12.7: Summary

In this chapter we have covered the basic concepts of analog-to-digital and digital-to-analog conversion. By placing a signal in the digital domain, a variety of new analysis, storage, and transmission techniques become available.

The process of transforming an analog signal into a digital representation is referred to as quantization, or more simply, as digitizing or sampling. One of the most popular methods used is PCM, pulse code modulation. In this scheme, the input signal is measured, or sampled, at a constant rate. Each sample point is represented as a digital word. The sequence of words describes the input signal and is used to recreate it, if necessary. The ultimate accuracy of the conversion is dependent on the resolution and sampling rate of the system. Resolution refers to the number of bits present in the digital word. An 8-bit word can represent 256 steps, whereas a 16-bit word is much finer, offering 65,536 discrete steps. With so many steps, round-off error is much less of a problem in 16-bit systems than in, say, 8- or 12-bit systems.

The minimum allowable sampling rate is twice the highest input frequency. In other words, at least two samples per cycle are required for the highest harmonic in the input signal. Another way of stating this is that no input frequency component can exceed the Nyquist frequency, which is defined as one-half of the sampling frequency. If the input exceeds this limit, alias distortion may occur. Aliases are nonharmonically-related frequencies that are effectively created by improper sampling. In order to remove all possibility of alias distortion, special high-order low-pass filters, called anti-alias filters, are normally placed before the sampling circuitry.

Two popular methods of analog-to-digital conversion are the flash and successive approximation techniques. Flash converters are very fast, but require one comparator per output step, so they are not normally used where high resolution is required. Successive approximation takes longer than flash conversion, but can produce resolutions in excess of 16 bits. In either case, the conversion process is not instantaneous, and any fluctuation of the input signal during the conversion can produce errors. In order to alleviate this difficulty, special track-and-hold amplifiers are used.
between the anti-alias filters and the AD converter to create a non-varying signal.

Once the signal has been digitized, it may be stored in RAM or some other media for future use, or directly analyzed. Often, a personal computer can prove to be very useful for waveform analysis.

To reconstruct the waveform, a digital-to-analog converter is used. In essence, this is usually little more than a weighted summing amplifier. Due to accuracy and construction constraints, an \( R/2R \) ladder technique is often employed. In order to smooth out the resulting waveform and remove any remaining digital “glitches”, the signal is passed through a reconstruction filter. This is a low-pass filter and is often called a smoothing filter. Generally, the process of digital-to-analog conversion is much faster than analog-to-digital conversion. Indeed, the successive approximation analog-to-digital scheme requires an internal digital-to-analog converter.

Applications for AD and DA converters range from laboratory instruments such as digital sampling oscilloscopes and hand-held digital multimeters, to industrial instrument and device control. AD and DA systems have found a large application market in the commercial sector. Uses include the popular music CD player and musical keyboard equipment.