Sampling

In the world of analog audio, signals are passed, recorded, stored, and reproduced as changes in voltage levels that continuously change over time (Figure 6.2). The digital recording process, on the other hand, doesn't operate in a continuous manner; rather, digital recording takes periodic *samples* of a changing audio waveform (Figure 6.3) and transforms these sampled signal levels into a representative stream of binary words that can be manipulated or stored for later processing and/or reproduction.

Within a digital audio system, the *sampling rate* is defined as the number of measurements (samples) that are taken of an analog signal in one second. Its reciprocal (sampling time) is the elapsed time that occurs between each sampling period. For example, a sample rate of 48 kHz corresponds to a sample time of 1/48,000th of a second. Because sampling is tied directly to the component of time, the sampling rate of a system determines its overall bandwidth (Figure 6.4), meaning that a system with higher sample rates is capable of storing more frequencies at its upper limit.

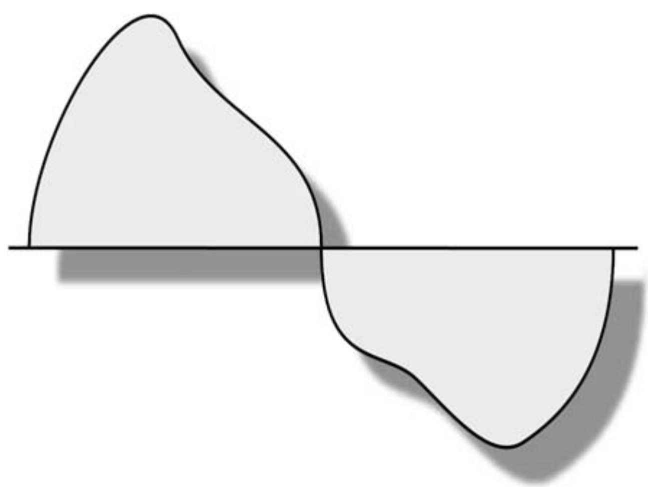

Figure 6.2. *An analog signal is continuous in nature.*

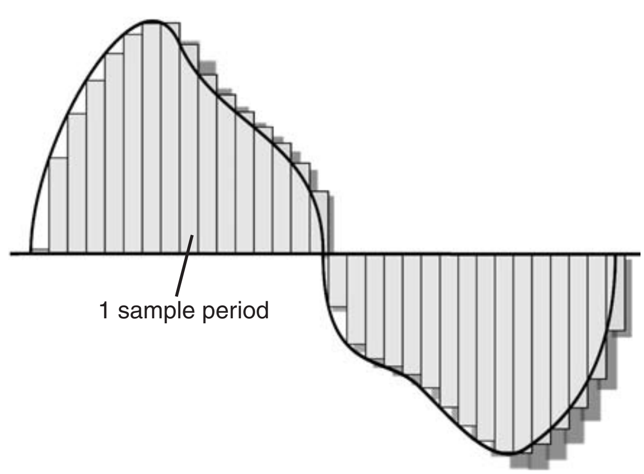

Figure 6.3. *A digital signal makes use of periodic sampling to encode information.*

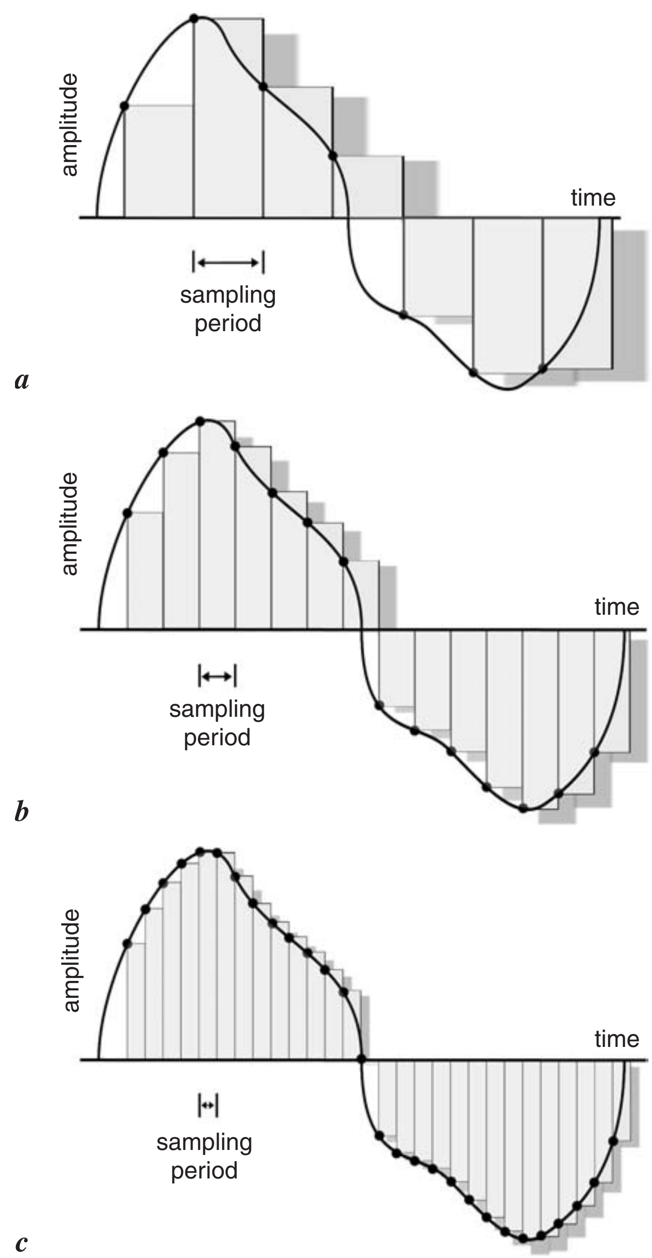

Figure 6.4. *Discrete time sampling. (a) Whenever the sample rate is set too low, important data between sample periods will be lost. (b) As the rate is increased, more frequency-related data can be encoded. (c) Increasing the sampling frequency further can encode the recorded signal with an even higher bandwidth range.*

As you might expect, the sampling process can be likened to a photographer who takes a series of shots of an action sequence. As the number of pictures taken in a second increases, the accuracy of the captured event will likewise increase . . . until the resolution is so great, that you can't tell that the successive pictures have turned into a (hopefully) compelling movie.

During the sampling process (Figure 6.5), an incoming analog signal is sampled at discrete and precisely timed intervals (as determined by the sample rate). At each interval, this analog

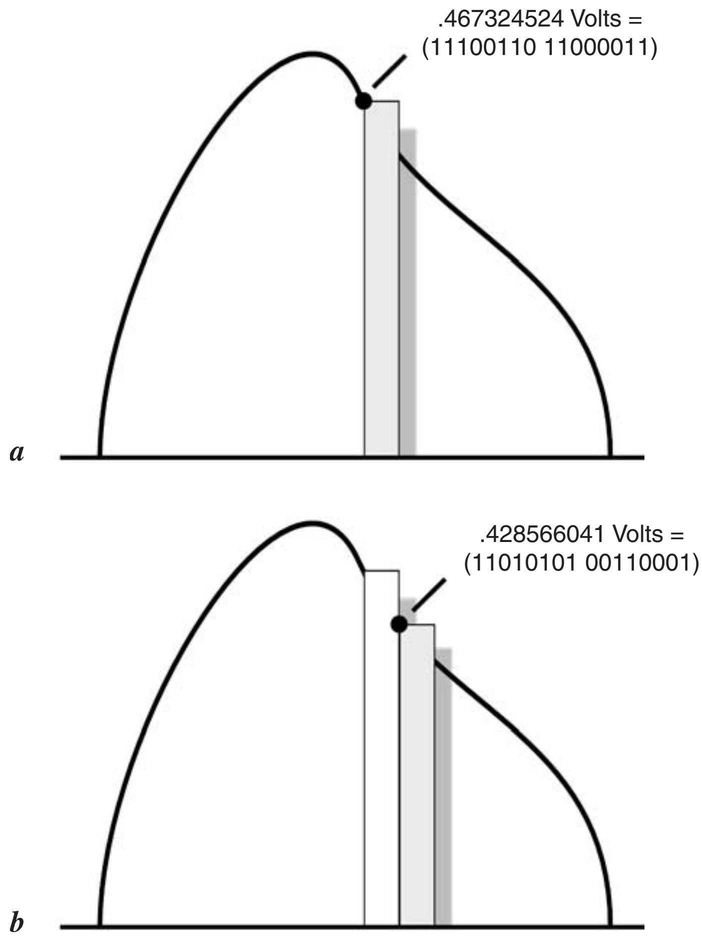

Figure 6.5. *The sampling process. (a) The analog signal is momentarily "held" (frozen in time), while the converter goes about the process of determining the voltage level at that point in time and then converting that level into a binary-encoded word that's numerically equivalent to the sampled level. (b) After the converter has stored the representative word into a memory medium, the sample is released and the next sample is held, as the system again goes about the task of determining the level of the next sampled voltage . . . and so forth, and so forth, and so forth over time.*

signal is momentarily ''held'' (frozen in time), while the converter goes about the process of determining what the voltage level actually is, with a degree of accuracy that's defined by the converter's circuitry (Figures 6.6 and 6.7) and the chosen bit rate. The converter then generates a binary-encoded word that's numerically equivalent to the analog level currently being sampled. Once done, the converter can store the representative word into a memory medium (tape, disk, disc, etc.), release its hold, and then go about the task of determining the level of the next sampled voltage. The process is then continuously repeated throughout the recording process.

The Nyquist Theorem

According to the *Nyquist theorem*, in order for the desired frequency bandwidth to be faithfully encoded in the digital domain, *the selected sample rate must be at least twice as high as the highest frequency to be recorded* (sample rate $\geq 2 \times$ highest frequency). Thus, an audio signal with a bandwidth of 20 kHz would require that the sampling rate be at least 40 kHz samples/second.