VOIP QUALITY OF SERVICE AND VOIP SIGNALING PROTOCOLS

A common issue that affects the QoS of packetized audio is jitter. Voice data requires a constant packet interarrival rate at receivers to convert data into a proper analog signal for playback. The variations in the packet interarrival rate lead to jitter, which results in improper signal reconstruction at the receiver. Typically, an unstable sine wave reproduced at the receiver results from the jitter in the signal. Buffering packets can help control the interarrival rate. The buffering scheme can be used to output the data packets at a fixed rate. The buffering scheme works well when the arrival time of the next packet is not very long. Buffering can also introduce a certain amount of delay.

Another issue having a great impact on real-time transmission quality is network latency, or delay, which is a measure of the time required for a data packet to travel from a sender to a receiver. For telephone networks, a round-trip delay that is too large can result in an echo in the earpiece. Delay can be controlled in networks by assigning a higher priority for voice packets. In such cases, routers and intermediate switches in the network transport these high-priority packets before processing lower-priority data packets.

Congestion in networks can be a major disruption for IP telephony. Congestion can be controlled to a certain extent by implementing weighted random early discard, whereby routers begin to intelligently discard lower-priority packets before congestion occurs. The drop in packets results in a subsequent decrease in the window size in TCP, which relieves congestion to a certain extent.

A VoIP connection has several QoS factors:

- Packet loss is accepted to a certain extent.
- Packet delay is normally unacceptable.
- Jitter, as the variation in packet arrival time, is not acceptable after a certain limit.

Packet loss is a direct result of the queueing scheme used in routers. VoIP can use priority queueing, weighted fair queuing, or class-based weighted fair queuing, whereby traffic amounts are also assigned to classes of data traffic. Besides these well-known queueing schemes, voice traffic can be handled by a custom queuing, in which a certain amount of channel bandwidth for voice traffic is reserved.

Although the packet loss is tolerable to a certain extent, packet delay may not be tolerable in most cases. The variability in the time of arrival of packets in packet-switched networks gives rise to jitter variations. This and other QoS issues have to be handled differently than in conventional packet-switched networks. QoS must also consider connectivity of packet-voice environment when it is combined with traditional telephone networks.

VoIP Signaling Protocols

The IP telephone system must be able to handle signalings for call setup, conversion of phone number to IP address mapping, and proper call termination. Signaling is required for call setup, call management, and call termination. In the standard telephone network, signaling involves identifying the user's location given a phone number, finding a route between a calling and a called party, and handling the issue of call forwarding and other call features.

IP telephone systems can use either a distributed or a centralized signaling scheme. The distributed approach enables two IP telephones to communicate using a client/ server model, as most Internet applications do. The distributed approach works well with VoIP networks within a single company. The centralized approach uses the conventional model and can provide some level of guarantee. Three well-known signaling protocols are

1. Session Initiation Protocol (SIP)
2. H.323 protocols
3. Media Gateway Control Protocol (MGCP)

Figure 7.2 shows the placement of VoIP in the five-layer TCP/IP model. SIP, H.323, and MGCP run over TCP or UDP; real-time data transmission protocols, such as RTP, typically run over UDP. Real-time transmissions of audio and video traffic are implemented over UDP, since the real-time data requires lower delay and less overhead. Our focus in this chapter is on the two signaling protocols SIP and H.323 and on RTP and RTCP.

Figure 7.2. Main protocols for VoIP and corresponding layers of operation

Layer	Protocol							
	SIP		H.323					
	Other Signals	Media Transport	Registration	Media Transport: Voice Codec	Media Transport: Video Codec	Security	Signaling	Data
5			H.225.0-RAS	G.711 H.263 G.722 G.723 G.728	H.261 H.323	H.235	H.225.9-Q.931 H.250 H.245 H.250	T.120
		RTP, RTCP		RTP, RTCP				
4	UDP						TCP	
3	IP, RSVP, and IGMP							

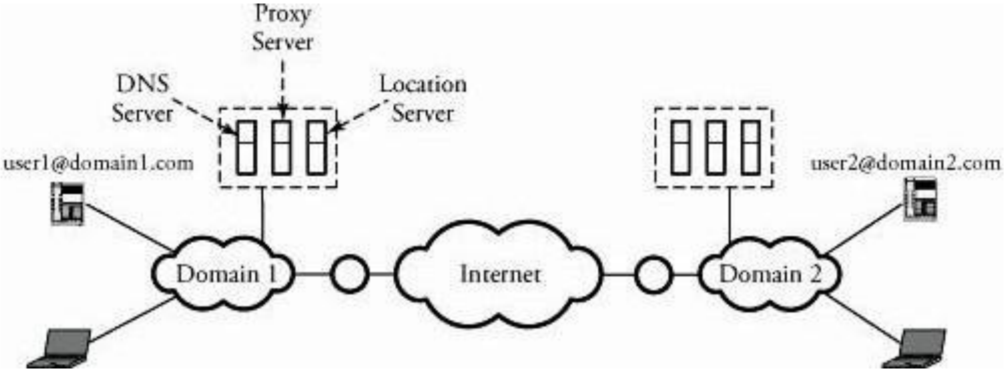
7.2.1. Session Initiation Protocol (SIP)

The Session Initiation Protocol (SIP) is one of the most important VoIP signaling protocols operating in the application layer in the five-layer TCP/IP model. SIP can perform both unicast and multicast sessions and supports user mobility. SIP handles signals and identifies user location, call setup, call termination, and busy signals. SIP can use multicast to support conference calls and uses the Session Description Protocol (SDP) to negotiate parameters.

SIP Components

Figure 7.3 shows an overview of SIP. A call is initiated from a user agent: the user's IP telephone system, which is similar to a conventional phone. A user agent assists in initiating or terminating a phone call in VoIP networks. A user agent can be implemented in a standard telephone or in a laptop with a microphone that runs some software. A user agent is identified using its associated domain. For example, user1@domain1.com refers to user 1, who is associated with the domain1.com network.

Figure 7.3. Overview of SIP



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