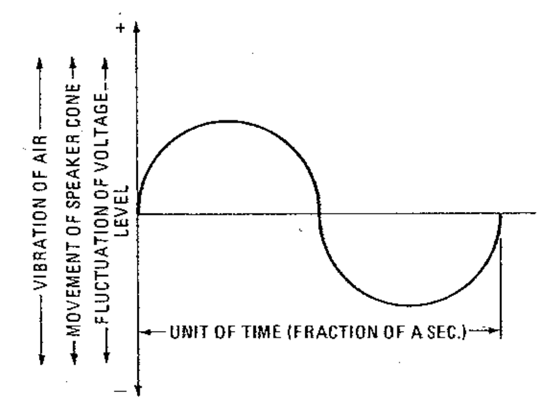
ANALOG VERSUS DIGITAL

To understand the advantages and advancements that computer music and related developments encompass, we must distinguish between **analog** and **digital** technology.

Tape manipulation, voltage control, and all the other resources introduced in Chapter 8 depend on *analog* circuitry, which produces varying levels of voltage, corresponding directly to varying levels of amplitude or frequency in the sounds being generated. The waveform in Figure 17–1 will help in visualizing this correspondence. The waveshape depicted might represent a sound wave—a rapid fluctuation of air molecules whose pattern is perceived by the ear as a tone with a particular frequency, amplitude, and timbre. As the vertical axis shows, however, the same waveshape could just as accurately represent the motion of a loudspeaker cone creating that sound wave, and it could also represent equally well the fluctuation of voltage in a synthesizer inducing that motion in the speaker cone. There is, in other words, a direct analogy between the acoustic sound and its electronic source.

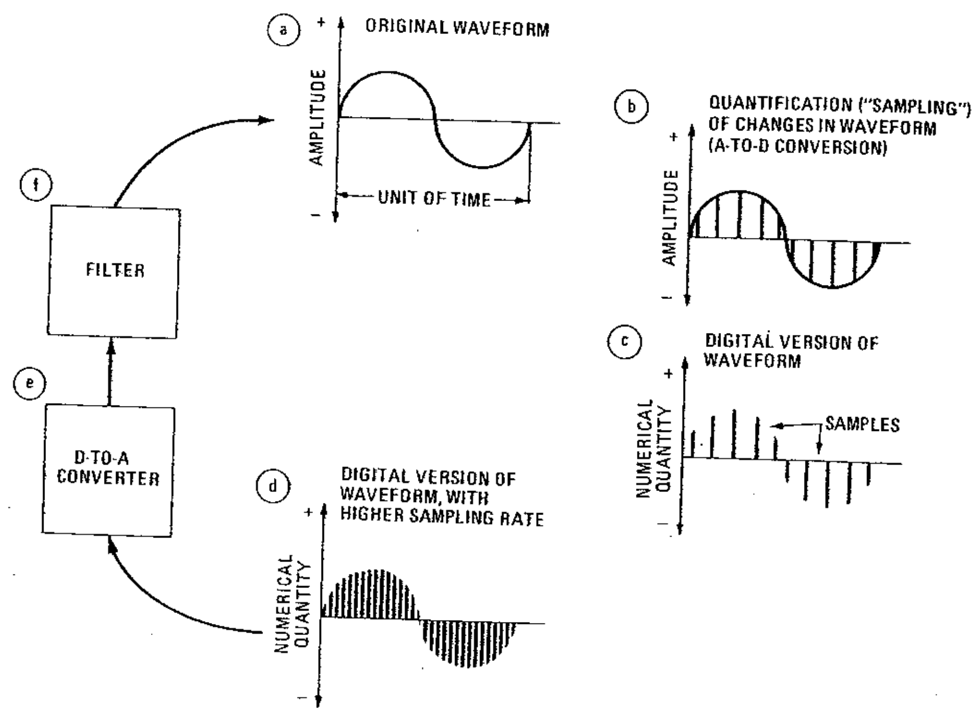
On the other hand, *digital* hardware (of which the computer is an example) deals not with voltage levels but with numbers. The basic steps in digital sound synthesis are depicted in Figure 17–2. A computer generates a waveshape not with continually rising and falling voltage but with a

FIGURE 17-1.



Three guises of a simple waveform

FIGURE 17-2.



Digital conversion of a simple waveform

series of numbers that go up and down to approximate the desired waveform. Figure 17–2a is a hypothetical waveform to be generated; as Figure 17–2b illustrates, the computer constructs it by representing its amplitude instant by instant with a number or quantity (each quantity shown here as a vertical bar). As seen in Figure 17–2c, the result is a series of such quantities that approximates the intended shape of the wave.

Each quantity is called a *sample*, and the higher the sampling rate (i.e., the number of samples per time unit), the more closely this series of quantities will represent its acoustic goal. This can be seen in figure 17–2d, where the higher number of samples (suggested by the more frequent vertical bars) outlines a smoother shape, one closer to the desired waveform. (In large mainframe computers, sampling rates of fifty thousand cycles per second are common.) In any case, the digital form of the wave, as we can see, is really just a string of numbers stored in the computer's memory.

To be realized in sound, however, this string of numbers must be transformed into an electrical signal—an analog fluctuation of voltage that can move the speaker cone. A *digital-to-analog* (or D-to-A) *converter* (Figure 17–2e) is used to accomplish this transformation. The typical computer music facility of the 1960s and 1970s required both a mainframe computer and a large bank of such converters. Ideally, the conversion process should transform discrete increments of change like those in Figure 17–2d into a continuous waveform like that in Figure 17–2a. In reality, however, the quantifying and converting processes produce acoustical by-products—unwanted partials and other extraneous noise—which must be filtered out (Figure 17–2f) to achieve acceptable results.

The advantages of *digital synthesis* over analog synthesis are easy to see. A string of numbers can be modified and manipulated with much more precision and flexibility than voltage-controlled devices or magnetic tape, especially considering that a single tone may be represented by as many as fifty thousand numbers (samples) for every second of its duration. Digital hardware and software can manage these numbers to simulate (but with greater sophistication) all the sound-processing functions we have already encountered with older, analog technology—timbral modification, envelope shaping, sequencing (i.e., ordering), mixing, and the like.

Note, however, that the practice of converting digital information into sound can be reversed. Once acoustic sounds are turned into analog electrical signals by means of a microphone, they can then be digitized (i.e., changed into numbers) by *analog-to-digital* (or A-to-D) *converters*. This process is called *digital sampling*, and as with digital synthesis, it too depends on very high sampling rates for the best results. Figure 17–2 could be used to visualize digital sampling as well, the only difference being that the analog wave form sampled (Figure 17–2a) would be produced by a real sound rather than the composer's imagination.

In either case, however, the result is a string of numbers that can be stored in computer memory, manipulated at will, and then transformed into sound by digital-to-analog conversion. Hence, digital sampling makes possible a kind of high-tech *musique concrète* wherein sounds can first be digitally recorded, then fragmented, reordered, timbrally altered or otherwise varied through sophisticated computer programming, and finally reconverted into a remarkable transmutation of the original.

As we shall soon note, the musical power of digital technology has not been limited to large mainframe computers. Since 1980 the musical community has seen a proliferation of smaller, more efficient, and more affordable digital equipment—devices that can be installed in small studios (or even used on the concert stage) and that require little or no computer know-how, while surpassing the musical capabilities of their mainframe ancestors.