Because an IP internet is not isochronous, additional protocol support is required when sending digitized real-time data. In addition to basic sequence information that allows detection of duplicate or reordered packets, each packet must carry a separate timestamp that tells the receiver the exact time at which the data in the packet should be played.

28.4 Jitter And Playback Delay

How can a receiver recreate a signal accurately if the network introduces jitter? The receiver must implement a *playback buffer*† as Figure 28.1 illustrates.

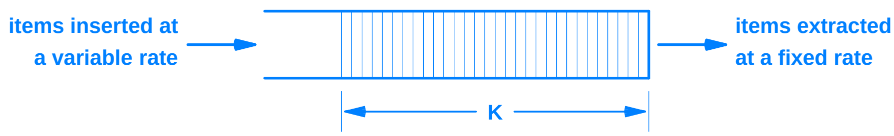

Figure 28.1 The conceptual organization of a playback buffer that compensates for jitter. The buffer holds *K* time units of data.

When a session begins, the receiver delays playback and places incoming data in the buffer. When data in the buffer reaches a predetermined threshold, known as the *playback point*, output begins. The playback point, labeled *K* in the figure, is measured in time units of data to be played. Thus, playback begins when a receiver has accumulated *K* time units' worth of data.

As playback proceeds, datagrams continue to arrive. If there is no jitter, new data will arrive at exactly the same rate old data is being extracted and played, meaning the buffer will always contain exactly *K* time units of unplayed data. If a datagram experiences a small delay, playback is unaffected. The buffer size decreases steadily as data is extracted, and playback continues uninterrupted for *K* time units. When a delayed datagram arrives, the buffer is refilled.

Of course, a playback buffer cannot compensate for datagram loss. In such cases, playback eventually reaches an unfilled position in the buffer, and output pauses for a time period corresponding to the missing data. Furthermore, the choice of *K* is a compromise between loss and delay‡. If *K* is too small, a small amount of jitter causes the system to exhaust the playback buffer before the needed data arrives. If *K* is too large, the system remains immune to jitter, but the extra delay, when added to the transmission delay in the underlying network, may be noticeable to users. Despite the disadvantages, most applications that send real-time data across an IP internet depend on playback buffering as the primary solution for jitter.

†A playback buffer is also called a *jitter buffer*.

‡Although network delay and jitter can be used to determine a value for *K* dynamically, many playback buffering schemes use a constant.