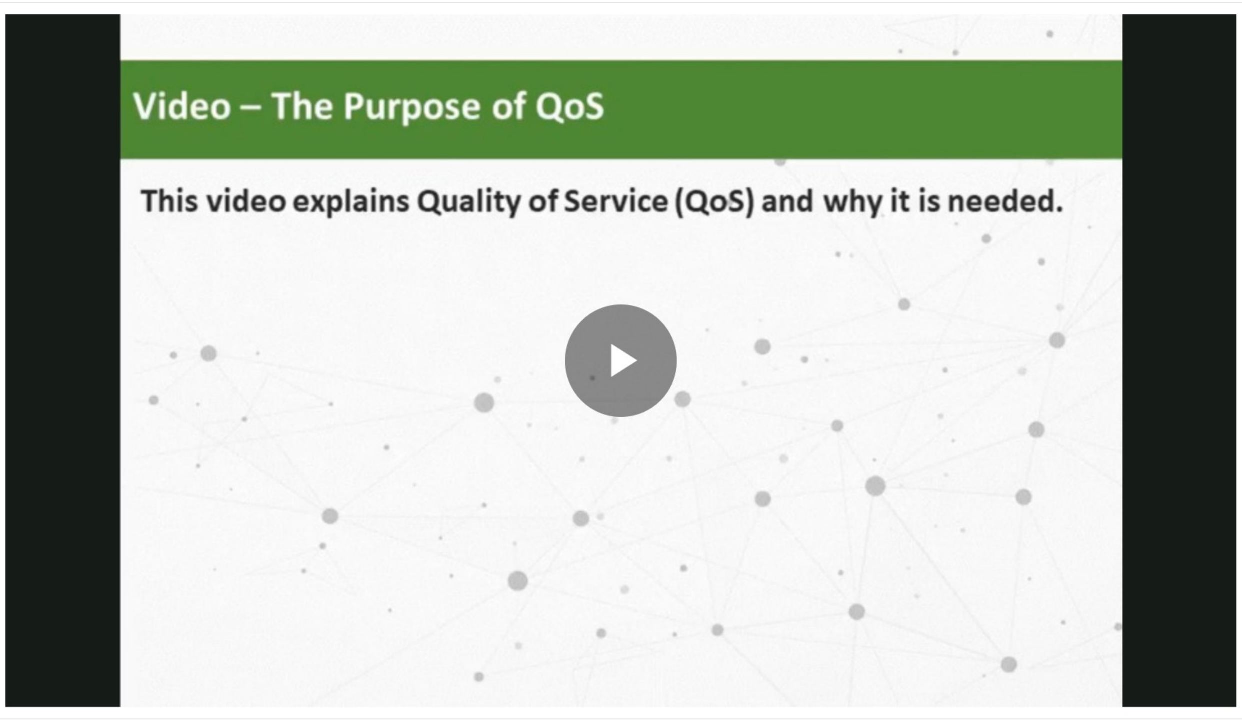
Network Transmission Quality

9.1.1

Video Tutorial - The Purpose of QoS

Click Play for a brief explanation of the purpose of QoS.



9.1.2

Prioritizing Traffic

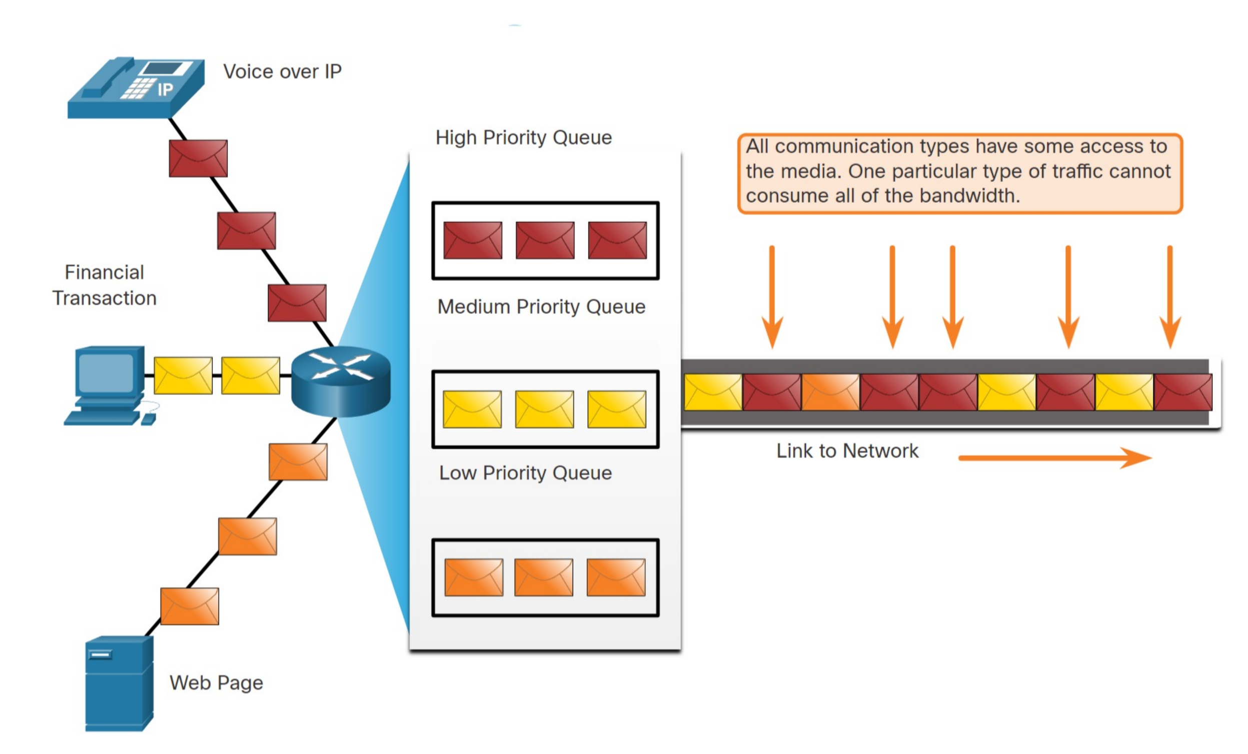
In the previous video, you learned about the purpose of Quality of Service (QoS). QoS is an ever-increasing requirement of networks today. New applications, such as voice and live video transmissions, create higher expectations for quality delivery among users.

Congestion occurs when multiple communication lines aggregate onto a single device such as a router, and then much of that data is placed on just a few outbound interfaces, or onto a slower interface. Congestion can also occur when large data packets prevent smaller packets from being transmitted in a timely manner.

When the volume of traffic is greater than what can be transported across the network, devices queue (hold) the packets in memory until resources become available to transmit them. Queuing packets causes delay because new packets cannot be transmitted until previous packets have been processed. If the number of packets to be queued continues to increase, the memory within the device fills up and packets are dropped. One QoS technique that can help with this problem is to classify data into multiple queues, as shown in the figure.

Note: A device implements QoS only when it is experiencing some type of congestion.

Using Queues to Priorities Communications



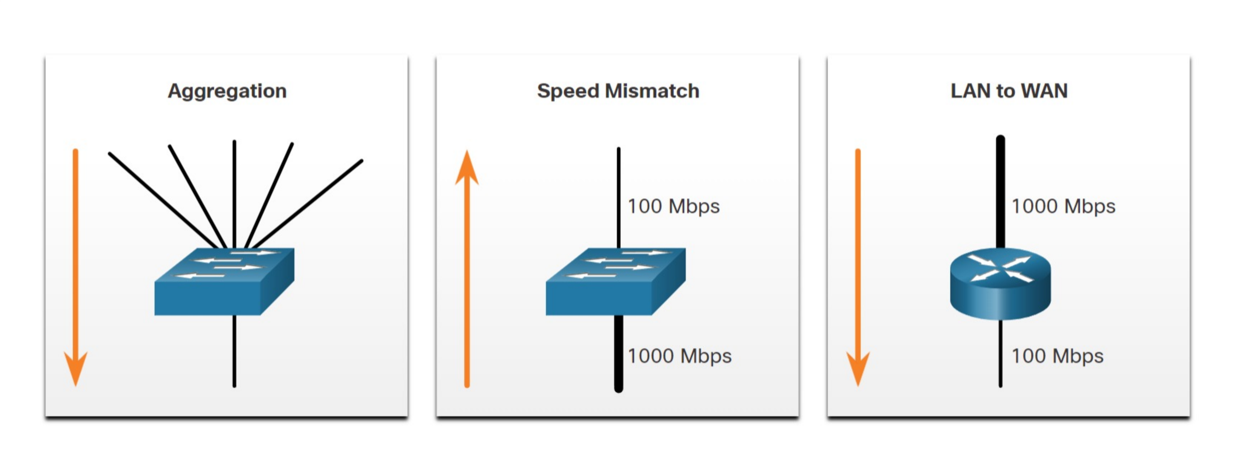
9.1.3

Bandwidth, Congestion, Delay, and Jitter

Network bandwidth is measured in the number of bits that can be transmitted in a single second, or bits per second (bps). For example, a network device may be described as having the capability to perform at 10 gigabits per second (Gbps).

Network congestion causes delay. An interface experiences congestion when it is presented with more traffic than it can handle. Network congestion points are ideal candidates for QoS mechanisms. The figure shows three examples of typical congestion points.

Examples of Congestion Points



Delay or latency refers to the time it takes for a packet to travel from the source to the destination. Two types of delays are fixed and variable. A fixed delay is a specific amount of time a specific process takes, such as how long it takes to place a bit on the transmission media. A variable delay takes an unspecified amount of time and is affected by factors such as how much traffic is being processed.

The sources of delay are summarized in the table.

Sources of Delay

Delay	Description
Code delay	The fixed amount of time it takes to compress data at the source before transmitting to the first internetworking device, usually a switch.
Packetization delay	The fixed time it takes to encapsulate a packet with all the necessary header information.
Queuing delay	The variable amount of time a frame or packet waits to be transmitted on the link.
Serialization delay	The fixed amount of time it takes to transmit a frame onto the wire.
Propagation delay	The variable amount of time it takes for the frame to travel between the source and destination.
De-jitter delay	The fixed amount of time it takes to buffer a flow of packets and then send them out in evenly spaced intervals.

Jitter is the variation in the delay of received packets. At the sending side, packets are sent in a continuous stream with the packets spaced evenly apart. Due to network congestion, improper queuing, or configuration errors, the delay between each packet can vary instead of remaining constant. Both delay and jitter need to be controlled and minimized to support real-time and interactive traffic.

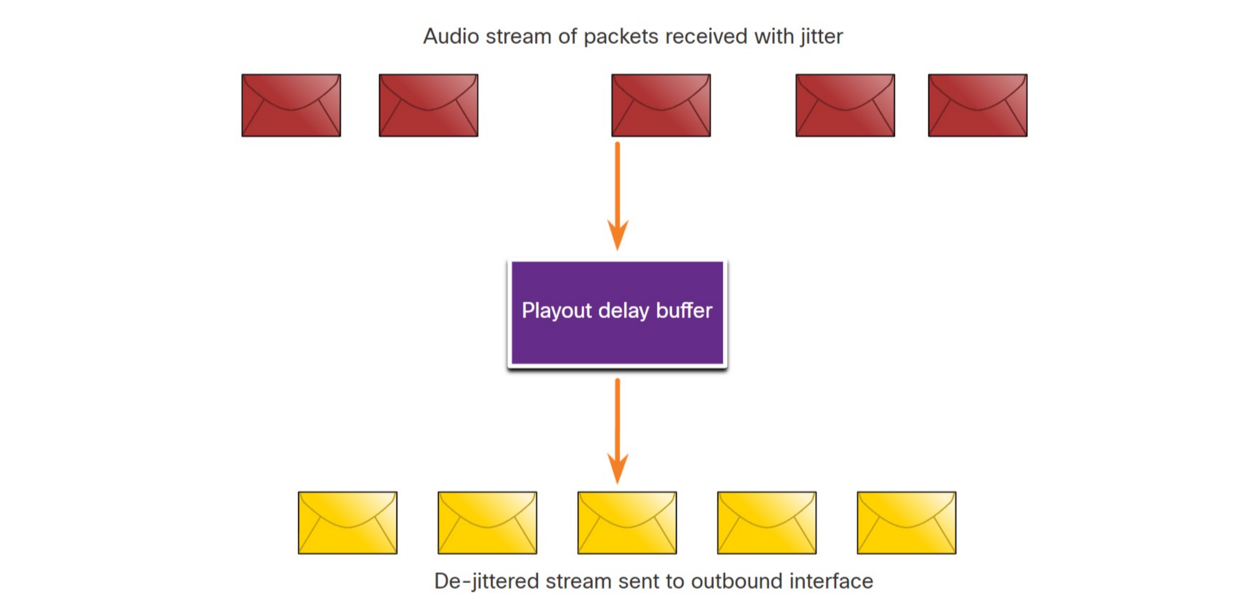
9.1.4

Packet Loss

Without any QoS mechanisms in place, packets are processed in the order in which they are received. When congestion occurs, network devices such as routers and switches can drop packets. This means that time-sensitive packets, such as real-time video and voice, will be dropped with the same frequency as data that is not time-sensitive, such as email and web browsing.

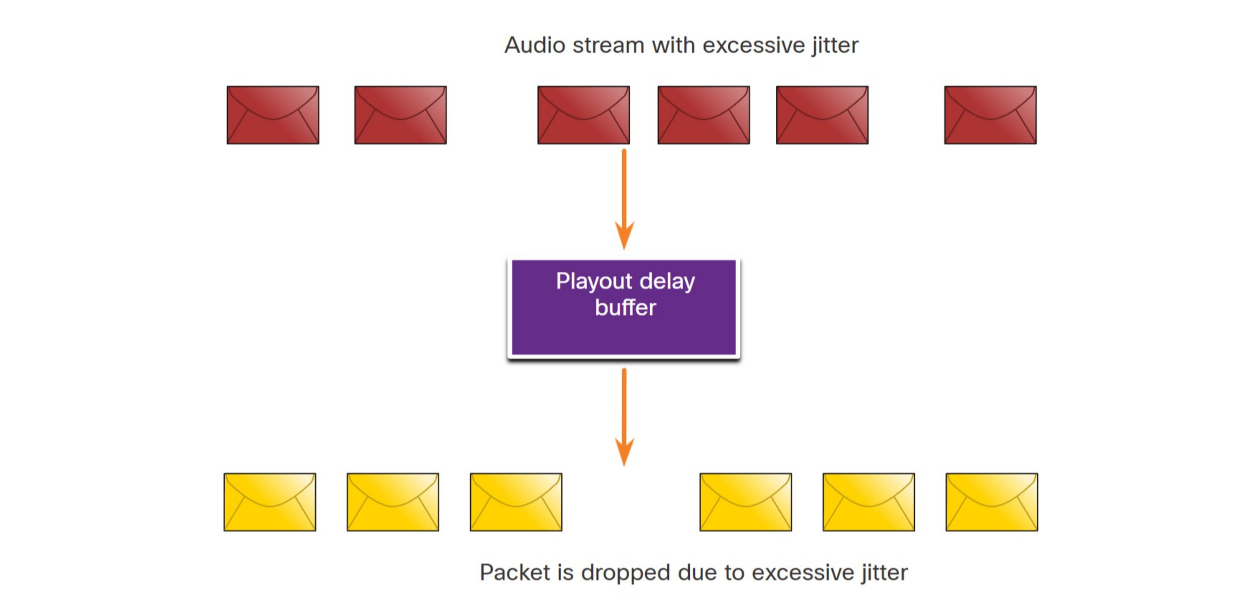
When a router receives a Real-Time Protocol (RTP) digital audio stream for Voice over IP (VoIP), it must compensate for the jitter that is encountered. The mechanism that handles this function is the playout delay buffer. The playout delay buffer must buffer these packets and then play them out in a steady stream, as shown in the figure. The digital packets are later converted back to an analog audio stream.

Playout Delay Buffer Compensates for Jitter



If the jitter is so large that it causes packets to be received out of the range of this buffer, the out-of-range packets are discarded and dropouts are heard in the audio, as shown in the figure.

Packet Dropped Due to Excessive Jitter



For losses as small as one packet, the digital signal processor (DSP) interpolates what it thinks the audio should be and no problem is audible to the user. However, when jitter exceeds what the DSP can do to make up for the missing packets, audio problems are heard.

Packet loss is a very common cause of voice quality problems on an IP network. In a properly designed network, packet loss should be near zero. The voice codecs used by the DSP can tolerate some degree of packet loss without a dramatic effect on voice quality. Network engineers use QoS mechanisms to classify voice packets for zero packet loss. Bandwidth is guaranteed for the voice calls by giving priority to voice traffic over traffic that is not sensitive to delays.

9.1.5

Check Your Understanding - Network Transmission Quality

Check your understanding of network transmission quality by choosing the BEST answer to the following questions.

1. What is the variable amount of time it takes for a frame to traverse the links between the source and destination?
   - [ ] serialization delay
   - [x] propagation delay
   - [ ] code delay

2. What happens when congestion occurs?
   - [x] packet loss
   - [ ] jitter
   - [ ] code delay

3. What is the fixed amount of time it takes to transmit a frame from the NIC to the wire?
   - [x] serialization delay
   - [ ] jitter
   - [ ] code delay

4. What is caused by variation in delay?
   - [ ] congestion
   - [ ] packet loss
   - [x] jitter

Check

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