

QoS and Good Design

Need for QoS Mechanisms

This topic describes problems associated with transmitting voice over a data network and the need for QoS in such a network.

What Is QoS and Why Is It Needed?

- Delay
- Delay variation (jitter)
- Packet loss

The diagram shows two blue human figures representing users at IP telephones. They are connected to two blue routers. A cloud labeled 'VoIP QoS' is positioned between the routers. A speech bubble from the left user says 'Hello'. A speech bubble from the right user says 'H&e~lo', indicating a corrupted message due to network issues. The background is a light greenish-yellow.

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Real-time applications, such as voice applications, have different characteristics and requirements than traditional data applications. Voice applications tolerate little variation in the amount of delay. This delay variation affects delivery of voice packets. Packet loss and jitter degrade the quality of the voice transmission that is delivered to the recipient. The figure shows how these problems can affect a voice message.

Objectives of QoS

To ensure that VoIP is a realistic replacement for standard PSTN telephony services, customers must receive the same consistently high quality of voice transmission that they receive with basic telephone services. This topic discusses how QoS can help you achieve this objective.

Objectives of QoS

QoS has the following objectives:

- **Supporting dedicated bandwidth**
- **Improving loss characteristics**
- **Avoiding and managing network congestion**
- **Shaping network traffic**
- **Setting traffic priorities across the network**

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Like other real-time applications, VoIP is extremely sensitive to issues related to bandwidth and delay. To ensure that VoIP transmissions are intelligible to the receiver, voice packets cannot be dropped, excessively delayed, or subject to variations in delay, or jitter.

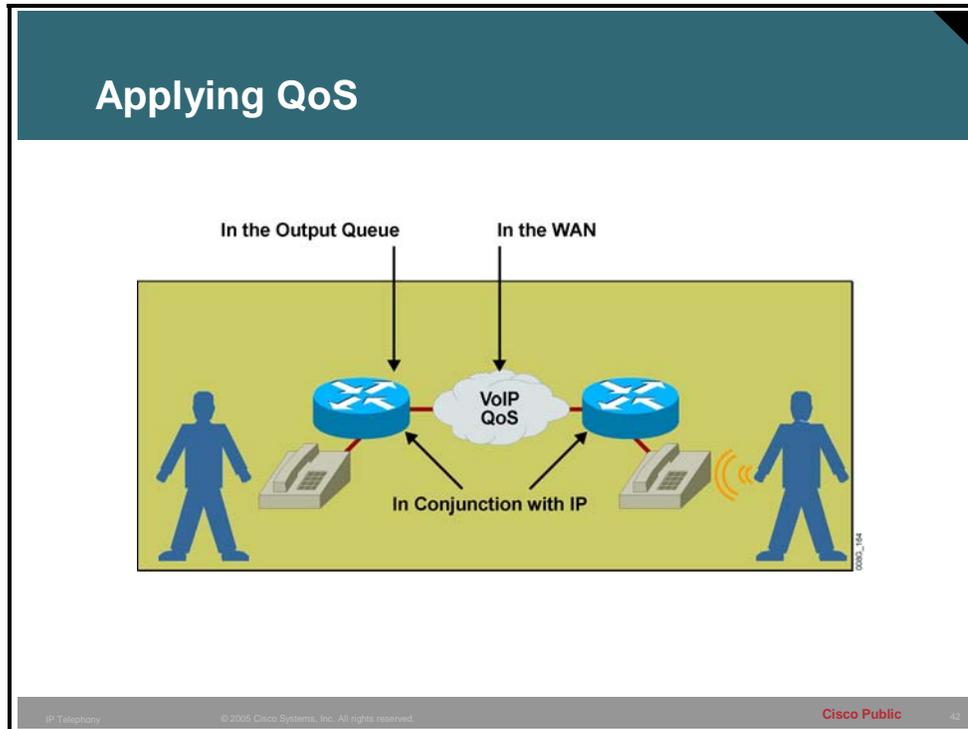
Example: QoS Objectives

VoIP guarantees high-quality voice transmission only if the signaling and audio channel packets have priority over other kinds of network traffic. To deploy VoIP, you must provide an acceptable level of voice quality by meeting VoIP traffic requirements for issues related to bandwidth, latency, and jitter. QoS provides better, more predictable network service by performing the following:

- **Supporting dedicated bandwidth:** Designing the network such that speeds and feeds can support the desired voice and data traffic
- **Improving loss characteristics:** Designing the Frame Relay network such that discard eligibility is not a factor, keeping voice below committed information rate (CIR)
- **Avoiding and managing network congestion:** Ensuring that the LAN and WAN infrastructure can support the volume of data traffic and voice calls
- **Shaping network traffic:** Using Cisco traffic-shaping tools to ensure smooth and consistent delivery of frames to the WAN
- **Setting traffic priorities across the network:** Marking the voice traffic as priority and queuing it first

Applying QoS for End-to-End Improvement of Voice Quality

Voice features for Cisco IOS QoS are deployed at different points in the network and designed for use with other QoS features to achieve specific goals, such as control over jitter and delay. This topic lists the network areas in which Cisco IOS QoS is implemented.



Cisco IOS software includes a complete set of features for delivering QoS throughout the network. Following are Cisco IOS features that address the voice packet delivery requirements of end-to-end QoS and service differentiation:

- In the output queue of the router:
 - **Class-based weighted fair queuing (CBWFQ):** Extends the standard weighted fair queuing (WFQ) functionality by providing support for user-defined traffic classes. You can create a specific class for voice traffic by using CBWFQ.
 - **Low Latency Queuing (LLQ):** Provides strict priority queuing on ATM VCs and serial interfaces. LLQ configures the priority status for a class within CBWFQ and is not limited to UDP port numbers (as in IP RTP priority). LLQ is considered a “best practice” by the Cisco Enterprise Solutions Engineering (ESE) group for delivering voice QoS services over a WAN.
 - **WFQ and distributed weighted fair queuing (DWFQ):** Segregates traffic into flows and then schedules traffic onto the outputs to meet specified bandwidth allocation or delay bounds.
 - **Weighted random early detection (WRED) and distributed weighted random early detection (DWRED):** Provides differentiated performance characteristics for different classes of service. This classification allows preferential handling of voice traffic under congestion conditions without worsening the congestion.

- In the WAN or WAN protocol:
 - **Committed access rate (CAR):** Provides a rate-limiting feature for allocating bandwidth commitments and bandwidth limitations to traffic sources and destinations. At the same time, it specifies policies for handling the traffic that may exceed bandwidth allocation.
 - **Frame Relay traffic shaping (FRTS):** Delays excess traffic by using a buffer or queuing mechanism to hold packets and shape the flow when the data rate of the source is higher than expected.
 - **Frame Relay Forum Standard 12 (FRF.12):** Ensures predictability for voice traffic by providing better throughput on low-speed Frame Relay links. FRF.12 interleaves delay-sensitive voice traffic on one VC with fragments of a long frame on another VC that is using the same interface.
 - **IP to ATM class of service (CoS):** Includes a feature suite that maps CoS characteristics between the IP and ATM. It also offers differential service classes across the entire WAN—not just the routed portion—and gives mission-critical applications exceptional service during periods of high network usage and congestion.
 - **Multilink PPP (MLP) with link fragmentation and interleaving (LFI):** Allows large packets to be multilink-encapsulated and fragmented so that they are small enough to satisfy the delay requirements of real-time traffic. LFI also provides a special transmit queue for smaller, delay-sensitive packets, enabling them to be sent earlier than other flows.
- In conjunction with the IP operation:
 - **Compressed Real-Time Transport Protocol (CRTP):** Compresses the extensive RTP header when used in conjunction with RTP. The result is decreased consumption of available bandwidth for voice traffic and a corresponding reduction in delay.
 - **Resource Reservation Protocol (RSVP):** Supports the reservation of resources across an IP network, allowing end systems to request QoS guarantees from the network. For networks that support VoIP, RSVP—in conjunction with features that provide queuing, traffic shaping, and voice call signaling—provides Call Admission Control (CAC) for voice traffic.
 - **QoS policy propagation on Border Gateway Protocol (BGP):** Steadies BGP to distribute QoS policy to remote routers in a network. It allows classification of packets and then uses other QoS features, such as CAR and WRED, to specify and enforce business policies to fit a business model.