with CBWFQ or WFQ.)

IP RTP Priority closely polices use of bandwidth for the priority queue, ensuring that the allocated amount is not exceeded. In fact, IP RTP Priority polices the flow every second. IP RTP Priority prohibits transmission of additional packets once the allocated bandwidth is consumed. If it discovers that the configured amount of bandwidth is exceeded, IP RTP Priority will drop packets, an event that is poorly tolerated by voice traffic. Close policing allows for fair treatment of other data packets enqueued in other CBWFQ or WFQ queues.

You should also consider that the IP RTP Priority admission control policy disregards voice packet compression (IP RTP Priority does not take Compressed Real-Time Protocol (CRTP) compression into account). Suppose you use priority queueing within CBWFQ and you reserve 24-kbps bandwidth for the voice priority queue, but the voice packets are compressed and compression reduces the flow to 12 kbps. In this case, admission control would not double the number of voice packets it would let through. Rather, the unused bandwidth would be distributed among the other CBWFQ classes. This precept also holds true for use of IP RTP Priority with WFQ.

The IP RTP Priority bandwidth management feature stipulates that the sum of all bandwidth allocation for voice and data flows on the interface is not to exceed 75 percent of the total available bandwidth. The remaining 25 percent of bandwidth is used for other overhead, including the Layer 2 header.

Typically, you apportion the 75 percent of available bandwidth to all classes and flows, including the CBWFQ voice class assigned the priority queue or the WFQ voice flow that uses the priority queue. Bandwidth allocation for voice packets takes into account the payload plus the IP, RTP, and UDP headers—but, again, not the Layer 2 header. The remaining 25 percent of bandwidth that is used for other overhead includes the Layer 2 header and best-effort traffic. Bandwidth for the CBWFQ default class, for instance, is taken from the remaining 25 percent. Allowing 25 percent bandwidth for other overhead is conservative and safe. On a PPP link, for instance, overhead for Layer 2 headers assumes 4 kbps.

If you know how much bandwidth is required for additional overhead on a link, under aggressive circumstances in which you want to give voice traffic as much bandwidth as possible, you can override the 75 percent maximum allocation for the bandwidth sum allocated to all classes or flows. The IP RTP Priority feature includes a command called **max-reserved-bandwidth** that you can use to override the 75 percent limitation. If you override the fixed amount of bandwidth, exercise caution and ensure that you allow enough remaining bandwidth to support best-effort and control traffic.

An another alternative, if the importance of voice traffic far exceeds that of data, is that you can allocate most of the 75 percent bandwidth used for flows and classes to the voice priority queue. Unused bandwidth at any given point is made available to the other flows or classes.

Because the **ip rtp priority** gives absolute priority over other traffic, it should be used with care. If the traffic exceeds the configured bandwidth, then all the excess traffic is dropped.

The IP RTP Priority feature runs on these platforms:

- Cisco 1003
- Cisco 1004
- Cisco 1005
- Cisco 1600 series
- Cisco 2500 series
- Cisco 2600 series
- Cisco 3600 series
- Cisco 3800 series
- Cisco 4000 series (Cisco 4000, 4000-M, 4500, 4500-M, 4700, 4700-M)
- Cisco 5200 series
- Cisco AS5300
- Cisco 7000 series
- Cisco 7200 series
- Cisco 7500 series

Priority Queueing within CBWFQ

The priority queueing within CBWFQ feature brings the strict priority queueing functionality of IP RTP Priority required for delay-sensitive, real-time traffic, such as voice, to CBWFQ. Priority queueing within CBWFQ allows for use of a strict priority queue created when WFQ is enabled but not available for use until CBWFQ is enabled. Although it is possible to enqueue various types of real-time traffic to the strict priority queue, we strongly recommend that you direct only voice traffic to it. This recommendation is made because voice traffic is well-behaved, whereas other types of real-time traffic are not. Moreover, voice traffic requires that delay be nonvariable in order to avoid jitter. Other real-time traffic such as video could introduce variation in delay thereby thwarting the constancy of delay required by voice traffic. Priority queueing within CBWFQ satisfies the criteria for and implements the extended forwarding functionality of distributed services.

Priority queueing within CBWFQ allows for a broad range of ways in which to specify the traffic to be guaranteed strict priority delivery. To indicate the voice flow to be enqueued to the strict priority queue, you are not limited to use of the UDP port range as you are with IP RTP Priority. Rather, you can use all of the mechanisms implicit to class definition: you can match on access lists, protocols, input interfaces as well as IP Distributed Services Code Point (DSCP) values.

To understand how priority queueing within CBWFQ works, it helps to consider certain aspects of the combined features—the IP RTP Priority feature and the CBWFQ feature—and then to consider how they are used together, including the few differences introduced when strict priority queueing for delay-sensitive traffic is applied to a CBWFQ class within a policy map. This section takes that approach.

About CBWFQ and Strict Priority

Without use of IP RTP Priority, CBWFQ provides true WFQ based on defined classes with no strict priority queue available for real-time traffic.

Recall that CBWFQ has two main parts:

- Named classes, which you set up and which are mapped to existing entities such as protocols, access control lists (ACLs), or input interfaces.
- Policy maps, which you define to include one or more classes. After you specify a class within the policy map, you assign it values, such as the bandwidth available for its traffic and the individual queue limit.

Packets satisfying the match criteria for a class constitute the traffic for that class. A queue is reserved for each class, and traffic belonging to a class is directed to the queue of that class.

For CBWFQ, the weight assigned to the packets of a class determines the order in which packets are sent. All packets are serviced fairly based on weight; no class of packets may be granted strict priority. This scheme poses problems for voice traffic which is largely intolerant of delay, especially variation in delay. (For voice traffic, variations in delay introduce irregularities of transmission manifesting as jitter in the heard conversation.)

The **priority** command, which configures the IP RTP Priority feature for a CBWFQ class, enables use of a single, strict priority queue within CBWFQ at the class level. Priority queueing within CBWFQ allows you to direct traffic belonging to a class to the WFQ strict priority queue. To enqueue class traffic to the strict priority queue, you configure the **priority** command for the class after you specify the named class within a policy map. (Classes to which the **priority** command is applied are considered priority classes.) Within a policy map, you can give one or more classes priority status. When multiple classes within a single policy map are configured as priority classes, all traffic from these classes is enqueued to the same, single, strict priority queue.

One way in which the IP RTP Priority feature used within CBWFQ differs from its use outside CBWFQ is in the parameters it takes. Outside CBWFQ, you specify the range of UDP ports whose voice traffic flows are to be given priority service. Because you can configure the priority status for a class within CBWFQ, you are no longer limited to a UDP port name to stipulate priority flows. Instead, all of the valid match criteria used to specify traffic for a class now applies to priority traffic. These methods of specifying traffic for a class include matching on access lists, protocols, and input interfaces. Moreover, within an access list you can specify that traffic matches are allowed based on the IP DSCP value that is set using the first six bits of the ToS byte in the IP header.

Guaranteed Bandwidth

When you specify the **priority** command for a class, it takes a bandwidth argument that gives bandwidth in kbps. You use this parameter to specify the amount of bandwidth allocated for packets belonging to the class configured with the **priority** command. The bandwidth parameter both guarantees bandwidth to the priority class and restrains the flow of packets from the priority class.

When the bandwidth is exceeded, tail drop is used to drop packets. Voice traffic enqueued to the priority queue is UDP-based and therefore not adaptive to the early packet drop characteristic of WRED. Because WRED is ineffective, you cannot use the WRED **random-detect** command with the priority command.

When congestion occurs, traffic destined for the priority queue is metered to ensure that the bandwidth allocation configured for the class to which the traffic belongs is not exceeded.

Priority traffic metering has the following features:

- It is much like the rate limiting in CAR, except that priority traffic metering is only performed under congestion conditions.
- It is performed on a per-packet basis and tokens are replenished as packets are sent. If not enough tokens are available to send the packet, it is dropped.
- It restrains priority traffic to its allocated bandwidth to ensure that nonpriority traffic, such as routing packets and other data, is not starved.

With metering, the classes are policed and rate-limited individually. That is, although a single policy map might contain four priority classes, all of which are enqueued in a single priority queue, they are each treated as separate flows with separate bandwidth allocations and constraints.

It is important to note that because bandwidth for the priority class is specified as a parameter to the **priority** command, you cannot also configure the **bandwidth** command for a priority class. To do so is a configuration violation that would only introduce confusion in relation to the amount of bandwidth to allocate.

The bandwidth allocated for a priority queue always includes the Layer 2 encapsulation header. However, it does not include other headers, such as ATM cell-tax overheads. Here are three primary factors you must allow for:

- When you calculate the amount of bandwidth to allocate for a given priority class, account for the fact the Layer 2 headers are included.
- When ATM is used, account for the fact that ATM cell-tax overhead is not included.
- You must also allow bandwidth for the possibility of jitter introduced by routers in the voice path.

An Example

Consider this case that uses ATM. Suppose a voice stream of 60 bytes emitting 50 packets per second is encoded using G.729. Prior to converting the voice stream to cells, the meter for the priority queue used for the voice stream assesses the length of the packet after the L2 LLC headers have been added.

Given the 8-byte Layer 2 LLC header, the meter will take into account a 68-byte packet. Because ATM cells are a standard 53 bytes long, before the 68-byte packet is emitted on the line, it is divided into two 53-byte ATM cells. Thus, the bandwidth consumed by this flow is 106 bytes per packet.

For this case, then, you must configure the bandwidth to be at least 25.6 kbps (68*50*8 = 27.2 kbps). However, recall that you must also allow for the cell tax overhead, which is not accounted for by the configured bandwidth. In other words, the sum of the bandwidths for all classes must be less than the interface bandwidth by at least (106-68)*50*8 = 15.2 kbps. You should also remember to allow bandwidth for router-introduced jitter.

Note In general, whether or not you use CBWFQ and strict priority, the amount of bandwidth you allocate per PVC or WAN link for voice traffic alone or sharing the PVC or link with data traffic should not exceed 75 percent of the available bandwidth of the link or PVC. The remaining bandwidth is required for overhead, such as routing messages. If your network usage shows a preference for voice traffic over data traffic, you could allocate all bandwidth up to the 75 percent allowable ceiling to voice traffic.

Link Efficiency Mechanisms

This section describes the following Cisco IOS QoS link efficiency mechanisms that you can use to give voice traffic priority treatment if your network carries voice and data traffic concurrently:

- FRF.12 (Frame Relay Forum.12)
- MLPPP LFI (Multilink Point-to-Point Protocol Link Fragmentation and Interleaving)
- IP MTU Size Reduction (IP Maximum Transmission Unit Size Reduction)
- CRTP (Compressed Real-Time Protocol)

In conjunction with these link efficiency mechanisms, you should use queueing and traffic shaping features to provide greater QoS.

There are a number of ways to transport data and voice traffic throughout an enterprise. For Frame Relay, you can run packetized voice alongside data on a single PVC to maximize network bandwidth use. Most data applications attempt to send frames up to the allowed MTU for the link, but voice packets are usually fairly small (80 to 256 bytes). Over LAN media, large data packets, called jumbograms, mixed with small voice packets is not an issue in most cases, but over slow-speed WAN links mixing of jumbograms and smaller voice packets can lead to additional delay and jitter. Consequently, voice over frame-based media is susceptible to increased latency and jitter on slow-speed links when large frames such as LAN-to-LAN FTP Telnet transfers traversing a WAN link are queued for transmission and voice packets are detained behind them.

To limit the delay of real-time packets on relatively slow bandwidth links—links such as 56-kbps Frame Relay or 64-kbps ISDN B channels—a method for fragmenting larger packets and queueing smaller packets between fragments of the large packet is needed.

Large frames cause excessive delay to smaller real-time voice traffic if the large packets are sent as single units. IP-based datagram transmission techniques for voice transmission do not adequately or always address the problems posed by limited bandwidth and the very stringent telephony delay bound, especially on slow links. (The complete end-to-end delay target for real-time traffic such as voice packets is 150 ms one way.) Moreover, unacceptable queueing delays for voice packets exist regardless of use of QoS features such as RSVP and WFQ and use of voice compression algorithms such as Compressed Encoding for Linear Prediction (CLEP), even though CLEP reduces the inherent bit rate from 64 kbps to as low as 8 kbps. Additional techniques are required to improve the efficiency and predictability of voice packet transmission in relation to coexistent data flows sharing the link.

For each link efficiency mechanism, Table 2-4 shows the versions of the Cisco IOS software that support the feature, the switching mode used, and the platforms the feature runs on.

Terms used in Table 2-5 are explained as follows:

- "All Cisco IOS platforms" refers to this set of platforms: 1000, 1600 series, 1720, 2500 series, 2600 series, 3600 series, 4500 series, 4700 series, 7200 series, and RSP in the 7500.
- The following abbreviations are used to represent various switching modes in this table:
  - P = Process
  - F = Fast
  - N = NetFlow
  - O = Optimum
  - CEF = Cisco Express Forwarding
  - d = distributed (VIP)
  - dCEF = distributed CEF

Table 2-5 Cisco IOS Version, Switching Modes, and Platform Support for Link Efficiency Mechanisms

Feature	Cisco IOS Version and Switching Mode							Platform Support
	10.3	11.0	11.1	11.2	11.1CC	11.3	12.0	
Link Efficiency Mechanisms								
Link Fragmentation and Interleaving	—	—	—	—	—	P	P	All Cisco IOS platforms
Real-Time Protocol Header Compression (RTP-HC)	—	—	—	P	—	P	P	All Cisco IOS platforms
Frame Relay Fragmentation (FRF12)	—	—	—	—	—	—	—	All Cisco IOS platforms 12.0(4)T

Why Fragment Data Packets?

To avoid excessive delay, you should fragment the larger data packets and interleave them with the smaller voice packets. FRF.12, LFI, and MTU Size Reduction—the three fragmentation methods you can use in conjunction with VoIP traffic—improve link efficiency by segmenting data packets into sequences of shorter packets or frames called fragments and interleaving low-delay traffic with the resulting smaller packets. These fragments are of a specified size such that a receiving device can reassemble them into the original frame. Interleaving fragments with voice traffic ensures predictable delay for voice traffic and creates an even flow of shortened frames and digitized voice packets.

Which Fragmentation Method Should You Use?

If appropriate for the link configuration, you should use one of the fragmentation mechanisms that operate at Layer 2, such as FRF.12 or LFI. If using one of these methods is not feasible, then as a last resort you should consider using the IP MTU Size Reduction feature, which fragments IP packets at Layer 3.

You might want to use IP MTU Size Reduction to reduce the MTU size for IP packets that you are sending from a Frame Relay link across an ATM link to a Frame Relay endpoint. FRF.12 works well for links whose endpoints are both Frame Relay because Frame Relay reassembles fragments at the receiving end. However, it does not work well when the sending end of the link is Frame Relay and the receiving end is ATM because ATM does not reassemble fragments.

For instance, suppose a 1500-byte data packet is fragmented and sent across a link that uses FRF.12 and MLPPP-LFI; the packet would be fragmented and sent across the Frame Relay link and reassembled at the Frame Relay link on the router that receives the fragments. Suppose the packet were fragmented and sent across a link that uses IP MTU Size Reduction. The packet would be fragmented at the Frame Relay source and traverse the link in fragments that would not be reassembled but would continue in transmission as fragments for the life of the packet.

Before you use any fragmentation method, you should determine whether fragmentation can benefit your configuration. In large networks, you should calculate the voice delay budget. Fragmentation is generally not needed on links whose speed exceeds 768 kbps. However, where use of fragmentation is warranted, you should configure the size of the fragments to break large jumbograms into. The optimum size of the fragment you should specify depends on the queueing delay caused by the large frames at a given speed.

If you find that fragmentation is required but you do not use it, voice quality will suffer and no amount of queueing will restore it.

For various link speeds, Table 2-6 shows the time it takes in microseconds or milliseconds to send a certain number of bytes. (The abbreviation “us” indicates microseconds: 1000 us = 1 ms.) Notice that as the link speed increases, the transmission time decreases to the degree that fragmentation might not be desirable.

Table 2-6 Byte Count and Transmission Time for Link Speed

	Number of Bytes and Transmission Time for Link Speed						
Link Speed	**1 Byte**	**64 Bytes**	**128 Bytes**	**256 Bytes**	**512 Bytes**	**1024 Bytes**	**1500 Bytes**
56 kbps	142 us	9 ms	18 ms	36 ms	72 ms	144 ms	214 ms
64 kbps	125 us	8 ms	16 ms	32 ms	64 ms	128 ms	187 ms
128 kbps	62.5 us	4 ms	8 ms	16 ms	32 ms	64 ms	93 ms
256 kbps	31 us	2 ms	4 ms	8 ms	16 ms	32 ms	46 ms
512 kbps	15.5 us	1 ms	2 ms	4 ms	8 ms	16 ms	23 ms
768 kbps	10 us	640 us	1.28 ms	2.56 ms	5.12 ms	10.24 ms	15 ms
1536 kbps	5 us	320 us	640 us	1.28 ms	2.56 ms	5.12 ms	7.5 ms

When specifying the fragment size, the basic premise to understand is that delay depends on the bandwidth of the link speed, or trunk, and the size of the frame, or data packet. Table 2-7 recommends the fragment sizes you should specify for frame transmission at various link speeds. These sizes help to ensure that the delay incurred by voice packets is no greater than 10 ms; that is, the recommended values assume a 10-ms blocking delay per fragment.

Table 2-7 Recommended Fragment Sizes

Link Speed (Bandwidth)	Fragment Size
56 kbps	70 bytes
64 kbps	80 bytes

Table 2-7 Recommended Fragment Sizes (continued)

Link Speed (Bandwidth)	Fragment Size
128 kbps	160 bytes
256 kbps	320 bytes
512 kbps	640 bytes
768 kbps	1000 bytes
1536 kbps	Fragmentation not necessary

Here is how you can calculate the appropriate fragment size to use:

```
fragment size = 10 ms / time for 1 byte at bandwidth
```

To calculate the worst-case queueing delay for this scenario, you can use the following equation

```
Q = (Pv*N/C) + LFI
```

where:

Q = Worst case queueing delay of the voice packet in ms.

Pv = Size of a voice packet in bits at Layer 1.

N = Number of calls.

C = Link capacity in bps.

LFI = Fragment size queue delay in ms.

As an example, consider a voice application that uses codec G.729. Assume that there are four calls made on a 128-kbps circuit. The fragment blocking delay is 10 ms (160 bytes). Here is how you calculate the worst-case queueing delay for this scenario:

```
Q = (480 bits * 4/128000) + 10 ms = 25 ms
```

The worst-case queueing delay is 25 ms.

For a rigorous determination of the allowable delay for a voice packet, perform the following tasks:

Step 1 Determine the worst-case route for a voice packet through the network.

Step 2 For the route you identified in Step 1, add the worst-case delays due to queueing and propagation delays and dejitter.

Step 3 Subtract the result of Step 2 from the budgeted delay for voice across the network, usually 150 to 200 ms.

The resulting figure will indicate the allowable delay due to fragmentation.

With or without fragmentation, to optimize bandwidth utilization on oversubscribed output links, for instance, your edge routers can perform RTP header compression using CRTP. CRTP improves link efficiency by compressing voice packets. Bandwidth is at a premium on slow-speed WAN links, and compressed traffic uses less of it. Moreover, unless voice traffic is compressed, VoIP scalability will not be possible because the large size of the IP/UDP/RTP headers consumes an equally large amount of bandwidth. Consider this case: a G.729 (8K codec) VoIP call will consume 24 kbps when the IP/UDP/RTP headers are added. When you add to this amount the 5 to 7 bytes of overhead required for

Layer 2, the bandwidth cost of the call could be up to 26 kbps. CRTP can compress the IP/UDP/RTP header down to as little as 2 bytes, which, for example, results in a 12K G.729 call—a cost that is acceptable to most users.

FRF.12

FRF.12 is the standard implementation agreement for Frame Relay fragmentation ratified by the Frame Relay Forum in 1998. FRF.12 specifies a common way for different vendors to segment data packets into fragments. FRF.12 ensures predictability and QoS for voice traffic, aiming to provide better throughput on low-speed Frame Relay links by interleaving delay-sensitive voice traffic on one VC with fragments of a long frame on another VC utilizing the same interface. FRF.12 has all the advantages of MLPPP

- It incurs low overhead.
- It can be applied at the link level or the PVC level.
- You can set the fragmentation threshold.
- It is protocol transparent.
- It can only be applied on a Frame Relay trunk.

FRF.12 is available on Cisco 2600, 3600, 3810. and 7200 routers beginning with release 12.0(4) T.

Cisco IOS software supports three types of end-to-end fragmentation:

- Pure fragmentation.
- FRF.11 Annex C fragmentation.
- Cisco proprietary fragmentation.

We recommend using pure end-to-end fragmentation for VoIP because data packets that are smaller than the configured fragmentation size do not contain the fragmentation header. For pure end-to-end fragmentation, if the VoIP packet is larger than the fragment size, the packet will include the header. For FRF.11 Annex C and Cisco proprietary fragmentation, all data packets include the fragmentation header under all conditions.

Note We recommend that you not mix VoIP and VoFR traffic on the same PVC.

FRF.12 has the following features:

- It incurs low overhead.
- It can be applied at the PVC level. (You can use a single PVC in a Frame Relay network to carry both voice and data.)
- It allows you to set the fragmentation threshold.
- It is protocol-transparent.
- It does not support concurrent use of RSVP.

When FRF.12 is used on a PVC, WFQ is automatically used as the queueing technique.

FRF.12 preserves the initial characteristics of an isochronous flow when different types of traffic share the same physical interface.

FRF.12 over VoIP has the following features:

- It uses RTP to reserve bandwidth.
- It is IP-Precedence-based.
- It is CODEC-compression-based (G.729 or G.711).
- It uses CRTP to compress packet headers.

You should use FRF.12 as the fragmentation type for a PVC that does not carry voice, but that shares the link with other PVCs that carry voice. When FRF.12 is used, short, whole (nonfragmented) voice frames bypass already-queued data fragments. That is, queues containing voice packets are dequeued exhaustively, thereby granting whole, small, voice packets priority transmission over queues containing fragmented data packets.

FRF.12 allows you to fragment data on links only if the access rate requires it. Moreover, the FRF.12 fragmentation header is included only for frames that are greater than the configured fragment size. That is, in end-to-end fragmentation, Frame Relay frames with a payload size less than the fragmentation size configured for that PVC are not fragmented, and thus are sent without the fragmentation header. Voice packets do not include the FRF.12 header, provided the size of the VoIP packet is smaller than the fragment size configured.

FRF.12 allows fragmentation on nonfragmentable protocols and on IP messages with the do-not-fragment bit set. You set the fragmentation threshold on a per-PVC level. You can configure fragmentation through a map class, which you can apply to one or many PVCs, depending on how the class is assigned.

When FRF.12 is configured, the following conditions and restrictions apply:

- Hardware compression is not supported.
- Priority queueing for WFQ (PQ-WFQ) is the only queueing strategy that can be used at the PVC level.
- VoIP frames are not fragmented unless they are bigger than the configured fragment size.

With FRF.12, PQ-WFQ is used on input on a per-PVC basis; therefore, there is no weight associated with VoIP flows:

- If fragments arrive out of sequence, packets are dropped.
- FRF.12 works with Frame Relay Traffic Shaping (FRTS) only. It does not work with Generic Traffic Shaping.

Here is an approximate outline of how FRF.12 works:

1. WFQ scheduling of packets occurs before the packets are fragmented.
2. Packets are fragmented as they are dequeued from the PVC; at this point, voice packets and data fragments are interleaved.
3. After the packets are dequeued, they are sent to one of the dual FIFO interface queues.

   These two FIFO queues are the point where traffic from multiple PVCs interact.

   – All voice packets that are smaller than the fragment size configured for the PVC and LMI packets are enqueued in the high-priority FIFO queue. (We advise against mixing VoIP and VoFR traffic on the same PVC. However, if VoFR is also used on the PVC, VoFR packets are placed in the high-priority FIFO queue.) This queue is used exclusively for nonfragmented packets. None of the packets in this queue has a fragmentation header.

     This high-priority FIFO queue might also hold other nonfragmented packets, such as keepalives, which could introduce unevenness in the delay experienced by voice traffic. This queue is an exhaustive queue, similar in behavior to a PQ-WFQ queue. Enqueueing small, unfragmented VoIP packets into an exhaustive queue ensures their priority delivery over packets in the low-priority queue. (An exhaustive queue is one whose packets are dequeued exhaustively, that is, entirely, whenever the queue contains packets and before packets from the other FIFO queue are serviced.) Use of an exhaustive FIFO queue for voice packets protects voice flows from heavy traffic sent on PVCs using the same interface. It helps to prevent traffic sent on other PVCs from detrimentally affecting nonfragmented voice packets in the event that the other PVCs are oversubscribed or their traffic shaping parameters are misconfigured.

– The low-priority FIFO queue is used for all other packets, including other small unfragmented packets.

Figure 2-10 shows FRF.12 enabled on a per-PVC basis for two PVCs: PVC 1 and PVC 2. Notice that for both PVCs, FRTS is also enabled. WFQ is the only supported queueing mechanism for FRF.12. (All other queueing types currently supported on a PVC are not allowed.)

Note When there is no congestion on a PVC, voice traffic is not shaped. Thus, it is not enqueued to the PQ-WFQ queue. Rather, it is placed in the interface queue of the transmit (TX) ring or sent directly to the TX ring, depending on whether there is traffic in the TX queue.

Each PVC has its own WFQ structure, which is used to schedule packets appropriately for dequeueing based on their IP Precedences. Dual-FIFO interface queueing is used at the interface level; these two FIFO queues are set up automatically when FRTS is configured.

Figure 2-10 A Conceptual View of FRF.12 with Multiple PVCs

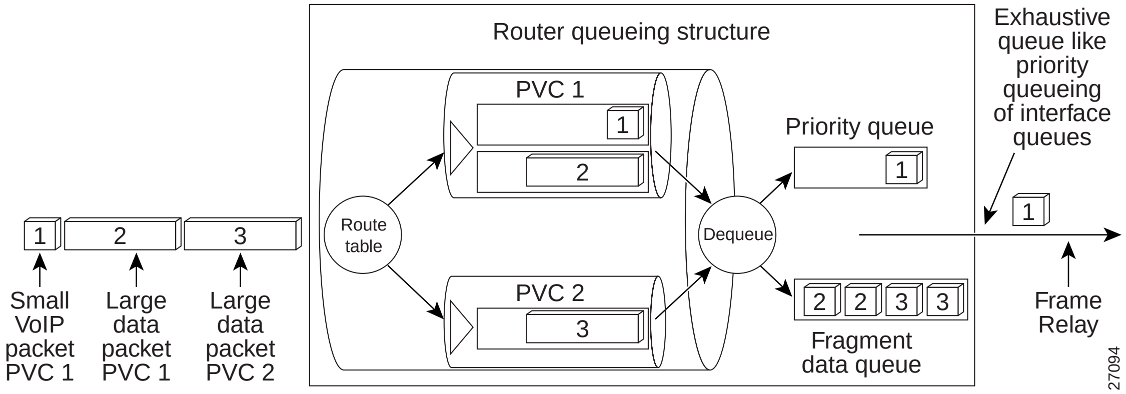
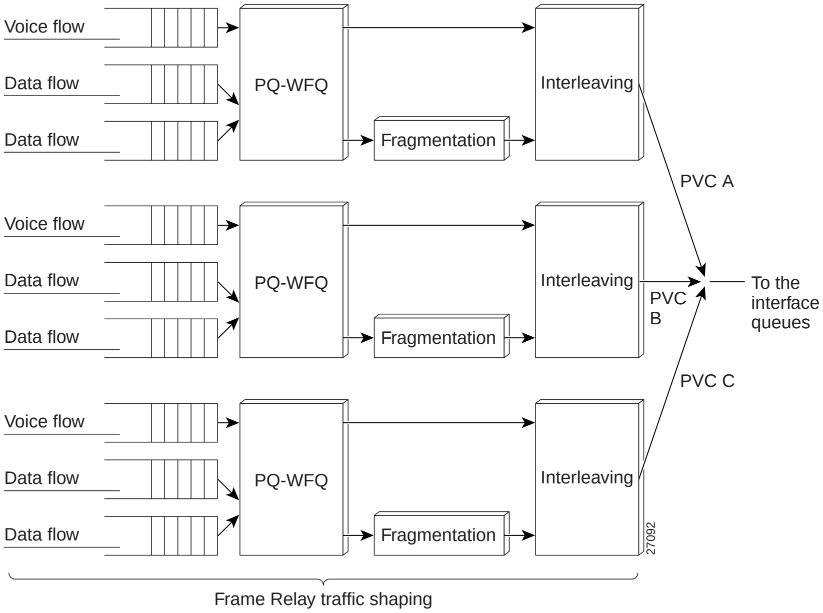


Figure 2-11 gives another view of FRF.12 used for three PVCs in which each PVC carries one voice flow and two data flows. Notice that before the large data frames are fragmented, WFQ is used to dequeue the frames, or packets, from each set of flows—WFQ gives priority to small packets with low weight. After they are fragmented, the data frames are sent in order; VoIP packets are interleaved between the fragments of the jumbogram data frames. The interleaved single flows from each PVC are sent to the interface queues for enqueueing in the high-priority or low-priority FIFO queues. The Frame Relay frame is compressed before it is fragmented, and it is decompressed after it is reassembled at the other end of the link.

Figure 2-11 PVC Queueing and Fragmentation for FRF.12



MLPPP LFI

MLPPP LFI provides a method of splitting, recombining, and sequencing datagrams across multiple logical links.

MLPPP LFI is used on slow-speed PPP links to fragment these large data packets and interleave the resulting fragments with delay-sensitive voice packets in order to provide priority service to voice traffic. Large packets are fragmented for transmission onto the PPP leased-line trunk, then reassembled on receipt from the trunk. They do not travel through the entire network as fragments.

To ensure correct order of transmission and reassembly, LFI adds multilink headers to the datagram fragments after the packets are dequeued and ready to send. Small delay-sensitive packets are not multilink encapsulated, but are interleaved between fragments of the large datagram. LFI also provides a special transmit queue for the smaller, delay-sensitive packets, enabling them to be sent earlier than other flows.

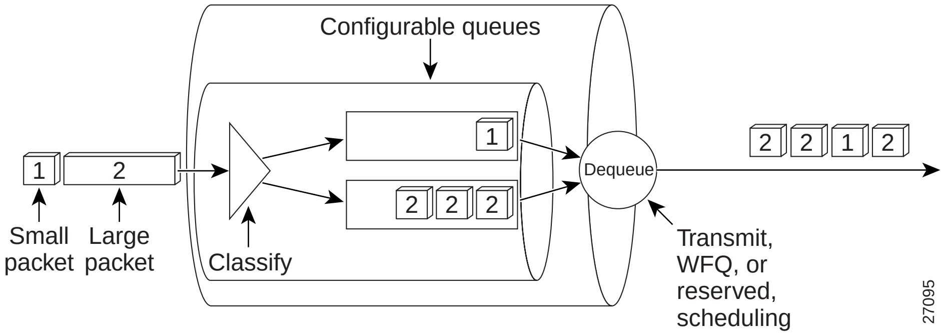
MLPPP allows the fragmented packets to be sent at the same time over multiple point-to-point links to the same remote address. The multiple links come up in response to a dialer load threshold that you define. The load can be calculated on inbound traffic, outbound traffic, or on either, as needed for the traffic between the specific sites. MLPPP provides bandwidth on demand and reduces transmission latency across WAN links.

MLPPP alone works at the packet level, not at the level of multilink fragments. Thus, without use of LFI, if a small real-time voice packet gets queued behind a large data packet and no special queue has been reserved for voice packets, the small voice packet will be scheduled for transmission only after all the fragments of the larger data packet are scheduled for transmission.

LFI allows reserve queues to be set up so that RTP streams can be mapped into a higher priority queue in the configured weighted fair queue.

As Figure 2-12 shows, LFI also provides a special transmit queue for the smaller, delay-sensitive packets, enabling them to be sent earlier than other flows.

Figure 2-12 MLPPP LFI



IP MTU Size Reduction

The IP MTU Size Reduction feature allows you to reduce the size of IP packets at Layer 3. Using this feature, you can fragment large packets and interleave them with VoIP packets. The fragments are not reassembled at the other side of the link, but are sent as fragments until they reach their final destination. IP MTU size reduction offers a good fragmentation method when you need to send fragmented and interleaved frames, for instance, from a Frame Relay network across an ATM network and WAN cloud to a destination Frame Relay link. However, use of IP MTU size reduction is not ideal for the following reasons, which you should take into account when deciding whether to use it:

- Large frames belonging to protocols other than IP are not fragmented. If your network is multiprotocol in its support, large, unfragmented frames could pose a problem. For instance, if large non-IP frames are sent, their transmission will vary the delay, a situation intolerable to voice, which requires delay constancy.
- Small fragments can lead to performance inefficiencies. Reducing the IP MTU size below 300 bytes may adversely affect an application as well as router endpoint performance. For instance, on a 56-kbps circuit, the MTU size is commonly set to 300 bytes to achieve a minimum blocking delay of 42 ms.
- Fragmentation can lead to a condition colloquially known as “packet bloat.” For example, consider the case of packet bloat that occurs in which a 1500-byte frame is broken into 300-byte fragments, each fragment now carrying a 40-byte header. The overhead incurred by the necessary headers for each discrete fragment results in consumption of additional bandwidth. Because the packets travel the entire course of the internetwork as fragments with headers, this overhead persists for the transmission duration.

CRTP

RTP, the Internet-standard protocol for the transport of real-time data, provides end-to-end network transport functions suitable for sending voice traffic over multicast or unicast network services. CRTP is used in conjunction with RTP to reduce, or compress, the extensive RTP header, resulting in decreased consumption of available bandwidth for voice traffic manifesting as a corresponding reduction in delay. You use CRTP on a link-by-link basis. CRTP is currently process switched.

Bandwidth is at a premium on low-speed WAN links, necessitating that you compress any voice traffic you intend to send over these links. CRTP is required to ensure VoIP scalability which would otherwise be impossible due to the extensive overhead incurred by the size of VoIP packets, which include the IP header, the UDP header, and the RTP header, in addition to the voice payload.

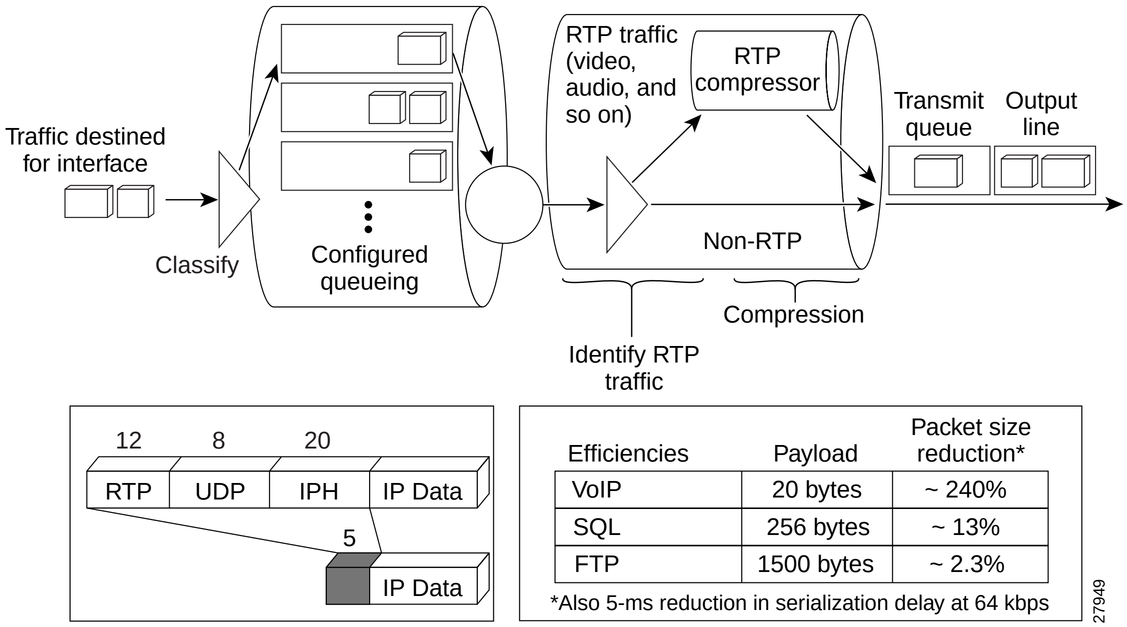
For example, a G.729 (8 K codec) VoIP call expands to 24 kbps when the IP/UDP/RTP headers are added. Considering that another 5 to 7 bytes of overhead are incurred per packet at Layer 2, a VoIP call could require up to 26 kbps. CRTP can compress the IP/UDP/RTP headers to as little as 2 bytes, resulting in a 12-kbps G.729 call.

CRTP is especially beneficial when the RTP payload size is small, as is the case for compressed voice payloads. The payload of a typical voice packet using RTP is 20 bytes, while the header is often twice this size—the minimal 12 bytes of the RTP header, combined with 20 bytes of IP header (IPH) and 8 bytes of UDP header, create a 40-byte IP/UDP/RTP header. Given the size of the IP/UDP/RTP header combinations, it is inefficient to send the IP/UDP/RTP header without compressing it.

Note The voice payload size is usually configurable in 10-byte increments to a maximum of 240 bytes; the limits are codec-specific. Increasing the payload size reduces the overall voice channel bandwidth. However, the improvement in bandwidth is gained at the expense of voice connection delay.

Figure 2-13 shows the point at which CRTP header compression occurs in the transmission of a voice packet. It also notes the degree of compression for VoIP packets.

Figure 2-13 Compression of Real-Time Protocol Packet Header



Efficiencies	Payload	Packet size reduction*
VoIP	20 bytes	~ 240%
SQL	256 bytes	~ 13%
FTP	1500 bytes	~ 2.3%

*Also 5-ms reduction in serialization delay at 64 kbps

A full-rate voice call consumes about 84 kbps of network bandwidth. To avoid unnecessary consumption of available bandwidth, you should use CRTP on a link-by-link basis to reduce the RTP header to 2 to 4 bytes. Reducing the bandwidth necessary for each voice call increases the number of calls that can be sent on a given trunk.

Table 2-8 shows some combinations of codec, payload size, CRTP, and voice activity detection (VAD) used to compress data. G.729B includes an integral VAD function but otherwise has identical performance to G.729. The following assumptions are made:

- The RTP/UDP/IP header is 40 bytes. (CRTP reduces this amount to 2 bytes but MLPPP adds 6 bytes.)

- FRF.12 adds 6 bytes to the header, including the Frame Relay header and checksum. (Thus, the values for FRF.12 are the same as those for MLPPP.)
- VAD is assumed to reduce utilization to 65 percent of the full rate.

Table 2-8 Data Compression and Codec, Payload Size, and Use of QoS Features

Compression Technique	Payload Size	Bandwidth at Full Rate with MLPPP or FRF.12	Bandwidth with CRTP and MLPPP	Bandwidth with VAD and MLPP or FRF.12	Bandwidth with CRTP, VAD, and MLPPP or FRF.12
G.711 (64 kbps)	240 bytes	76 kbps	66 kbps	50 kbps	43 kbps
G.711 (64 kbps)	120 bytes	89 kbps	68 kbps	58 kbps	44 kbps
G.726 (32 kbps)	120 bytes	44 kbps	34 kbps	29 kbps	22 kbps
G.726 (32 kbps)	60 bytes	57 kbps	36 kbps	37 kbps	24 kbps
G.726 (24 kbps)	80 bytes	38 kbps	27 kbps	25 kbps	17 kbps
G.726 (24 kbps)	40 bytes	52 kbps	29 kbps	34 kbps	19 kbps
G.728 (16 kbps)	80 bytes	25 kbps	18 kbps	17 kbps	12 kbps
G.728 (16 kbps)	40 bytes	35 kbps	19 kbps	23 kbps	13 kbps
G.729 (8 kbps)	40 bytes	17.2 kbps	9.6 kbps	11.2 kbps	6.3 kbps
G.729 (8 kbps)	20 bytes	26.4 kbps	11.2 kbps	17.2 kbps	7.3 kbps
G.723.1 (6.3 kbps)	48 bytes	12.3 kbps	7.4 kbps	8.0 kbps	4.8 kbps
G.723.1 (6.3 kbps)	24 bytes	18.4 kbps	8.4 kbps	12.0 kbps	5.5 kbps
G.723.1 (5.3 kbps)	40 bytes	11.4 kbps	6.4 kbps	7.4 kbps	4.1 kbps
G.723.1 (5.3 kbps)	20 bytes	17.5 kbps	7.4 kbps	11.4 kbps	4.8 kbps

You should use CRTP on any slow-speed WAN interface where bandwidth is a concern and there is a high portion of RTP traffic. However, you should avoid using CRTP on high-speed interfaces—that is, links whose speed exceeds the T1 speed—because the trade-offs are undesirable.

CRTP is supported on serial lines using Frame Relay, HDLC, or PPP encapsulation. It is also supported over Integrated Services Digital Network (ISDN) interfaces.

Congestion Avoidance

Congestion avoidance techniques attempt to anticipate and avoid congestion at common network bottlenecks. Congestion most often occurs on WAN links. Global WAN congestion lends greater cause for concern than does campus congestion. This section describes the WRED Cisco IOS QoS feature and its variations. Though not applicable to voice traffic in most cases because it is essentially a packet-drop mechanism best used in TCP environments, WRED can be used for links sending voice traffic under certain situations. This section describes those situations.

- Table 2-9 shows the versions of the Cisco IOS software that support the WRED feature, the switching mode used, and the platforms WRED runs on.“All Cisco IOS platforms” refers to this set of platforms: 1000, 1600 series, 1720, 2500 series, 2600 series, 3600 series, 4500 series, 4700 series, 7200 series, and RSP in the 7500.

- The following abbreviations are used to represent various switching modes in this table:
  - P = Process
  - F = Fast
  - N = NetFlow
  - O = Optimum
  - CEF = Cisco Express Forwarding
  - d = distributed (VIP)
  - dCEF = distributed CEF

Table 2-9 Cisco IOS Versions, Switching Modes, and Platform Support for Congestion Avoidance Features

Feature	Cisco IOS Version and Switching Mode							Platform Support
	10.3	11.0	11.1	11.2	11.1CC	11.3	12.0	
Congestion Avoidance								
Weighted Random Early Detection (WRED)	—	—	—	P, F, N, O	—	P, F, N, O	P, F, N, O, C	All Cisco IOS platforms: Catalyst 5000 series, Catalyst 6000 series, 12000
Distributed Weighted Random Early Detection (DWRED)	—	—	—	—	dCEF	—	dCEF	VIP distributed
Flow WRED	—	—	—	—	—	—	P, F, N, O, C	All Cisco IOS platforms

WRED

The WRED QoS feature combines IP Precedence and standard Random Early Detection (RED) capabilities to provide differentiated performance characteristics for different classes of service—thus allowing for preferential handling of voice traffic under congestion conditions without exacerbating the congestion. WRED is especially useful in high-speed transit networks.

How Does WRED Apply to Voice Traffic?

Although WRED provides good congestion avoidance capabilities designed to increase aggregate network throughput maximizing capacity utilization while carefully managing packet discard, it does not give voice traffic the strict priority that it requires. WRED can only provide preferential treatment for high priority traffic such as voice packets during congestion situations, minimizing voice packet loss and delay while discarding standard traffic earlier and more aggressively. That understood, configuration of WRED is directly or indirectly useful for voice traffic under the following conditions:

- WRED can be used to classify traffic on GSR platforms forming the core of a network. WRED can also serve as the queueing discipline. WRED uses and interprets IP Precedence to give priority to voice traffic over data traffic, dropping only data packets.
- When there are concurrent TCP flows on a link that supports voice traffic, WRED should be used for the TCP flows, whereas strict priority should be configured for the voice flow, if the feature is available.

Overview

WRED has the following features:

- It distinguishes between temporary traffic bursts that can be accommodated by the network and excessive offered load likely to monopolize network resources.
- It provides fair bandwidth reduction to curtail traffic sources in proportion to the bandwidth being utilized.
- It provides you with considerable flexibility including parameters to set minimum and maximum queue depth thresholds and packet drop probability.

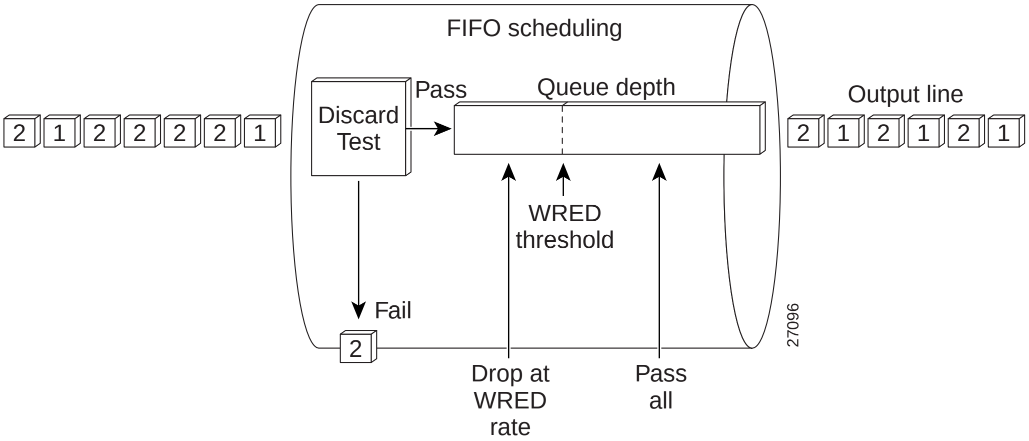
Using WRED, you can define minimum and maximum queue depth thresholds and drop probabilities for each class of service. You can use WRED to benefit VoIP traffic by specifying that WRED should not drop voice packets when the queue depth is exceeded, if voice packet drop is avoidable.

However, implicit in WRED are certain limitations in regard to voice traffic that you should understand if you plan to use it. For instance:

- WRED used alone does not give bandwidth guarantees to voice.
- WRED works best in conjunction with robust transport protocols such as TCP that react to packet drop. VoIP relies on UDP, which does not react to packet drop. In other words, WRED does not strictly protect against flows such as UDP that do not react to packet drop. Consequently, an extremely heavy UDP flow could overflow a WRED queue, resulting in loss of voice packets.

Figure 2-14 illustrates how WRED activity is carried out to determine which packets are enqueued and then dropped or sent during periods of congestion affecting the queue depth threshold.

Figure 2-14 Weighted Random Early Detection



When WRED is not configured, router and switch behavior allows output buffers to fill during periods of congestion. By default, tail drop is used to address the buffer congestion.

If a link is oversubscribed, a potentially large number of packets from numerous connections are discarded, leading to a condition called global synchronization. WRED handles this situation proactively. That is, when WRED is configured, instead of waiting for buffers to fill and tail drop to go into effect, WRED anticipates congestion by intelligent queue occupancy monitoring. It checks the buffer depth and performs early discards on selected packets and selected connections. Early discard improves the overall utilization of the link because it allows connections to slow down gradually.

WRED adds to its congestion avoidance features the ability to apply different discarding algorithms depending on the precedence. Thus, WRED can begin to discard traffic of precedence 0 (best effort) at a relatively low buffer occupancy, while it will discard traffic from precedence 1 only at a much higher buffer occupancy. Thus best-effort traffic will be slowed down much sooner than guaranteed traffic such as voice.

You can configure WRED on a per-PVC level so that it runs on each PVC queue to ensure low loss for high-precedence voice traffic on that queue.

Traffic Shaping and Policing

Traffic shaping provides the ability to limit the transmit rate of traffic out an interface to a specific peak/average value—a rate that is less than the link speed and therefore well tolerated. Policing, such as is enacted by the rate limiting feature of CAR, allows you to allocate bandwidth for traffic sources and specify policies for traffic that exceeds the configured bandwidth amount. Policing with CAR and shaping with FRTS both use the token bucket mechanism.

Traffic shaping is only applicable to Frame Relay or ATM networks. If the media is a leased line, FRTS does not apply. Use of FRTS is essential for Frame Relay to address a number of conditions, such as link (circuit) speed mismatch and oversubscription, both of which are described in this section. FRTS is also essential in cases for which you want to prohibit bursting above the committed information rate (CIR). Moreover, FRTS is required to support FRF.12.

- For each congestion avoidance feature, Table 2-10 shows the versions of the Cisco IOS software that support the feature, the switching mode used, and the platforms the feature runs on. “All Cisco IOS platforms” refers to this set of platforms: 1000, 1600 series, 1720, 2500 series, 2600 series, 3600 series, 4500 series, 4700 series, 7200 series, and RSP in the 7500.
- The following abbreviations are used to represent various switching modes in this table:
  - P = Process
  - F = Fast
  - N = NetFlow
  - O = Optimum
  - CEF = Cisco Express Forwarding
  - d = distributed (VIP)
  - dCEF = distributed CEF

Table 2-10 Cisco IOS Versions, Switching Modes, and Platform Support for Traffic Shaping and Policing Features

Feature	Cisco IOS Version and Switching Mode							Platform Support
	10.3	11.0	11.1	11.2	11.1CC	11.3	12.0	
Traffic Shaping and Policing								
Frame Relay Traffic Shaping (FRTS)	—	—	—	P, F, 11.2(9)	—	P, F	P, F, C	All Cisco IOS platforms
Committed Access Rate (CAR)	—	—	—	—	—	—	C	Cisco IOS 12.0: 2600 series, 3600 series, 4500 series, 4700 series, 7200 series Cisco IOS 12.0(4)T: 1600 series, 1720, 2500 series

Token Bucket Mechanism

A token bucket is used to manage a device that regulates the flow of data. For example, the regulator might be a traffic policer, such as CAR, or a traffic shaper, such as FRTS. A token bucket itself has no discard or priority policy. Rather, a token bucket discards tokens and leaves to the flow the problem of managing its transmission queue if the flow overdrives the regulator. (Neither CAR nor FRTS implement either a true token bucket or a true leaky bucket.)

A token bucket is a formal definition of a rate of transfer. It has three components: a burst size, a mean rate, and a time interval (Tc). Although the mean rate is generally represented as bits per second, any two values may be derived from the third by the relation shown as follows:

```
mean rate = (burst size)/(time interval)
```

Here are some definitions of these terms:

- Mean rate—Also called the CIR, it specifies how much data can be sent or forwarded per unit time on average.
- Burst size—Also called the Committed Burst (Bc) size, it specifies in bits per burst how much can be sent within a given unit of time to not create scheduling concerns.
- Time interval—Also called the measurement interval, it specifies the time quantum in seconds per burst.

By definition, over any integral multiple of the interval, the bit rate of the interface will not exceed the mean rate. The bit rate, may, however, be arbitrarily fast within the interval.

In the token bucket metaphor, tokens are put into the bucket at a certain rate. The bucket itself has a specified capacity. If the bucket fills to capacity, newly arriving tokens are discarded. Each token is granted permission for the source to send a certain number of bits into the network. To send a packet, the regulator must remove from the bucket a number of tokens equal in representation to the packet size.

If not enough tokens are in the bucket to send a packet, the packet either waits until the bucket has enough tokens or the packet is discarded. If the bucket is already full of tokens, incoming tokens overflow and are not available to future packets. Thus, at any time, the largest burst a source can send into the network is roughly proportional to the size of the bucket.

Note that the token bucket used for traffic shaping has both a token bucket and a data buffer, or queue; if it did not have a data buffer, it would be a policer. For traffic shaping, packets that arrive that cannot be sent immediately are delayed in the data buffer.

For traffic shaping, a token bucket permits burstiness but bounds it. It guarantees that the burstiness is bounded so that the flow will never send faster than the capacity of the token bucket plus the time interval divided by the established rate at which tokens are placed in the bucket. It also guarantees that the long-term transmission rate will not exceed the established rate at which tokens are placed in the bucket.

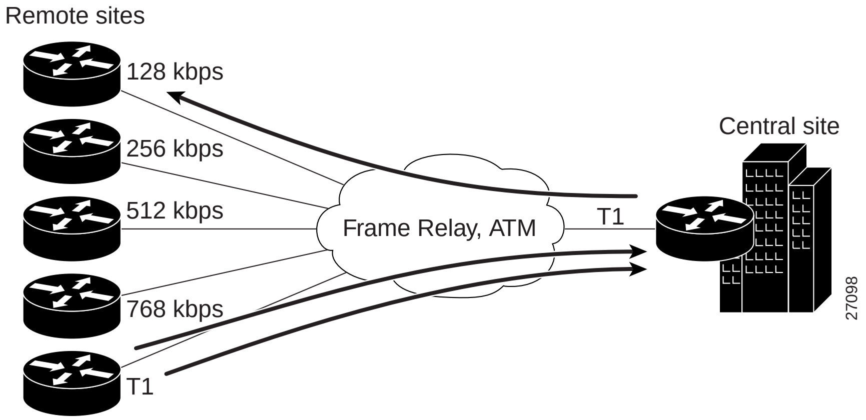
FRTS

FRTS uses queueing on a Frame Relay network to limit surges that can cause congestion. FRTS for voice traffic uses WFQ as an internal queue. FRTS allows you to control the traffic going out an interface in order to match its flow to the speed of the remote, target interface and to ensure that the traffic conforms to policies contracted for it. Using FRTS, you can ensure that traffic adhering to a particular profile can be shaped to meet downstream requirements, thereby eliminating bottlenecks in topologies with data-rate mismatches.

FRTS lets you specify an average bandwidth size for Frame Relay VCs. You can also define the bandwidth allocated for bursty traffic and control whether the circuit responds to notifications from the network that the circuit is becoming congested. Using FRTS, you can define minimum rate commitment for the virtual circuit and accommodate the occasional need for greater bandwidth.

As Figure 2-15 shows, Frame Relay allows many remote sites to aggregate upon a single circuit through use of VCs. Data networking is bursty in nature and Frame Relay takes advantage of oversubscription in relation to this burstiness. Not every remote site may be sending data toward the central site at the same time. However, the aggregate of all the remote site circuit speeds or PVC speeds may be greater than the central site circuit speed.

Figure 2-15 Frame Relay Links from Remote Sites Aggregating upon a Single Circuit at a Central Site



Overview

Packet loss can result in detrimental consequences for real-time and interactive applications such as voice applications. Using FRTS to shape traffic prevents packet loss. Use of FRTS is especially important in Frame Relay networks because the switch cannot determine which packets take precedence, and therefore which packets should be dropped when congestion occurs. Moreover, real-time traffic such as voice traffic requires that latency be bounded, thereby bounding the amount of traffic and traffic loss in the data-link network at any given time by keeping the data in the router that is making the guarantees. Retaining the data in the router allows the router to prioritize traffic according to the guarantees it is making.

Consider a Frame Relay network topology that consists of a high-speed (T1 line speed) connection at the central site and low-speed (less than 56 kbps) connections at the branch sites. This kind of topology involves a speed mismatch that often produces a bottleneck for traffic on a VC when the faster central site connection tries to communicate with the slower branch site connection. A bottleneck such as this introduces poor response times for interactive traffic such as voice traffic, especially when voice packets must wait behind a large FTP packet on a low-speed line.

FRTS can eliminate bottlenecks in Frame Relay networks that have high-speed connections at the central site and low-speed connections at branch sites. To limit the rate at which data is sent on the VC at the central site, you can configure rate enforcement (a peak rate configured to limit outbound traffic) on a per-VC basis to either the CIR or some other defined value such as the excess information rate (EIR). Rate enforcement can also be used in conjunction with the existing DLCI prioritization feature to further improve performance in this situation.

When a Frame Relay network is constructed with many VCs to different locations on a single physical line into the network, these VCs send traffic as fast as the physical line speed allows. The rate enforcement capability enables you to control the transmission speed used by the router by other criteria such as CIR or EIR, providing a mechanism for sharing media by multiple VCs. FRTS can preallocate the bandwidth each VC receives on the physical line into the network. This effectively creates a virtual time-division multiplexing (TDM) network.

A shaper, such as FRTS, typically delays excess traffic using a buffer, or queueing mechanism, to hold packets and shape the flow when the data rate of the source is higher than expected.

Advanced Concepts: Link Speed Mismatches

Link speed (circuit) mismatches can occur between high-speed links to low-speed links when data is sent from a fast central site link to a slower remote site link or when a central site is oversubscribed by remote sites. This section discusses the first case and how traffic shaping can reduce ensuing problems. The “Advanced Concepts: Traffic Shaping and Oversubscription” section discusses the second case.

For data-only networks that rely on TCP, TCP slow-starts allow a TCP flow between two endpoints to increase its effective flow rate gradually in order to determine the minimum bandwidth link between the two endpoints. This process negates the need for traffic shaping in data-only networks. This is not the case for voice transmission that relies on UDP networks.

Traffic sent out an interface may be shaped in such a way that the transmission rate may be different depending on the destination. This capability is necessary to address the differing remote site link speeds for circumstances, for instance, in which a central site router must shape traffic destined for one remote site at a rate of 56 kbps and shape traffic destined for another remote site at 128 kbps.

Consider, for example, that traffic leaving a high-speed interface such as a T1 line at a central site often terminates at a remote site that may have a circuit speed of, say, 56 kbps. By default, a router will transmit on an outgoing interface at a data rate that is equal to the clock rate of the circuit itself. This means that a T1 interface on a router will send data out at a T1 rate even if the remote site has a clock rate of 56 kbps. (This situation is analogous to four lanes on a highway merging into two lanes.) Congestion prevails, resulting from the slowdown of high-speed to slow-speed links. A consequence of this kind of congestion is that packets are queued in buffers, which manifests as delay. Buffers are finite storage entities that fill up to capacity and overflow, resulting in packet loss. Buffering works well for data frames, but the delay it incurs is detrimental to voice. Applying traffic shaping to a condition such as this one enables the queues to handle the T1 transmission, narrowing it down to fit the 56-kbps limitations without frame buildup in the queues, and thus eliminating delays or drops.

Consider another similar case that involves an FTP file transfer. Suppose a voice flow exists between two IP phones, one at a T1 site and one at a 56-kbps site while the FTP-based file is being transferred. The file transfer is started and congestion occurs in the Frame Relay network; the congestion causes buffering of voice packets in the transmission from the T1 line to the 56-kbps line. Even if voice traffic is given priority above the data traffic at the T1 router, buffering delay occurs in the Frame Relay network, and both data and voice are affected equally. Both flows are affected because the Frame Relay network operates at Layer 2, and therefore it has no regard for or knowledge of which frames are voice frames. Poor voice quality results under these circumstances as a consequence of dropped packets or delays. The severity of voice degradation incurred depends on the amount of congestion and buffering that exists in the Frame Relay network.

This possibility of WAN congestion due to link, or circuit, speed mismatches (high speed to low speed) makes evident the need for the edge router to shape the traffic before it enters the network, which will help to avoid congestion in the Frame Relay network. In this example—a T1 line to a 56-kbps line—you would shape the traffic to a rate lower than 56 kbps so that the rate would equal the CIR.

Advanced Concepts: Traffic Shaping and Oversubscription

A remote-to-central site configuration is deemed to be oversubscribed any time the aggregate transmit ability of the remote site exceeds the link bandwidth of the central site. Any oversubscribed situation is at risk of having voice negatively impacted depending on network conditions. Oversubscription is a commonly used design characteristic on data-only environments in which often many remote sites feed a central site. In this situation, the aggregate link speeds or CIRs of all the remote sites can exceed the link speed of the central site. Even if the CIRs of all the remote sites collectively are less than the central site link speed, oversubscription can occur when bursting of all the remote sites exceeds the central site link speed.

If you run voice traffic over a network design that allows for oversubscribed configuration, you must use traffic shaping to avoid the degradation to voice quality that can occur when the amount of traffic from the remote site exceeds the central site’s circuit speed buffering. For instance, if the burst period during which the remote site transmits data to the central site is extensive, packet drop could occur.

Because it is beneficial to data traffic, oversubscription is a common Frame Relay design implementation, and therefore it is impossible to eliminate it to properly facilitate voice. You should understand the risks to voice traffic involved in use of oversubscription and consider how to compensate for them.

If you want to allow data to be oversubscribed and concurrently guarantee voice quality, Cisco recommends that you configure the voice bandwidth to be the same as the minimum CIR (mincir) so that during congestion—that is, when bursting occurs—voice traffic will not be delayed or dropped.

Advanced Concepts: Bursting

On Frame Relay networks, the CIR specifies the guaranteed amount of information carried during periods of congestion. Bursting over the CIR—a beneficial, common, and intentional practice in a data-only Frame Relay network—creates a situation in which voice packets are sent out into the Frame Relay network essentially in a best-effort manner, subjecting them to packet drop.

Typical Frame Relay circuits have a link speed and an associated CIR that is in most cases less than the line rate. Many network providers allow applications to burst above the guaranteed CIR all the way up to the line rate. It is well understood that in the event of congestion, bursting about the CIR incurs the risk of packet loss because any data sent over the CIR is considered best-effort traffic.

If you run voice traffic across a network designed to support bursting, it is likely that voice will be included in the excess bandwidth above the CIR because Frame Relay has no way of differentiating between voice traffic and data traffic. If global congestion ensues, delay (caused by buffering) and packet drop could occur. Within a Frame Relay network, if frames must be dropped, those with the discard eligible (DE) bit set will be dropped first.

It is imperative to accept the premise that for a Frame Relay network to carry voice along with data, you must relinquish some of the benefits that bursting can give to data traffic because it is so detrimental to voice. Most importantly, to curtail the risks implicit in bursting, you should use traffic shaping on each PVC to ensure that the outbound traffic rate of traffic leaving the router going into the Frame Relay network does not exceed the CIR. Although this condition could reduce throughput of other data, it guarantees throughput and constant delay to voice traffic even when congestion occurs.

That said, there are a number of ways you can configure the Frame Relay network to handle traffic that bursts above the CIR when congestion occurs:

- You can drop frames that are sent above the CIR.
- You can drop frames with the DE set.
- You can mark frames above the CIR as DE and buffer (delay) them, but still send them.

Deploying QoS traffic regulation mechanisms such as policing and shaping throughout a network ensures that the correct QoS is rendered to a packet and that the data source adheres to the stipulated contract.

CAR

CAR provides a rate-limiting feature that allows you to allocate both bandwidth commitments and bandwidth limitations to traffic sources and destinations while specifying policies for handling traffic that exceeds the bandwidth allocation.

How Does CAR Apply to Voice Traffic?

Despite the advent of strict priority queueing for voice traffic, there remain some circumstances in which either the ability of CAR to police traffic through rate limiting or its IP Precedence traffic-marking ability is still useful for voice traffic. For instance, here are cases when you should use CAR for voice traffic:

- Rate limiting

  If the queueing and scheduling discipline applied to the link is First-In First-Out (FIFO), you should configure CAR to rate limit traffic. To give priority to voice traffic, you should rate limit data traffic, but not voice traffic. In order to police traffic advantageously for voice traffic, CAR must be able to differentiate between voice traffic and other forms of traffic. Therefore, you must mark the traffic with IP Precedence.

- Traffic marking

  In later releases of Cisco (IOS 12.0(5)T and beyond) that support use of a strict priority queue within WFQ, you no longer need to mark traffic with IP Precedence to satisfy voice traffic requirements. If you configure strict priority, voice traffic is enqueued in the priority queue and scheduled for transmission before any other traffic. However, use of strict priority for voice traffic is limited to Cisco IOS platforms. Moreover, there is no persistence at the packet level that indicates the packet is entitled to strict priority treatment; when strict priority is used, the packet is not marked with priority as it is when IP Precedence is used to set priority.

Therefore, although strict priority meets the queueing and scheduling requirements of voice traffic, largely replacing the need for WFQ and traffic marking, it is still important that you mark voice packets with an IP Precedence of 5 if that traffic will be sent across an internetwork. This requirement exists because platforms such as GSR routers, which are heavily used to form the core (or backbone) of networks such as ISP networks, do not support strict priority.

In some cases, WRED is used as the scheduling mechanism on GSR platforms, and WRED relies on IP Precedence to determine which packets to drop. Even traffic assigned to a priority voice class within CBWFQ should be marked with IP Precedence so that voice traffic can be differentiated from data traffic in the core of another network. You can use CAR to mark packets at the edge of the network.

Overview

CAR is used to enforce a maximum transmit rate (rate limit) for IP traffic only. Non-IP traffic is not rate limited. As a traffic policer, the rate limiting feature of CAR is most commonly configured on interfaces at the edge of a network to limit traffic into or out of the network. CAR does not smooth or shape traffic and thus does no buffering and adds no delay. CAR is highly optimized to run on high-speed links in distributed mode on VIPs on the Cisco 7500 series.

The CAR rate-limiting feature uses token bucket filters to measure the traffic load and limit sources to bandwidth allocations while accommodating the inherently bursty nature of IP traffic. For traffic that exceeds allocated bandwidth, CAR utilizes Extended ACLs to define policies, including bandwidth utilization thresholds under which packet priority is modified or packets are dropped.

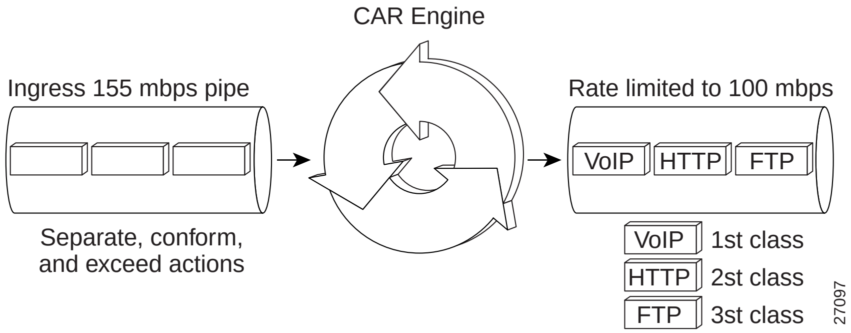
Using the rate limiting functionality of CAR, you can do the following:

- Define Layer 3 aggregate (matching all of the packets on an interface or subinterface) or granular (matching a particular type of traffic based on precedence, MAC address, or other parameters) access or egress bandwidth rate limits.
- Specify traffic handling policies when the traffic either conforms to or exceeds the specified rate limits.
- Control the maximum rate of traffic sent or received on an interface.

You can specify CAR rate-limiting policies based on criteria including physical port, packet classification, IP address, MAC address, application flow, or other criteria specifiable by access lists or extended access lists. You can implement CAR rate limits in either input or output interfaces or subinterfaces that support dCEF including Frame Relay and ATM subinterfaces.

Figure 2-16 conceptualizes the CAR rate-limiting process.

Figure 2-16 Committed Access Rate



Note dCEF switching must be enabled on any interface that uses VIP-Distributed CAR, even when only output CAR is configured. For dCEF configuration information, see the *Cisco IOS Switching Services Configuration Guide*. A VIP2-50 interface processor is strongly recommended when the aggregate line rate of the port adapters on the VIP is greater than DS3. A VIP2-50 interface processor is required for OC-3 rate PAs.

Advanced Concepts

CAR measures Layer 3 bandwidth utilization using a token bucket filter. Because CAR uses a token bucket, it can pass temporary bursts that exceed the rate limit as long as tokens are available. Thus, the token bucket can accommodate the bursty nature of IP traffic while still enforcing a specified rate limit.

Once a packet has been classified as conforming to or exceeding a particular rate limit, the router performs one of the following actions on the packet:

- Transmit—The packet is sent.
- Drop—The packet is discarded.
- Set precedence and transmit—The IP Precedence (ToS) bits in the packet header are rewritten. The packet is then sent.
- Continue—The packet is evaluated using the next rate policy in a chain of rate limits. If there is not another rate policy, the packet is sent.

CAR does not smooth or shape traffic and thus does no buffering and adds no delay.

You can also use CAR to specify more complex bandwidth management policies; to do so you can cascade or chain together a series of rate limits for specifying granular traffic control policies—the sequence of rate limits is evaluated and conform or exceed actions are executed until a packet is either sent or discarded. Packets falling through to the bottom of the list of rate limits are sent. Use of cascaded rate limits gives you very fine-grained network resource control.

Rate-limiting application examples include the following:

- Application-based rate limits: For example, limit HTTP web traffic to 50 percent of the link bandwidth, thus ensuring capacity for voice applications.
- Subrate IP services. For example, the network delivers a physical T1/E1 or T3/E3 to the customer but offers a less expensive subrate service, that is, 1 Mbps on an E1 or 10 Mbps on a T3. The customer only pays for the subrate bandwidth and may be upgraded to additional access bandwidth over time based on demand. CAR limits the traffic rate available to the customer and delivered to the network to the agreed upon rate limit and limits the ability to temporarily burst over the limit. The network operator may upgrade the service without any physical network rearrangement.
- Web hosting service. For example, a service provider hosts a web server farm providing outsourced services for multiple customers. Customers payment levels are set based on both bandwidth capacity and traffic prioritization. Customers may sign up for maximum delivered bandwidth of 2 Mbps and medium traffic prioritization. The service provider uses CAR functionality to limit the web server of the customer as the medium for appropriate traffic handling under congestion.

Classification

Packet classification enables network managers to specify policies that identify network traffic in order to partition, or classify, that traffic into multiple priority levels or CoS. Cisco IOS QoS for VoIP features include these technologies that you can use to classify packets:

- BGP Policy Propagation to set IP Precedence. Cisco IP Phones are a family of full-featured phones that can be plugged directly into the LAN. They include two models: 30VIP and 12SP+.
- Dial Peers, which can be used to set IP Precedence for voice packets.
- BGP Policy to set IP Precedence of QoS group ID

After classification, routers at the network edge ensure that packets within a class receive appropriate service levels, such as allocated rates and loss probability.

Table 2-11 shows the Cisco IOS software versions that support BGP Policy Propagation, the switching mode used, and the platforms it runs on.

Terms used in Table 2-11 are explained as follows:

- "All Cisco IOS platforms" refers to this set of platforms: 1000, 1600 series, 1720, 2500 series, 2600 series, 3600 series, 4500 series, 4700 series, 7200 series, and RSP in the 7500.
- "VIP distributed" refers to this set of platforms: 7000, 7500, and the Router Switch Module (RSM).
- The following abbreviations are used to represent various switching modes in this table:
  - P = Process
  - F = Fast
  - N = NetFlow
  - O = Optimum
  - CEF = Cisco Express Forwarding
  - d = distributed (VIP)
  - dCEF = distributed CEF

Table 2-11 Cisco IOS Versions, Switching Modes, and Platform Support for Classification Features

Feature	Cisco IOS Version and Switching Mode							Platform Support
	10.3	11.0	11.1	11.2	11.1CC	11.3	12.0	
Classification								
BGP Policy Propagation	—	—	—	—	dCEF, CEF	—	—	7500 VIP or RSP, 7200

BGP Policy Propagation

BGP is an interdomain routing protocol that exchanges reachability information with other BGP systems. The QoS policy propagation via Border Gateway Protocol (BGP) feature allows you to classify packets and then use other QoS features such as CAR and WRED to specify and enforce business policies to fit your business model.

Overview

BGP policy propagation leverages BGP to distribute QoS policy to remote routers in your network. It allows ingress routers to prioritize traffic. BGP policy propagation is defined by RFC 1163.

You can filter routing updates by specifying an access list on both incoming and outbound updates based on the BGP autonomous system path in the following ways:

- By setting IP Precedence for a packet.
- By tagging the packet with a QoS class unidentified (QoS group ID) internal to the router.
- A QoS group ID is a user-specified number that is assigned to a packet when that packet matches user-specified criteria. You set an internal QoS group ID, classify the packet with the number, then use that number to perform rate-limiting based on QoS group ID.
- Through source and destination address lookup.

You can specify whether the IP Precedence level or QoS group ID used is obtained from the source (input) address or destination (output) address entry in the route table. By setting the QoS group ID in addition to the IP Precedence, you can have more than eight classes on which to perform rate limiting.

For the QoS Policy Propagation via BGP feature to work, you must enable BGP and CEF/dCEF on the router. Subinterfaces on an ATM interface that have the **bgp-policy** command enabled must use CEF mode because distributed CEF is not supported. (Note that dCEF uses the VIP rather than the RSP to perform forwarding functions.)

IP to ATM CoS

Cisco IOS QoS software includes a feature suite that maps QoS characteristics between IP and ATM.

Two phases of this feature suite, characterized as follows, are highly useful for VoIP traffic:

- Phase 2, which encompasses two main parts:
  - IP to ATM CoS support for a single ATM VC, which allows you to use existing features, such as CAR or policy-based routing, to classify and mark different IP traffic by modifying the IP Precedence field in the IPv4 packet header. Subsequently, WRED or Distributed WRED can be configured on a per-VC basis so that the IP traffic is subject to different drop probabilities (and therefore priorities) as IP traffic coming into a router competes for bandwidth on a particular VC.
  - ATM VC bundle management allows you to configure multiple VCs that have different QoS characteristics between any pair of ATM-connected routers. These VCs are grouped in a bundle and are referred to as bundle members.
- Phase 3 IP to ATM CoS, per-VC WFQ, and CBWFQ allows you to apply CBWFQ functionality—normally applicable at the interface or subinterface levels only—to an individual VC configured for IP to ATM CoS. You can use this extension to IP to ATM CoS to apply either CBWFQ class-based WFQ or flow-based WFQ on a per-VC basis.

This section describes both of these IP to ATM CoS features. For each feature, Table 2-12 shows the versions of the Cisco IOS software that support the feature, the switching mode used, and the platforms it runs on.

- The following abbreviations are used to represent various switching modes in this table:
  - P = Process
  - F = Fast
  - CEF = Cisco Express Forwarding

Table 2-12 Cisco IOS Versions, Switching Mode, and Platform Support for IP to ATM CoS Features

Feature	Cisco IOS Version and Switching Mode							Platform Support
	10.3	11.0	11.1	11.2	11.1CC	11.3	12.0	
IP to ATM Class of Service								
VC Bundling (Phase II)	—	—	—	—	—	—	P, F, C	12.0(3)T: 7200 and 7500
Per-PVC PQ within CBWFQ	—	—	—	—	—	—	—	12.0(7)T: 7200 12.0(7)XE: 7500

IP to ATM, per-VC WFQ, and VC Bundling (Phase 2)

The IP to ATM Class of Service (IP to ATM CoS) feature implements a solution for coarse-grained mapping of QoS characteristics between IP and ATM, using Cisco PA-A3 ATM port adapters on Cisco 7200 and 7500 series routers. (This category of coarse-grained QoS is often referred to as CoS). The resulting feature makes it possible to support differential services in network service provider environments.

IP to ATM CoS provides a true working solution to class-based services, without the investment of new ATM network infrastructures. Using this feature, you can do the following:

- Offer different service classes (sometimes termed “differential service classes”) across the entire WAN, not just the routed portion.
- Give mission-critical applications exceptional service during periods of high network usage and congestion. In addition, noncritical traffic can be restricted in its network usage, which ensures greater QoS for more important traffic and user types.

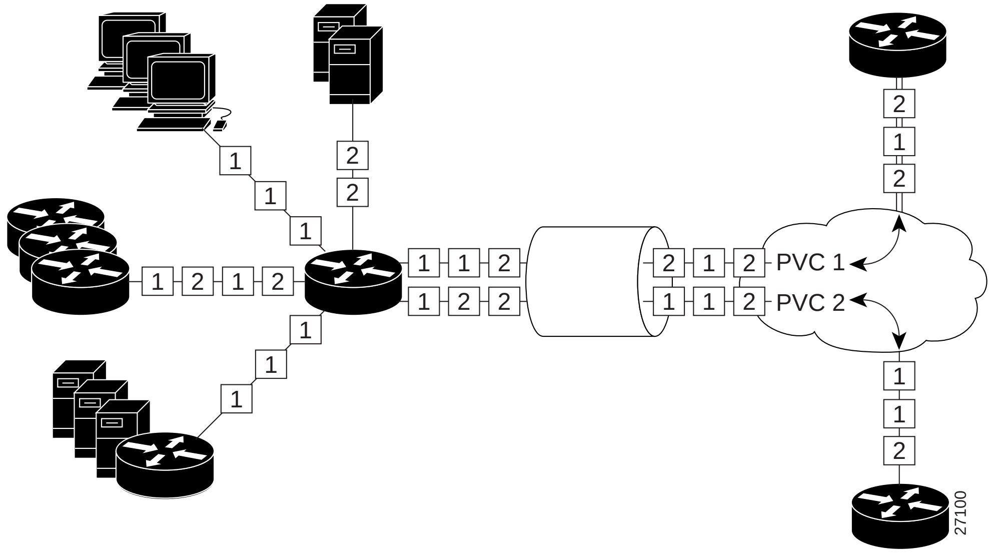
IP to ATM CoS supports configuration of both a single ATM VC and VC bundles. It does the following:

- Ensures effective differential classes over IP and traditional ATM networks. For instance, the VC bundle management feature provides for differentiated QoS by allowing for the coexistence of multiple VCs with different QoS characteristics from the same source to the same destination.
- Uses existing ATM infrastructures.
- Implements solutions for coarse-grained mapping of QoS characteristics called CoS between IP and ATM.
- Employs a high-performance design benefiting from distributed processing on the Cisco 7500 series routers and VIP.
- Uses the Cisco advanced PA-A3 ATM port adapter (PA), which supports traffic shaping and has rich ATM Service Category support. This PA is supported on the Cisco 7500+VIP and 7200 series routers.
- Provides per-VC queueing on the PA, per-VC back pressure, and per-VC WRED VIP queueing, which bring stability to a network by ensuring that system packets—such as BGP and Intermediate System-to-Intermediate System (ISIS)—are never dropped.
- Provides flexible management of the VC bundle on PVC failure.

IP to ATM CoS support for a single ATM VC allows you to use existing features, such as CAR or policy-based routing to classify and mark different IP traffic by modifying the IP Precedence field in the IPv4 packet header (PBR). Subsequently, WRED or Distributed WRED can be configured on a per-VC basis so that the IP traffic is subject to different drop probabilities (and therefore priorities) as IP traffic coming into a router competes for bandwidth on a particular VC.

Figure 2-17 shows two PVCs carrying traffic whose QoS is mapped to ATM VCs with equivalent QoS.

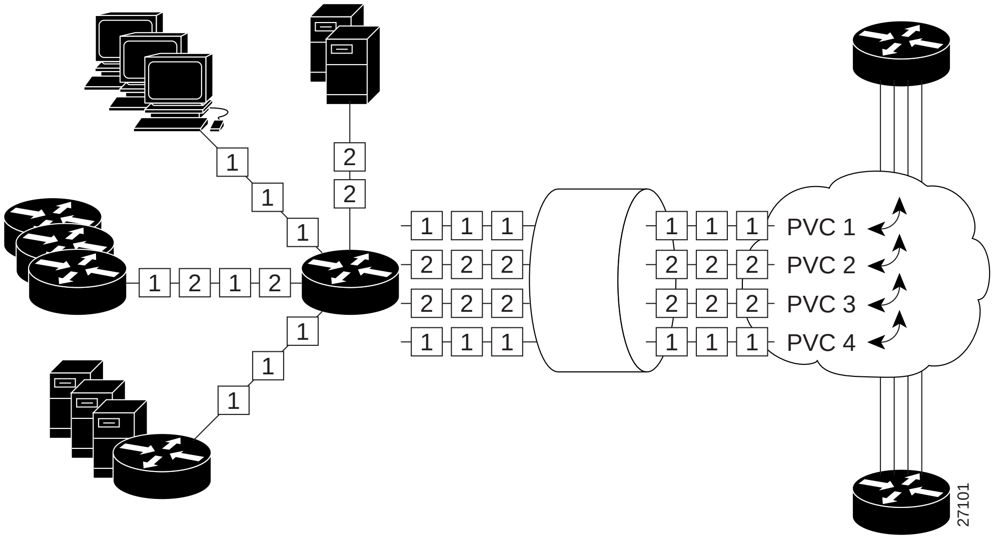
Figure 2-17 IP to ATM CoS Support for Single ATM VCs



PA-A3 ATM port adapters provide the ability to shape traffic on each VC according to the ATM service category and traffic parameters employed. When you use the IP to ATM CoS feature, congestion is managed entirely at the IP layer by WRED running on the routers at the edge of the ATM network.

IP to ATM CoS VC bundle management allows you to configure multiple VCs that have different QoS characteristics between any pair of ATM-connected routers. These VCs are grouped in a bundle and are referred to as bundle members. Figure 2-18 shows four PVCs whose traffic is mapped to ATM VCs with equivalent QoS characteristics.

Figure 2-18 IP to ATM CoS Support for VC Bundles



ATM VC bundle management allows you to define an ATM VC bundle and add VCs to it. Each VC of a bundle has its own ATM traffic class and ATM traffic parameters. You can apply attributes and characteristics to discrete VC bundle members or you can apply them collectively at the bundle level.

To use IP to ATM CoS VC bundle for VoIP traffic, you should mark voice traffic with an IP Precedence of 5, then map the voice traffic to a particular VC.

Using VC bundles, you can create differentiated service by flexibly distributing IP Precedence levels over the different VC bundle members. You can map a single precedence level or a range of levels to each discrete VC in the bundle, thereby enabling individual VCs in the bundle to carry packets marked with different precedence levels. Some VCs in a bundle could carry voice traffic, while others could carry non-voice data traffic. For mixed traffic, you can use WRED (or DWRED) to further differentiate service across traffic that has different IP Precedence levels but that uses the same VC in a bundle.

To determine which VC in the bundle to use to forward a packet to its destination, the ATM VC bundle management software matches precedence levels between packets and VCs. IP traffic is sent to the next hop address for the bundle because all VCs in a bundle share the same destination, but the VC used to carry a packet depends on the value set for that packet in the IP Precedence bits of the ToS byte of its header. The ATM VC bundle management software matches the IP Precedence of the packet to the IP Precedence value or range of values assigned to a VC, sending the packet out on the appropriate VC. Moreover, the ATM VC bundle management feature allows you to configure how traffic will be redirected when the VC the packet was matched to goes down.

The support of multiple parallel ATM VCs allows you to create stronger service differentiation at the IP layer. For instance, you might want to provide IP traffic belonging to VoIP traffic on an ATM VC with variable bit rate (VBR-rt PVC), for example, while transporting traffic other than real-time traffic over a more elastic ATM available bit rate (ABR) PVC. Using a configuration such as this would allow you to fully utilize your network capacity. You could also elect to transport best-effort IP traffic over an uncommitted bit rate (UBR) PVC—UBR is effectively the ATM version of best-effort service.

IP to ATM CoS, per-VC WFQ, and CBWFQ (Phase 3)

The IP to ATM CoS, per-VC WFQ, and CBWFQ feature with strict priority queueing allows you to apply CBWFQ functionality—normally applicable at the interface or subinterface levels only—to an individual VC configured for IP to ATM CoS and within CBWFQ to enable use of the strict priority queue for VoIP traffic. You can apply strict priority queueing to a CBWFQ class so that the class gets strict priority—that is, all of its data is sent before other classes sharing the interface are serviced.

Before looking at IP to ATM CoS, per-VC WFQ, and CBWFQ functions, consider briefly how CBWFQ and IP to ATM CoS work independently.

CBWFQ extends the flow-based WFQ functionality to provide support for user-defined classes. CBWFQ allows you to define traffic classes that are based on certain match criteria such as access control lists, input interfaces names, protocols, and QoS labels. Once a class has been defined according to its match criteria, you can assign it characteristics.

On an ATM network, you can use a VBR VC, which gives a form of strict priority, or you can use a strict priority class for voice traffic. For strict priority classes, you do not assign bandwidth to the class, as you would otherwise. To characterize a class, you specify the queue limit for that class, which is the maximum number of packets allowed to accumulate in its queue. Packets belonging to a class are subject to the queue limits that characterize the class as well as the bandwidth assigned when you enable strict priority for the class.

After you define traffic classes, you can configure one or more of them in a policy map to be attached as a service policy. CBWFQ allows you to create policy maps and attach them to interfaces or subinterfaces as service policies. The IP to ATM CoS, per-VC WFQ, and CBWFQ feature allows you to do the following:

- Create a policy map using standard CBWFQ.
- Within the policy map, apply the priority queueing feature, described in the “Priority Queueing within CBWFQ” section in this chapter, to one or more classes to be used for voice traffic.
- Apply the policy map to a VC to be used as a service policy for that VC.

Per-VC WFQ and CBWFQ allows you to differentiate the use of individual VCs within a bundle. For instance, you can apply one service policy to one VC belonging to a VC bundle and apply a different service policy to another VC belonging to the same bundle. You can also apply the same policy map to multiple VCs—whether standalone or bundle members—but each VC can have only one service policy. To concatenate service policies, you must create a third policy map and include in it all the classes that you want to use from policy maps you would have concatenated.

The IP to ATM CoS, per-VC WFQ, and CBWFQ feature allows you to apply a policy map to a VC to specify a service policy for that VC so that all traffic sent on that VC is categorized according to the classes and their match criteria defined by the service policy. In other words, IP to ATM CoS, per-VC WFQ, and CBWFQ takes the functionality defined for standard CBWFQ and for use of strict priority within CBWFQ and makes it available for application and use at the discrete VC level.

IP to ATM CoS allows you to configure a single, standalone VC or individual VCs belonging to a bundle. You also can configure collectively all VCs belonging to a bundle. However, for the IP to ATM CoS, per-VC WFQ, and CBWFQ feature, you can configure individual VCs only. That is, you can configure a standalone VC or a VC that belongs to a bundle, but you cannot use per-VC WFQ and CBWFQ to configure a bundle of VCs collectively.

Note For ATM, you cannot use RSVP or IP RTP Priority to give voice traffic strict priority queueing; these constraints render strict priority queueing with CBWFQ all the more requisite for voice traffic.

Although CBWFQ allows you to guarantee bandwidth allocation to a voice class that is a strict-priority class, this act alone will not eliminate jitter because CBWFQ will eventually service data queues. If one of those data queues contained a large packet—for instance, a 1500-byte packet—onset servicing of the data packet after the strict priority queue contents were exhausted might detain delivery of voice traffic subsequently enqueued to the strict priority queue. Moreover, on ATM networks, you cannot fragment large, delay-inducing packets.

If your network usage shows a preference for voice over data at the expense of data, you could allocate all available bandwidth up to 75 percent to the voice class or queue. This circumvention is considered an imperfect solution because eventually the data traffic would be sent in a best-effort fashion, introducing some jitter to voice traffic.

Signalling

True end-to-end QoS requires that every element in the network path—switch, router, firewall, host, client, and so forth—deliver its part of QoS, and that all of these entities be coordinated with QoS signalling. To achieve end-to-end QoS, signalling must span the entire network.

An aspect of providing QoS signalling for VoIP is not to allow the call to be passed onto the IP network unless sufficient resources are available. This feature is called Call Admission Control (CAC). Mechanisms that implement CAC involve the router maintaining a knowledge of the IP network and determining if there is enough bandwidth to allow the call to be placed as a VoIP call. If the bandwidth is not sufficient, the router would signal back to the originating device that the call should be placed via alternative means.

This section describes RSVP, which is a signalling and resource reservation protocol that also provides CAC. This section also describes the problems that exist today in relation to use of RSVP for voice applications.

Tips If you choose not to use RSVP, you should ensure that voice call patterns do not oversubscribe the available bandwidth of the network. The usual way to ensure this is to overlay the worst-case scenario, the calling pattern of the busiest hour, onto the network and ensure that the required traffic can be accommodated by network trunks. If the network can accommodate the traffic, CAC is unnecessary; thus RSVP is not needed. Alternatively, you could design your network to limit the number of voice ports so that there can never be too many voice calls for the carrying capacity of any particular trunk.

RSVP

RSVP, which runs over IP, allows applications to dynamically request and reserve network resources that meet their specific and stringent QoS requirements. RSVP is the first significant industry-standard protocol for dynamically setting up end-to-end QoS across a heterogeneous network. RSVP provides guaranteed service and admission control. It uses out-of-band signalling, a method of signalling in which hosts send additional packets beyond data packets—that is, out-of-band packets—to indicate that a certain QoS service is desired for a particular traffic flow.

RSVP is a setup protocol used by a host to request specific qualities of service from the network for particular application flows. RSVP is also used by routers to deliver QoS requests to other routers (or other types of nodes) along the path(s) of a flow. RSVP requests generally result in resources being reserved in each node along the path.

For networks supporting VoIP, RSVP—in conjunction with features that provide queueing, traffic shaping, and voice call signalling—can provide call admission control for voice traffic. Call admission control determines whether a voice call should be accepted or denied based on available resources. At the present time, there are a number of problems pertaining to use of RSVP for voice applications. After these problems are explained, this section describes how RSVP works.

Current Limitations of RSVP in Relation to Voice Applications

Here are the three main problems that exist in relation to use of RSVP for voice applications:

- Resource-based admission control is not synchronized with voice call signalling.

  The resource-based admission control process is not initiated until Q.931 signalling within H.323 or Session Initiation Protocol (SIP) signalling makes known the destination gateway port numbers. Because of this sequence of events, the voice call signalling process is almost always completed before the resource-based admission control process can commence.

  If the required resources could not be reserved, the resource-based admission control process would fail. Given the sequence of events, RSVP notification of failure may occur after the destination phone had been picked up or while it was ringing. In either case, it would be too late to tear down the link for the voice call. The quality of the call would be poor, because it would have gone through without the guarantees of RSVP. The poor quality might exist for the duration of the call. If, during the call, the RSVP reservation were accepted, the call quality would improve. However, even if the RSVP reservation were initiated at a later stage while the call was still active, it could fail again due to insufficient resources.

- Bandwidth made available through CRTP is not recognized by RSVP. Thus, additional voice calls that could have been accommodated would be rejected or they would receive best-effort service.

  The terminating gateway or the end station requests for full bandwidth for the voice call are based on the negotiated codec. If RTP header compression occurs through use of CRTP on any links throughout the end-to-end path, the bandwidth made available through flow compression is not recognized. Although the bandwidth is available for other uses, loss of use of it for the voice application can have considerable, negative effect, especially on low-bandwidth access links. That is, based on the bandwidth made available through CRTP compression but not accessible to the voice application, a certain number of calls are rejected or given best-effort service due to admission control. On low bandwidth links, it is important to make use of the entire (actual) available bandwidth to put calls through, not just the bandwidth amount specified on the RSVP RESV (reservation) message.

- RSVP uses WFQ and does not give strict priority to voice traffic.

  Voice traffic requires strict priority so as to minimize delay and jitter. The end-to-end delay restriction for voice traffic is 150 ms. Although WFQ includes a strict priority queue, this queue is not yet used by RSVP. Rather, RSVP relies on WFQ to provide fairness among flows, assigning a low weight to the queue for the voice flow to give it priority. This priority is insufficient to minimize the jitter for voice traffic.

These restrictions on the ability of RSVP to fully utilize bandwidth resulting from compression and to provide voice applications with the strict priority service it requires are being addressed and will be resolved in future release of Cisco IOS software.

How RSVP Works

RSVP works in conjunction with mechanisms at end systems to request services. To ensure that QoS conditions are met, RSVP clients provide the intermediate network nodes an estimate of the data traffic they will generate. This information is provided with a traffic specification and a service-request specification.

RSVP uses WFQ. When an intermediate router agrees to an RSVP request, RSVP sets up a queue for traffic within the reserved conversation and allocates enough weight to the queue to give absolute priority over the interface to the bandwidth requested. If the reserved traffic does not use all of the bandwidth, other traffic on the interface will be able to use it.

RSVP offers these benefits:

- Bandwidth guarantee. An RSVP reservation, if accepted, will guarantee that those reserved resources will continue to be available while the reservation is in place.
- Media independent reservation. An end-to-end RSVP reservation can span arbitrary lower layer media types.
- A topology-aware reservation. Once an RSVP reservation is accepted, the subsequent RSVP flows belonging to that reservation will follow the reserved network path.
- Data classification. While a reservation is in place, data packets belonging to that RSVP flow are separated from other packets and forwarded as part of the reserved flow.
- Data policing. Data packets belonging to an RSVP flow that exceed the reserved bandwidth size are marked with a lower packet precedence.

RSVP is used by a host to request specific qualities of service from the network for a particular traffic classification or application flow. RSVP is also used by routers to deliver QoS to other routers or nodes along the path of a flow. RSVP requests generally result in resources being reserved in each node along a path.

RSVP operates in tandem with unicast and multicast routing protocols. RSVP, itself, is not a routing protocol. A routing protocol determines where packets get forwarded; RSVP is concerned with the QoS those packets receive. RSVP consults the local unicast or multicast routing database to obtain routes.

RSVP requests for a specific QoS are made by the receiver, not the source. Putting the onus on the receiver allows RSVP to scale to very large multicast groups. That is, receiver-oriented reservation requests can more easily accommodate large groups, dynamic group membership, and heterogeneous receiver requirements. Receiver-oriented reservation requests merge as they progress up the multicast tree.

RSVP is a setup protocol that allows applications, such as voice applications, that require guaranteed bandwidth for successful operation to request it. The need for network resource reservations differs for voice traffic and data traffic in these ways:

- Voice traffic experiences problems when operating over datagram services. Voice traffic sends an almost constant flow of information, so the transmission must embody constancy. Some guarantee must be provided that service between real-time hosts will not vary. Routers operating on a FIFO basis risk unrecoverable disruption of the real-time information that is being sent.

  RSVP mechanisms enable real-time traffic to reserve resources necessary for consistent latency. RSVP checks and repeats reservations at regular intervals. By this process, RSVP can adjust and alter the path between RSVP end systems from router changes. However, RSVP entails a number of limitations in regard to voice traffic. For details, see the “Current Limitations of RSVP in Relation to Voice Applications” section in this chapter.

- Data traffic seldom needs reserved bandwidth because internetworks provide datagram services for it. This asynchronous packet switching does not need guarantees of quality of service. Routers can operate in a FIFO manner for data traffic packets. End-to-end controls between data traffic senders and receivers help ensure adequate transmission of bursts of information.

For RSVP, a set of packets is treated like a flow and the manner in which its treatment is specified is known as a flow specification, referred to as a flowspec. The flowspec describes the traffic sent and the service requirements of an application, which results in a request for the desired QoS. The flowspec can be considered to represent a reservation request.

In addition to a flowspec, reservation requests also consist of a filterspec. The filterspec specifies those packets that will be serviced by the flowspec. Think of a flowspec as defining the QoS and a filterspec as qualifying certain traffic for the QoS.

RSVP implements QoS for a particular flow using mechanisms collectively called traffic control. These mechanisms include:

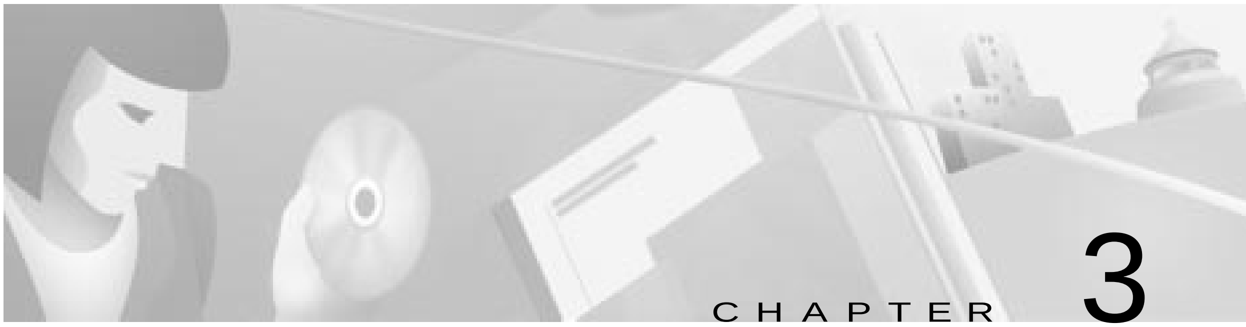
- A packet classifier that determines the QoS class, and perhaps the router, for each packet.
- An admission control function that determines whether the node has sufficient available resources to supply the requested QoS.
- A packet scheduler that determines when particular packets are forwarded to meet QoS requirements of a flow.

Consider how these aspects of RSVP work together to create a reservation. A host uses RSVP to request a specific QoS service from the network on behalf of an application data stream. To ensure that QoS conditions are met, RSVP clients provide the intermediate network nodes an estimate of the data traffic they will generate.

RSVP requests the particular QoS, but it is up to the interface queueing mechanism to implement the reservation. RSVP carries the request through the network, visiting each node the network uses to carry the stream. At each node, RSVP attempts to make a resource reservation for the stream using its own admission control module, exclusive to RSVP, which determines whether the node has sufficient available resources to supply the requested QoS.

If either resources are unavailable or the user is denied administrative permission, the RSVP program returns an error notification to the application process that originated the request. If both attempts succeed, the RSVP daemon sets parameters in a packet classifier and packet scheduler to obtain the desired QoS. The packet classifier determines the QoS class for each packet and the scheduler orders packet transmission to achieve the promised QoS for each stream.

WFQ (or WRED) sets up the packet classification and the scheduling required for the reserved flows. Using WFQ, RSVP can deliver an integrated services guaranteed rate service. Using WRED, RSVP can deliver a controlled load service.



CHAPTER 3

Providing Branch Office Site Users and Telecommuters VoIP Access to the Main Campus

This chapter explores the quality of service (QoS) for Voice over IP (VoIP) features deployed for the various links of an end-to-end VoIP path throughout a corporate network that encompasses a main enterprise campus network with remote branch offices and telecommuters. The chapter explains how to configure these QoS features for optimum voice quality delivery.

This chapter includes the following sections:

- Scenario Description
- Providing Small Branch Office Users Access Across 64-K Links
- Providing Medium Size Branch Office Users QoS for VoIP Access to the Main Campus Using T1 Links

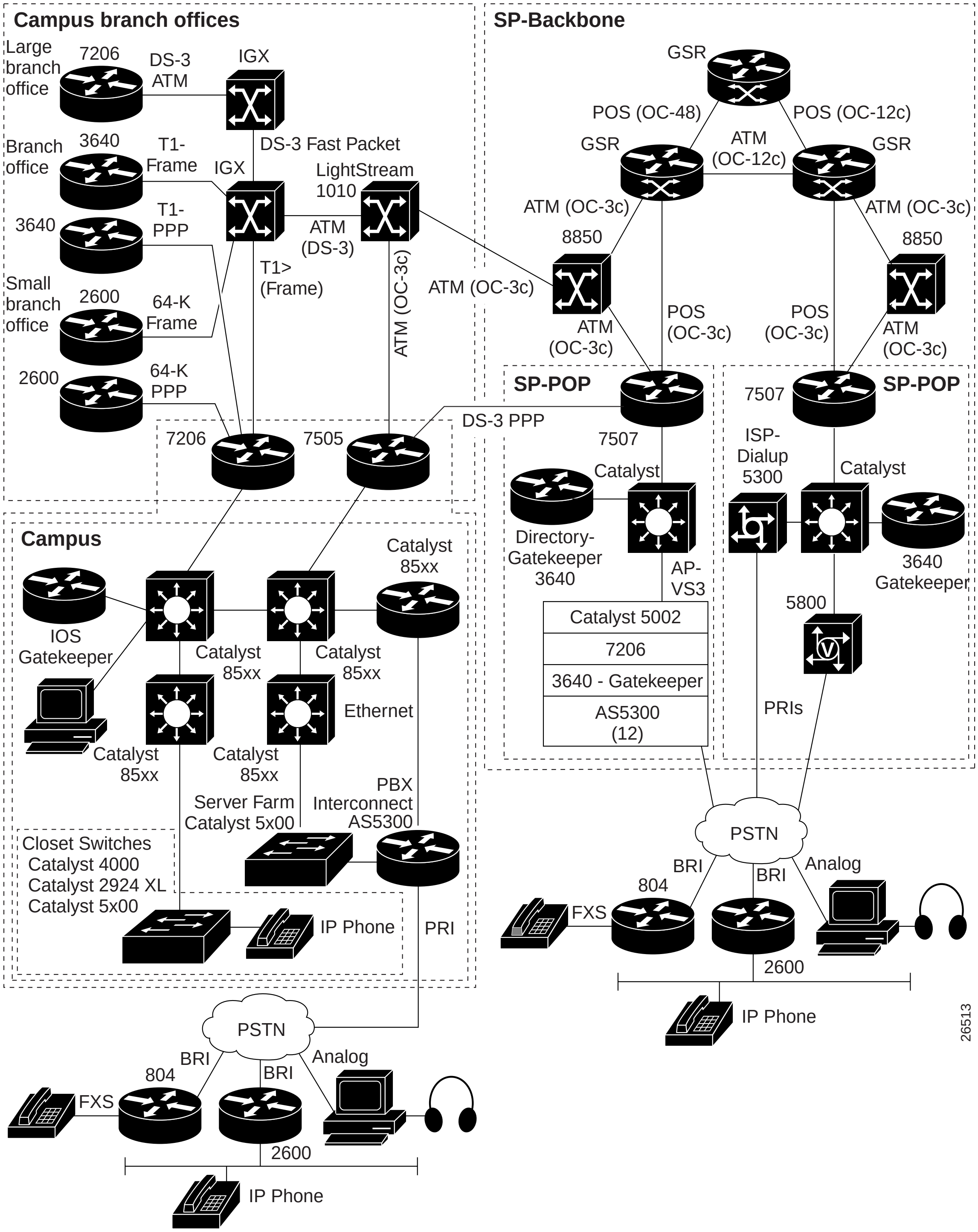
Scenario Description

The chapter addresses a scenario that illustrates how to deploy QoS features for the links comprising segments of an end-to-end VoIP path through a corporate campus (enterprise) network that has been rearchitected to carry packetized voice in addition to regular data. (QoS for VoIP configuration for segments of the end-to-end path are described in their own sections in this chapter.) The complete end-to-end path gives VoIP access to all corporate members. All employees—those who work at the main campus, the branch offices, and at home—use VoIP for telephone communication with other employees.

The internetwork that includes the campus network encompasses Internet Service Provider (ISP) and Wide Area Network (WAN) networks. The campus network remote branch offices are connected to the main campus through WAN links, which also interconnect the campus network to the ISP. This chapter focuses exclusively on the end-to-end path across the campus network, but users could make calls to and receive them from other users across the internetwork.

Figure 3-1 shows the complete end-to-end QoS for VoIP path throughout the corporate campus—but in this case the path is shown in relation to the entire internetwork of which it is a part.

Figure 3-1 QoS for VoIP Path for Main Campus, Branches, and Telecommuters within the Complete Internetwork



Although all voice communication occurs over VoIP, the underlying link technology differs for the various segments of the end-to-end VoIP path into and out from the campus. Table 3-1identifies each of the path segments and gives their link characteristics

.

Table 3-1 Link Type for Segments of the Campus QoS for VoIP End-to-End Path

Use of Segment	Link Type
Access to and from small branch office	• 64-K Point-to-Point Protocol (PPP) or • 64-K Frame Relay
Access to and from medium branch office	• T1 (1.544 Mbps) Frame Relay or • T1 (1.544 Mb) PPP
Access to and from telecommuters	• Access over a PSTN using ISDN or analog dial connectivity.

The end-to-end QoS for VoIP path throughout the corporate network allows for all variations of caller-to-called relationships. For instance, a call could be placed by a user at the small branch office or the medium size branch office and terminate at a telecommuter's home phone. Here is a summary of the relationships the QoS for VoIP end-to-end path enables within the campus network:

- Branch office users can call other branch office users, users in the main campus, and telecommuters.
- Telecommuters can call other telecommuters, users in the main campus, and branch office users.
- Main campus users can call other main campus users, branch office users, and telecommuters.

Table 3-2 shows the salient hardware features for the corporation's network.

Table 3-2 Campus Routers and Switches Used in End-to-End QoS for VoIP Path

Router	Location in End-to-End Path	Use
Cisco 2600 router	Small branch office	VoIP gateway.Converts digital phone calls into packetized (analog) voice traffic, and the reverse.
Cisco 3640 router	Medium size branch office	VoIP gateway. Converts digital phone calls into packetized (analog) voice traffic, and the reverse.
Cisco 7206 router	Campus access from and to the branch offices	Aggregate router for corporate leased lines and packet-switched lines.
Catalyst 8500 switches	Packet switching within the main campus	Core and distribution routers within the main campus network.
Cisco 5300 router	PBX interconnect from and to the PSTN used by telecommuters.	VoIP gateway. Supports ISDN PRI channels and analog phone line calls from and to the PSTN.

Providing Small Branch Office Users Access Across 64-K Links

This section describes the QoS for VoIP configuration for the following two types of links, either of which could be used to provide VoIP access to and from the small branch office users so that they can communicate with one another, main campus users, and telecommuters:

- Using a 64-K PPP Link
- Using a 64-K Frame Relay Link

This topology configuration assumes that the small branch office site accommodates 100 or fewer users.

Using a 64-K PPP Link

This section describes the link type and the QoS features for VoIP you should use for a 64-K PPP link and configuration.

64-K PPP Link Type and Configuration

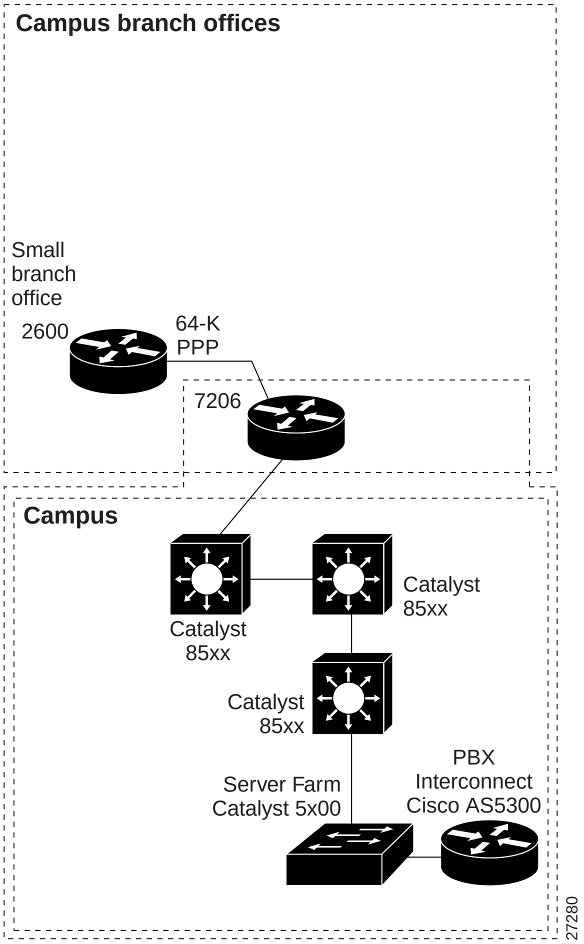
The campus (corporate) network provides a 64-K PPP link for access to and from the small branch office and main campus. This link is directly connected to the Cisco 7206 router giving access to the main campus. Because the 64-K PPP link is a slow-speed link, it requires use of specific QoS features different from those used for a faster speed link, such as a T1 line. For instance, to obviate (or at least control) delay and jitter, a slow-speed link requires that large data packets be fragmented and interleaved with voice packets.

Most of the same QoS features used for the 64-K Frame Relay link (described in “Using a 64-K Frame Relay Link” section on page 3-10) are also used for PPP. Both PPP and Frame Relay 64-K links require use of priority queueing in order to grant voice the strict priority it requires. They both also require use of CRTP to compress the large RTP header to provide greater throughput. Fragmentation of large data packets is done on both links, but using different QoS technologies. Link Fragmentation and Interleaving (LFI) is supported on PPP links, whereas FRF.12 is supported on Frame Relay links for the same purpose.

Figure 3-2 shows the path from the 2600 router at the small branch office which is directly connected via the 64-K PPP link to the 7206 campus access router into the main campus. The 2600 router at the small branch office is configured as a VoIP gateway that allows for codec conversion (using G.729a) of analog voice to packetized digital voice and back. Users throughout the branch office can make phone calls to users throughout the main campus and telecommuters. Routed through the gateway, their analog calls are converted to packetized digital data transmitted using VoIP.

This portion of the configuration focuses on the path segment from the 2600 router to 7206 router. However, Figure 3-2 shows the full path through the campus to the PBX interconnect 5300 router.

Figure 3-2 End-to-End QoS for VoIP Path Between a Small Branch Office Router and Campus Access Over a 64-K PPP Link



Configuring QoS for VoIP on a 64 K PPP Link

This section describes the configuration for deploying the following QoS for VoIP features for the 64-K PPP link:

- Compressed Real-Time Protocol (CRTP)—Used to compress the large RTP header.
- Link Fragmentation and Interleaving (LFI)—Used as the large data packet fragmentation and interleaving mechanism.
- IP RTP Priority queueing (Also referred to as Priority Queueing-Weighted Fair Queueing (PQWFQ))—This feature is used as the queueing mechanism to give voice traffic strict priority service.

Example 3-1 shows the configuration commands for the 2600 router 64-K PPP link with the commands for the QoS features called out in bold.

Example 3-1 64-K PPP Link Configuration with QoS for VoIP

```
 interface Loopback 0
 ip address 10.14.26.26 255.255.255.0
 h323-gateway voip interface
 h323-gateway voip id r1-3640a-gk.cisco.com ipaddr 10.12.254.3 1719
 h323-gateway voip h323-id r1-2600@cisco.com
!
interface Serial0/1
 bandwidth 64
 ip address 10.14.97.2 255.255.252
 encapsulation PPP
 ppp multilink
 multilink-group 1
!
interface multilink 1
 ip unnumbered Serial0/1
 ip rtp header-compression iphc-format
 ip tcp header-compression iphc-format
 fair-queue 64 256 0
 ppp multilink
 ppp multilink fragment-delay 20
 ppp multilink interleave
 ip rtp priority 16384 16384 48
```

Configuring a Multilink Interface with QoS for VoIP Features

This section describes the QoS features you configure for the PPP link using a multilink interface. Most of these QoS features are the same as those that you configure for a Frame Relay 64-K link. Therefore, instead of repeating conceptual information, these steps refer to the related Frame Relay sections that cover configuration steps for common features.

Note This configuration process provides two alternative ways of configuring strict priority queueing for VoIP traffic. In this set of steps, both methods are presented as Step 2. The first Step 2 shows how to configure strict priority using IP RTP Priority. The second (alternative) Step 2 shows how to configure strict priority queueing using strict priority within CBWFQ.

Step 1: Configure CRTP for the PPP Link

To enable RTP header compression, perform the following task in interface configuration mode (you need to enable compression on both ends of the connection).:

Command	Purpose
ip rtp header-compression iphc-format	Enables RTP header compression for VoIP packets.

For background information on CRTP, see the “Step 1: Configure CRTP for a 64-K Frame Relay Link” section. Also, see Chapter 2, “About QoS Features for Voice.”

Step 2: Configure WFQ and Enable the Strict Priority Queue for VoIP Packets on the PPP Link

When WFQ is enabled, it creates a strict priority queue that exists potentially for use by delay-sensitive traffic such as voice traffic. However, the strict priority queue cannot be used until it is enabled through configuration of the **ip rtp priority** command. This section gives the command syntax for the **fair-queue** and **ip rtp priority** commands that you use to enable WFQ and its strict priority queue.

To enable WFQ for the PPP interface shown in Example 3-1, set the congestion threshold after which messages for high-bandwidth conversations are dropped, and specify the number of dynamic and reservable queues, perform the following task in interface configuration mode after specifying the interface:

Command	Purpose
fair-queue 64 256 0	Configures the PPP interface to use weighted fair queueing with a congestive discard threshold of 64, 256 dynamic queues, and no reservable queues.

To reserve a strict priority queue for a set of RTP voice packet flows belonging to a range of UDP destination ports, use the following command in interface configuration mode:

Command	Purpose
ip rtp priority 16384 16383 48	Reserves a strict priority queue for the VoIP packet flow using destination ports in the range of 16384 and 16383. VoIP flows for the example PPP link whose UDP destination ports are within the range of 16384 and 16383 are granted strict priority with a bandwidth guarantee of 48 kbps. The example configuration is designed to service 4 voice calls at the cost of 12 K per call. For your configuration, assume that each call consumes 12 K and set the bandwidth parameter to the result of the following equation: *bandwidth* = 12K x *number-of-calls* The IP RTP Priority feature does not require that you know the port of a voice call. Rather, the feature gives you the ability to identify a range of ports whose traffic is put into the priority queue. Moreover, you can specify the entire voice port range—16384 to 32767—to ensure that all voice traffic is given strict priority service.

For background information on WFQ and the strict priority feature, see the "Step 2: Configure WFQ and Enable the Strict Priority Queue for VoIP Packets on the PPP Link" section. Also, see Chapter 2, "About QoS Features for Voice."

Step 2: Alternative Configuration: Configure WFQ and Enable the Strict Priority Queue within CBWFQ for a 64 K PPP Link

This step provides an alternative method of giving voice traffic strict priority queueing. It describes how you can use the same feature described in the first Step 2, but within CBWFQ to apply strict priority queueing to a CBWFQ class used for voice traffic.

Priority queueing within CBWFQ enables use of the strict priority queue implicit to WFQ. Although it is possible to enqueue various types of real-time traffic to the strict priority queue, it is strongly recommended that you direct only voice traffic to it. This recommendation is made because voice traffic is well-behaved, whereas other types of real-time traffic are not. Moreover, voice traffic requires that delay be nonvariable in order to avoid jitter. Real-time traffic such as video could introduce variation in delay thereby thwarting the steadiness of delay required for successful voice traffic transmission. For complete information the Priority Queueing within CBWFQ feature, see Chapter 2, "QoS Features for Voice over IP."

This section gives the command syntax for the **fair-queue** and **ip rtp priority** commands that you use to enable WFQ and its strict priority queue.

To enable weighted fair queueing for the PPP interface shown in Example 3-1, set the congestion threshold after which messages for high-bandwidth conversations are dropped, and specify the number of dynamic and reservable queues, perform the following task in interface configuration mode after specifying the interface:

Command	Purpose
fair-queue 64 256 0	Configures the PPP interface to use weighted fair queueing with a congestive discard threshold of 64, 256 dynamic queues, and no reservable queues.

To create a class for voice called "class voice" and configure that class as a strict priority class with a bandwidth allocation of 50 kbps, use the following commands beginning in interface configuration mode:

Command	Purpose
class-map voice	Creates a class and names it voice.
match access-group 102	Specifies that all traffic belonging to the numbered access group 102 will belong to the voice class.
policy-map voiceplus	Creates a policy map called voice to be applied to the interface.
class voice	Adds the class called voice to the policy map voiceplus.
priority 50	Enables the strict priority queue for the class called voice and sets the bandwidth for the voice traffic to 50 kbps.

Step 3: Configure Link Fragmentation and Interleaving for a 64-K PPP Link

Because voice packets are small in size and can be detained between large data packets sent out the same interface, you should use LFI on slow-speed PPP links. (When you enable LFI, large data packets are fragmented and the small voice packets are interleaved between the data fragments.)

To configure LFI on a 64-K PPP link, use the following commands:

Command	Purpose
ppp multilink	Enables MLP on the PPP link.
ppp multilink fragment-delay 20	Configures a maximum fragment delay of 20 ms. This gives the voice stream a maximum bound on delay of 20 ms. In this case MLP will choose a fragment size based on the configured value.
ppp multilink interleave	Enables real-time packet interleaving on the PPP link

Note At higher link rates, the bandwidth savings of CRTP may be outweighed by additional CPU load. For slow-speed links, CPU utilization is generally not very high. However, you can monitor CPU utilization to ensure that it is not above 75% when CRTP is configured.

Outcome Measurements for a 64-K PPP Link QoS for VoIP Configuration

To convey a sense of how application of QoS features affects the quality of voice transmission, this section provides measurements that show a set of results produced when the following criteria, shown in Table 3-3, vary:

- QoS Feature Used. This column identifies which QoS features were configured to satisfy voice traffic requirements and improve voice quality.
- Concurrent TCP Data Traffic (noise level). This column identifies the amount of data traffic transmitted on the same line as voice.
- Number of Voice Calls: This column gives the throughput—the number of voice calls put through for the test.
- Average PSQM Score

Table 3-3 Voice Call and Quality Measurements for 64-K PPP

QoS Feature Used	Concurrent TCP Data Traffic	Number of Voice Calls	PSQM Score Average
Without QoS **(to establish baseline)**	none (baseline)	1	1.69542
		2	1.74596
		3	2.0772
		4	2.08331
WFQ	55 kbps 19 pps 5% CPU 2 TCP streams	1	2.73794
		2	3.38811
		3	3.76196
		4	4.99414

Table 3-3 Voice Call and Quality Measurements for 64-K PPP (continued)

QoS Feature Used	Concurrent TCP Data Traffic	Number of Voice Calls	PSQM Score Average
CBWFQ	55 kbps	1	2.87017
	19 pps	2	2.40119
	5% CPU	3	4.21419
	2 TCP streams	4	n/a
PQWFQ **(Priority Queueing within CBWFQ)**	55 kbps	1	2.062
	19 pps	2	2.12987
	5% CPU	3	1.91315
	2 TCP streams	4	2.3743

Using a 64-K Frame Relay Link

This section describes the link type, the QoS features for VoIP you should use for this type of link and circumstance, and how to handle conditions you might encounter that could negatively affect voice traffic.

64-K Frame Relay Link Type and Configuration

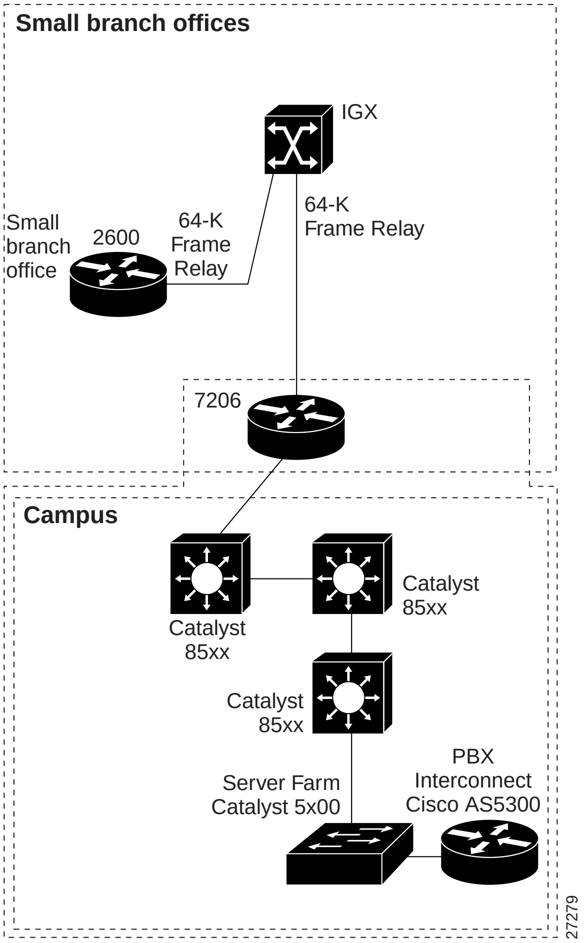
The campus (corporate) network provides a 64-K Frame Relay link for access to and from the small branch office and main campus. This link connects from the small branch office 2600 router through the IGX switch to the T1 Frame Relay link. The T1 Frame Relay link gives access to the campus through the 7206 router. Although not shown in the illustrations depicting this scenario or path segment, connected to the small branch office 7206 router is the network used by the 100 or so employees comprising the small branch office group. Let's assume that the small branch office network uses fast-speed (10-to-100 Mb Ethernet) links. Because the Ethernet links provide greater bandwidth (and, thus, faster speed) than does the 64-K Frame Relay link out from the campus, the 64-K Frame Relay link is the bottleneck in this path. All of the QoS for VoIP features used in the configuration muster against occurrence of congestion.

Because it is a slow-speed link, the 64-K Frame Relay link requires use of specific QoS features different from those used for a faster speed link, such as a T1. For instance, to obviate (or at least control) delay and jitter, a slow-speed link requires that large data packets be fragmented and interleaved with voice packets. Moreover, traffic coming into the network across the fast-speed LAN links must be shaped as it leaves the router to accommodate the slower-speed 64-K line and to ensure that bursting does not occur. (For slow-speed Frame Relay links, you control packet drop by configuring FRTS to ensure against bursting.)

Figure 3-3 shows the path from the 2600 router at the small branch office which connects through the IGX switch to the 7206 campus access router, then into the main campus and out through the 5300 router across the PSTN, giving voice access to all corporate users. The 2600 router at the small branch office is configured as a VoIP gateway that allows for codec conversion (using G.729a) of analog voice to packetized digital voice and back. This QoS for VoIP configuration focuses on the 64-K Frame Relay link. Figure 3-3 shows the full path from the 2600 router through the IGX switch and 7206 router into the campus, then to the PBX interconnect 5300 router.

The main campus network encompasses Catalyst 8500 series switches with fast-speed (10-to-100 Mbps Ethernet) links. Traffic from multiple applications aggregates across the campus Ethernet links.

Figure 3-3 Path Between a Small Branch Office and the Main Campus Router Across a 64-K Frame Relay Link



Configuring QoS for VoIP on a 64-K Frame Relay Link

This section describes the configuration for deploying the following QoS for VoIP features on the outbound interface and at the PVC level for the 64-K Frame Relay link:

- Compressed Real-Time Protocol (CRTP)—Used to compress the large RTP header.
- Frame Relay Traffic Shaping (FRTS)—Used to shape traffic leaving the 2600 router to meet the 64-K Frame Relay link constraints in order to avoid packet loss and smooth out bursts.
- Frame Relay Fragmentation 12 (FRF.12)—Used as the fragmentation and interleaving mechanism.
- IP RTP Priority queueing—Also referred to as Priority Queueing-Weighted Fair Queueing (PQWFQ), is used as the queueing mechanism to give voice traffic strict priority service and service data traffic with WFQ.

Example 3-2 shows the configuration commands for the 2600 router with the commands for the QoS features called out in bold. In this example, subinterface Serial10/1.64 is configured for the Frame Relay PVC.

To register the router as a gateway with the gatekeeper, the router is configured with a loopback interface. The loopback is used so that the gateway registration with the gatekeeper remains relatively stable even if the gatekeeper cannot contact the gateway router when updating its tables.

The interface to the 64-K Frame Relay link at the 2600 router in the small branch office and the 7206 router at the campus access have mirroring configurations, using the same QoS features. Because they are largely the same, this section describes the QoS configuration for the 2600 router only. CRTP and FRTS must be enabled at the interface level. Though you enable FRTS at the interface level, you configure it at the PVC level. At the PVC level, FRF.12, IP RTP Priority queueing, the Frame Relay committed information rate (CIR), the Frame Relay minimum committed information rate (MINCIR), and WFQ parameters are configured; these QoS features are configured within a Frame Relay map class called voice that is applied to the PVC.

Example 3-2 64-K Frame Relay Link Configuration with QoS for VoIP

```
interface Loopback 0
 ip address 10.14.26.26.255.255.255.0
 h323-gateway voip interface
 h323-gateway voip id r1-3649a-gk.Cisco.com ipaddr 10.12.254.3 1719
 h323-gateway voip h323-id r1-2600@Cisco.com
!
interface Serial0/1
 no ip address
 encapsulation frame-relay
 ip rtp header-compression iphc-format
 ip tcp header-compression iphc-format
 frame-relay traffic-shaping
!
interface Serial0/1.64 point-to-point
 ip address 10.14.97.2 255.255.255.252
 frame-relay interface-dlci 64
 frame-relay class voice
!
map-class frame-relay voice
 frame-relay cir 64000
 frame-relay mincir 48
 frame-relay fragment 80
 frame-relay fair-queue
 ip rtp priority 16384 16383 48
```

Step 1: Configure CRTP for a 64-K Frame Relay Link

In a VoIP network, compression reduces the bandwidth necessary for each voice call, thus increasing the number of calls that can be transmitted over a given link. Enabling compression is especially necessary on a slow-speed (low-bandwidth) link in order to yield greater throughput. (You enable CRTP on both ends of the link.)

The two portions—payload and header—of a voice packet can each be compressed. Payload compression is determined by the codec used. CRTP is used to compress the RTP header. (CRTP compression is applied to the voice packet header after its payload has already been reduced through codec compression, which both converts analog to digital data and compresses the data.)

To gain a sense of the bandwidth savings based on codec compression, consider this case: uncompressed voice traffic converted by a gateway that uses codec G.711 PCM has a bandwidth requirement of 64 kbps before the RTP header overhead is accounted for. Cisco VoIP gateways can compress the voice signal down to 8 kbps using the G.729 codec. Although the number of bytes of voice payload can be specified as part of the codec selection, this has no effect on the RTP header added to each voice packet, which is considerable.

To compress the IP/UDP/RTP header down to as little as 2 bytes, you use CRTP. For a G.729 call, for instance, this results in a cost of 12K—a cost that is acceptable to most users.

Before you can enable RTP header compression, you must have configured the Frame Relay line, as shown previously in Example 3-2.

To enable RTP header compression, perform the following task in interface configuration mode (you need to enable compression on both ends of the connection):

Command	Purpose
ip rtp header-compression iphc-format	Enables RTP header compression for VoIP packets.

Step 2: Enable FRTS at the Interface Level for a 64-K Frame Relay Link

FRTS shapes traffic to ensure that traffic conforms to the policies contracted for it. In terms of the configuration shown in Example 3-2, the contracted policy is a link speed of 64 K. In this sense, traffic shaping is essentially a means of limiting a router's transmit rate toward a target destination to a specific peak/average value that is lower than the line rate of the circuit (link speed). To this end, FRTS is used to eliminate the possibility of traffic bursting about the committed bandwidth so as to fend against packet loss. You enable FRTS at the interface level and configure it at the PVC level.

Use of FRTS is essential for voice traffic, which is UDP based, to prohibit bursting above the committed information rate (CIR). For data-only networks that rely on TCP, TCP incorporates a slow-start mechanism that allows a TCP flow between two end points to increase its effective flow rate slowly in order to determine the minimum bandwidth link between the two endpoints. While this TCP process negates the need for traffic shaping in data-only networks, the absence of a similar process in UDP-based networks necessitates use of FRTS to shape traffic.

In the configuration shown in Example 3-2, FRTS is enabled for interface Serial10/1 and configured for DLCI 64. For the PVC, the Frame Relay CIR parameter is set to 64 K (the link speed) so that traffic will not burst above the link's bandwidth causing loss, which would degrade voice quality. If the PVC is congested, the traffic flow is reduced to the MINCIR configured for that PVC.

The **frame-relay fair-queue** command is given to enable FRTS WFQ at the PVC level. WFQ is used as the queueing mechanism for all traffic on this PVC other than voice. WFQ does not provide strict priority required for voice. Therefore, to give voice traffic the strict priority queueing it requires, IP RTP Priority is configured for the PVC. (For more information, see the "About IP RTP Priority" later in this chapter.)

At the output interface on the 7206 campus access router, traffic coming from the small branch office fast-speed Ethernet links (ranging from 10 Mb to 100 Mb) is shaped before it leaves the router to match the flow of the slower-speed 64-K Frame Relay link. If traffic were not shaped at the router, a bottleneck condition could occur.

It is good policy to use FRTS for all links that carry both voice and data, regardless of the link bandwidth capacity. Moreover, you must use traffic shaping to handle these three conditions, discussed more fully in Chapter 2:

- link speed mismatches

Link speed mismatches result in congestion during which packets are queued in buffers causing delays.

- oversubscription

  Oversubscription occurs when many remote sites feed a central site. A remote-to-central site configuration is oversubscribed when aggregate transmission from remote sites exceeds the central site's link bandwidth capacity. Even if the CIRs of all the remote sites collectively are less than the central site's link speed, oversubscription can occur when bursting of the remote sites exceeds the central site link speed. The topology, shown in Figure 3-1, for the example scenario, indicates that traffic from the following three links aggregates at the 7206 campus access router:

  – The 64-K point-to-point link from the small branch office

  – The T1 link from the medium branch office, switched through the IGX to the 7206 router.

  – The DS-3 ATM link switched through the IGX to a T1 Frame Relay link connecting to the 7206 router

- bursting

  Frame Relay circuits have a link speed and an associated CIR that is in most cases less than the line rate. The CIR is the amount of traffic that is transmitted when there is no congestion on the PVC. When traffic bursts occur or when congestion is reported, or both events occur, the amount of traffic transmitted is determined by the configured MINCIR, and this becomes the guaranteed minimum committed information rate.

  Many network providers allow applications to burst above the guaranteed CIR all the way up to the line rate. It is well understand that in the event of congestion, bursting about the CIR incurs the risk of packet loss because any data sent over the CIR is considered best-effort traffic. If you run voice traffic across a network designed to support bursting, it is likely that voice will be included in the excess bandwidth above the CIR because Frame Relay has no way of differentiating between voice traffic and data traffic. If global congestion ensues, delay (caused by buffering) and packet drop could occur. To stem the problems for voice traffic resulting from bursting, you must use FRTS on each PVC that carries voice to ensure that the outbound traffic rate does not exceed the link's configured CIR. Although this could potentially reduce data throughput, it guarantees throughput and constant delay to voice traffic even when congestion occurs.

Enabling Frame Relay traffic shaping on an interface enables both traffic shaping and per-virtual circuit queuing on all the interface's PVCs and SVCs. Traffic shaping enables the router to control the circuit's output rate and react to congestion notification information if also configured.

To enable Frame Relay traffic shaping on the specified interface, complete the following task in interface configuration mode:

Command	Purpose
frame-relay traffic-shaping	Enables Frame Relay traffic shaping and per-virtual circuit queuing.

Step 3: Configure a Map Class and Apply It to the PVC for a 64-K Frame Relay Link

To configure a Frame Relay link for voice traffic, you must create a Frame Relay map class and configure it to support voice traffic. The voice bandwidth, fragmentation size, traffic shaping attributes, and strict priority service for voice are configured on the map class. These attributes are required for sending voice traffic on the PVC.

In the configuration shown in Example 3-2, the data fragment size is 80 bytes, and the BE and BC traffic shaping parameters assume their default values. For the PVC, the Frame Relay CIR parameter—which specifies the upper bound on bandwidth allocation—is set to the link speed (64 K) so that traffic will not burst above the link's bandwidth causing loss, which would degrade voice quality. Also for the PVC, the Frame Relay MINCIR—which establishes the lower bound on bandwidth allocation—is set 48 K, the amount of bandwidth guaranteed to be delivered to voice traffic under congestion conditions.

Note You must set the bandwidth amount allocated for strict priority for voice to be less than or equal to the MINCIR. (You use the **ip rtp priority** command within the map class for voice to configure strict priority.)

To configure QoS for VoIP at the PVC level for the 64-K Frame Relay link, you create a Frame Relay map class. The attributes required for sending voice on the PVC are configured in the map class, which is applied to the Frame Relay DLCI used for voice traffic through the **frame-relay class voice** command.

The following topics, underlying use of these commands, are explored in this step after the configuration commands are listed:

- About the CIR
- FRF.12
- FRF and ATM
- Using WFQ as the Data Queueing Strategy
- Why WFQ Is Inadequate for Voice
- About IP RTP Priority

To configure the frame-relay map class called voice, use the following commands:

Command	Purpose
map-class frame-relay voice	Specifies the Frame Relay map class name—voice, in this case—and enters map class configuration mode.
frame-relay cir 64000	Specifies the committed information rate (CIR) in bits per second (bps) for the PVC. For the example configuration, the CIR is 64000 bps.
frame-relay mincir 48	Specifies the minimum committed information rate (MINCIR) (guaranteed for voice traffic under congestion conditions) in bits per second (bps) for the PVC. For the example configuration, the MINCIR is 48 bps.
frame-relay fragment 80	Configures Frame Relay fragmentation for the map class. The *fragment_size*, which is 80 bytes for the example configuration, defines the payload size of a fragment, and excludes the Frame Relay headers and any Frame Relay fragmentation header. The valid range is from 16 to 1600 bytes, and the default is 53. (See "Determining If You Need to Fragment Data Packets and Frames" in Chapter 2 for optimum sizes per link type.) The *fragment_size* should be less than or equal to the MTU size. You should set the fragmentation size such that the largest data packet is not larger than the voice packets.

Command	Purpose
frame-relay fair-queue	Enables weighted fair queuing for the map class. When used in conjunction with WFQ, as in the example configuration, IP RTP Priority provides strict priority to voice traffic and WFQ scheduling among all other traffic. (This command together with the **ip rtp priority** command are also referred to as PQWFQ.)
ip rtp priority 16384 16383 48	Reserves a strict priority queue and bandwidth for the set of RTP voice packet flows belonging to a range of UDP destination ports. For the example configuration, RTP (voice) packet flows using ports in the range of 16384 and 16383 are granted strict priority with a bandwidth guarantee of 48 kbps. The example configuration is designed to service 4 voice calls at the cost of 12 K per call. For your configuration, assume that each call consumes 12 K and set the bandwidth parameter to the result of the following equation: *bandwidth* `= 12 K x` *number-of-calls* The IP RTP Priority feature does not require that you know the port of a voice call. Rather, the feature gives you the ability to identify a range of ports whose traffic is put into the priority queue. Moreover, you can specify the entire voice port range—16384 to 32767—to ensure that all voice traffic is given strict priority service. (This command together with the **frame-relay fair-queue** command are also referred to as PQWFQ.)

About the CIR

Frame Relay uses the concept of CIR on each PVC to specify the amount of traffic that is transmitted when there is no congestion on the PVC. When traffic bursts occur or when congestion is reported, or both events occur, the amount of traffic transmitted is determined by the configured MINCIR, and this becomes the guaranteed minimum committed information rate.

The MINCIR is guaranteed in the presence of global congestion or when congestion results from multiple remote sites transmitting concurrently. Under periods of no congestion, remote sites may burst above the MINCIR up to the line rate of the circuit with a maximum throughput of the minimum line speed the PVC traverses. (The CIR value establishes the upper bound.) However, bursting above the guaranteed MINCIR, which is done in many cases to twice its value, creates a best-effort traffic situation in which traffic might be dropped by the Frame Relay network if congestion occurs. Bursting is discouraged for networks that support voice transmission because loss of packets resulting from bursting is not well tolerated by voice traffic. Therefore, to prohibit bursting, use of FRTS is essential for a link that carries voice.

FRF.12

Giving voice traffic strict priority over steeper data flows, such as FTP flows, does not sufficiently ensure satisfaction of voice traffic requirements. When large, lower-priority data packets up to 1500 bytes in size (jumbograms) are sent out the interface, their transmission can take up to 241 ms on a slow-speed link such as a 64-K link. (Considering overhead, a 64-K link gives an actual speed of a 56 kbps line).This condition incurs delays unacceptable for interactive two-way conversations. To curtain this problem, you should configure FRF.12 on the slow-speed Frame Relay link.

Note For information on how to assess whether you need to use fragmentation for a specific type of link, see the section “Why Fragment Data Packets?” in Chapter 2, “QoS Features for Voice over IP.”

FRF.12 fragments large frames, or packets, and interleaves VoIP frames with them at the link layer, reassembling the fragments at the other end of the link. This is the optimal method for preventing excessive delay to voice frames incurred on slow Frame Relay links by jumbograms. You should use this method for Frame Relay-to-Frame Relay endpoints. When FRF.12 is used, a VoIP packet will not include the FRF.12 header if the VoIP packet is smaller than the configured fragment size.

You configure FRF.12 at the PVC level. In the example configuration, FRF.12 is configured within the voice map class that is applied to the PVC used for voice traffic. In order to configure FRF.12, FRTS must be configured for the PVC.

To configure FRF.12, you specify the fragment size. The fragment size specifies the amount of data in bytes that large frames are broken into. The fragment size defines the fragment’s payload, excluding the Frame Relay headers and any Frame Relay fragmentation header. Valid sizes range from 16 to 1600 bytes (with a default size of 53 bytes). In our example configuration (Example 3-2), a fragment size of 80 bytes is configured. An 80-byte payload is the optimum fragment size for 64-kbps link.The fragment size you choose should always be less than or equal to the size of the maximum transmission unit (MTU).

FRF and ATM

Suppose you wanted to use Service Interworking between Frame Relay and an ATM cloud (FRF.8). In particular, suppose you wanted to configure the 64-K Frame Relay link for VoIP communication from the 2600 small branch office router to interwork with an ATM network across an IGX switch. In this case, translation of Frame Relay packets to ATM cells cannot occur unless the internetworking software is able to discern the content of the packets. This imposes the preliminary requirement of reassembly (that is, prior to translation).

To accommodate voice traffic for this type of scenario so that voice packets are not queued behind jumbograms, you could use IP MTU size reduction. IP MTU size reduction reduces the MTU size for IP packets sent from a Frame Relay link across an ATM link to a Frame Relay endpoint. IP MTU size reduction fragments packets at Layer 3. Layer 3 packets are transmitted across internetworks; therefore, they are transmitted seamlessly across both ATM and Frame Relay links. If you decide to use IP MTU size reduction, take into account the fact that small fragments can lead to performance inefficiencies. Each fragment carries the IP header, which can incur considerable transmission overhead across the internetwork. Reducing the IP MTU size below 300 bytes may adversely affect an application as well as router endpoint performance. Moreover, use of IP MTU size reduction is restricted to IP traffic; you cannot use it if your internetwork carries other types of packets such as SNA traffic.

Note Although conditions intrinsic to Service Internetworking between Frame Relay and an ATM cloud necessitate use of IP MTU size reduction, these problems do not exist if you are using network interworking between FRF.12 and FRF.5 because there is no preliminary packet translation requirement. You can configure for network interworking using FRF.12 and FRF.5 using applicable QoS features.

Using WFQ as the Data Queueing Strategy

WFQ offers a solution that provides consistent, fair response time, based on weights, to heavy and light traffic alike without adding excessive bandwidth. Implicit within WFQ is a strict priority queue that is created when WFQ is enabled. However, this queue cannot be used until the IP RTP Priority feature is enabled. Moreover. because WFQ fairly shares bandwidth, it cannot offer voice traffic guaranteed bandwidth unless use of the strict priority queue is enabled. Therefore, to ensure that VoIP traffic always gets the bandwidth it requires, you must use the **ip rtp priority** command to configure the strict queue for use by voice traffic; all other traffic will use fair queueing. IP RTP Priority is especially useful on slow-speed links whose speed is less than 1.544 Mbps.

Why WFQ Is Inadequate for Voice

One of the intrinsic values of WFQ in the data-only world is its fairness. WFQ is designed to fairly share available resources between bursty traffic types. This very aspect of WFQ so beneficial to data is what renders it inadequate for packetized voice traffic. Although voice traffic can be assigned an IP Precedence of 5 to give it a weight that grants it greater priority than other flows, if a large number of competing flows exist, voice traffic may not be allocated sufficient bandwidth to maintain the kind of quality—the controlled constancy of delay and packet loss—it requires all the time, regardless of the priority (weight) assigned to voice packets.

Consider some instances that illustrate why voice traffic could, but would not always, get the priority it requires when WFQ is used as the queueing strategy. To understand these examples, first consider how the amount of bandwidth allocated to a flow whose packets are marked with IP Precedence is calculated, as illustrated by the equation shown in Example 3-3. In this equation, which calculates the bandwidth for flowA, individual flow parts are given the weight of 1 plus their assigned IP Precedence value.

Example 3-3 Calculating Flow Bandwidth Allocation Using WFQ/IP Precedence

```
flowA_Bandwidth = (flowA_parts/sum_of_all_flow_ parts) x circuit_bandwidth
```

Table 3-4 identifies the weight given to a flow based on the IP Precedence assigned to the flow. (Recall that weight is equivalent to the IP Precedence value plus 1.)

Table 3-4 IP Precedence and Flow Parts

IP Precedence Value	Weight
0	1
1	2
2	3
3	4
4	5
5	6

Table 3-4 IP Precedence and Flow Parts (continued)

IP Precedence Value	Weight
6	7
7	8

Take the example of four flows of equal precedence—all flows have a precedence of 0—that must share a 56-kbps link. Two of the flows, called flowA and flowB, are VoIP flows, each requiring 24 kbps. The other two flows, called flowC and flowD, are FTP flows, each requiring 56 kbps. Using the prototype equation shown in Example 3-3, the following equation shows that each of the VoIP flows would be granted 14 kbps of bandwidth, which is unacceptable because it does not meet the stringent requirement of 24 kbps.

```
14 kbps = (1/4) x 56 kbps
```

Consider a variation on the same example in which IP Precedence is used to give the VoIP flows a high precedence of 5. Again, the two VoIP flows called flowA and flowB each require 24 kbps of the 56 kbps link. As in the previous example, the two FTP flows called flowC and flowD each require 56 kbps; the IP Precedence for each FTP flow is still 0. Using the prototype equation shown in Example 3-3, the following equation shows that each of the VoIP flows would be granted its required 24 kbps of bandwidth.

```
24 kbps = (6/14) x 56 kbps
```

Even though VoIP traffic is given the bandwidth it requires when the number of flows and values factored into the previous example pertain, only strict priority service can guarantee satisfaction of voice traffic requirements because strict priority ensures that voice packets are serviced first. Strict priority means that if packets exist in the priority queue, they are dequeued and sent first—that is, before packets in other queues are dequeued, and this service ensures against delays. WFQ, even when adequate bandwidth is allocated for voice, cannot ensure against delay because of its fair queueing and servicing policy.

About IP RTP Priority

To ensure that VoIP traffic always gets the bandwidth it requires, you must use the IP RTP Priority feature for voice traffic and WFQ or CBWFQ for all other traffic. IP RTP Priority is especially useful on slow-speed links whose speed is less than 1.544 Mbps.

To use the IP RTP Priority feature, you specify a range of ports and a strict bandwidth limitation that specifies the amount of bandwidth guaranteed to voice traffic. Traffic matching the range of ports specified for the priority queue is guaranteed strict priority over other CBWFQ classes or WFQ flows; voice packets in the priority queue are always serviced first.

IP RTP Priority closely polices use of bandwidth for the priority queue, ensuring that the allocated amount is not exceeded. IP RTP Priority prohibits transmission of additional packets once the allocated bandwidth is consumed and drops packets if the bandwidth is exceeded. For this reason, you should enable debugging to watch for packet drop as voice traffic does not tolerate it well.

Note IP RTP Priority can be used in conjunction with either WFQ or CBWFQ on the same outgoing interface. When used in conjunction with WFQ, the **ip rtp priority** command provides strict priority to voice traffic and WFQ scheduling among other traffic flows. When used in conjunction with CBWFQ, IP RTP Priority provides strict priority to voice and CBWFQ can be used to set up classes for other types of traffic, such as SNA, that needs dedicated bandwidth and needs to be treated better than best-effort and not as strict priority. CBWFQ can also support flow-based WFQ within

To avoid packet drop, be certain to allocate to the priority queue the most optimum amount of bandwidth, taking into consideration the type of codec used and interface characteristics. It is always safest to allocate to the priority queue slightly more than the known required amount of bandwidth. For example, suppose you allocated 24 kbps bandwidth, the standard amount required for voice transmission, to the priority queue. This allocation seems safe because transmission of voice packets occurs at a constant bit rate. However, because the network and the router or switch can use some of the bandwidth and introduce jitter and delay, allocating slightly more than the required amount of bandwidth (such as 25 kbps) ensures constancy and availability.

Note The IP RTP Priority admission control policy disregards voice packet compression—that is, it does not account for CRTP compression. Suppose you use IP RTP Priority with WFQ and, as in our example configuration, you reserve 48 kbps of bandwidth for the voice priority queue to service four calls each requiring 12 kbps, but the voice packets are compressed and compression reduces the flow to a total requirement of 24 kbps for all four calls. In this case, admission control would not double the number of voice packets it would let through. Rather, the unused bandwidth would be distributed among the other WFQ flows.

Outcome Measurements for a 64 K Frame Relay Link QoS for VoIP Configuration

To convey a sense of how application of QoS features affects the quality of voice transmission, this section provides measurements that show a set of results produced when the following criteria, given in Table 3-5, vary:

- QoS Feature Used. This column identifies which QoS features were configured so as to satisfy voice traffic requirements and improve voice quality.
- Concurrent TCP Data Traffic (noise level). This column identifies the amount of data traffic transmitted on the same line as voice.
- Number of Voice Calls: This column gives the throughput—the number of voice calls put through for the test.
- Average PSQM Score

Table 3-5 Voice Call and Quality Measurements for 64-K Frame Relay

QoS Feature Used	Concurrent TCP Data Traffic	Number of Voice Calls	PSQM Score Average
Without QoS **(to establish baseline)**	none (baseline)	1	1.75424
		2	1.95946
		3	2.30772
		4	2.44013

Table 3-5 Voice Call and Quality Measurements for 64-K Frame Relay (continued)

QoS Feature Used	Concurrent TCP Data Traffic	Number of Voice Calls	PSQM Score Average
PQWFQ **(Priority within CBWFQ)**	78 kbps 34 pps 10% CPU 2 TCP streams	1	2.2062
		2	2.2197
		3	2.35315
		4	2.45143

Providing Medium Size Branch Office Users QoS for VoIP Access to the Main Campus Using T1 Links

This section describes the QoS for VoIP configuration for the following two types of links, either of which could be used to provide VoIP access to and from the users at the medium-size branch office so that they can communicate with one another, main campus users, and telecommuters:

- Medium Branch to Campus Access Using a T1 PPP Link
- Medium Branch to Campus Access Using a T1 Frame Relay Link

This topology configuration assumes that the medium branch office site accommodates 2000 or fewer users.

Medium Branch to Campus Access Using a T1 PPP Link

This section describes the link type and the QoS features for VoIP you should use for a T1 PPP link and configuration.

About the Link Type and Configuration

The campus (corporate) network provides a T1 PPP link for access to and from the medium branch office and main campus. This link is directly connected to the 7206 router giving access to the main campus.

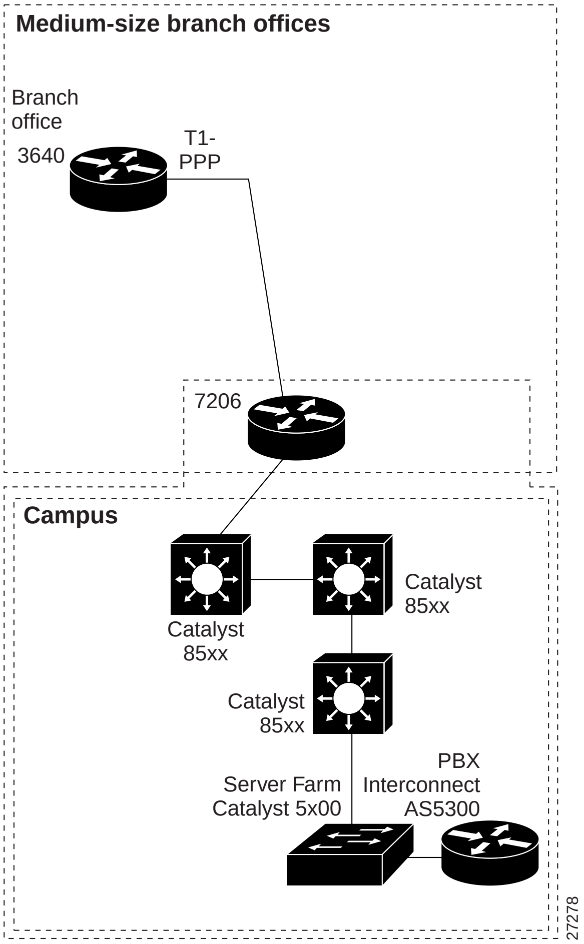
Apart from LFI, the same QoS features used for a 64-K PPP link are also used for the T1 link. (They are also used for the 64 K Frame Relay link.) (For background information on these features, see “Using a 64-K Frame Relay Link” earlier in this chapter.)

Both PPP and Frame Relay T1 links require use of priority queueing in order to grant voice the strict priority it requires. They both also require use of CRTP to compress the large RTP header to provide greater throughput.

Figure 3-4 shows the path from the 3640 router at the medium branch office which is directly connected via the T1 PPP link to the 7206 campus access router into the main campus. The 3640 router at the medium branch office is configured as a VoIP gateway that allows for codec conversion (using G.729a) of analog voice to packetized digital voice and back. Users throughout the branch office can make phone calls to users throughout the main campus and telecommuters. Routed through the gateway, their analog calls are converted to packetized digital data transmitted using VoIP.

This portion of the configuration focuses on the path segment from the 3640 router to 7206 router. Figure 3-4 shows the full path through the campus to the PBX interconnect 5300 router.

Figure 3-4 Path Between a Medium Branch Office Router and Campus Access Over a T1 PPP Link



Configuring QoS for VoIP on a T1 PPP Link

This section describes the configuration for deploying the following QoS for VoIP features for the T1 PPP link:

- Compressed Real-Time Protocol (CRTP)—Used to compress the large RTP header.
- IP RTP Priority queueing (Also referred to as Priority Queueing-Weighted Fair Queueing (PQWFQ))—This feature is used as the queueing mechanism to give voice traffic strict priority service.

Example 3-4 shows the configuration commands for the 3640 router T1 PPP link with the commands for the QoS features called out in boldface. The QoS for VoIP features are configured within a virtual template called Virtual-Template1 which is applied to interface Serial10/1.

Example 3-4 T1 PPP Link Configuration with QoS for VoIP

```
interface Loopback 0
 ip address 10.14.26.26 255.255.255.0
 h323-gateway voip interface
 h323-gateway voip id r1-3640a-gk.cisco.com ipaddr 10.12.254.3 1719
 h323-gateway voip h323-id r1-2600@cisco.com
!
interface Serial0/1
 bandwidth 15440
 ip address 10.14.97.2 255.255.252
 encapsulation PPP
 ppp multilink
 multilink-group 1
!
interface multilink 1
 ip unnumbered Serial0/1
 ip rtp header-compression iphc-format
 ip tcp header-compression iphc-format
 fair-queue 64 256 0
 ppp multilink
 ip rtp priority 16384 16384 48
```

Configuring a Multilink Interface with QoS for VoIP Features for a T1 PPP Link

This section describes the QoS features you configure for the PPP link.Most of these QoS features are the same as those that you configure for a PPP 64-K link, a Frame Relay 64-K link, and a Frame Relay T1 link. Therefore, instead of repeating conceptual information, these steps refer to the related 64-K Frame Relay sections that fully cover background information for these features common to all configurations.

Note This configuration process provides two alternative ways of configuring strict priority queueing for VoIP traffic. In this set of steps, both methods are presented as Step 2. The first Step 2 shows how to configure strict priority using IP RTP Priority. The second, (alternative) Step 2 shows how to configure strict priority queueing using strict priority within CBWFQ.

Step 1: Configure CRTP for the T1 PPP Link

To enable RTP header compression, perform the following task in interface configuration mode (you need to enable compression on both ends of the connection):

Command	Purpose
ip rtp header-compression iphc-format	Enables RTP header compression for VoIP packets.

For background information on CRTP, see “Step 1: Configure CRTP for a 64-K Frame Relay Link” earlier in this chapter. Also, see Chapter 2, “About QoS Features for Voice.”

Step 2: Configure WFQ and Enable the Strict Priority Queue for VoIP Packets on the T1 PPP Link

When WFQ is enabled, it creates a strict priority queue that exists potentially for use by delay-sensitive traffic such as voice traffic. However, the strict priority queue cannot be used until it is enabled through configuration of the **ip rtp priority** command. This section gives the command syntax for the **fair-queue** and **ip rtp priority** commands that you use to enable WFQ and its strict priority queue.

To enable weighted fair queueing for the PPP interface shown in Example 3-4, set the congestion threshold after which messages for high-bandwidth conversations are dropped, and specify the number of dynamic and reservable queues, perform the following task in interface configuration mode after specifying the interface:

Command	Purpose
fair-queue 64 256 0	Configures the PPP interface to use weighted fair queueing with a congestive discard threshold of 64, 256 dynamic queues, and no reservable queues.

To reserve a strict priority queue for a set of RTP voice packet flows belonging to a range of UDP destination ports, use the following command in interface configuration mode:

Command	Purpose
ip rtp priority 16384 16383 288	Reserves a strict priority queue for the VoIP packet flow using destination ports in the range of 16384 and 16383. VoIP flows for the example PPP link whose UDP destination ports are within the range of 16384 and 16383 are granted strict priority with a bandwidth guarantee of 48 kbps. The example configuration is designed to service 4 voice calls at the cost of 12 kbps per call. For your configuration, assume that each call consumes 12 K and set the bandwidth parameter to the result of the following equation: *bandwidth* `= 12 K x` *number-of-calls* The IP RTP Priority feature does not require that you know the port of a voice call. Rather, the feature gives you the ability to identify a range of ports whose traffic is put into the priority queue. Moreover, you can specify the entire voice port range—16384 to 32767—to ensure that all voice traffic is given strict priority service.

For background information on WFQ and the strict priority feature, see the "Step 2: Configure WFQ and Enable the Strict Priority Queue for VoIP Packets on the PPP Link" earlier in this chapter. Also, see Chapter 2, "About QoS Features for Voice."

Step 2: Alternative Configuration: Configure WFQ and Enable the Strict Priority Queue within CBWFQ

This step provides an alternative method of giving voice traffic strict priority queueing. It describes how you can use the same feature described in the first Step 2, but within CBWFQ to apply strict priority queueing to a CBWFQ class used for voice traffic.

Priority queueing within CBWFQ enables use of the strict priority queue implicit to WFQ. Although it is possible to enqueue various types of real-time traffic to the strict priority queue, it is strongly recommended that you direct only voice traffic to it. This recommendation is made because voice traffic is well-behaved, whereas other types of real-time traffic are not. Moreover, voice traffic requires that delay be nonvariable in order to avoid jitter. Real-time traffic such as video could introduce variation in delay thereby thwarting the steadiness of delay required for successful voice traffic transmission. For complete information the Priority Queueing within CBWFQ feature, see Chapter 2, “QoS Features for Voice over IP.”

This section gives the command syntax for the **fair-queue** and **ip rtp priority** commands that you use to enable WFQ and its strict priority queue.

To enable weighted fair queueing for the PPP interface shown in Example 3-4, set the congestion threshold after which messages for high-bandwidth conversations are dropped, and specify the number of dynamic and reservable queues, perform the following task in interface configuration mode after specifying the interface:

Command	Purpose
fair-queue 64 256 0	Configures the PPP interface to use weighted fair queueing with a congestive discard threshold of 64, 256 dynamic queues, and no reservable queues.

To create a class for voice called “class voice” and configure that class as a strict priority class with a bandwidth allocation of 288 kbps, use the following commands beginning in interface configuration mode:

Command	Purpose
class-map voice	Creates a class and names it voice.
match access-group 102	Specifies that all traffic belonging to the numbered access group 102 will belong to the voice class.
policy-map voiceplus	Creates a policy map called voiceplus to be applied to the interface.
class voice	Adds the class called voice to the policy map voiceplus.
priority 288	Enables the strict priority queue for the class called voice and sets the bandwidth for the voice traffic to 288 kbps.

Outcome Measurements for a T1 PPP Link QoS for VoIP Configuration

To convey a sense of how application of QoS features affects the quality of voice transmission, this section provides measurements that show a set of results produced when the following criteria, shown in Table 3-5, vary:

- QoS Feature Used. This column identifies which QoS features were configured so as to satisfy voice traffic requirements and improve voice quality.
- Concurrent TCP Data Traffic (noise level). This column identifies the amount of data traffic transmitted on the same line as voice.

- Number of Voice Calls. This column gives the throughput—the number of voice calls put through for the test.
- Average PSQM Score

Table 3-6 Voice Call and Quality Measurements for T-1 PPP

QoS Feature Used	Concurrent TCP Data Traffic	Number of Voice Calls	PSQM Score Average
Without QoS **(to establish baseline)**	none	1	1.86154
		6	2.00987
		12	2.21582
		23	2.42247
WFQ	1967 kbps 723 pps 18% CPU 24 TCP streams	6	2.75
		12	3.13926
		23	2.98119
CBWFQ	1967 kbps 723 pps 18% CPU 24 TCP streams	6	1.86772
		12	1.87503
		23	1.90786

Medium Branch to Campus Access Using a T1 Frame Relay Link

This section describes the link type, the QoS features for VoIP you should use for this type of link and circumstance, and how to handle conditions you might encounter that could negatively affect voice traffic.

About the Link Type and Configuration

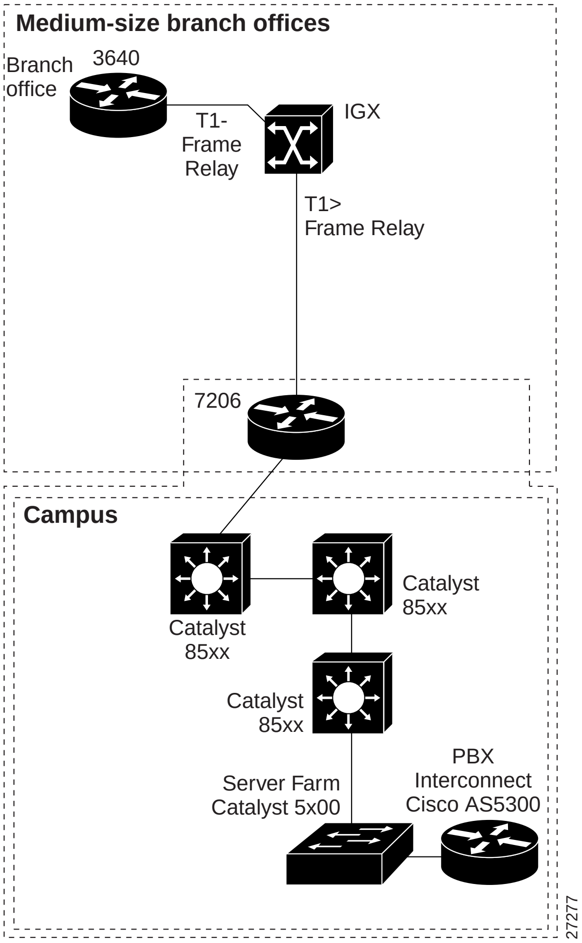
The campus (corporate) network provides a T1 Frame Relay link for access to and from the medium branch office and main campus. This link connects from the medium branch office 3640 router across a T1 line to the IGX switch and on through to the T1 Frame Relay link. The T1 Frame Relay link gives access to the campus through the 7206 router. Although not shown in the illustrations depicting this scenario or path segment, connected to the medium branch office 3640 router is the branch office user network.

The T1 link, which is high speed, requires with one exception the same QoS features used for the Frame Relay slow-speed link described in the “Using a 64-K Frame Relay Link”earlier in this chapter. Because the T1 line is high-speed, it is not necessary to fragment large data packets to interleave with voice packets.

Figure 3-5 shows the path from the 3640 router at the medium branch office which connects across the T1 line through the IGX switch to the 7206 campus access router, then into the main campus and out through the 5300 router across the PSTN, giving voice access to all corporate users. The 3640 router at the small branch office is configured as a VoIP gateway that allows for codec conversion (using G.729a) of analog voice to packetized digital voice and back.

The main campus network contains Catalyst 8500 series switches with fast-speed (10-to-100 mbps Ethernet) links. Traffic from multiple applications aggregates across the campus Ethernet links.

Figure 3-5 Path Between a Medium-Size Branch Office and the Main Campus Across a T1 Frame Relay Link



Configuring QoS for VoIP on a T1 Frame Relay Link

This section describes the configuration for deploying the following QoS for VoIP features on the outbound interface and at the PVC level for the T1 Frame Relay link:

- Compressed Real-Time Protocol (CRTP)—Used to compress the large RTP header.
- Frame Relay Traffic Shaping (FRTS)—Used to shape traffic leaving the 3640 router to meet the T1 Frame Relay link constraints in order to avoid packet loss and smooth out bursts.
- IP RTP Priority queueing (Also referred to as Priority Queueing-Weighted Fair Queueing (PQWFQ))—Is used as the queueing mechanism to give voice traffic strict priority service and service data traffic with WFQ. (Alternatively, you could use the strict priority feature within CBWFQ to give voice traffic the priority it requires.)

Example 3-5 shows the configuration commands for the 3640 router with the commands for the QoS features called out in bold. In this example, subinterface Serial10/1.64 is configured for the Frame Relay PVC.

To register the router as a gateway with the gatekeeper, the router is configured with a loopback interface. The loopback is used so that the gateway registration with the gatekeeper remains relatively stable even if the gatekeeper cannot contact the gateway router when updating its tables.

The interfaces to the T1 Frame Relay link at the 3640 router in the medium branch office and the 7206 router at the campus access have mirroring configurations, using the same QoS features. Because they are largely the same, this section describes the QoS configuration for the 3640 router only. CRTP and FRTS must be enabled at the interface level. Though you enable FRTS at the interface level, you configure it at the PVC level. At the PVC level, priority queueing configured through the **ip rtp priority** command, the Frame Relay CIR, the Frame Relay MINCIR, and WFQ parameters are configured; these QoS features are configured within a Frame Relay map class called voice that is applied to the PVC.

Example 3-5 T1 Frame Relay Link Configuration with QoS for VoIP

```
interface Loopback 0
 ip address 10.14.26.26.255.255.255.0
 h323-gateway voip interface
 h323-gateway voip id r1-3649a-gk.Cisco.com ipaddr 10.12.254.3 1719
 h323-gateway voip h323-id r1-2600@Cisco.com
!
interface Serial0/1
 no ip address
 encapsulation frame-relay
 ip rtp header-compression iphc-format
 ip tcp header-compression iphc-format
 frame-relay traffic-shaping
!
interface Serial0/1.64 point-to-point

 ip address 10.14.97.2 255.255.255.252
 frame-relay interface-dlci 64
 frame-relay class voice
!
map-class frame-relay voice
 frame-relay cir 1544000
 frame-relay mincir 288
 frame-relay fair-queue
 ip rtp priority 16384 16383 288
```

Step 1: Configure CRTP for a T1 Link

For a VoIP network, compression reduces the bandwidth necessary for each voice call, thus increasing the number of calls that can be transmitted over a given link.

The two portions—payload and header—of a voice packet can each be compressed. Payload compression is determined by the codec used. CRTP is used to compress the RTP header. (CRTP compression is applied to the voice packet header after its payload has already been reduced through codec compression, which both converts analog to digital data and compresses the data.)

To compress the IP/UDP/RTP header down to as few as 2 bytes, you use CRTP. For a G.729 call, for instance, this results in a cost of 12K—a cost that is acceptable to most users.

Before you can enable RTP header compression, you must have configured the Frame Relay line, as shown previously in Example 3-5.

To enable RTP header compression, perform the following task in interface configuration mode (you need to enable compression on both ends of the connection):

Command	Purpose
ip rtp header-compression iphc-format	Enables RTP header compression for VoIP packets.

Step 2: Enable FRTS at the Interface Level for a FR T1 Link

FRTS shapes traffic to ensure that traffic conforms to the policies contracted for it. In the configuration shown in Example 3-5, FRTS is enabled for interface Serial10/1 and configured for DLCI 64.

For the PVC, the Frame Relay CIR parameter—which specifies the upper bound on bandwidth allocation—is set to the T1 link speed so that traffic will not burst above the link's bandwidth causing loss, which would degrade voice quality. Also for the PVC, the Frame Relay MINCIR—which establishes the lower bound on bandwidth allocation—is set to the amount of bandwidth guaranteed to be delivered to voice traffic under congestion conditions.

Note You must set the bandwidth amount allocated for strict priority for voice to be less than or equal to the MINCIR. (You use the **ip rtp priority** command within the map class for voice to configure strict priority.)

The **frame-relay fair-queue** command is given to enable FRTS WFQ at the PVC level. WFQ is used as the queueing mechanism for all traffic on this PVC other than voice. WFQ does not provide strict priority required for voice. Therefore, to give voice traffic the strict priority queueing it requires, IP RTP Priority is configured for the PVC. (For more information, see the "About IP RTP Priority" earlier in this chapter.)

For more background information on use of FRTS for this step, see "Step 2: Enable FRTS at the Interface Level for a 64-K Frame Relay Link" earlier in this chapter.

Enabling Frame Relay traffic shaping on an interface enables both traffic shaping and per-virtual circuit queuing on all the interface's PVCs and SVCs. Traffic shaping enables the router to control the circuit's output rate and react to congestion notification information if also configured.

To enable Frame Relay traffic shaping on the specified interface, complete the following task in interface configuration mode:

Command	Purpose
frame-relay traffic-shaping	Enables Frame Relay traffic shaping and per-virtual circuit queuing.

Step 3: Configure a Map Class and Apply It to the PVC on a Frame Relay T1 Link

To configure a Frame Relay link for voice traffic, you must create a Frame Relay map class and configure it to support voice traffic. The voice bandwidth, traffic shaping attributes, and strict priority service for voice are configured on the map class. These attributes are required for sending voice traffic on the PVC. The map class is applied to the Frame Relay DLCI used for voice traffic through the **frame-relay class voice** command.

To configure the frame-relay map class called voice, use the following commands:

Command	Purpose
map-class frame-relay voice	Specifies the Frame Relay map class name—voice, in this case—and enters map class configuration mode
frame-relay cir 154400	Specifies the committed information rate (CIR) in bits per second (bps) for the PVC. For the example configuration, the CIR is 1.544 bps.
frame-relay mincir 288	Specifies the minimum committed information rate (MINCIR) (guaranteed for voice traffic under congestion conditions) in bits per second (bps) for the PVC. For the example configuration, the MINCIR is 288 bps.
frame-relay fair-queue	Enables weighted fair queuing for the map class. When used in conjunction with WFQ, as in the example configuration, IP RTP Priority provides strict priority to voice traffic and WFQ scheduling among all other traffic. (This command together with the **ip rtp priority** command are also referred to as PQWFQ.)
ip rtp priority 16384 16383 288	Reserves a strict priority queue and bandwidth for the set of RTP voice packet flows belonging to a range of UDP destination ports. For the example configuration, RTP (voice) packet flows using ports in the range of 16384 and 16383 are granted strict priority with a bandwidth guarantee of 288 kbps. The example configuration is designed to service 24 voice calls at the cost of 12 kbps per call. For your configuration, assume that each call consumes 12 kbps and set the bandwidth parameter to the result of the following equation: *bandwidth* `= 12 K x` *number-of-calls* The IP RTP Priority feature does not require that you know the port of a voice call. Rather, the feature gives you the ability to identify a range of ports whose traffic is put into the priority queue. Moreover, you can specify the entire voice port range—16384 to 32767—to ensure that all voice traffic is given strict priority service. (This command together with the **frame-relay fair-queue** command are also referred to as PQWFQ.)

Outcome Measurements for a T1 Frame Relay Link QoS for VoIP Configuration

To convey a sense of how application of QoS features affects the quality of voice transmission, this section provides measurements that show a set of results produced when the following criteria, shown in Table 3-7, vary:

- QoS Feature Used. This column identifies which QoS features were configured so as to satisfy voice traffic requirements and improve voice quality.
- Concurrent TCP Data Traffic (noise level). This column identifies the amount of data traffic transmitted on the same line as voice.

- Number of Voice Calls: This column gives the throughput—the number of voice calls put through for the test.
- Average PSQM Score.

Table 3-7 Voice Call and Quality Measurements for T-1 Frame Relay

QoS Feature Used	Concurrent TCP Data Traffic	Number of Voice Calls	PSQM Score Average
PQWFQ **(Priority Queueing within CBWFQ)**	none	6	2.4097
		12	2.41582
		23	2.4224
	1967 kbps 723 pps 18% CPU 24 TCP streams	6	2.46772
		12	2.47503
		23	2.50786