CHAPTER 9

Multimedia Networking

While lounging in bed or riding buses and subways, people in all corners of the world are currently using the Internet to watch movies and television shows on demand. Internet movie and television distribution companies such as Netflix and Amazon in North America and Youku and Kankan in China have practically become household names. But people are not only watching Internet videos, they are using sites like YouTube to upload and distribute their own user-generated content, becoming Internet video producers as well as consumers. Moreover, network applications such as Skype, Google Talk, and WeChat (enormously popular in China) allow people to not only make "telephone calls" over the Internet, but to also enhance those calls with video and multi-person conferencing. In fact, we predict that by the end of the current decade most of the video consumption and voice conversations will take place end-to-end over the Internet, more typically to wireless devices connected to the Internet via cellular and WiFi access networks. Traditional telephony and broadcast television are quickly becoming obsolete.

We begin this chapter with a taxonomy of multimedia applications in Section 9.1. We'll see that a multimedia application can be classified as either *streaming stored audio/video*, *conversational voice/video-over-IP*, or *streaming live audio/video*. We'll see that each of these classes of applications has its own unique service requirements that differ significantly from those of traditional elastic applications such as e-mail, Web browsing, and remote login. In Section 9.2, we'll examine video streaming in some detail. We'll explore many of the underlying principles behind

video streaming, including client buffering, prefetching, and adapting video quality to available bandwidth. In Section 9.3, we investigate conversational voice and video, which, unlike elastic applications, are highly sensitive to end-to-end delay but can tolerate occasional loss of data. Here we'll examine how techniques such as adaptive playout, forward error correction, and error concealment can mitigate against network-induced packet loss and delay. We'll also examine Skype as a case study. In Section 9.4, we'll study RTP and SIP, two popular protocols for real-time conversational voice and video applications. In Section 9.5, we'll investigate mechanisms within the network that can be used to distinguish one class of traffic (e.g., delay-sensitive applications such as conversational voice) from another (e.g., elastic applications such as browsing Web pages), and provide differentiated service among multiple classes of traffic.

9.1 Multimedia Networking Applications

We define a multimedia network application as any network application that employs audio or video. In this section, we provide a taxonomy of multimedia applications. We'll see that each class of applications in the taxonomy has its own unique set of service requirements and design issues. But before diving into an in-depth discussion of Internet multimedia applications, it is useful to consider the intrinsic characteristics of the audio and video media themselves.

9.1.1 Properties of Video

Perhaps the most salient characteristic of video is its **high bit rate**. Video distributed over the Internet typically ranges from 100 kbps for low-quality video conferencing to over 3 Mbps for streaming high-definition movies. To get a sense of how video bandwidth demands compare with those of other Internet applications, let's briefly consider three different users, each using a different Internet application. Our first user, Frank, is going quickly through photos posted on his friends' Facebook pages. Let's assume that Frank is looking at a new photo every 10 seconds, and that photos are on average 200 Kbytes in size. (As usual, throughout this discussion we make the simplifying assumption that 1 Kbyte = 8,000 bits.) Our second user, Martha, is streaming music from the Internet ("the cloud") to her smartphone. Let's assume Martha is using a service such as Spotify to listen to many MP3 songs, one after the other, each encoded at a rate of 128 kbps. Our third user, Victor, is watching a video that has been encoded at 2 Mbps. Finally, let's suppose that the session length for all three users is 4,000 seconds (approximately 67 minutes). Table 9.1 compares the bit rates and the total bytes transferred for these three users. We see that video streaming consumes by far the most bandwidth, having a bit rate of more than ten times greater than that of the Facebook and music-streaming applications. Therefore, when design-

	Bit rate	Bytes transferred in 67 min
Facebook Frank	160 kbps	80 Mbytes
Martha Music	128 kbps	64 Mbytes
Victor Video	2 Mbps	1 Gbyte

Table 9.1 ♦ Comparison of bit-rate requirements of three Internet applications

ing networked video applications, the first thing we must keep in mind is the high bit-rate requirements of video. Given the popularity of video and its high bit rate, it is perhaps not surprising that Cisco predicts [Cisco 2015] that streaming and stored video will be approximately 80 percent of global consumer Internet traffic by 2019.

Another important characteristic of video is that it can be compressed, thereby trading off video quality with bit rate. A video is a sequence of images, typically being displayed at a constant rate, for example, at 24 or 30 images per second. An uncompressed, digitally encoded image consists of an array of pixels, with each pixel encoded into a number of bits to represent luminance and color. There are two types of redundancy in video, both of which can be exploited by **video compression**. *Spatial redundancy* is the redundancy within a given image. Intuitively, an image that consists of mostly white space has a high degree of redundancy and can be efficiently compressed without significantly sacrificing image quality. *Temporal redundancy* reflects repetition from image to subsequent image. If, for example, an image and the subsequent image are exactly the same, there is no reason to re-encode the subsequent image; it is instead more efficient simply to indicate during encoding that the subsequent image is exactly the same. Today's off-the-shelf compression algorithms can compress a video to essentially any bit rate desired. Of course, the higher the bit rate, the better the image quality and the better the overall user viewing experience.

We can also use compression to create **multiple versions** of the same video, each at a different quality level. For example, we can use compression to create, say, three versions of the same video, at rates of 300 kbps, 1 Mbps, and 3 Mbps. Users can then decide which version they want to watch as a function of their current available bandwidth. Users with high-speed Internet connections might choose the 3 Mbps version; users watching the video over 3G with a smartphone might choose the 300 kbps version. Similarly, the video in a video conference application can be compressed "on-the-fly" to provide the best video quality given the available end-to-end bandwidth between conversing users.

9.1.2 Properties of Audio

Digital audio (including digitized speech and music) has significantly lower bandwidth requirements than video. Digital audio, however, has its own unique properties that must be considered when designing multimedia network applications.

To understand these properties, let's first consider how analog audio (which humans and musical instruments generate) is converted to a digital signal:

- The analog audio signal is sampled at some fixed rate, for example, at 8,000 samples per second. The value of each sample will be some real number.
- Each of the samples is then rounded to one of a finite number of values. This operation is referred to as **quantization**. The number of such finite values—called quantization values—is typically a power of two, for example, 256 quantization values.
- Each of the quantization values is represented by a fixed number of bits. For example, if there are 256 quantization values, then each value—and hence each audio sample—is represented by one byte. The bit representations of all the samples are then concatenated together to form the digital representation of the signal. As an example, if an analog audio signal is sampled at 8,000 samples per second and each sample is quantized and represented by 8 bits, then the resulting digital signal will have a rate of 64,000 bits per second. For playback through audio speakers, the digital signal can then be converted back—that is, decoded—to an analog signal. However, the decoded analog signal is only an approximation of the original signal, and the sound quality may be noticeably degraded (for example, high-frequency sounds may be missing in the decoded signal). By increasing the sampling rate and the number of quantization values, the decoded signal can better approximate the original analog signal. Thus (as with video), there is a trade-off between the quality of the decoded signal and the bit-rate and storage requirements of the digital signal.

The basic encoding technique that we just described is called **pulse code modulation (PCM)**. Speech encoding often uses PCM, with a sampling rate of 8,000 samples per second and 8 bits per sample, resulting in a rate of 64 kbps. The audio compact disk (CD) also uses PCM, with a sampling rate of 44,100 samples per second with 16 bits per sample; this gives a rate of 705.6 kbps for mono and 1.411 Mbps for stereo.

PCM-encoded speech and music, however, are rarely used in the Internet. Instead, as with video, compression techniques are used to reduce the bit rates of the stream. Human speech can be compressed to less than 10 kbps and still be intelligible. A popular compression technique for near CD-quality stereo music is **MPEG 1 layer 3**, more commonly known as **MP3**. MP3 encoders can compress to many different rates; 128 kbps is the most common encoding rate and produces very little sound degradation. A related standard is **Advanced Audio Coding (AAC)**, which has been popularized by Apple. As with video, multiple versions of a prerecorded audio stream can be created, each at a different bit rate.

Although audio bit rates are generally much less than those of video, users are generally much more sensitive to audio glitches than video glitches. Consider, for example, a video conference taking place over the Internet. If, from time to time, the video signal is lost for a few seconds, the video conference can likely proceed

without too much user frustration. If, however, the audio signal is frequently lost, the users may have to terminate the session.

9.1.3 Types of Multimedia Network Applications

The Internet supports a large variety of useful and entertaining multimedia applications. In this subsection, we classify multimedia applications into three broad categories: *(i) streaming stored audio/video*, *(ii) conversational voice/video-over-IP*, and *(iii) streaming live audio/video*. As we will soon see, each of these application categories has its own set of service requirements and design issues.

Streaming Stored Audio and Video

To keep the discussion concrete, we focus here on streaming stored video, which typically combines video and audio components. Streaming stored audio (such as Spotify's streaming music service) is very similar to streaming stored video, although the bit rates are typically much lower.

In this class of applications, the underlying medium is prerecorded video, such as a movie, a television show, a prerecorded sporting event, or a prerecorded user-generated video (such as those commonly seen on YouTube). These prerecorded videos are placed on servers, and users send requests to the servers to view the videos *on demand*. Many Internet companies today provide streaming video, including YouTube (Google), Netflix, Amazon, and Hulu. Streaming stored video has three key distinguishing features.

- *Streaming.* In a streaming stored video application, the client typically begins video playout within a few seconds after it begins receiving the video from the server. This means that the client will be playing out from one location in the video while at the same time receiving later parts of the video from the server. This technique, known as **streaming**, avoids having to download the entire video file (and incurring a potentially long delay) before playout begins.
- *Interactivity.* Because the media is prerecorded, the user may pause, reposition forward, reposition backward, fast-forward, and so on through the video content. The time from when the user makes such a request until the action manifests itself at the client should be less than a few seconds for acceptable responsiveness.
- *Continuous playout.* Once playout of the video begins, it should proceed according to the original timing of the recording. Therefore, data must be received from the server in time for its playout at the client; otherwise, users experience video frame freezing (when the client waits for the delayed frames) or frame skipping (when the client skips over delayed frames).

By far, the most important performance measure for streaming video is average throughput. In order to provide continuous playout, the network must provide an

average throughput to the streaming application that is at least as large the bit rate of the video itself. As we will see in Section 9.2, by using buffering and prefetching, it is possible to provide continuous playout even when the throughput fluctuates, as long as the average throughput (averaged over 5–10 seconds) remains above the video rate [Wang 2008].

For many streaming video applications, prerecorded video is stored on, and streamed from, a CDN rather than from a single data center. There are also many P2P video streaming applications for which the video is stored on users' hosts (peers), with different chunks of video arriving from different peers that may spread around the globe. Given the prominence of Internet video streaming, we will explore video streaming in some depth in Section 9.2, paying particular attention to client buffering, prefetching, adapting quality to bandwidth availability, and CDN distribution.

Conversational Voice- and Video-over-IP

Real-time conversational voice over the Internet is often referred to as **Internet telephony**, since, from the user's perspective, it is similar to the traditional circuit-switched telephone service. It is also commonly called **Voice-over-IP (VoIP)**. Conversational video is similar, except that it includes the video of the participants as well as their voices. Most of today's voice and video conversational systems allow users to create conferences with three or more participants. Conversational voice and video are widely used in the Internet today, with the Internet companies Skype, QQ, and Google Talk boasting hundreds of millions of daily users.

In our discussion of application service requirements in Chapter 2 (Figure 2.4), we identified a number of axes along which application requirements can be classified. Two of these axes—timing considerations and tolerance of data loss—are particularly important for conversational voice and video applications. Timing considerations are important because audio and video conversational applications are highly **delay-sensitive**. For a conversation with two or more interacting speakers, the delay from when a user speaks or moves until the action is manifested at the other end should be less than a few hundred milliseconds. For voice, delays smaller than 150 milliseconds are not perceived by a human listener, delays between 150 and 400 milliseconds can be acceptable, and delays exceeding 400 milliseconds can result in frustrating, if not completely unintelligible, voice conversations.

On the other hand, conversational multimedia applications are **loss-tolerant**—occasional loss only causes occasional glitches in audio/video playback, and these losses can often be partially or fully concealed. These delay-sensitive but loss-tolerant characteristics are clearly different from those of elastic data applications such as Web browsing, e-mail, social networks, and remote login. For elastic applications, long delays are annoying but not particularly harmful; the completeness and integrity of the transferred data, however, are of paramount importance. We will explore conversational voice and video in more depth in Section 9.3, paying particular attention

to how adaptive playout, forward error correction, and error concealment can mitigate against network-induced packet loss and delay.

Streaming Live Audio and Video

This third class of applications is similar to traditional broadcast radio and television, except that transmission takes place over the Internet. These applications allow a user to receive a *live* radio or television transmission—such as a live sporting event or an ongoing news event—transmitted from any corner of the world. Today, thousands of radio and television stations around the world are broadcasting content over the Internet.

Live, broadcast-like applications often have many users who receive the same audio/video program at the same time. In the Internet today, this is typically done with CDNs (Section 2.6). As with streaming stored multimedia, the network must provide each live multimedia flow with an average throughput that is larger than the video consumption rate. Because the event is live, delay can also be an issue, although the timing constraints are much less stringent than those for conversational voice. Delays of up to ten seconds or so from when the user chooses to view a live transmission to when playout begins can be tolerated. We will not cover streaming live media in this book because many of the techniques used for streaming live media—initial buffering delay, adaptive bandwidth use, and CDN distribution—are similar to those for streaming stored media.

9.2 Streaming Stored Video

For streaming video applications, prerecorded videos are placed on servers, and users send requests to these servers to view the videos on demand. The user may watch the video from beginning to end without interruption, may stop watching the video well before it ends, or interact with the video by pausing or repositioning to a future or past scene. Streaming video systems can be classified into three categories: **UDP streaming**, **HTTP streaming**, and **adaptive HTTP streaming** (see Section 2.6). Although all three types of systems are used in practice, the majority of today's systems employ HTTP streaming and adaptive HTTP streaming.

A common characteristic of all three forms of video streaming is the extensive use of client-side application buffering to mitigate the effects of varying end-to-end delays and varying amounts of available bandwidth between server and client. For streaming video (both stored and live), users generally can tolerate a small several-second initial delay between when the client requests a video and when video playout begins at the client. Consequently, when the video starts to arrive at the client, the client need not immediately begin playout, but can instead build up a reserve of video in an application buffer. Once the client has built up a reserve of several seconds of

buffered-but-not-yet-played video, the client can then begin video playout. There are two important advantages provided by such **client buffering**. First, client-side buffering can absorb variations in server-to-client delay. If a particular piece of video data is delayed, as long as it arrives before the reserve of received-but-not-yet-played video is exhausted, this long delay will not be noticed. Second, if the server-to-client bandwidth briefly drops below the video consumption rate, a user can continue to enjoy continuous playback, again as long as the client application buffer does not become completely drained.

Figure 9.1 illustrates client-side buffering. In this simple example, suppose that video is encoded at a fixed bit rate, and thus each video block contains video frames that are to be played out over the same fixed amount of time, Δ. The server transmits the first video block at t_0, the second block at $t_0 + \Delta$, the third block at $t_0 + 2\Delta$, and so on. Once the client begins playout, each block should be played out Δ time units after the previous block in order to reproduce the timing of the original recorded video. Because of the variable end-to-end network delays, different video blocks experience different delays. The first video block arrives at the client at t_1 and the second block arrives at t_2. The network delay for the ith block is the horizontal distance between the time the block was transmitted by the server and the time it is received at the client; note that the network delay varies from one video block to another. In this example, if the client were to begin playout as soon as the first block arrived at t_1, then the second block would not have arrived in time to be played out at out at $t_1 + \Delta$. In this case, video playout would either have to stall (waiting for block 2 to arrive) or block 2 could be skipped—both resulting in undesirable playout impairments. Instead, if the client were to delay the start of playout until t_3, when blocks 1 through 6 have all arrived, periodic playout can proceed with *all* blocks having been received before their playout time.

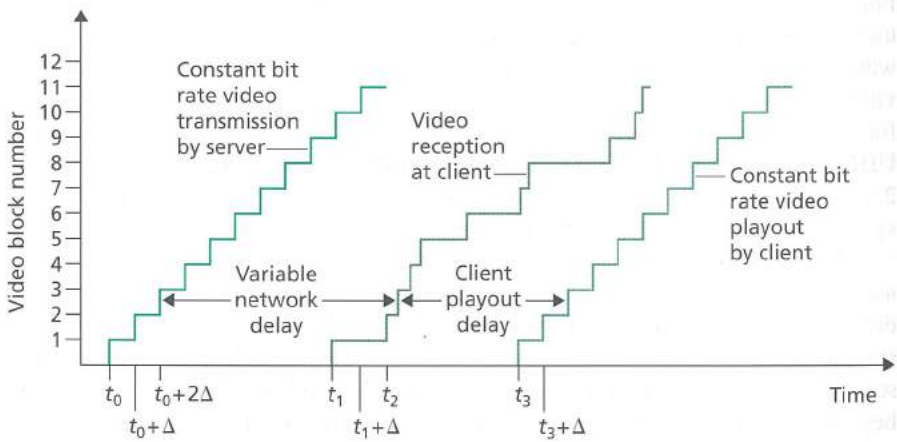

Figure 9.1 ♦ Client playout delay in video streaming

9.2.1 UDP Streaming

We only briefly discuss UDP streaming here, referring the reader to more in-depth discussions of the protocols behind these systems where appropriate. With UDP streaming, the server transmits video at a rate that matches the client's video consumption rate by clocking out the video chunks over UDP at a steady rate. For example, if the video consumption rate is 2 Mbps and each UDP packet carries 8,000 bits of video, then the server would transmit one UDP packet into its socket every (8000 bits)/(2 Mbps) = 4 msec. As we learned in Chapter 3, because UDP does not employ a congestion-control mechanism, the server can push packets into the network at the consumption rate of the video without the rate-control restrictions of TCP. UDP streaming typically uses a small client-side buffer, big enough to hold less than a second of video.

Before passing the video chunks to UDP, the server will encapsulate the video chunks within transport packets specially designed for transporting audio and video, using the Real-Time Transport Protocol (RTP) [RFC 3550] or a similar (possibly proprietary) scheme. We delay our coverage of RTP until Section 9.3, where we discuss RTP in the context of conversational voice and video systems.

Another distinguishing property of UDP streaming is that in addition to the server-to-client video stream, the client and server also maintain, in parallel, a separate control connection over which the client sends commands regarding session state changes (such as pause, resume, reposition, and so on). The Real-Time Streaming Protocol (RTSP) [RFC 2326], explained in some detail in the Web site for this textbook, is a popular open protocol for such a control connection.

Although UDP streaming has been employed in many open-source systems and proprietary products, it suffers from three significant drawbacks. First, due to the unpredictable and varying amount of available bandwidth between server and client, constant-rate UDP streaming can fail to provide continuous playout. For example, consider the scenario where the video consumption rate is 1 Mbps and the server-to-client available bandwidth is usually more than 1 Mbps, but every few minutes the available bandwidth drops below 1 Mbps for several seconds. In such a scenario, a UDP streaming system that transmits video at a constant rate of 1 Mbps over RTP/UDP would likely provide a poor user experience, with freezing or skipped frames soon after the available bandwidth falls below 1 Mbps. The second drawback of UDP streaming is that it requires a media control server, such as an RTSP server, to process client-to-server interactivity requests and to track client state (e.g., the client's playout point in the video, whether the video is being paused or played, and so on) for *each* ongoing client session. This increases the overall cost and complexity of deploying a large-scale video-on-demand system. The third drawback is that many firewalls are configured to block UDP traffic, preventing the users behind these firewalls from receiving UDP video.

9.2.2 HTTP Streaming

In HTTP streaming, the video is simply stored in an HTTP server as an ordinary file with a specific URL. When a user wants to see the video, the client establishes a TCP connection with the server and issues an HTTP GET request for that URL. The server then sends the video file, within an HTTP response message, as quickly as possible, that is, as quickly as TCP congestion control and flow control will allow. On the client side, the bytes are collected in a client application buffer. Once the number of bytes in this buffer exceeds a predetermined threshold, the client application begins playback—specifically, it periodically grabs video frames from the client application buffer, decompresses the frames, and displays them on the user's screen.

We learned in Chapter 3 that when transferring a file over TCP, the server-to-client transmission rate can vary significantly due to TCP's congestion control mechanism. In particular, it is not uncommon for the transmission rate to vary in a "saw-tooth" manner associated with TCP congestion control. Furthermore, packets can also be significantly delayed due to TCP's retransmission mechanism. Because of these characteristics of TCP, the conventional wisdom in the 1990s was that video streaming would never work well over TCP. Over time, however, designers of streaming video systems learned that TCP's congestion control and reliable-data transfer mechanisms do not necessarily preclude continuous playout when client buffering and prefetching (discussed in the next section) are used.

The use of HTTP over TCP also allows the video to traverse firewalls and NATs more easily (which are often configured to block most UDP traffic but to allow most HTTP traffic). Streaming over HTTP also obviates the need for a media control server, such as an RTSP server, reducing the cost of a large-scale deployment over the Internet. Due to all of these advantages, most video streaming applications today—including YouTube and Netflix—use HTTP streaming (over TCP) as its underlying streaming protocol.

Prefetching Video

As we just learned, client-side buffering can be used to mitigate the effects of varying end-to-end delays and varying available bandwidth. In our earlier example in Figure 9.1, the server transmits video at the rate at which the video is to be played out. However, for streaming *stored* video, the client can attempt to download the video at a rate *higher* than the consumption rate, thereby **prefetching** video frames that are to be consumed in the future. This prefetched video is naturally stored in the client application buffer. Such prefetching occurs naturally with TCP streaming, since TCP's congestion avoidance mechanism will attempt to use all of the available bandwidth between server and client.

To gain some insight into prefetching, let's take a look at a simple example. Suppose the video consumption rate is 1 Mbps but the network is capable of delivering the video from server to client at a constant rate of 1.5 Mbps. Then the client will

not only be able to play out the video with a very small playout delay, but will also be able to increase the amount of buffered video data by 500 Kbits every second. In this manner, if in the future the client receives data at a rate of less than 1 Mbps for a brief period of time, the client will be able to continue to provide continuous playback due to the reserve in its buffer. [Wang 2008] shows that when the average TCP throughput is roughly twice the media bit rate, streaming over TCP results in minimal starvation and low buffering delays.

Client Application Buffer and TCP Buffers

Figure 9.2 illustrates the interaction between client and server for HTTP streaming. At the server side, the portion of the video file in white has already been sent into the server's socket, while the darkened portion is what remains to be sent. After "passing through the socket door," the bytes are placed in the TCP send buffer before being transmitted into the Internet, as described in Chapter 3. In Figure 9.2, because the TCP send buffer at the server side is shown to be full, the server is momentarily prevented from sending more bytes from the video file into the socket. On the client side, the client application (media player) reads bytes from the TCP receive buffer (through its client socket) and places the bytes into the client application buffer. At the same time, the client application periodically grabs video frames from the client application buffer, decompresses the frames, and displays them on the user's screen. Note that if the client application buffer is larger than the video file, then the whole process of moving bytes from the server's storage to the client's application buffer is equivalent to an ordinary file download over HTTP—the client simply pulls the video off the server as fast as TCP will allow!

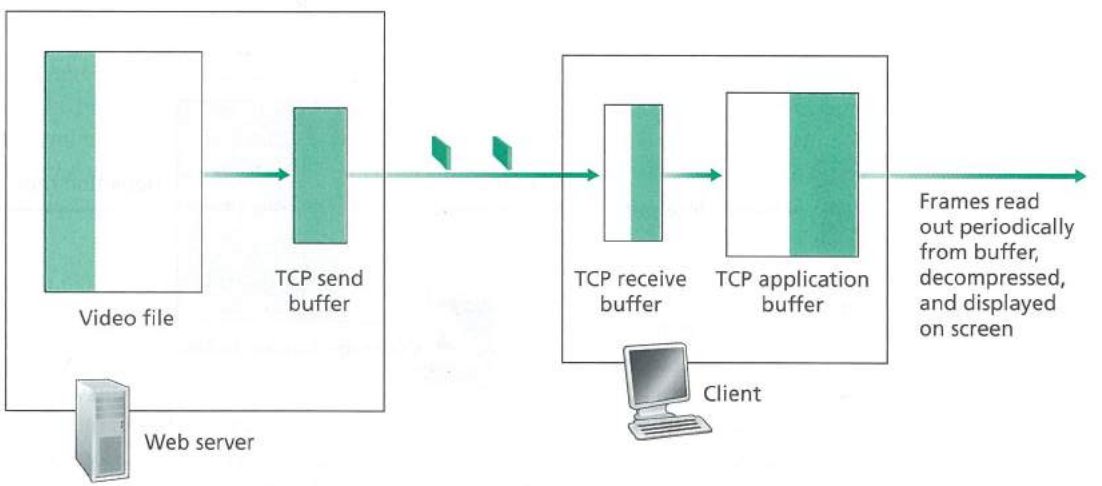

Figure 9.2 ♦ Streaming stored video over HTTP/TCP

Consider now what happens when the user pauses the video during the streaming process. During the pause period, bits are not removed from the client application buffer, even though bits continue to enter the buffer from the server. If the client application buffer is finite, it may eventually become full, which will cause "back pressure" all the way back to the server. Specifically, once the client application buffer becomes full, bytes can no longer be removed from the client TCP receive buffer, so it too becomes full. Once the client receive TCP buffer becomes full, bytes can no longer be removed from the server TCP send buffer, so it also becomes full. Once the TCP becomes full, the server cannot send any more bytes into the socket. Thus, if the user pauses the video, the server may be forced to stop transmitting, in which case the server will be blocked until the user resumes the video.

In fact, even during regular playback (that is, without pausing), if the client application buffer becomes full, back pressure will cause the TCP buffers to become full, which will force the server to reduce its rate. To determine the resulting rate, note that when the client application removes f bits, it creates room for f bits in the client application buffer, which in turn allows the server to send f additional bits. Thus, the server send rate can be no higher than the video consumption rate at the client. Therefore, *a full client application buffer indirectly imposes a limit on the rate that video can be sent from server to client when streaming over HTTP.*

Analysis of Video Streaming

Some simple modeling will provide more insight into initial playout delay and freezing due to application buffer depletion. As shown in Figure 9.3, let B denote the size

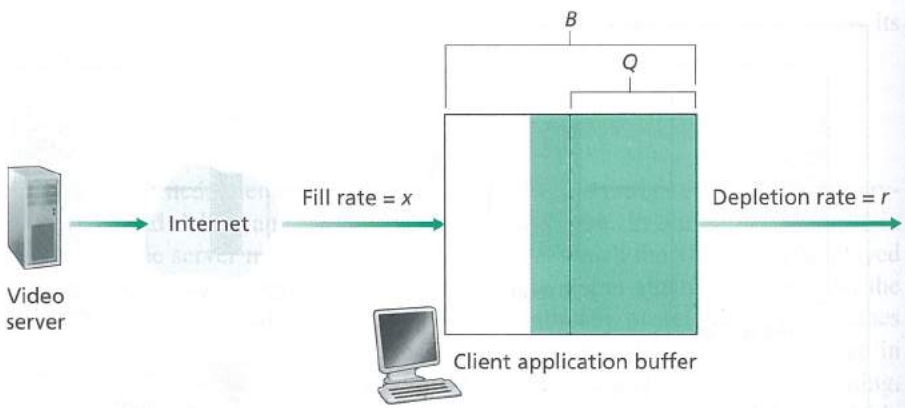

Figure 9.3 ♦ Analysis of client-side buffering for video streaming

(in bits) of the client's application buffer, and let Q denote the number of bits that must be buffered before the client application begins playout. (Of course, $Q < B$.) Let r denote the video consumption rate—the rate at which the client draws bits out of the client application buffer during playback. So, for example, if the video's frame rate is 30 frames/sec, and each (compressed) frame is 100,000 bits, then $r = 3$ Mbps. To see the forest through the trees, we'll ignore TCP's send and receive buffers.

Let's assume that the server sends bits at a constant rate x whenever the client buffer is not full. (This is a gross simplification, since TCP's send rate varies due to congestion control; we'll examine more realistic time-dependent rates $x(t)$ in the problems at the end of this chapter.) Suppose at time $t = 0$, the application buffer is empty and video begins arriving to the client application buffer. We now ask at what time $t = t_p$ does playout begin? And while we are at it, at what time $t = t_f$ does the client application buffer become full?

First, let's determine t_p, the time when Q bits have entered the application buffer and playout begins. Recall that bits arrive to the client application buffer at rate x and *no* bits are removed from this buffer before playout begins. Thus, the amount of time required to build up Q bits (the initial buffering delay) is $t_p = Q/x$.

Now let's determine t_f, the point in time when the client application buffer becomes full. We first observe that if $x < r$ (that is, if the server send rate is less than the video consumption rate), then the client buffer will never become full! Indeed, starting at time t_p, the buffer will be depleted at rate r and will only be filled at rate $x < r$. Eventually the client buffer will empty out entirely, at which time the video will freeze on the screen while the client buffer waits another t_p seconds to build up Q bits of video. *Thus, when the available rate in the network is less than the video rate, playout will alternate between periods of continuous playout and periods of freezing.* In a homework problem, you will be asked to determine the length of each continuous playout and freezing period as a function of Q, r, and x. Now let's determine t_f for when $x > r$. In this case, starting at time t_p, the buffer increases from Q to B at rate $x - r$ since bits are being depleted at rate r but are arriving at rate x, as shown in Figure 9.3. Given these hints, you will be asked in a homework problem to determine t_f, the time the client buffer becomes full. Note that *when the available rate in the network is more than the video rate, after the initial buffering delay, the user will enjoy continuous playout until the video ends.*

Early Termination and Repositioning the Video

HTTP streaming systems often make use of the **HTTP byte-range header** in the HTTP GET request message, which specifies the specific range of bytes the client currently wants to retrieve from the desired video. This is particularly useful when the user wants to reposition (that is, jump) to a future point in time in the video. When the user repositions to a new position, the client sends a new HTTP request, indicating with the byte-range header from which byte in the file should the server send data. When the server receives the new HTTP request, it can forget about any earlier request and instead send bytes beginning with the byte indicated in the byte-range request.

While we are on the subject of repositioning, we briefly mention that when a user repositions to a future point in the video or terminates the video early, some prefetched-but-not-yet-viewed data transmitted by the server will go unwatched—a waste of network bandwidth and server resources. For example, suppose that the client buffer is full with B bits at some time t_0 into the video, and at this time the user repositions to some instant $t > t_0 + B/r$ into the video, and then watches the video to completion from that point on. In this case, all B bits in the buffer will be unwatched and the bandwidth and server resources that were used to transmit those B bits have been completely wasted. There is significant wasted bandwidth in the Internet due to early termination, which can be quite costly, particularly for wireless links [Ihm 2011]. For this reason, many streaming systems use only a moderate-size client application buffer, or will limit the amount of prefetched video using the byte-range header in HTTP requests [Rao 2011].

Repositioning and early termination are analogous to cooking a large meal, eating only a portion of it, and throwing the rest away, thereby wasting food. So the next time your parents criticize you for wasting food by not eating all your dinner, you can quickly retort by saying they are wasting bandwidth and server resources when they reposition while watching movies over the Internet! But, of course, two wrongs do not make a right—both food and bandwidth are not to be wasted!

In Sections 9.2.1 and 9.2.2, we covered UDP streaming and HTTP streaming, respectively. A third type of streaming is Dynamic Adaptive Streaming over HTTP (DASH), which uses multiple versions of the video, each compressed at a different rate. DASH is discussed in detail in Section 2.6.2. CDNs are often used to distribute stored and live video. CDNs are discussed in detail in Section 2.6.3.

9.3 Voice-over-IP

Real-time conversational voice over the Internet is often referred to as **Internet telephony**, since, from the user's perspective, it is similar to the traditional circuit-switched telephone service. It is also commonly called **Voice-over-IP (VoIP)**. In this section we describe the principles and protocols underlying VoIP. Conversational video is similar in many respects to VoIP, except that it includes the video of the participants as well as their voices. To keep the discussion focused and concrete, we focus here only on voice in this section rather than combined voice and video.

9.3.1 Limitations of the Best-Effort IP Service

The Internet's network-layer protocol, IP, provides best-effort service. That is to say the service makes its best effort to move each datagram from source to destination as quickly as possible but makes no promises whatsoever about getting the packet

to the destination within some delay bound or about a limit on the percentage of packets lost. The lack of such guarantees poses significant challenges to the design of real-time conversational applications, which are acutely sensitive to packet delay, jitter, and loss.

In this section, we'll cover several ways in which the performance of VoIP over a best-effort network can be enhanced. Our focus will be on application-layer techniques, that is, approaches that do not require any changes in the network core or even in the transport layer at the end hosts. To keep the discussion concrete, we'll discuss the limitations of best-effort IP service in the context of a specific VoIP example. The sender generates bytes at a rate of 8,000 bytes per second; every 20 msecs the sender gathers these bytes into a chunk. A chunk and a special header (discussed below) are encapsulated in a UDP segment, via a call to the socket interface. Thus, the number of bytes in a chunk is (20 msecs) · (8,000 bytes/sec) = 160 bytes, and a UDP segment is sent every 20 msecs.

If each packet makes it to the receiver with a constant end-to-end delay, then packets arrive at the receiver periodically every 20 msecs. In these ideal conditions, the receiver can simply play back each chunk as soon as it arrives. But unfortunately, some packets can be lost and most packets will not have the same end-to-end delay, even in a lightly congested Internet. For this reason, the receiver must take more care in determining (1) when to play back a chunk, and (2) what to do with a missing chunk.

Packet Loss

Consider one of the UDP segments generated by our VoIP application. The UDP segment is encapsulated in an IP datagram. As the datagram wanders through the network, it passes through router buffers (that is, queues) while waiting for transmission on outbound links. It is possible that one or more of the buffers in the path from sender to receiver is full, in which case the arriving IP datagram may be discarded, never to arrive at the receiving application.

Loss could be eliminated by sending the packets over TCP (which provides for reliable data transfer) rather than over UDP. However, retransmission mechanisms are often considered unacceptable for conversational real-time audio applications such as VoIP, because they increase end-to-end delay [Bolot 1996]. Furthermore, due to TCP congestion control, packet loss may result in a reduction of the TCP sender's transmission rate to a rate that is lower than the receiver's drain rate, possibly leading to buffer starvation. This can have a severe impact on voice intelligibility at the receiver. For these reasons, most existing VoIP applications run over UDP by default. [Baset 2006] reports that UDP is used by Skype unless a user is behind a NAT or firewall that blocks UDP segments (in which case TCP is used).

But losing packets is not necessarily as disastrous as one might think. Indeed, packet loss rates between 1 and 20 percent can be tolerated, depending on how voice is encoded and transmitted, and on how the loss is concealed at the receiver. For example, forward error correction (FEC) can help conceal packet loss. We'll see

below that with FEC, redundant information is transmitted along with the original information so that some of the lost original data can be recovered from the redundant information. Nevertheless, if one or more of the links between sender and receiver is severely congested, and packet loss exceeds 10 to 20 percent (for example, on a wireless link), then there is really nothing that can be done to achieve acceptable audio quality. Clearly, best-effort service has its limitations.

End-to-End Delay

End-to-end delay is the accumulation of transmission, processing, and queuing delays in routers; propagation delays in links; and end-system processing delays. For real-time conversational applications, such as VoIP, end-to-end delays smaller than 150 msecs are not perceived by a human listener; delays between 150 and 400 msecs can be acceptable but are not ideal; and delays exceeding 400 msecs can seriously hinder the interactivity in voice conversations. The receiving side of a VoIP application will typically disregard any packets that are delayed more than a certain threshold, for example, more than 400 msecs. Thus, packets that are delayed by more than the threshold are effectively lost.

Packet Jitter

A crucial component of end-to-end delay is the varying queuing delays that a packet experiences in the network's routers. Because of these varying delays, the time from when a packet is generated at the source until it is received at the receiver can fluctuate from packet to packet, as shown in Figure 9.1. This phenomenon is called **jitter**. As an example, consider two consecutive packets in our VoIP application. The sender sends the second packet 20 msecs after sending the first packet. But at the receiver, the spacing between these packets can become greater than 20 msecs. To see this, suppose the first packet arrives at a nearly empty queue at a router, but just before the second packet arrives at the queue a large number of packets from other sources arrive at the same queue. Because the first packet experiences a small queuing delay and the second packet suffers a large queuing delay at this router, the first and second packets become spaced by more than 20 msecs. The spacing between consecutive packets can also become less than 20 msecs. To see this, again consider two consecutive packets. Suppose the first packet joins the end of a queue with a large number of packets, and the second packet arrives at the queue before this first packet is transmitted and before any packets from other sources arrive at the queue. In this case, our two packets find themselves one right after the other in the queue. If the time it takes to transmit a packet on the router's outbound link is less than 20 msecs, then the spacing between first and second packets becomes less than 20 msecs.

The situation is analogous to driving cars on roads. Suppose you and your friend are each driving in your own cars from San Diego to Phoenix. Suppose you and your

friend have similar driving styles, and that you both drive at 100 km/hour, traffic permitting. If your friend starts out one hour before you, depending on intervening traffic, you may arrive at Phoenix more or less than one hour after your friend.

If the receiver ignores the presence of jitter and plays out chunks as soon as they arrive, then the resulting audio quality can easily become unintelligible at the receiver. Fortunately, jitter can often be removed by using **sequence numbers**, **timestamps**, and a **playout delay**, as discussed below.

9.3.2 Removing Jitter at the Receiver for Audio

For our VoIP application, where packets are being generated periodically, the receiver should attempt to provide periodic playout of voice chunks in the presence of random network jitter. This is typically done by combining the following two mechanisms:

- *Prepending each chunk with a **timestamp***. The sender stamps each chunk with the time at which the chunk was generated.
- ***Delaying playout** of chunks at the receiver*. As we saw in our earlier discussion of Figure 9.1, the playout delay of the received audio chunks must be long enough so that most of the packets are received before their scheduled playout times. This playout delay can either be fixed throughout the duration of the audio session or vary adaptively during the audio session lifetime.

We now discuss how these three mechanisms, when combined, can alleviate or even eliminate the effects of jitter. We examine two playback strategies: fixed playout delay and adaptive playout delay.

Fixed Playout Delay

With the fixed-delay strategy, the receiver attempts to play out each chunk exactly q msecs after the chunk is generated. So if a chunk is timestamped at the sender at time t, the receiver plays out the chunk at time $t + q$, assuming the chunk has arrived by that time. Packets that arrive after their scheduled playout times are discarded and considered lost.

What is a good choice for q? VoIP can support delays up to about 400 msecs, although a more satisfying conversational experience is achieved with smaller values of q. On the other hand, if q is made much smaller than 400 msecs, then many packets may miss their scheduled playback times due to the network-induced packet jitter. Roughly speaking, if large variations in end-to-end delay are typical, it is preferable to use a large q; on the other hand, if delay is small and variations in delay are also small, it is preferable to use a small q, perhaps less than 150 msecs.

The trade-off between the playback delay and packet loss is illustrated in Figure 9.4. The figure shows the times at which packets are generated and played

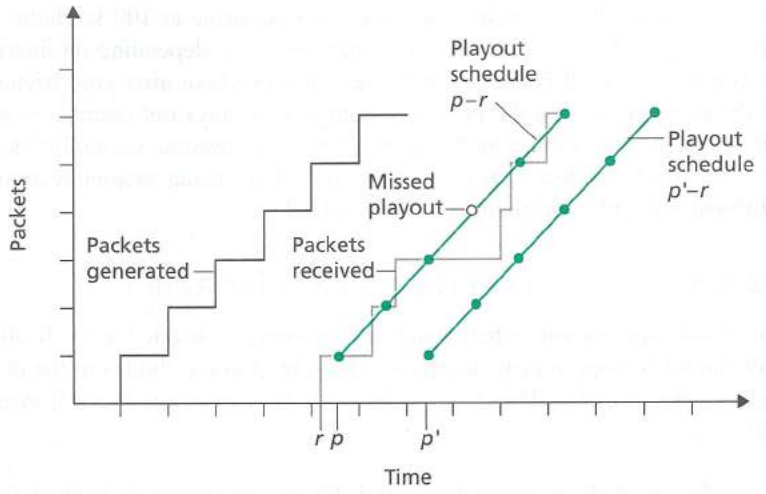

Figure 9.4 ♦ Packet loss for different fixed playout delays

out for a single talk spurt. Two distinct initial playout delays are considered. As shown by the leftmost staircase, the sender generates packets at regular intervals—say, every 20 msecs. The first packet in this talk spurt is received at time r. As shown in the figure, the arrivals of subsequent packets are not evenly spaced due to the network jitter.

For the first playout schedule, the fixed initial playout delay is set to $p - r$. With this schedule, the fourth packet does not arrive by its scheduled playout time, and the receiver considers it lost. For the second playout schedule, the fixed initial playout delay is set to $p' - r$. For this schedule, all packets arrive before their scheduled playout times, and there is therefore no loss.

Adaptive Playout Delay

The previous example demonstrates an important delay-loss trade-off that arises when designing a playout strategy with fixed playout delays. By making the initial playout delay large, most packets will make their deadlines and there will therefore be negligible loss; however, for conversational services such as VoIP, long delays can become bothersome if not intolerable. Ideally, we would like the playout delay to be minimized subject to the constraint that the loss be below a few percent.

The natural way to deal with this trade-off is to estimate the network delay and the variance of the network delay, and to adjust the playout delay accordingly at the beginning of each talk spurt. This adaptive adjustment of playout delays at the beginning of the talk spurts will cause the sender's silent periods to be compressed and elongated; however, compression and elongation of silence by a small amount is not noticeable in speech.

Following [Ramjee 1994], we now describe a generic algorithm that the receiver can use to adaptively adjust its playout delays. To this end, let

t_i = the timestamp of the ith packet = the time the packet was generated by the sender

r_i = the time packet i is received by receiver

p_i = the time packet i is played at receiver

The end-to-end network delay of the ith packet is $r_i - t_i$. Due to network jitter, this delay will vary from packet to packet. Let d_i denote an estimate of the *average* network delay upon reception of the ith packet. This estimate is constructed from the timestamps as follows:

$$d_i = (1 - u)\, d_{i-1} + u\,(r_i - t_i)$$

where u is a fixed constant (for example, $u = 0.01$). Thus d_i is a smoothed average of the observed network delays $r_1 - t_1, \ldots, r_i - t_i$. The estimate places more weight on the recently observed network delays than on the observed network delays of the distant past. This form of estimate should not be completely unfamiliar; a similar idea is used to estimate round-trip times in TCP, as discussed in Chapter 3. Let v_i denote an estimate of the average deviation of the delay from the estimated average delay. This estimate is also constructed from the timestamps:

$$v_i = (1 - u)\, v_{i-1} + u\,|r_i - t_i - d_i|$$

The estimates d_i and v_i are calculated for every packet received, although they are used only to determine the playout point for the first packet in any talk spurt.

Once having calculated these estimates, the receiver employs the following algorithm for the playout of packets. If packet i is the first packet of a talk spurt, its playout time, p_i, is computed as:

$$p_i = t_i + d_i + Kv_i$$

where K is a positive constant (for example, $K = 4$). The purpose of the Kv_i term is to set the playout time far enough into the future so that only a small fraction of the arriving packets in the talk spurt will be lost due to late arrivals. The playout point for any subsequent packet in a talk spurt is computed as an offset from the point in time when the first packet in the talk spurt was played out. In particular, let

$$q_i = p_i - t_i$$

be the length of time from when the first packet in the talk spurt is generated until it is played out. If packet j also belongs to this talk spurt, it is played out at time

$$p_j = t_j + q_i$$

The algorithm just described makes perfect sense assuming that the receiver can tell whether a packet is the first packet in the talk spurt. This can be done by examining the signal energy in each received packet.

9.3.3 Recovering from Packet Loss

We have discussed in some detail how a VoIP application can deal with packet jitter. We now briefly describe several schemes that attempt to preserve acceptable audio quality in the presence of packet loss. Such schemes are called **loss recovery schemes**. Here we define packet loss in a broad sense: A packet is lost either if it never arrives at the receiver or if it arrives after its scheduled playout time. Our VoIP example will again serve as a context for describing loss recovery schemes.

As mentioned at the beginning of this section, retransmitting lost packets may not be feasible in a real-time conversational application such as VoIP. Indeed, retransmitting a packet that has missed its playout deadline serves absolutely no purpose. And retransmitting a packet that overflowed a router queue cannot normally be accomplished quickly enough. Because of these considerations, VoIP applications often use some type of loss anticipation scheme. Two types of loss anticipation schemes are **forward error correction (FEC) and interleaving**.

Forward Error Correction (FEC)

The basic idea of FEC is to add redundant information to the original packet stream. For the cost of marginally increasing the transmission rate, the redundant information can be used to reconstruct approximations or exact versions of some of the lost packets. Following [Bolot 1996] and [Perkins 1998], we now outline two simple FEC mechanisms. The first mechanism sends a redundant encoded chunk after every n chunks. The redundant chunk is obtained by exclusive OR-ing the n original chunks [Shacham 1990]. In this manner if any one packet of the group of $n + 1$ packets is lost, the receiver can fully reconstruct the lost packet. But if two or more packets in a group are lost, the receiver cannot reconstruct the lost packets. By keeping $n + 1$, the group size, small, a large fraction of the lost packets can be recovered when loss is not excessive. However, the smaller the group size, the greater the relative increase of the transmission rate. In particular, the transmission rate increases by a factor of $1/n$, so that, if $n = 3$, then the transmission rate increases by 33 percent. Furthermore, this simple scheme increases the playout delay, as the receiver must wait to receive the entire group of packets before it can

begin playout. For more practical details about how FEC works for multimedia transport see [RFC 5109].

The second FEC mechanism is to send a lower-resolution audio stream as the redundant information. For example, the sender might create a nominal audio stream and a corresponding low-resolution, low-bit rate audio stream. (The nominal stream could be a PCM encoding at 64 kbps, and the lower-quality stream could be a GSM encoding at 13 kbps.) The low-bit rate stream is referred to as the redundant stream. As shown in Figure 9.5, the sender constructs the nth packet by taking the nth chunk from the nominal stream and appending to it the $(n-1)$st chunk from the redundant stream. In this manner, whenever there is nonconsecutive packet loss, the receiver can conceal the loss by playing out the low-bit rate encoded chunk that arrives with the subsequent packet. Of course, low-bit rate chunks give lower quality than the nominal chunks. However, a stream of mostly high-quality chunks, occasional low-quality chunks, and no missing chunks gives good overall audio quality. Note that in this scheme, the receiver only has to receive two packets before playback, so that the increased playout delay is small. Furthermore, if the low-bit rate encoding is much less than the nominal encoding, then the marginal increase in the transmission rate will be small.

In order to cope with consecutive loss, we can use a simple variation. Instead of appending just the $(n-1)$st low-bit rate chunk to the nth nominal chunk, the sender can append the $(n-1)$st and $(n-2)$nd low-bit rate chunk, or append the $(n-1)$st and $(n-3)$rd low-bit rate chunk, and so on. By appending more low-bit rate chunks to each nominal chunk, the audio quality at the receiver becomes acceptable for a wider variety of harsh best-effort environments. On the other hand, the additional chunks increase the transmission bandwidth and the playout delay.

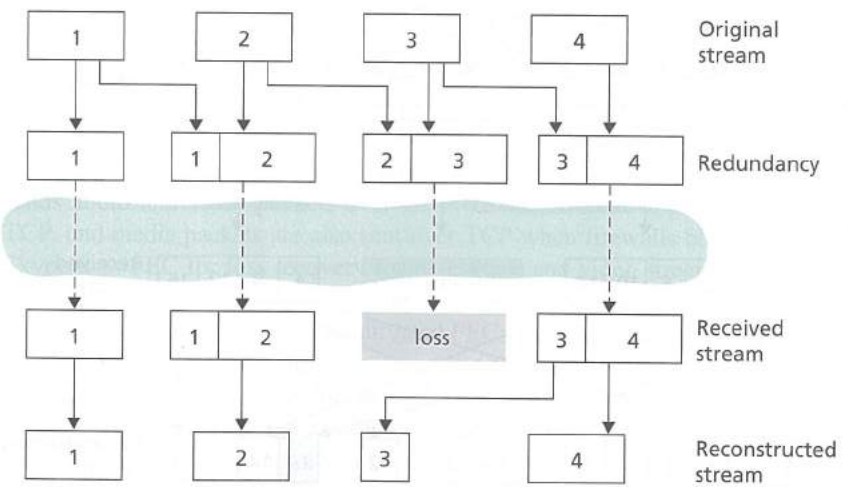

Figure 9.5 ♦ Piggybacking lower-quality redundant information

Interleaving

As an alternative to redundant transmission, a VoIP application can send interleaved audio. As shown in Figure 9.6, the sender resequences units of audio data before transmission, so that originally adjacent units are separated by a certain distance in the transmitted stream. Interleaving can mitigate the effect of packet losses. If, for example, units are 5 msecs in length and chunks are 20 msecs (that is, four units per chunk), then the first chunk could contain units 1, 5, 9, and 13; the second chunk could contain units 2, 6, 10, and 14; and so on. Figure 9.6 shows that the loss of a single packet from an interleaved stream results in multiple small gaps in the reconstructed stream, as opposed to the single large gap that would occur in a noninterleaved stream.

Interleaving can significantly improve the perceived quality of an audio stream [Perkins 1998]. It also has low overhead. The obvious disadvantage of interleaving is that it increases latency. This limits its use for conversational applications such as VoIP, although it can perform well for streaming stored audio. A major advantage of interleaving is that it does not increase the bandwidth requirements of a stream.

Error Concealment

Error concealment schemes attempt to produce a replacement for a lost packet that is similar to the original. As discussed in [Perkins 1998], this is possible since audio

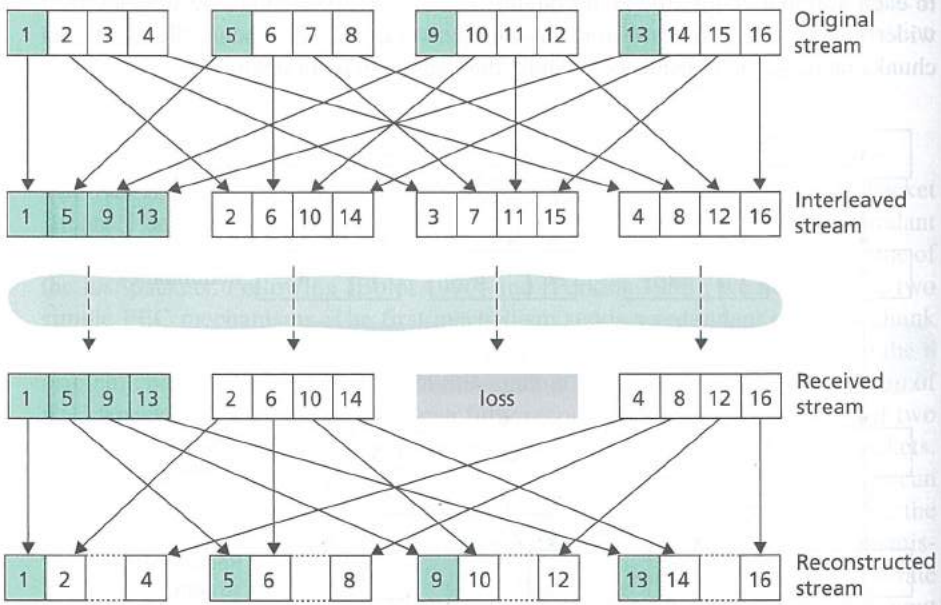

Figure 9.6 ♦ Sending interleaved audio

signals, and in particular speech, exhibit large amounts of short-term self-similarity. As such, these techniques work for relatively small loss rates (less than 15 percent), and for small packets (4–40 msecs). When the loss length approaches the length of a phoneme (5–100 msecs) these techniques break down, since whole phonemes may be missed by the listener.

Perhaps the simplest form of receiver-based recovery is packet repetition. Packet repetition replaces lost packets with copies of the packets that arrived immediately before the loss. It has low computational complexity and performs reasonably well. Another form of receiver-based recovery is interpolation, which uses audio before and after the loss to interpolate a suitable packet to cover the loss. Interpolation performs somewhat better than packet repetition but is significantly more computationally intensive [Perkins 1998].

9.3.4 Case Study: VoIP with Skype

Skype is an immensely popular VoIP application with over 50 million accounts active on a daily basis. In addition to providing host-to-host VoIP service, Skype offers host-to-phone services, phone-to-host services, and multi-party host-to-host video conferencing services. (Here, a host is again any Internet connected IP device, including PCs, tablets, and smartphones.) Skype was acquired by Microsoft in 2011.

Because the Skype protocol is proprietary, and because all Skype's control and media packets are encrypted, it is difficult to precisely determine how Skype operates. Nevertheless, from the Skype Web site and several measurement studies, researchers have learned how Skype generally works [Baset 2006; Guha 2006; Chen 2006; Suh 2006; Ren 2006; Zhang X 2012]. For both voice and video, the Skype clients have at their disposal many different codecs, which are capable of encoding the media at a wide range of rates and qualities. For example, video rates for Skype have been measured to be as low as 30 kbps for a low-quality session up to almost 1 Mbps for a high quality session [Zhang X 2012]. Typically, Skype's audio quality is better than the "POTS" (Plain Old Telephone Service) quality provided by the wire-line phone system. (Skype codecs typically sample voice at 16,000 samples/sec or higher, which provides richer tones than POTS, which samples at 8,000/sec.) By default, Skype sends audio and video packets over UDP. However, control packets are sent over TCP, and media packets are also sent over TCP when firewalls block UDP streams. Skype uses FEC for loss recovery for both voice and video streams sent over UDP. The Skype client also adapts the audio and video streams it sends to current network conditions, by changing video quality and FEC overhead [Zhang X 2012].

Skype uses P2P techniques in a number of innovative ways, nicely illustrating how P2P can be used in applications that go beyond content distribution and file sharing. As with instant messaging, host-to-host Internet telephony is inherently P2P since, at the heart of the application, pairs of users (that is, peers) communicate with each other in real time. But Skype also employs P2P techniques for two other important functions, namely, for user location and for NAT traversal.

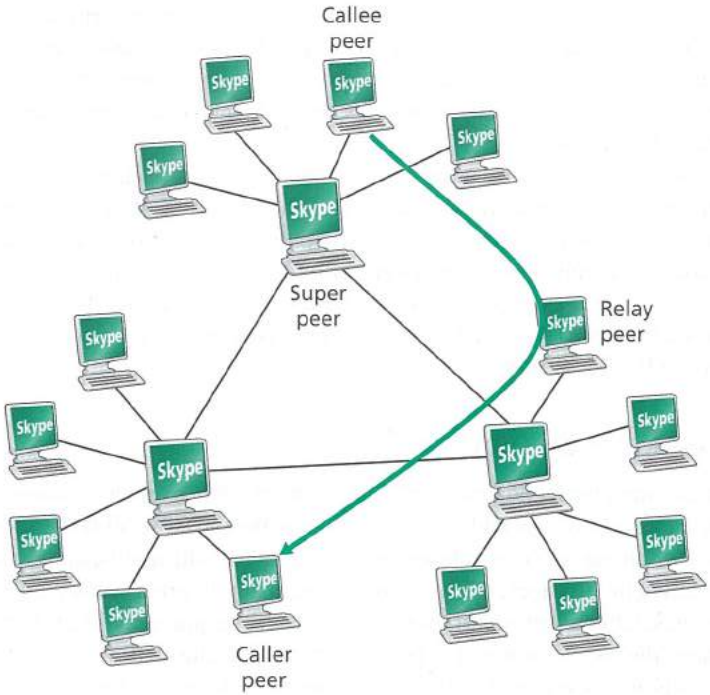

Figure 9.7 ♦ Skype peers

As shown in Figure 9.7, the peers (hosts) in Skype are organized into a hierarchical overlay network, with each peer classified as a super peer or an ordinary peer. Skype maintains an index that maps Skype usernames to current IP addresses (and port numbers). This index is distributed over the super peers. When Alice wants to call Bob, her Skype client searches the distributed index to determine Bob's current IP address. Because the Skype protocol is proprietary, it is currently not known how the index mappings are organized across the super peers, although some form of DHT organization is very possible.

P2P techniques are also used in Skype **relays**, which are useful for establishing calls between hosts in home networks. Many home network configurations provide access to the Internet through NATs, as discussed in Chapter 4. Recall that a NAT prevents a host from outside the home network from initiating a connection to a host within the home network. If *both* Skype callers have NATs, then there is a problem—neither can accept a call initiated by the other, making a call seemingly impossible. The clever use of super peers and relays nicely solves this problem. Suppose that when Alice signs in, she is assigned to a non-NATed super peer and initiates a session to that super peer. (Since Alice is initiating the session, her NAT permits this session.) This session allows Alice and her super peer to exchange

control messages. The same happens for Bob when he signs in. Now, when Alice wants to call Bob, she informs her super peer, who in turn informs Bob's super peer, who in turn informs Bob of Alice's incoming call. If Bob accepts the call, the two super peers select a third non-NATed super peer—the relay peer—whose job will be to relay data between Alice and Bob. Alice's and Bob's super peers then instruct Alice and Bob respectively to initiate a session with the relay. As shown in Figure 9.7, Alice then sends voice packets to the relay over the Alice-to-relay connection (which was initiated by Alice), and the relay then forwards these packets over the relay-to-Bob connection (which was initiated by Bob); packets from Bob to Alice flow over these same two relay connections in reverse. And *voila!*—Bob and Alice have an end-to-end connection even though neither can accept a session originating from outside.

Up to now, our discussion on Skype has focused on calls involving two persons. Now let's examine multi-party audio conference calls. With $N > 2$ participants, if each user were to send a copy of its audio stream to each of the $N - 1$ other users, then a total of $N(N - 1)$ audio streams would need to be sent into the network to support the audio conference. To reduce this bandwidth usage, Skype employs a clever distribution technique. Specifically, each user sends its audio stream to the conference initiator. The conference initiator combines the audio streams into one stream (basically by adding all the audio signals together) and then sends a copy of each combined stream to each of the other $N - 1$ participants. In this manner, the number of streams is reduced to $2(N - 1)$. For ordinary two-person video conversations, Skype routes the call peer-to-peer, unless NAT traversal is required, in which case the call is relayed through a non-NATed peer, as described earlier. For a video conference call involving $N > 2$ participants, due to the nature of the video medium, Skype does not combine the call into one stream at one location and then redistribute the stream to all the participants, as it does for voice calls. Instead, each participant's video stream is routed to a server cluster (located in Estonia as of 2011), which in turn relays to each participant the $N - 1$ streams of the $N - 1$ other participants [Zhang X 2012]. You may be wondering why each participant sends a copy to a server rather than directly sending a copy of its video stream to each of the other $N - 1$ participants? Indeed, for both approaches, $N(N - 1)$ video streams are being collectively received by the N participants in the conference. The reason is, because upstream link bandwidths are significantly lower than downstream link bandwidths in most access links, the upstream links may not be able to support the $N - 1$ streams with the P2P approach.

VoIP systems such as Skype, WeChat, and Google Talk introduce new privacy concerns. Specifically, when Alice and Bob communicate over VoIP, Alice can sniff Bob's IP address and then use geo-location services [MaxMind 2016; Quova 2016] to determine Bob's current location and ISP (for example, his work or home ISP). In fact, with Skype it is possible for Alice to block the transmission of certain packets during call establishment so that she obtains Bob's current IP address, say every hour, without Bob knowing that he is being tracked and without being on Bob's

contact list. Furthermore, the IP address discovered from Skype can be correlated with IP addresses found in BitTorrent, so that Alice can determine the files that Bob is downloading [LeBlond 2011]. Moreover, it is possible to partially decrypt a Skype call by doing a traffic analysis of the packet sizes in a stream [White 2011].

9.4 Protocols for Real-Time Conversational Applications

Real-time conversational applications, including VoIP and video conferencing, are compelling and very popular. It is therefore not surprising that standards bodies, such as the IETF and ITU, have been busy for many years (and continue to be busy!) at hammering out standards for this class of applications. With the appropriate standards in place for real-time conversational applications, independent companies are creating new products that interoperate with each other. In this section we examine RTP and SIP for real-time conversational applications. Both standards are enjoying widespread implementation in industry products.

9.4.1 RTP

In the previous section, we learned that the sender side of a VoIP application appends header fields to the audio chunks before passing them to the transport layer. These header fields include sequence numbers and timestamps. Since most multimedia networking applications can make use of sequence numbers and timestamps, it is convenient to have a standardized packet structure that includes fields for audio/video data, sequence number, and timestamp, as well as other potentially useful fields. RTP, defined in RFC 3550, is such a standard. RTP can be used for transporting common formats such as PCM, ACC, and MP3 for sound and MPEG and H.263 for video. It can also be used for transporting proprietary sound and video formats. Today, RTP enjoys widespread implementation in many products and research prototypes. It is also complementary to other important real-time interactive protocols, such as SIP.

In this section, we provide an introduction to RTP. We also encourage you to visit Henning Schulzrinne's RTP site [Schulzrinne-RTP 2012], which provides a wealth of information on the subject. Also, you may want to visit the RAT site [RAT 2012], which documents VoIP application that uses RTP.

RTP Basics

RTP typically runs on top of UDP. The sending side encapsulates a media chunk within an RTP packet, then encapsulates the packet in a UDP segment, and then

hands the segment to IP. The receiving side extracts the RTP packet from the UDP segment, then extracts the media chunk from the RTP packet, and then passes the chunk to the media player for decoding and rendering.

As an example, consider the use of RTP to transport voice. Suppose the voice source is PCM-encoded (that is, sampled, quantized, and digitized) at 64 kbps. Further suppose that the application collects the encoded data in 20-msec chunks, that is, 160 bytes in a chunk. The sending side precedes each chunk of the audio data with an **RTP header** that includes the type of audio encoding, a sequence number, and a timestamp. The RTP header is normally 12 bytes. The audio chunk along with the RTP header form the **RTP packet**. The RTP packet is then sent into the UDP socket interface. At the receiver side, the application receives the RTP packet from its socket interface. The application extracts the audio chunk from the RTP packet and uses the header fields of the RTP packet to properly decode and play back the audio chunk.

If an application incorporates RTP—instead of a proprietary scheme to provide payload type, sequence numbers, or timestamps—then the application will more easily interoperate with other networked multimedia applications. For example, if two different companies develop VoIP software and they both incorporate RTP into their product, there may be some hope that a user using one of the VoIP products will be able to communicate with a user using the other VoIP product. In Section 9.4.2, we'll see that RTP is often used in conjunction with SIP, an important standard for Internet telephony.

It should be emphasized that RTP does not provide any mechanism to ensure timely delivery of data or provide other quality-of-service (QoS) guarantees; it does not even guarantee delivery of packets or prevent out-of-order delivery of packets. Indeed, RTP encapsulation is seen only at the end systems. Routers do not distinguish between IP datagrams that carry RTP packets and IP datagrams that don't.

RTP allows each source (for example, a camera or a microphone) to be assigned its own independent RTP stream of packets. For example, for a video conference between two participants, four RTP streams could be opened—two streams for transmitting the audio (one in each direction) and two streams for transmitting the video (again, one in each direction). However, many popular encoding techniques—including MPEG 1 and MPEG 2—bundle the audio and video into a single stream during the encoding process. When the audio and video are bundled by the encoder, then only one RTP stream is generated in each direction.

RTP packets are not limited to unicast applications. They can also be sent over one-to-many and many-to-many multicast trees. For a many-to-many multicast session, all of the session's senders and sources typically use the same multicast group for sending their RTP streams. RTP multicast streams belonging together, such as audio and video streams emanating from multiple senders in a video conference application, belong to an **RTP session**.

Figure 9.8 ♦ RTP header fields

RTP Packet Header Fields

As shown in Figure 9.8, the four main RTP packet header fields are the payload type, sequence number, timestamp, and source identifier fields.

The payload type field in the RTP packet is 7 bits long. For an audio stream, the payload type field is used to indicate the type of audio encoding (for example, PCM, adaptive delta modulation, linear predictive encoding) that is being used. If a sender decides to change the encoding in the middle of a session, the sender can inform the receiver of the change through this payload type field. The sender may want to change the encoding in order to increase the audio quality or to decrease the RTP stream bit rate. Table 9.2 lists some of the audio payload types currently supported by RTP.

For a video stream, the payload type is used to indicate the type of video encoding (for example, motion JPEG, MPEG 1, MPEG 2, H.261). Again, the sender can change video encoding on the fly during a session. Table 9.3 lists some of the video payload types currently supported by RTP. The other important fields are the following:

- *Sequence number field.* The sequence number field is 16 bits long. The sequence number increments by one for each RTP packet sent, and may be used by the receiver to detect packet loss and to restore packet sequence. For example, if the receiver side of the application receives a stream of RTP packets with a gap between sequence numbers 86 and 89, then the receiver knows that packets 87 and 88 are missing. The receiver can then attempt to conceal the lost data.
- *Timestamp field.* The timestamp field is 32 bits long. It reflects the sampling instant of the first byte in the RTP data packet. As we saw in the preceding section, the receiver can use timestamps to remove packet jitter introduced in the network and to provide synchronous playout at the receiver. The timestamp is derived from a sampling clock at the sender. As an example, for audio the timestamp clock increments by one for each sampling period (for example, each 125 μsec for an 8 kHz sampling clock); if the audio application generates chunks consisting of 160 encoded samples, then the timestamp increases by 160 for each RTP packet when the source is active. The timestamp clock continues to increase at a constant rate even if the source is inactive.
- *Synchronization source identifier (SSRC).* The SSRC field is 32 bits long. It identifies the source of the RTP stream. Typically, each stream in an RTP session has a distinct SSRC. The SSRC is not the IP address of the sender, but instead is a number that the source assigns randomly when the new stream is started. The probability that two streams get assigned the same SSRC is very small. Should this happen, the two sources pick a new SSRC value.

Payload-Type Number	Audio Format	Sampling Rate	Rate
0	PCM μ-law	8 kHz	64 kbps
1	1016	8 kHz	4.8 kbps
3	GSM	8 kHz	13 kbps
7	LPC	8 kHz	2.4 kbps
9	G.722	16 kHz	48–64 kbps
14	MPEG Audio	90 kHz	—
15	G.728	8 kHz	16 kbps

Table 9.2 ♦ Audio payload types supported by RTP

Payload-Type Number	Video Format
26	Motion JPEG
31	H.261
32	MPEG 1 video
33	MPEG 2 video

Table 9.3 ♦ Some video payload types supported by RTP

9.4.2 SIP

The Session Initiation Protocol (SIP), defined in [RFC 3261; RFC 5411], is an open and lightweight protocol that does the following:

- It provides mechanisms for establishing calls between a caller and a callee over an IP network. It allows the caller to notify the callee that it wants to start a call. It allows the participants to agree on media encodings. It also allows participants to end calls.
- It provides mechanisms for the caller to determine the current IP address of the callee. Users do not have a single, fixed IP address because they may be assigned addresses dynamically (using DHCP) and because they may have multiple IP devices, each with a different IP address.
- It provides mechanisms for call management, such as adding new media streams during the call, changing the encoding during the call, inviting new participants during the call, call transfer, and call holding.