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 $t_1 + \triangle$. In this case, video playout would either have to stall (waiting for block 1 to arrive) or block 1 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.

7.2.1 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 7.3, 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). This control connection is in many ways analogous to the FTP control connection we studied in Chapter 2. The Real-Time Streaming Protocol (RTSP) [RFC 2326], explained in some detail in the companion 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