CHAPTER 46

Quality of Service (QoS) Networking

Network QoS Defined

QoS refers to the capability of a network to provide better service to selected network traffic over various technologies, including Frame Relay, Asynchronous Transfer Mode (ATM), Ethernet and 802.1 networks, SONET, and IP-routed networks that may use any or all of these underlying technologies. Primary goals of QoS include dedicated bandwidth, controlled jitter and latency (required by some real-time and interactive traffic), and improved loss characteristics. QoS technologies provide the elemental building blocks that will be used for future business applications in campus, WAN, and service provider networks. This chapter outlines the features and benefits of the QoS provided by the Cisco IOS QoS.

The Cisco IOS QoS software enables complex networks to control and predictably service a variety of networked applications and traffic types. Almost any network can take advantage of QoS for optimum efficiency, whether it is a small corporate network, an Internet service provider, or an enterprise network. The Cisco IOS QoS software provides these benefits:

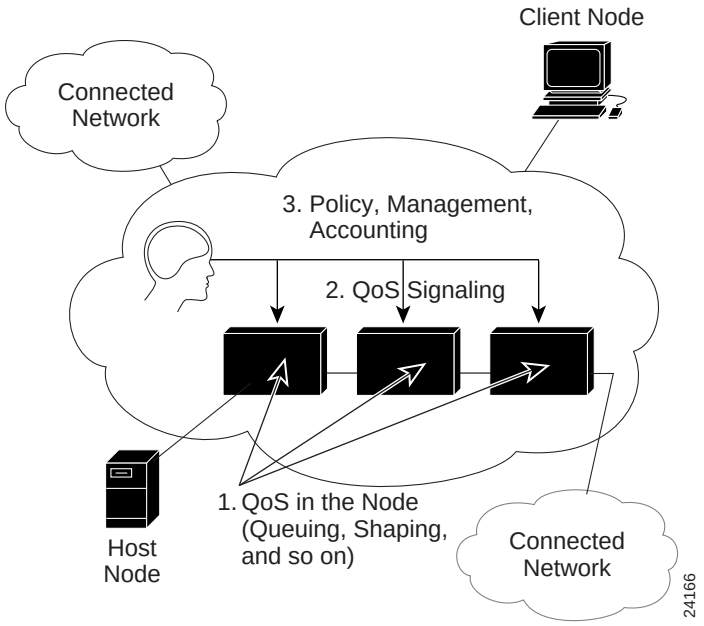
- *Control over resources*—You have control over which resources (bandwidth, equipment, wide-area facilities, and so on) are being used. For example, you can limit the bandwidth consumed over a backbone link by FTP transfers or give priority to an important database access.
- *More efficient use of network resources*—Using Cisco's network analysis management and accounting tools, you will know what your network is being used for and that you are servicing the most important traffic to your business.
- *Tailored services*—The control and visibility provided by QoS enables Internet service providers to offer carefully tailored grades of service differentiation to their customers.
- *Coexistence of mission-critical applications*—Cisco's QoS technologies make certain that your WAN is used efficiently by mission-critical applications that are most important to your business; that bandwidth and minimum delays required by time-sensitive multimedia and voice applications are available; and that other applications using the link get their fair service without interfering with mission-critical traffic.
- *Foundation for a fully integrated network in the future*—Implementing Cisco QoS technologies in your network now is a good first step toward the fully integrated multimedia network needed in the near future.

Basic QoS Architecture

The basic architecture introduces the three fundamental pieces for QoS implementation (see Figure 46-1):

- QoS within a single network element (for example, queuing, scheduling, and traffic shaping tools)
- QoS signaling techniques for coordinating QoS from end to end between network elements
- QoS policy, management, and accounting functions to control and administer end-to-end traffic across a network

Figure 46-1 A basic QoS implementation has three main components.



End-to-End QoS Levels

Service levels refer to the actual end-to-end QoS capabilities, meaning the ability of a network to deliver service needed by specific network traffic from end to end or edge to edge. The services differ in their level of "QoS strictness," which describes how tightly the service can be bound by specific bandwidth, delay, jitter, and loss characteristics.

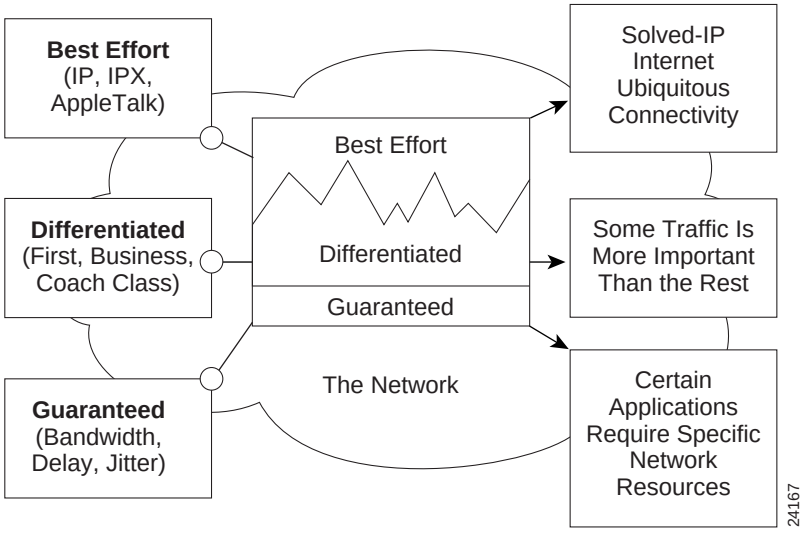
Three basic levels of end-to-end QoS can be provided across a heterogeneous network, as shown in Figure 46-2:

- *Best-effort service*—Also known as lack of QoS, best-effort service is basic connectivity with no guarantees.
- *Differentiated service (also called soft QoS)*—Some traffic is treated better than the rest (faster handling, more bandwidth on average, lower loss rate on average). This is a statistical preference, not a hard and fast guarantee.
- *Guaranteed service (also called hard QoS)*—An absolute reservation of network resources for specific traffic.

Deciding which type of service is appropriate to deploy in the network depends on several factors:

- The application or problem the customer is trying to solve. Each of the three types of service is appropriate for certain applications. This does not imply that a customer must migrate to differentiated and then to guaranteed service (although many probably eventually will). A differentiated service—or even best-effort service—may be appropriate depending on the customer application requirements.
- The rate at which customers can realistically upgrade their infrastructures. There is a natural upgrade path from the technology needed to provide differentiated services to that needed to provide guaranteed services, which is a superset of that needed for differentiated services.
- The cost of implementing and deploying guaranteed service is likely to be more than that for a differentiated service.

Figure 46-2 The three levels of end-to-end QoS are best-effort service, differentiated service, and guaranteed service.



Congestion Management Tools

One way network elements handle an overflow of arriving traffic is to use a queuing algorithm to sort the traffic, and then determine some method of prioritizing it onto an output link. Cisco IOS software includes the following queuing tools:

- First-in, first-out (FIFO) queuing
- Priority queuing (PQ)
- Custom queuing (CQ)
- Weighted fair queuing (WFQ)

Each queuing algorithm was designed to solve a specific network traffic problem and has a particular effect on network performance, as described in the following sections.

FIFO: Basic Store-and-Forward Capability

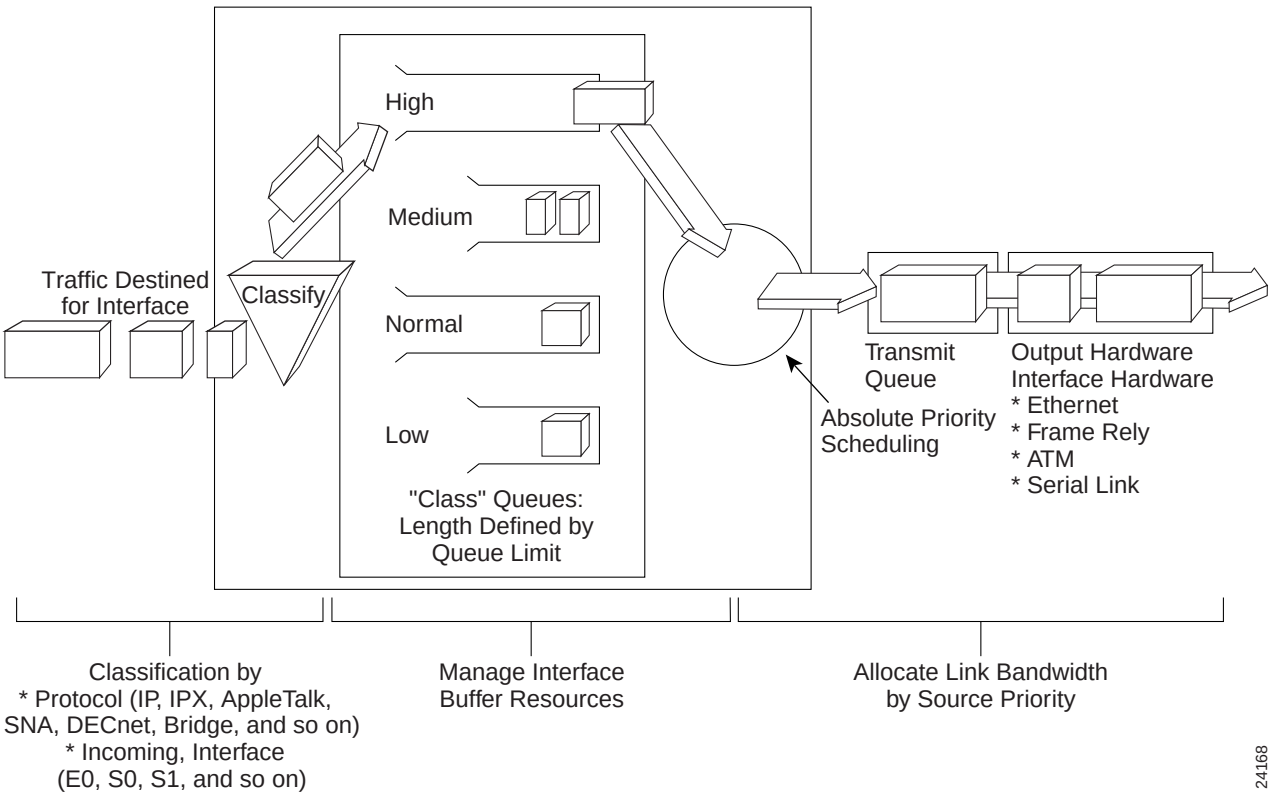
In its simplest form, FIFO queuing involves storing packets when the network is congested and forwarding them in order of arrival when the network is no longer congested. FIFO is the default queuing algorithm in some instances, thus requiring no configuration, but it has several

shortcomings. Most importantly, FIFO queuing makes no decision about packet priority; the order of arrival determines bandwidth, promptness, and buffer allocation. Nor does it provide protection against ill-behaved applications (sources). Bursty sources can cause long delays in delivering time-sensitive application traffic, and potentially to network control and signaling messages. FIFO queuing was a necessary first step in controlling network traffic, but today's intelligent networks need more sophisticated algorithms. Cisco IOS software implements queuing algorithms that avoid the shortcomings of FIFO queuing.

PQ: Prioritizing Traffic

PQ ensures that important traffic gets the fastest handling at each point where it is used. It was designed to give strict priority to important traffic. Priority queuing can flexibly prioritize according to network protocol (for example IP, IPX, or AppleTalk), incoming interface, packet size, source/destination address, and so on. In PQ, each packet is placed in one of four queues—high, medium, normal, or low—based on an assigned priority. Packets that are not classified by this priority-list mechanism fall into the normal queue; see Figure 46-3. During transmission, the algorithm gives higher-priority queues absolute preferential treatment over low-priority queues.

Figure 46-3 Priority queuing places data into four levels of queues: high, medium, normal, and low.

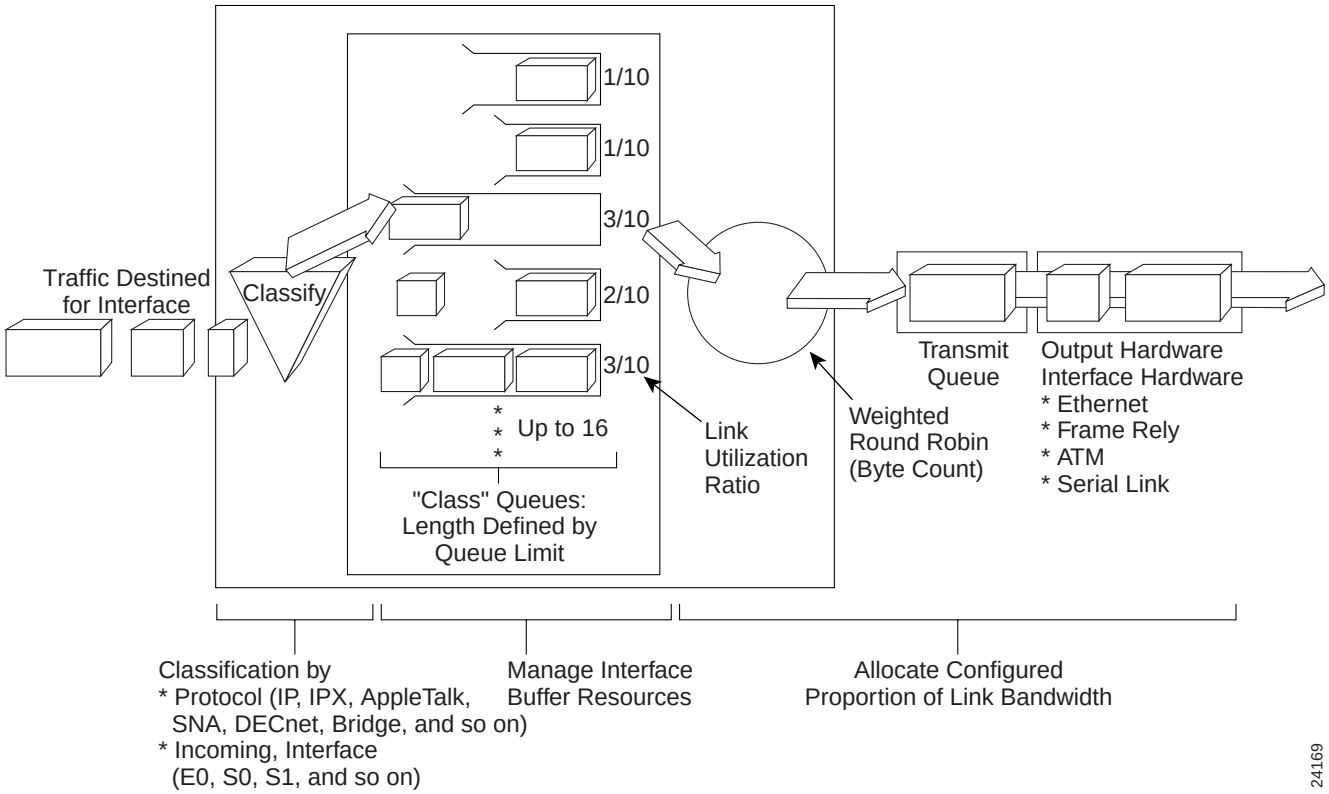


PQ is useful for making sure that mission-critical traffic traversing various WAN links gets priority treatment. For example, Cisco uses PQ to ensure that important Oracle-based sales reporting data gets to its destination ahead of other, less-critical traffic. PQ currently uses static configuration and thus does not automatically adapt to changing network requirements.

CQ: Guaranteeing Bandwidth

CQ was designed to allow various applications or organizations to share the network among applications with specific minimum bandwidth or latency requirements. In these environments, bandwidth must be shared proportionally between applications and users. You can use the Cisco CQ feature to provide guaranteed bandwidth at a potential congestion point, ensuring the specified traffic a fixed portion of available bandwidth and leaving the remaining bandwidth to other traffic. Custom queuing handles traffic by assigning a specified amount of queue space to each class of packets and then servicing the queues in a round-robin fashion (see Figure 46-4).

Figure 46-4 Custom queuing handles traffic by assigning a specified amount of queue space to each class of packets and then servicing up to 17 queues in a round-robin fashion.



As an example, encapsulated Systems Network Architecture (SNA) requires a guaranteed minimum level of service. You could reserve half of available bandwidth for SNA data, and allow the remaining half to be used by other protocols such as IP and Internetwork Packet Exchange (IPX).

The queuing algorithm places the messages in one of 17 queues (queue 0 holds system messages such as keepalives, signaling, and so on), and is emptied with weighted priority. The router services queues 1 through 16 in round-robin order, dequeuing a configured byte count from each queue in each cycle. This feature ensures that no application (or specified group of applications) achieves more than a predetermined proportion of overall capacity when the line is under stress. Like PQ, CQ is statically configured and does not automatically adapt to changing network conditions.

WFQ: Cisco's Intelligent Queuing Tool for Today's Networks

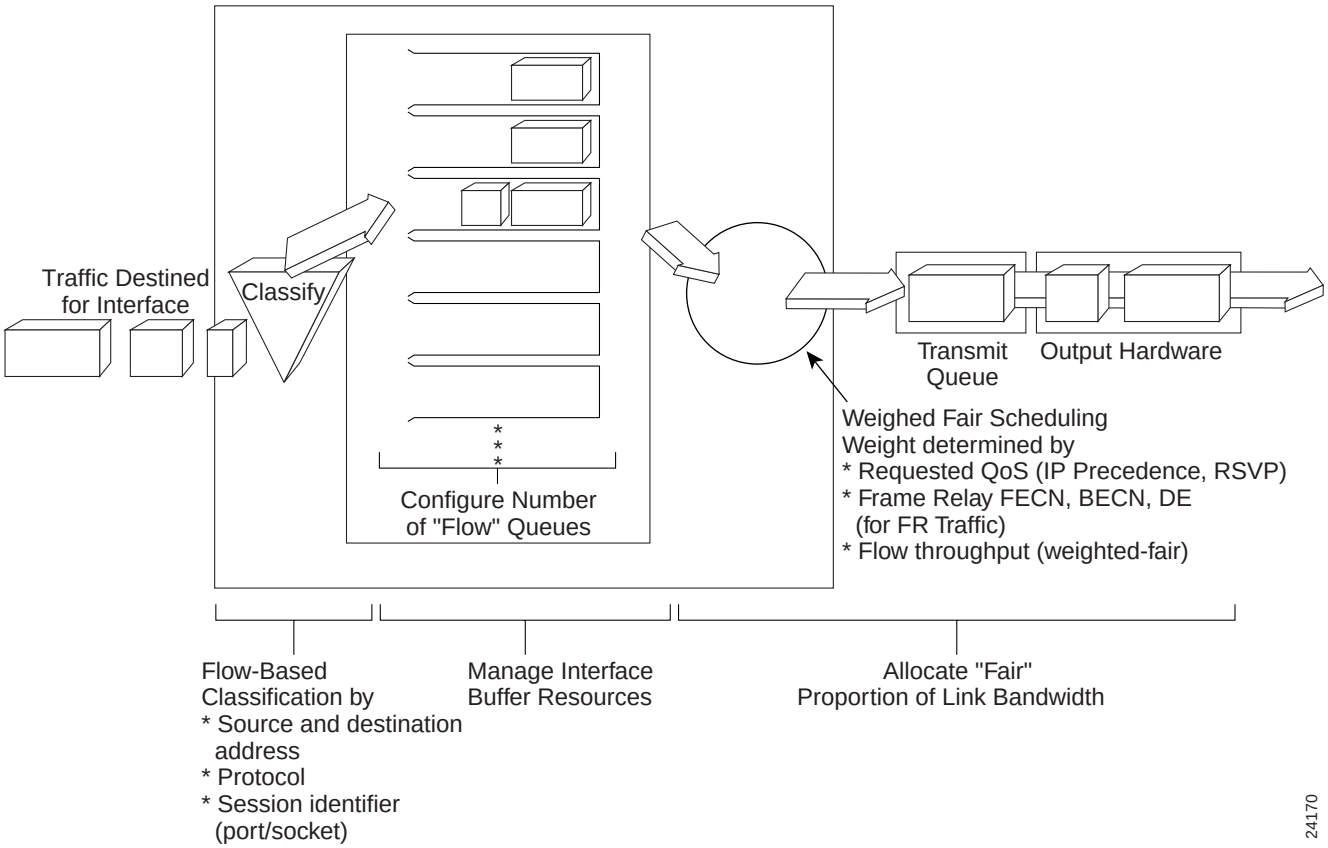
For situations in which it is desirable to provide consistent response time to heavy and light network users alike without adding excessive bandwidth, the solution is WFQ. WFQ is one of Cisco's premier queuing techniques. It is a flow-based queuing algorithm that does two things simultaneously: It schedules interactive traffic to the front of the queue to reduce response time, and it fairly shares the remaining bandwidth among high-bandwidth flows.

WFQ ensures that queues do not starve for bandwidth, and that traffic gets predictable service. Low-volume traffic streams—which comprise the majority of traffic—receive preferential service, transmitting their entire offered loads in a timely fashion. High-volume traffic streams share the remaining capacity proportionally between them, as shown in Figure 46-5.

WFQ is designed to minimize configuration effort and automatically adapts to changing network traffic conditions. In fact, WFQ does such a good job for most applications that it has been made the default queuing mode on most serial interfaces configured to run at or below E1 speeds (2.048 Mbps).

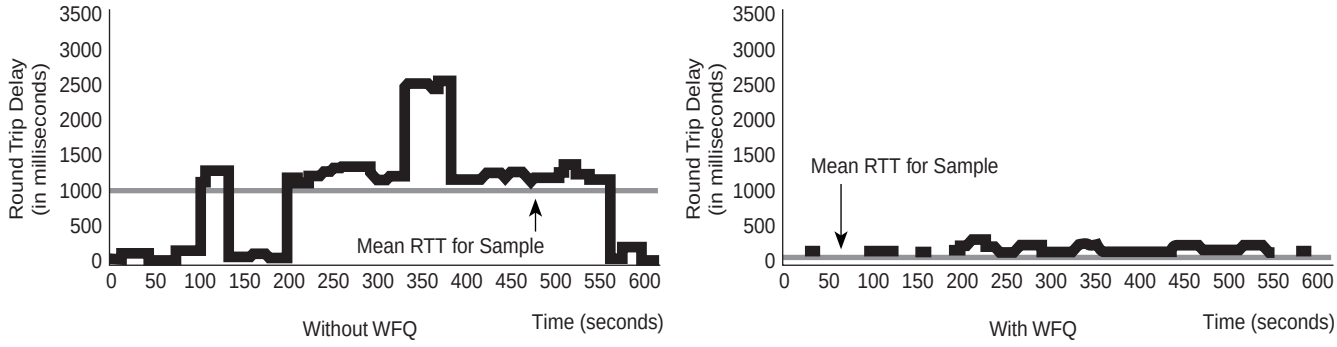
WFQ is efficient in that it uses whatever bandwidth is available to forward traffic from lower-priority flows if no traffic from higher-priority flows is present. This is different from time-division multiplexing (TDM), which simply carves up the bandwidth and lets it go unused if no traffic is present for a particular traffic type. WFQ works with both of Cisco's primary QoS signaling techniques—IP precedence and Resource Reservation Protocol (RSVP), described later in this chapter—to help provide differentiated QoS as well as guaranteed services.

Figure 46-5 With WFQ, if multiple high-volume conversations are active, their transfer rates and interarrival periods are made much more predictable.



The WFQ algorithm also addresses the problem of round-trip delay variability. If multiple high-volume conversations are active, their transfer rates and interarrival periods are made much more predictable. WFQ greatly enhances algorithms such as the SNA Logical Link Control (LLC) and the Transmission Control Protocol (TCP) congestion control and slow-start features. The result is more predictable throughput and response time for each active flow, as shown in Figure 46-6.

Figure 46-6 This diagram shows an example of interactive traffic delay (128-kbps Frame Relay WAN link).



Cooperation between WFQ and QoS Signaling Technologies

WFQ is IP precedence aware; that is, it is able to detect higher-priority packets marked with precedence by the IP forwarder and can schedule them faster, providing superior response time for this traffic. The IP Precedence field has values between 0 (the default) and 7. As the precedence value increases, the algorithm allocates more bandwidth to that conversation to make sure that it is served more quickly when congestion occurs. WFQ assigns a weight to each flow, which determines the transmit order for queued packets. In this scheme, 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.

For example, if you have one flow at each precedence level on an interface, each flow will get precedence + 1 parts of the link, as follows:

$1 + 2 + 3 + 4 + 5 + 6 + 7 + 8 = 36$

and the flows will get $^{8}/_{36}$, $^{7}/_{36}$, $^{6}/_{36}$, and $^{5}/_{36}$ of the link, and so on. However, if you have 18 precedence – 1 flows and 1 of each of the others, the formula looks like this:

$1 + 1\ 8 \times 2 + 3 + 4 + 5 + 6 + 7 + 8 = 36 – 2 + 18 \times 2 = 70$

and the flows will get $^{8}/_{70}$, $^{7}/_{70}$, $^{6}/_{70}$, $^{5}/_{70}$, $^{4}/_{70}$, $^{3}/_{70}$, $^{2}/_{70}$, and $^{1}/_{70}$ of the link, and 18 of the flows will get approximately $^{2}/_{70}$ of the link.

WFQ is also RSVP aware; RSVP uses WFQ to allocate buffer space and schedule packets, and guarantees bandwidth for reserved flows. Additionally, in a Frame Relay network, the presence of congestion is flagged by the forward explicit congestion notification (FECN) and backward explicit congestion notification (BECN) bits. WFQ weights are affected by Frame Relay discard eligible (DE), FECN, and BECN bits when the traffic is switched by the Frame Relay switching module. When congestion is flagged, the weights used by the algorithm are altered so that the conversation encountering the congestion transmits less frequently.

Congestion Avoidance Tools

Congestion avoidance techniques monitor network traffic loads in an effort to anticipate and avoid congestion at common network bottlenecks, as opposed to congestion management techniques that operate to control congestion after it occurs. The primary Cisco IOS congestion avoidance tool is weighted random early detection.

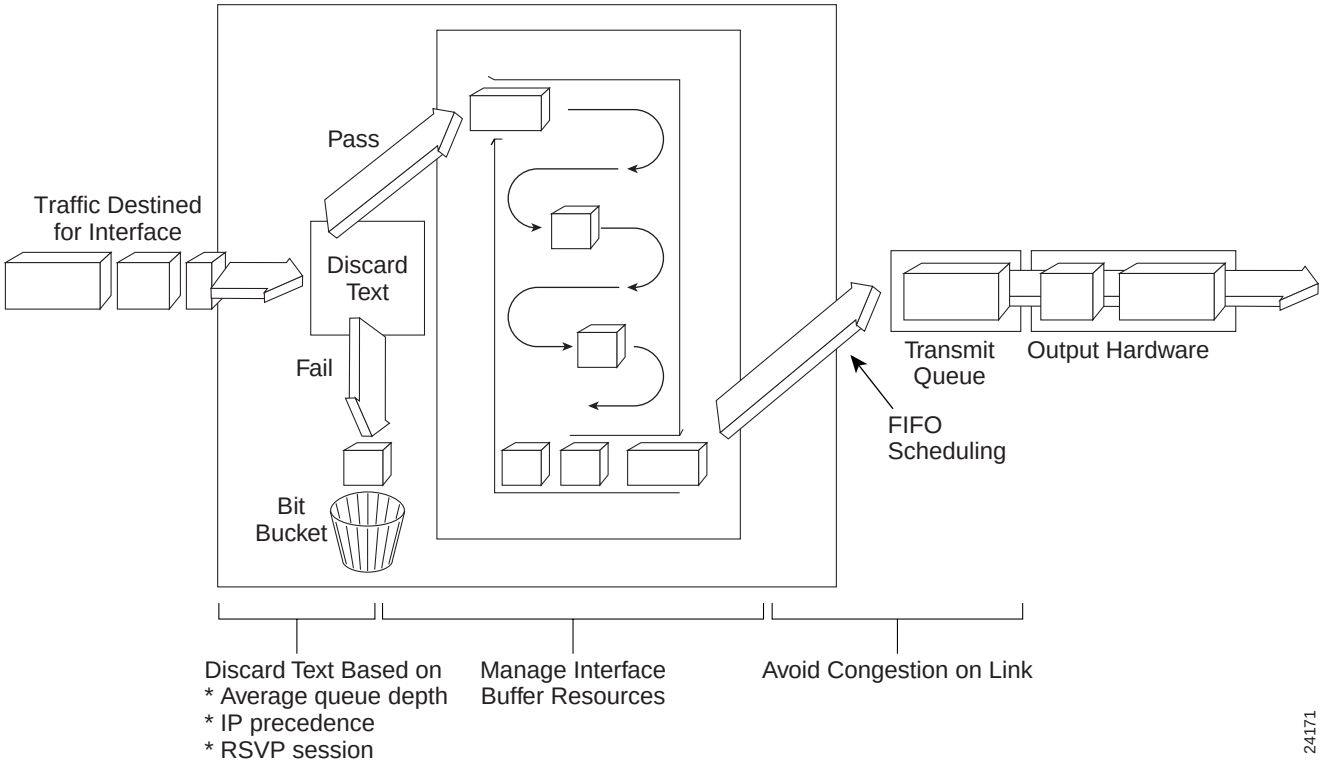
WRED: Avoiding Congestion

The random early detection (RED) algorithms are designed to avoid congestion in internetworks before it becomes a problem. RED works by monitoring traffic load at points in the network and stochastically discarding packets if the congestion begins to increase. The result of the drop is that the source detects the dropped traffic and slows its transmission. RED is primarily designed to work with TCP in IP internetwork environments.

WRED Cooperation with QoS Signaling Technologies

WRED combines the capabilities of the RED algorithm with IP precedence. This combination provides for preferential traffic handling for higher-priority packets. It can selectively discard lower-priority traffic when the interface starts to get congested and provide differentiated performance characteristics for different classes of service. See Figure 46-7. WRED is also RSVP aware, and can provide an integrated services controlled-load QoS.

Figure 46-7 WRED provides a method that stochastically discards packets if the congestion begins to increase.



D-WRED: Delivering High-Speed Differentiated Traffic on the 7500 Platform

Cisco IOS software also provides distributed weighted random early detection (D-WRED), a high-speed version of WRED that runs on VIP-distributed processors. The D-WRED algorithm provides functionality beyond what WRED provides, such as minimum and maximum queue depth thresholds and drop capabilities for each class of service.

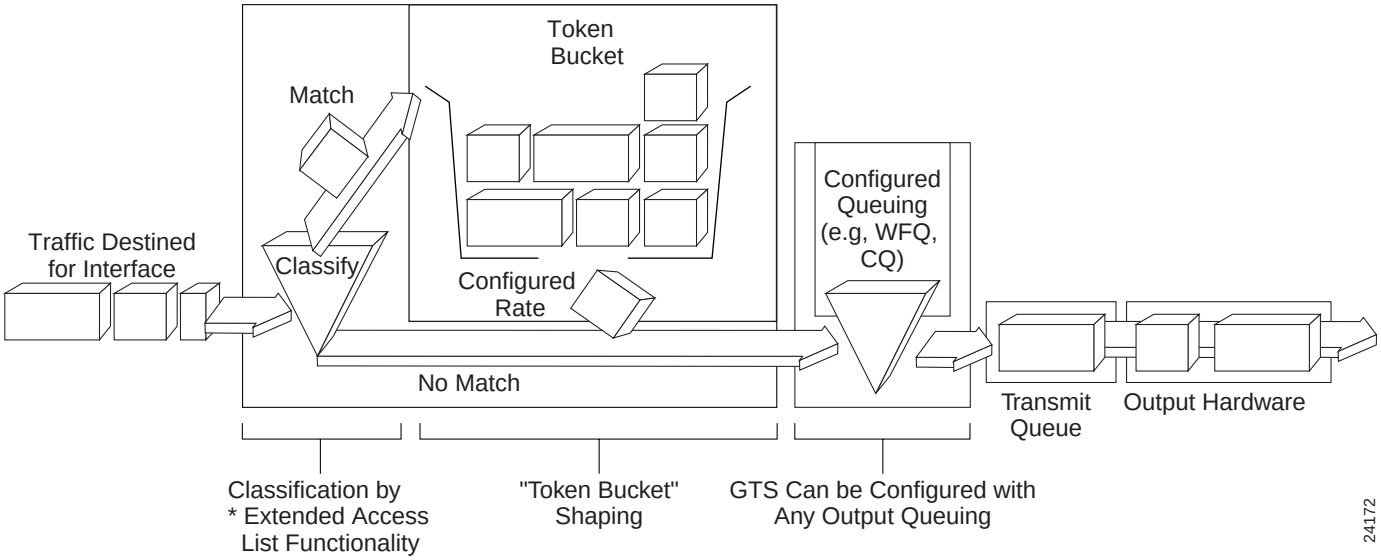
Traffic Shaping and Policing Tools

Cisco's QoS software solutions include two traffic shaping tools—generic traffic shaping (GTS) and Frame Relay traffic shaping (FRTS)—to manage traffic and congestion on the network.

GTS: Controlling Outbound Traffic Flow

GTS provides a mechanism to control the traffic flow on a particular interface. It reduces outbound traffic flow to avoid congestion by constraining specified traffic to a particular bit rate (also known as the token bucket approach), while queuing bursts of the specified traffic. Thus, traffic adhering to a particular profile can be shaped to meet downstream requirements, eliminating bottlenecks in topologies with data-rate mismatches. Figure 46-8 illustrates GTS.

Figure 46-8 Generic traffic shaping is applied on a per-interface basis.



GTS applies on a per-interface basis, can use access lists to select the traffic to shape, and works with a variety of Layer 2 technologies, including Frame Relay, ATM, Switched Multimegabit Data Service (SMDS), and Ethernet.

On a Frame Relay subinterface, GTS can be set up to adapt dynamically to available bandwidth by integrating BECN signals, or set up simply to shape to a prespecified rate. GTS can also be configured on an ATM/AIP (ATM Interface Processor) interface card to respond to RSVP signaled over statically configured ATM permanent virtual circuits (PVCs).

FRTS: Managing Frame Relay Traffic

FRTS provides parameters that are useful for managing network traffic congestion. These include committed information rate (CIR), FECN and BECN, and the DE bit. For some time, Cisco has provided support for FECN for DECnet, BECN for SNA traffic using direct LLC2 encapsulation via RFC 1490, and DE bit support. The FRTS feature builds on this Frame Relay support with additional capabilities that improve the scalability and performance of a Frame Relay network, increasing the density of virtual circuits and improving response time.

For example, you can configure rate enforcement—a peak rate configured to limit outbound traffic—to either the CIR or some other defined value, such as the excess information rate (EIR), on a per-virtual-circuit (VC) basis.

You can also define priority and custom queuing at the VC or subinterface level. This allows for finer granularity in the prioritization and queuing of traffic and provides more control over the traffic flow on an individual VC. If you combine CQ with the per-VC queuing and rate enforcement capabilities, you enable Frame Relay VCs to carry multiple traffic types such as IP, SNA, and IPX, with bandwidth guaranteed for each traffic type.

FRTS can eliminate bottlenecks in Frame Relay networks with high-speed connections at the central site and low-speed connections at the branch sites. You can configure rate enforcement to limit the rate at which data is sent on the VC at the central site. You can also use rate enforcement with the existing data-link connection identifier (DLCI) prioritization feature to further improve performance in this situation. FRTS applies only to Frame Relay PVCs and switched virtual circuits (SVCs).

Using information contained in BECN-tagged packets received from the network, FRTS can also dynamically throttle traffic. With BECN-based throttling, packets are held in the router's buffers to reduce the data flow from the router into the Frame Relay network. The throttling is done on a per-VC basis and the transmission rate is adjusted based on the number of BECN-tagged packets received.

FRTS also provides a mechanism for sharing media by multiple VCs. Rate enforcement allows the transmission speed used by the router to be controlled by criteria other than line speed, such as the CIR or EIR. The rate enforcement feature can also be used to preallocate bandwidth to each VC, creating a virtual TDM network. Finally, with the Cisco's FRTS feature, you can integrate StrataCom ATM Foresight closed loop congestion control to actively adapt to downstream congestion conditions.

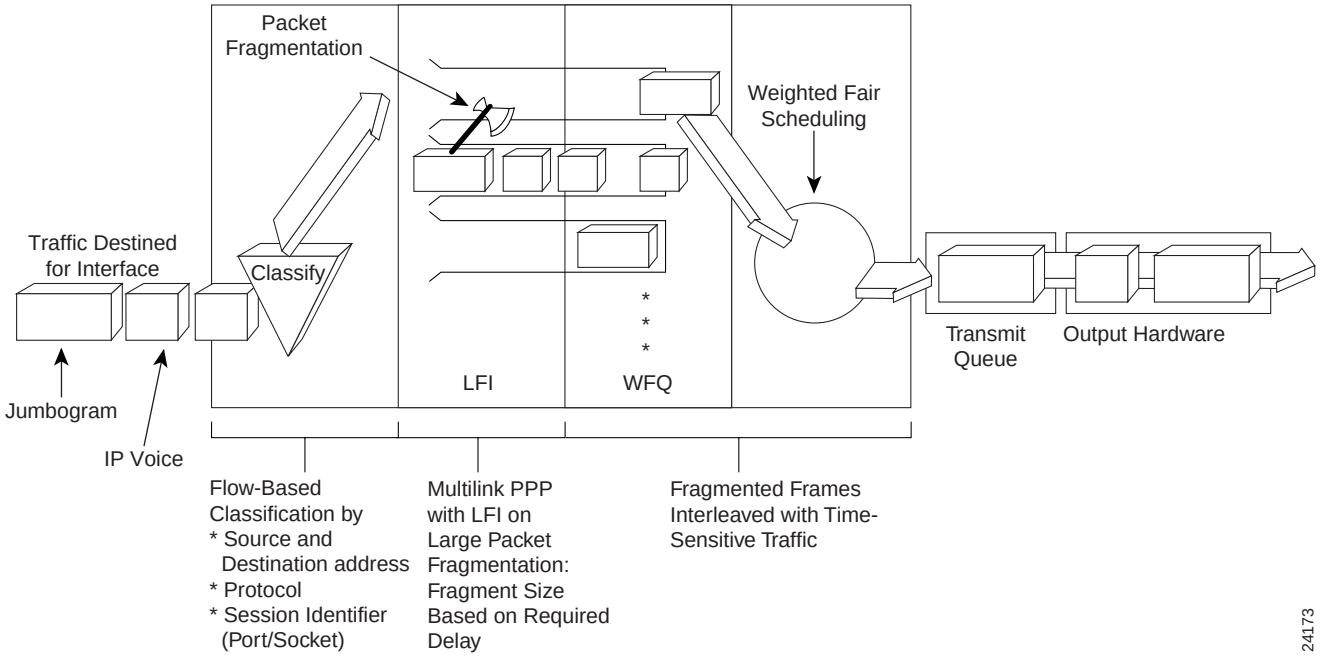
Link Efficiency Mechanisms

Currently, Cisco IOS software offers two link efficiency mechanisms—Link Fragmentation and Interleaving (LFI) and Real-Time Protocol Header Compression (RTP-HC)—which work with queuing and traffic shaping to improve the efficiency and predictability of the application service levels.

LFI: Fragmenting and Interleaving IP Traffic

Interactive traffic (Telnet, voice over IP, and the like) is susceptible to increased latency and jitter when the network processes large packets (for example LAN-to-LAN FTP transfers traversing a WAN link), especially as they are queued on slower links. The Cisco IOS LFI feature reduces delay and jitter on slower-speed links by breaking up large datagrams and interleaving low-delay traffic packets with the resulting smaller packets (see Figure 46-9).

Figure 46-9 By dividing large datagrams with the LFI feature, delay is reduced on slower speed links.

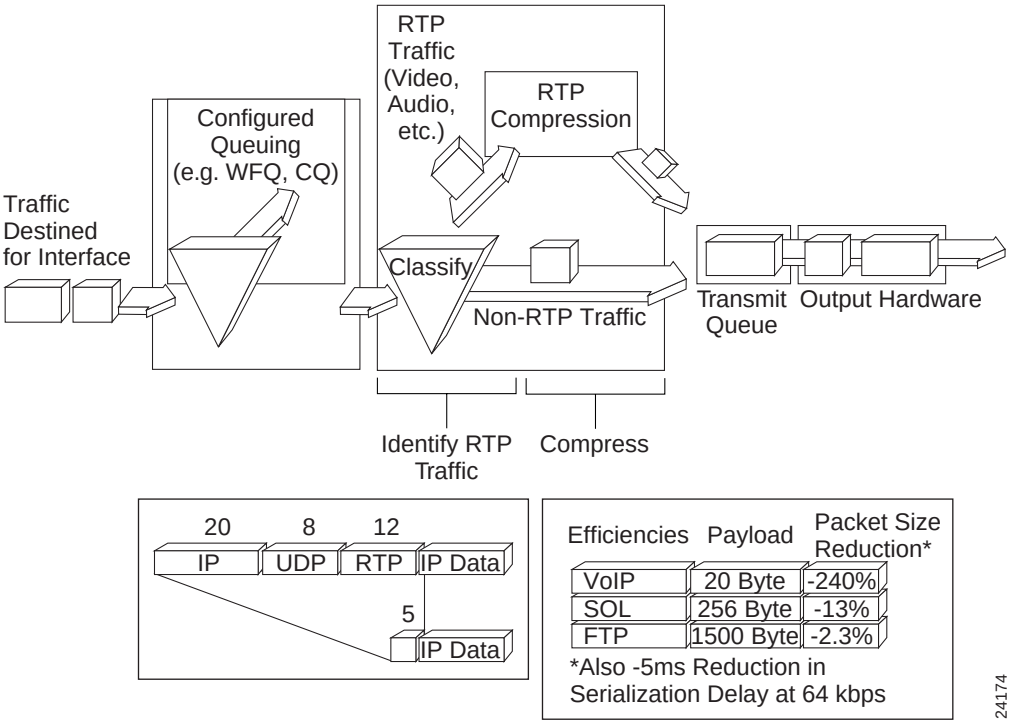


LFI was designed especially for lower-speed links in which serialization delay is significant. LFI requires that multilink Point-to-Point Protocol (PPP) be configured on the interface with interleaving turned on. A related IETF draft, titled Multiclass Extensions to Multilink PPP (MCML) implements almost the same function as LFI.

RTP Header Compression: Increasing Efficiency of Real-Time Traffic

Real-Time Transport Protocol is a host-to-host protocol used for carrying newer multimedia application traffic, including packetized audio and video, over an IP network. Real-Time Transport Protocol provides end-to-end network transport functions intended for applications transmitting real-time requirements, such as audio, video, or simulation data over multicast or unicast network services. Real-Time Transport Protocol header compression increases efficiency for many of the newer voice over IP or multimedia applications that take advantage of Real-Time Transport Protocol, especially on slow links. Figure 46-10 illustrates Real-Time Transport Protocol header compression.

Figure 46-10 This diagram illustrates Real-Time Transport Protocol header compression.



For compressed-payload audio applications, the RTP packet has a 40-byte header and typically a 20- to 150-byte payload. Given the size of the IP/UDP/Real-Time Transport Protocol header combination, it is inefficient to transmit an uncompressed header. Real-Time Transport Protocol header compression helps Real-Time Transport Protocol run more efficiently—especially over lower-speed links—by compressing the Real-Time Transport Protocol/UDP/IP header from 40 bytes to 2 to 5 bytes. This is especially beneficial for smaller packets (such as IP voice traffic) on slower links (385 kbps and below), where RTP header compression can reduce overhead and transmission delay significantly. Real-Time Transport Protocol header compression reduces line overhead for multimedia Real-Time Transport Protocol traffic with a corresponding reduction in delay, especially for traffic that uses short packets relative to header length.

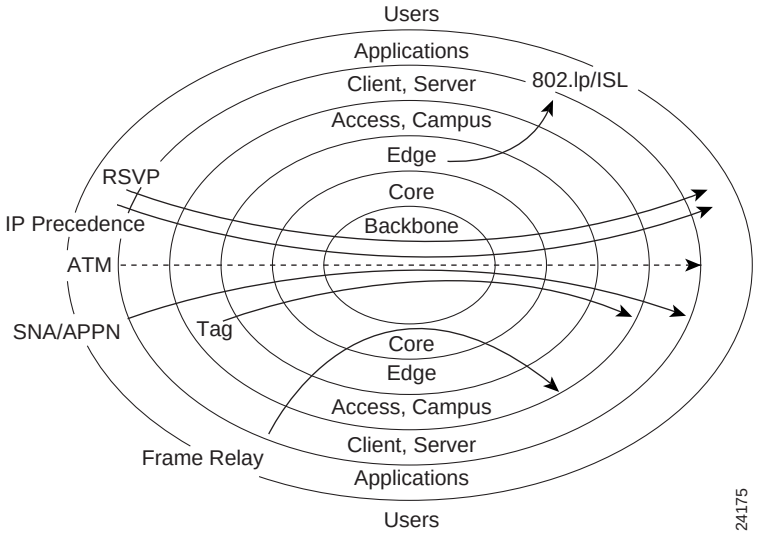
RTP header compression is supported on serial lines using Frame Relay, High-Level Data Link Control (HDLC), or PPP encapsulation. It is also supported over ISDN interfaces. A related IETF draft, titled Compressed RTP (CRTP), defines essentially the same functionality.

QoS Signaling

Think of QoS signaling as a form of network communication. It provides a way for an end station or a network element to signal certain requests to a neighbor. For example, an IP network can use part of the IP packet header to request special handling of priority or time-sensitive traffic. QoS signaling is useful for coordinating the traffic handling techniques described earlier in this chapter and has a key role in configuring successful end-to-end QoS across your network.

True end-to-end QoS requires that every element in the network path—switch, router, firewall, host, client, and so on—deliver its part of QoS, and it all must be coordinated with QoS signaling. However, the challenge is finding a robust QoS signaling solution that can operate end-to-end over heterogeneous network infrastructures. Although many viable QoS signaling solutions provide QoS at some places in the infrastructure, they often have limited scope across the network, as shown in Figure 46-11.

Figure 46-11 QoS signaling solutions provide QoS at some places in the infrastructure; they often have limited scope across the network.



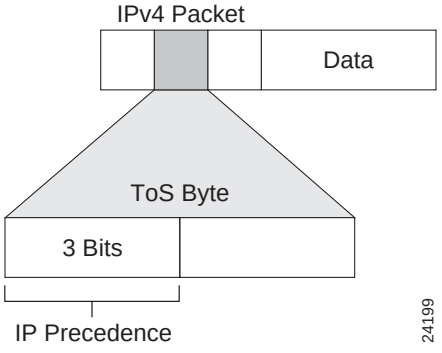
Cisco IOS software takes advantage of the end-to-end nature of IP to meet this challenge by overlaying Layer 2 technology-specific QoS signaling solutions with the Layer 3 IP QoS signaling methods of RSVP and IP precedence.

This section focuses on IP precedence and RSVP because both of these methods take advantage of the end-to-end nature of IP. As the majority of applications converge on the use of IP as the primary networking protocol, IP precedence and RSVP provide a powerful combination for QoS signaling—IP precedence signals for differentiated QoS, and RSVP for guaranteed QoS.

IP Precedence: Signaling Differentiated QoS

IP precedence utilizes the three precedence bits in the IPv4 header's ToS (Type of Service) field to specify class of service for each packet, as shown in Figure 46-12. You can partition traffic in up to six classes of service using IP precedence (two others are reserved for internal network use). The queuing technologies throughout the network can then use this signal to provide the appropriate expedited handling.

Figure 46-12 This diagram shows the IP precedence ToS field in an IP packet header.



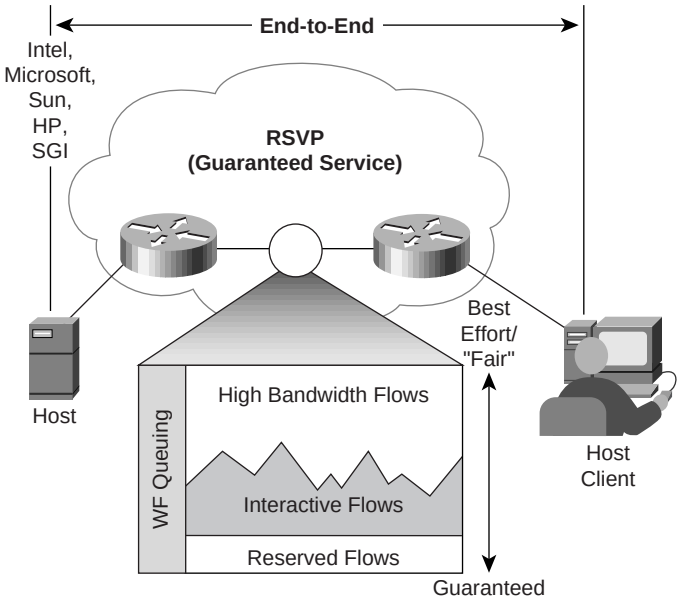
Features such as policy-based routing and committed access rate (CAR) can be used to set precedence based on extended access-list classification. This allows considerable flexibility for precedence assignment, including assignment by application or user, or by destination and source subnet, and so on. Typically this functionality is deployed as close to the edge of the network (or administrative domain) as possible, so that each subsequent network element can provide service based on the determined policy.

IP precedence can also be set in the host or network client, and this signaling can be used optionally; however, this can be overridden by policy within the network. IP precedence enables service classes to be established using existing network queuing mechanisms (for example, WFQ or WRED), with no changes to existing applications or complicated network requirements. Note that this same approach is easily extended to IPv6 using its Priority field.

RSVP: Guaranteeing QoS

RSVP is an IETF Internet Standard (RFC 2205) protocol for allowing an application to dynamically reserve network bandwidth. RSVP enables applications to request a specific QoS for a data flow, as shown in Figure 46-13. Cisco's implementation also allows RSVP to be initiated within the network, using configured proxy RSVP. Network managers can thereby take advantage of the benefits of RSVP in the network, even for non-RSVP-enabled applications and hosts.

Figure 46-13 This figure shows RSVP implemented in a Cisco-based router network.



Hosts and routers use RSVP to deliver QoS requests to the routers along the paths of the data stream and to maintain router and host state to provide the requested service, usually bandwidth and latency. RSVP uses a mean data rate, the largest amount of data the router will keep in queue, and minimum QoS to determine bandwidth reservation.

WFQ or WRED acts as the workhorse for RSVP, setting up the packet classification and scheduling required for the reserved flows. Using WFQ, RSVP can deliver an integrated services guaranteed service. Using WRED, it can deliver a controlled load service. WFQ continues to provide its advantageous handling of nonreserved traffic by expediting interactive traffic and fairly sharing the

remaining bandwidth between high-bandwidth flows, and WRED provides its commensurate advantages for non-RSVP flow traffic. RSVP can be deployed in existing networks with a software upgrade.

Tag Switching: Allowing Flexible Traffic Engineering

Cisco's tag switching feature contains the mechanisms to interoperate with and take advantage of both RSVP and IP precedence signaling. The tag switching header contains a 3-bit field that can be used as a traffic prioritization signal. It can also be used to map particular flows and classes of traffic along engineered tag switching paths to obtain the required QoS through the tag switching portion of a network.

Cisco's QoS Policy, Management, and Accounting Capabilities

Cisco IOS software provides technologies that enable policy control, management, and accounting of the QoS techniques described in this chapter. The following sections provide an overview of these technologies.

QoS Policy Control

The QoS policy control architecture is being developed as a key initial piece of the CiscoAssure policy networking initiative. This initiative leverages standards-based QoS policy control protocols and mechanisms to implement QoS policy from a single console interface.

At the infrastructure level, packet classification is a key capability for each policy technique that allows the appropriate packets traversing a network element or particular interface to be selected for QoS. These packets can then be marked for the appropriate IP precedence in some cases, or identified as an RSVP. Policy control also requires integration with underlying data link-layer network technologies or non-IP protocols.

SNA ToS

SNA ToS in conjunction with data-link switching plus (DLSw+), allows mapping of traditional SNA class of service (CoS) into IP differentiated service. This feature takes advantage of both QoS signaling and pieces of the architecture. DLSW+ opens four TCP sessions and maps each SNA ToS traffic into a different session. Each session is marked by IP precedence. Cisco's congestion control technologies (CQ, PQ, and WFQ) act on these sessions to provide a bandwidth guarantee or other improved handling across an intranet, as shown in Figure 46-14. This provides a migration path for traditional SNA customers onto an IP-based intranet, while preserving the performance characteristics expected of SNA.

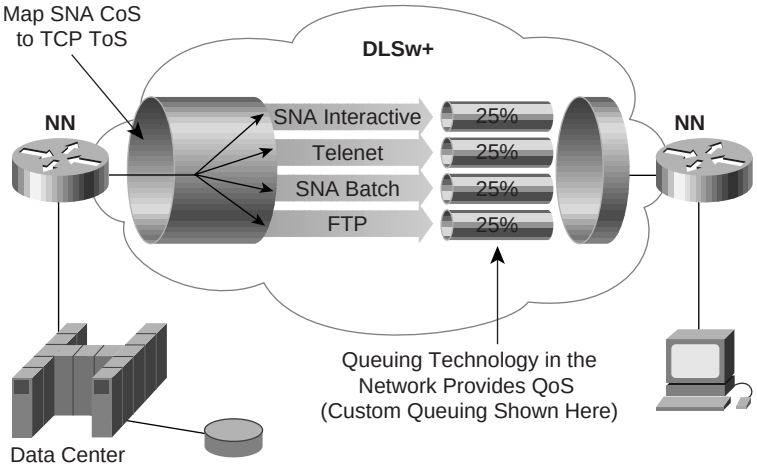
Thus, traditional mainframe-based, mission-critical applications can take advantage of evolving IP intranets and extranets without sacrificing the QoS capabilities historically provided by SNA networking.

QoS Policy Setting with Policy-Based Routing (PBR)

Cisco IOS PBR allows you to classify traffic based on extended access list criteria, set IP precedence bits, and even route to specific traffic-engineered paths that may be required to allow a specific QoS through the network. By setting precedence levels on incoming traffic and using them in

combination with the queuing tools described earlier in this chapter, you can create differentiated service. These tools provide powerful, simple, and flexible options for implementing QoS policies in your network.

Figure 46-14 SNA ToS, in conjunction with DLSw, allows mapping of SNA CoS into IP differentiated services.



You can also set up PBR as a way to route packets based on configured policies. Some applications or traffic can benefit from QoS-specific routing-transferring stock records to a corporate office (for example, on a higher-bandwidth, higher-cost link for a short time), while transmitting routine application data such as e-mail over a lower-bandwidth, lower-cost link. PBR can be used to direct packets to take different paths than the path derived from the routing protocols. It provides a more flexible mechanism for routing packets, complementing the existing mechanisms provided by routing protocols.

CAR: Managing Access Bandwidth Policy and Performing Policing

Similar in some ways to PBR, the CAR feature allows you to classify and police traffic on an incoming interface. It also allows specification of policies for handling traffic that exceeds a certain bandwidth allocation. CAR looks at traffic received on an interface, or a subset of that traffic selected by access list criteria, compares its rate to that of a configured token bucket, and then takes action based on the result (for example, drop or rewrite IP precedence).

QoS for Packetized Voice

One of the most promising uses for IP networks is to allow sharing of voice traffic with the traditional data and LAN-to-LAN traffic. Typically, this can help reduce transmission costs by reducing the number of network connections, sharing existing connections and infrastructure, and so on.

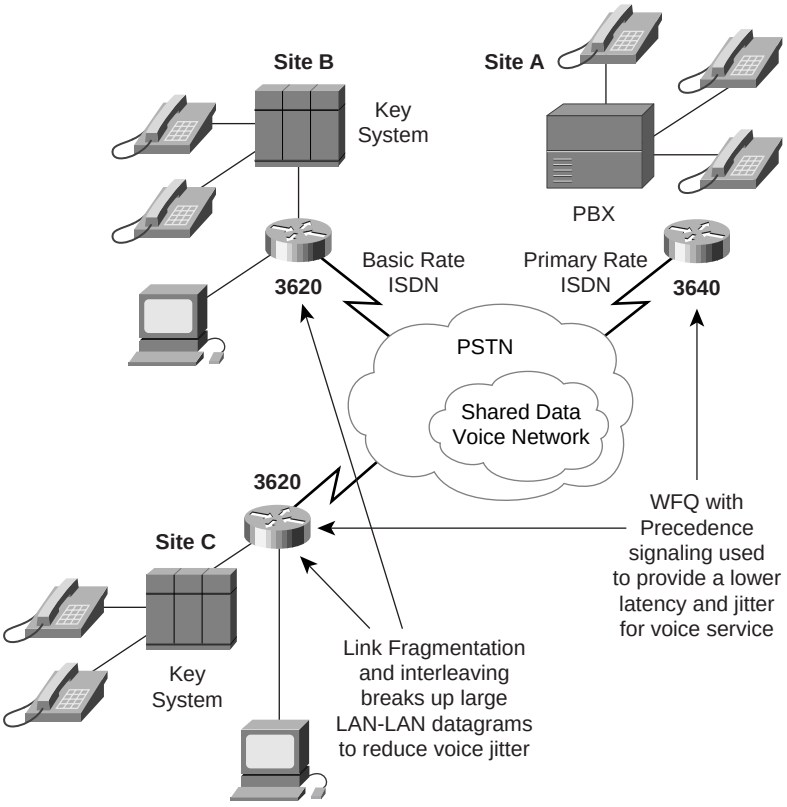
Cisco has a wide range of voice networking products and technologies, including a number of voice over IP (VoIP) solutions. To provide the required voice quality, however, QoS capability must be added to the traditional data-only network. Cisco IOS software QoS features give VoIP traffic the service it needs, while providing the traditional data traffic with the service it needs as well.

Figure 46-15 shows a business that has chosen to reduce some of its voice costs by combining voice traffic onto its existing IP network. Voice traffic at each office is digitized on voice modules on 3600 processors. This traffic is then routed via H.323 Gatekeeper, which also requests specific QoS for the

voice traffic. In this case, IP precedence is set to high for the voice traffic. WFQ is enabled on all the router interfaces for this network. WFQ automatically expedites the forwarding of high-precedence voice traffic out each interface, reducing delay and jitter for this traffic.

Because the IP network was originally handling LAN-to-LAN traffic, many datagrams traversing the network are large 1500-byte packets. On slow links (below T1/E1 speeds), voice packets may be forced to wait behind one of these large packets, adding tens or even hundreds of milliseconds to the delay. LFI is used in conjunction with WFQ to break up these "jumbograms" and interleave the voice traffic to reduce this delay as well as jitter.

Figure 46-15 This diagram provides an overview of a QoS VoIP solution.

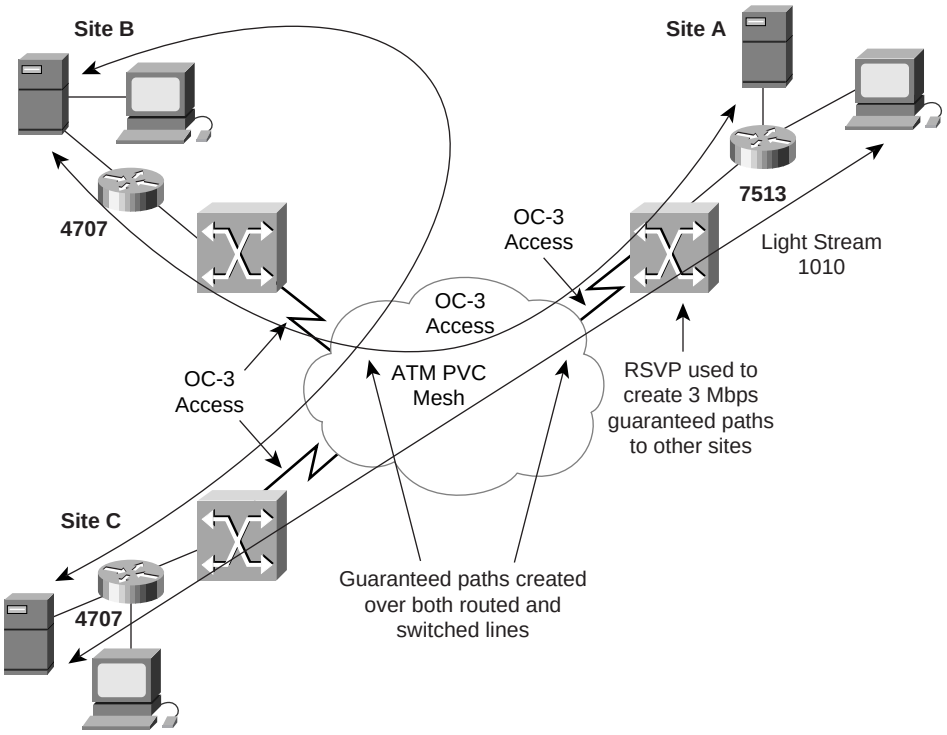


QoS for Streaming Video

One of the most significant challenges for IP-based networks, which have traditionally provided only best-effort service, has been to provide some type of service guarantees for different types of traffic. This has been a particular challenge for streaming video applications, which often require a significant amount of reserved bandwidth to be useful.

In the network shown in Figure 46-16, RSVP is used in conjunction with ATM PVCs to provide guaranteed bandwidth to a mesh of locations. RSVP is configured from within Cisco IOS to provide paths from the router networks, at the edges, and through the ATM core. Simulation traffic then uses these guaranteed paths to meet the constraints of geographically distributed real-time simulation. Video-enabled machines at the various sites also use this network to do live videoconferencing.

Figure 46-16 The network diagram shows the use of RSVP in a meshed ATM environment.



In this instance, OC-3 ATM links are configured with multiple 3 Mbps PVCs connecting to various remote sites. RSVP ensures that QoS from this PVC is extended to the appropriate application across the local routed network. In the future, Cisco IOS will extend this RSVP capability to dynamically set up ATM SVCs. This will reduce configuration complexity and add a great degree of automatic configuration.

QoS Looking Forward

In a continued evolution toward end-to-end services, Cisco is expanding QoS interworking to operate more seamlessly across heterogeneous link-layer technologies, and working closely with host platform partners to ensure interoperation between networks and end systems.