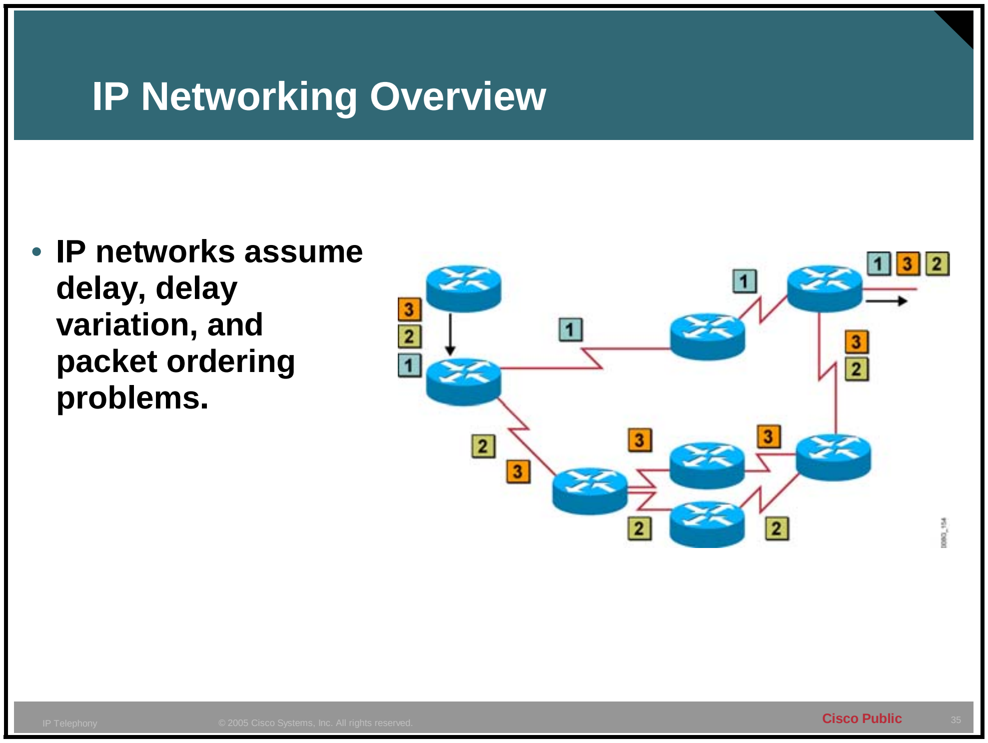
VoIP Challenges

IP Networking Overview

This topic provides an overview of IP networking and some of the inherent challenges when conveying voice over an IP network.



IP is a connectionless network protocol. Connectionless networks generally do not participate in signaling. The concept of session establishment exists between end systems, although the connectionless network remains unaware of the virtual circuit (VC).

IP resides at the network layer of the Open System Interconnection (OSI) protocol stack. Therefore, it can transport IP packets over deterministic and nondeterministic Layer 2 protocols, such as Frame Relay or ATM. IP can be used to communicate across any set of interconnected networks and is equally suited to both LAN and WAN communication.

IP information is transferred in a sequence of datagrams. A message is sent as a series of datagrams that are reassembled into the completed message at the receiving location. Because a voice conversation that is transported in IP can be considered a continuous audio file, all packets must be received in sequence immediately and without interpacket variable delay.

Traditionally, IP traffic transmits on a FIFO basis. Different packet types vary in size, allowing large file transfers to take advantage of the efficiency that is associated with larger packet sizes. FIFO queuing affects the way that voice packets transmit, causing delay and delay variation at the receiving end.

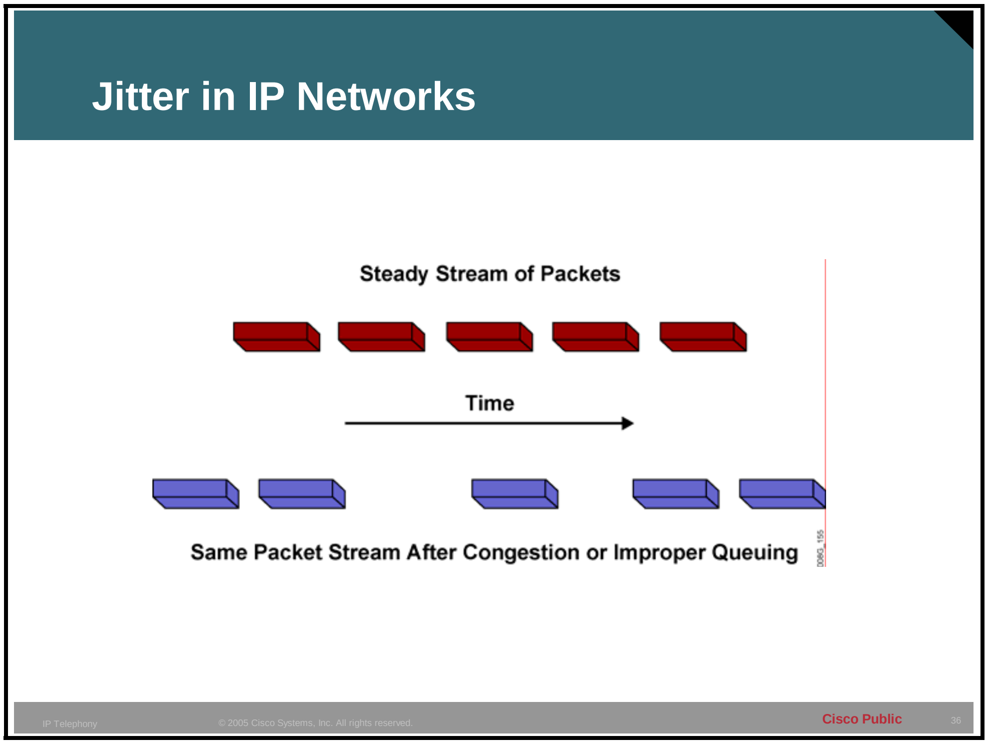
UDP is the connectionless transport layer protocol used for VoIP. UDP is a simple protocol that exchanges datagrams without acknowledgments or guaranteed delivery. UDP requires that other protocols handle error processing and retransmission. The figure shows how packets may be received out of sequence or become completely lost at the receiving end.

Example: IP Networking

Due to the very nature of IP networking, voice packets sent across IP will be subject to certain transmission problems. These problems include jitter, delay, and packet ordering. In the figure, packets sent from the originating router on the left are in sequence and sent with predictable transmission intervals. As they traverse the IP network, the routing protocol may send some of the packets through one path, while other packets traverse a different path. As the packets arrive at the destination router on the right, they arrive with varying delays and out of sequence. These problems must be addressed with QoS mechanisms explained further in this lesson.

Jitter

This topic describes the occurrence of jitter in IP networks and the Cisco Systems solution to this problem.



Jitter is defined as a variation in the delay of received packets. On the sending side, packets are sent in a continuous stream with the packets spaced evenly apart. Because of network congestion, improper queuing, or configuration errors, this steady stream can become lumpy, or the delay between each packet can vary instead of remaining constant, as displayed in the figure.

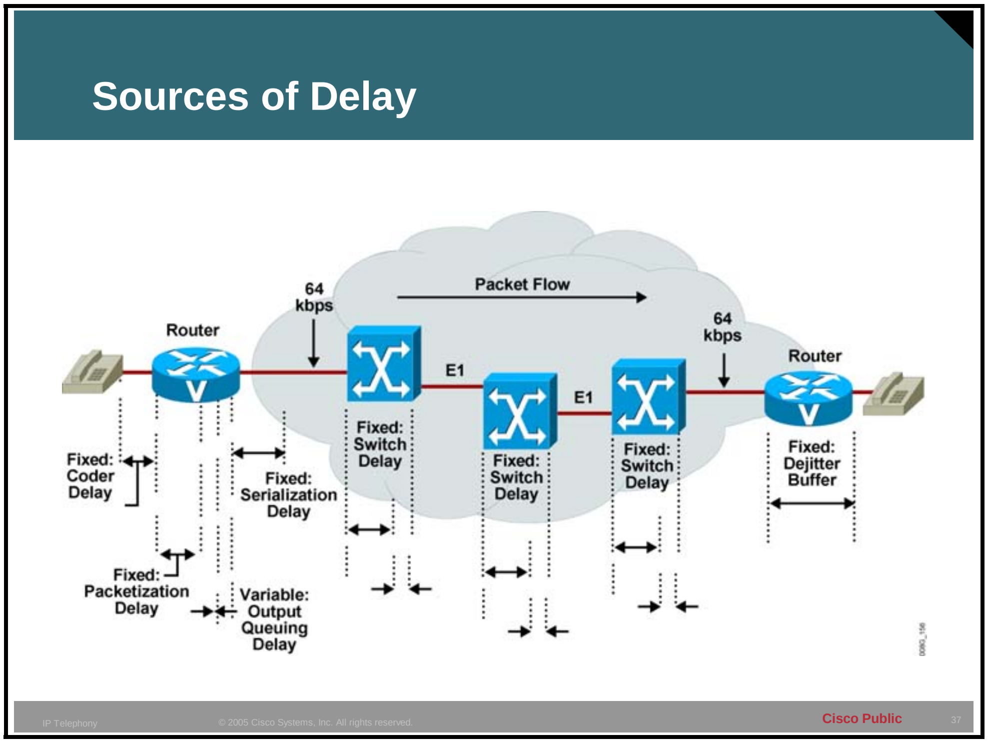
When a router receives an audio stream for VoIP, it must compensate for the jitter that is encountered. The mechanism that handles this function is the playout delay buffer, or dejitter buffer. The playout delay buffer must buffer these packets and then play them out in a steady stream to the digital signal processors (DSPs) to be converted back to an analog audio stream. The playout delay buffer, however, affects overall absolute delay.

Example: Jitter in Voice Networks

When a conversation is subjected to jitter, the results can be clearly heard. If the talker says, “Watson, come here. I want you,” the listener might hear, “Wat….s…on…….come here, I……wa……nt……..y……ou.” The variable arrival of the packets at the receiving end causes the speech to be delayed and garbled.

Delay

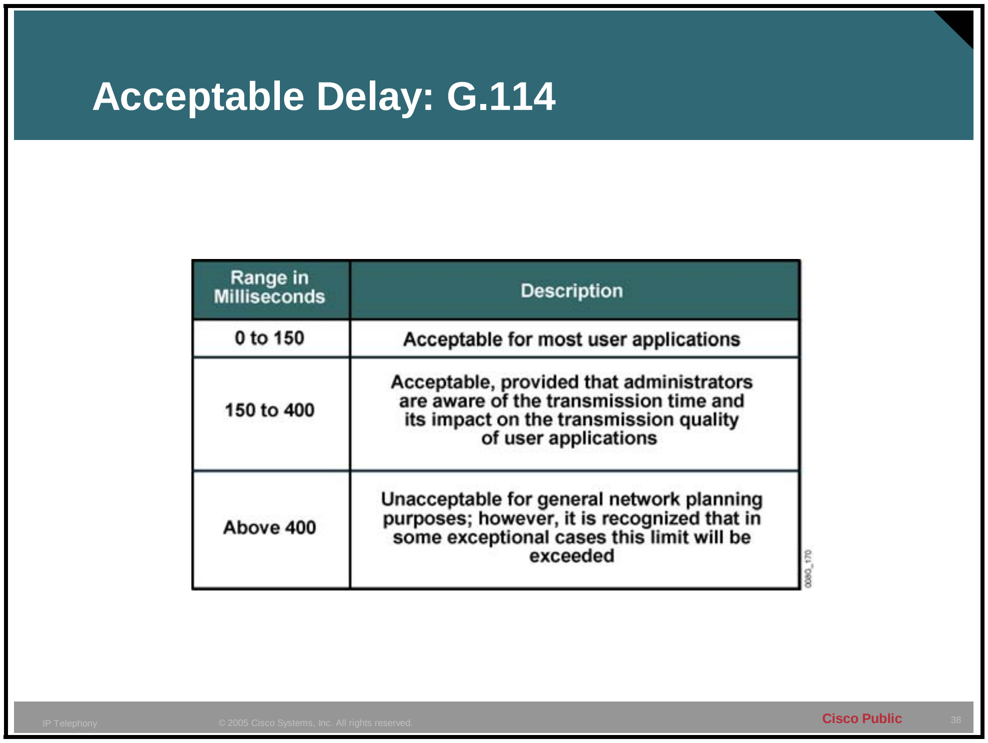
Overall or absolute delay can affect VoIP. You might have experienced delay in a telephone conversation with someone on a different continent. The delays can be very frustrating, causing words in the conversation to be cut off. This topic describes the causes of packet delay and the Cisco solution to this problem.



When you design a network that transports voice over packet, frame, or cell infrastructures, it is important to understand and account for the delay components in the network. You must also correctly account for all potential delays to ensure that overall network performance is acceptable. Overall voice quality is a function of many factors, including the compression algorithm, errors and frame loss, echo cancellation, and delay.

There are two distinct types of delay:

- Fixed-delay components add directly to the overall delay on the connection.
- Variable delays arise from queuing delays in the egress trunk buffers that are located on the serial port that is connected to the WAN. These buffers create variable delays, called jitter, across the network.



Range in Milliseconds	Description
0 to 150	Acceptable for most user applications
150 to 400	Acceptable, provided that administrators are aware of the transmission time and its impact on the transmission quality of user applications
Above 400	Unacceptable for general network planning purposes; however, it is recognized that in some exceptional cases this limit will be exceeded

The ITU considers network delay for voice applications in Recommendation G.114. This recommendation defines three bands of one-way delay, as shown in the table in the figure.

Note This recommendation is for connections with echo that are adequately controlled, implying that echo cancellers are used. Echo cancellers are required when one-way delay exceeds 25 ms (G.131).

This recommendation is oriented toward national telecommunications administrations, and therefore is more stringent than recommendations that would normally be applied in private voice networks. When the location and business needs of end users are well known to a network designer, more delay may prove acceptable. For private networks, a 200-ms delay is a reasonable goal and a 250-ms delay is a limit. This goal is what Cisco Systems proposes as reasonable as long as jitter does not impact voice quality. However, all networks must be engineered so that the maximum expected voice connection delay is known and minimized.

Example: Acceptable Delay

The G.114 recommendation is for one-way delay only and does not account for round-trip delay. Network design engineers must consider all delays, variable and fixed. Variable delays include queuing and network delays, while fixed delays include coder, packetization, serialization, and dejitter buffer delays. The table is an example of calculating delay budget.

Calculating Delay Budget

Delay Type	Fixed (ms)	Variable (ms)
Coder delay	18	
Packetization delay	30	
Queuing and buffering		8
Serialization (64 kbps)	5	
Network delay (public frame)	40	25
Dejitter buffer	45	
Totals	138	33