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.

7.1.3 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 streaming music) 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, and Hulu. By some estimates, streaming stored video makes up over 50 percent of the downstream traffic in the Internet access networks today [Cisco 2011]. 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 7.2, 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 7.2, 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 7.3, 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. Although the distribution of live audio/video to many receivers can be efficiently accomplished using the IP multicasting techniques described in Section 4.7, multicast distribution is more often accomplished today via application-layer multicast (using P2P networks or CDNs) or through multiple separate unicast streams. 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.

7.2 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**. 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 7.1 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, $\triangle$. The server transmits the first video block at t_0, the second block at $t_0 + \triangle$, the third block at $t_0 + 2\triangle$, and so on. Once the client begins playout, each block should be played out $\triangle$ 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

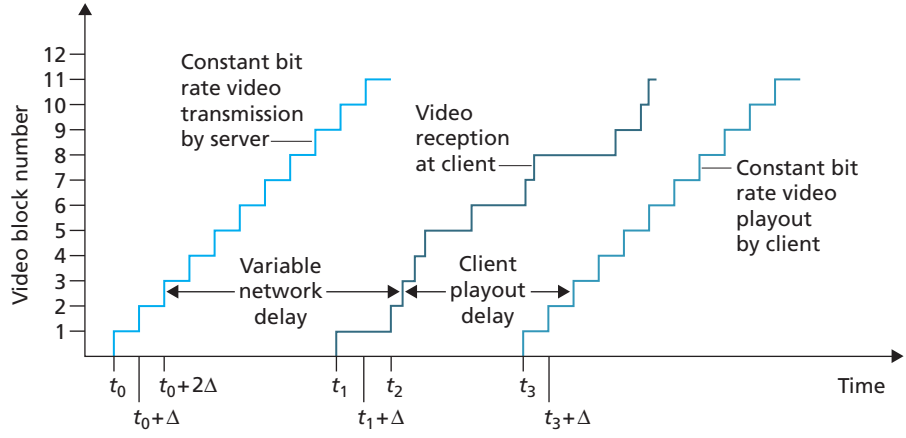

Figure 7.1 ♦ Client playout delay in video streaming