quencies generated. These devices are commonly used in digital audio equipment because of their low overall system cost and high resolution (16 to 24 bits) for frequencies below 100 kHz.

15.5 FILTERS IN DATA CONVERSION SYSTEMS

General noise-reduction filters are typically found on the power supplies feeding data conversion circuits, because less distortion of the analog signals results when there is less overall noise in the system. Purely digital systems are not immune to noise, but their tolerance threshold is much higher than for analog systems because of their highly quantized binary signals. It takes noise of greater magnitude to turn a 0 into a 1 than it does to distort a continuous analog signal.

In addition to general noise reduction, anti-aliasing filters are a key design aspect of data conversion systems. Filtering requirements dictated by the sampling rate and by the presence of undesired frequency content can be quite stringent. As the gap between the Nyquist frequency and the undesired frequencies decreases, more complex filters are necessary. The design of such complex filters requires a substantial set of analog design skills, the majority of which are outside the scope of this book. However, this chapter closes by identifying some of the issues that arise in anti-alias filtering so that you may be aware of them.

The first step in specifying an anti-aliasing filter is to identify how much attenuation is necessary in the stopband. An ADC or DAC can resolve an analog signal only to a finite resolution given by the number of bits that are supported. It is therefore unnecessary to attenuate high-frequency content beyond the point at which the circuit's inherent capabilities reach their limit. Attenuating unwanted signals to less than one-half of a voltage quantum renders them statistically insignificant. If a system represents an analog signal with N bits of resolution, this minimum attenuation, A_{MIN}, is given as

$$A_{MIN} = -20\log(2^{N+1}\sqrt{3})$$

One-half of a quantum is represented by the added power-of-two beyond that specified by the conversion resolution. The $\sqrt{3}$ term represents an allowance for the average quantization noise magnitude. As expected, a higher-resolution conversion requires greater attenuation, because the quantization noise is reduced. An 8-bit ADC, relatively low resolution by modern standards, requires a stopband gain of –59 dB to prevent distortion due to aliasing. A 12-bit ADC, quite moderate in resolution, requires attenuation of 83 dB.

Once the stopband gain is known, the filter's roll-off target can be determined by identifying the separation of the passband and stopband. Filter design is not a trivial process when minimal distortion is a design criterion. A real filter does not have perfectly uniform passband gain and therefore adds some distortion by attenuating some frequencies more than others even in the passband. When the passband and stopband are close together, a higher-order filter is called for to provide the necessary attenuation. Yet, filters with sharp roll-off can have the undesired side effect of exhibiting resonance around the cutoff frequency.

Additionally, filters do not simply change the amplitude of signals that pass through them but change their phase as well by introducing a finite time delay or phase shift. The problem is that filters do not have a uniform phase delay at all frequencies in the passband. Most real signals are not pure sine waves and therefore are composed of many frequencies. The result is distortion of the overall signal as the different frequency components are shifted relative to each other by slightly different delays.

The magnitude of these nonideal filter characteristics depends on each situation. A basic first-order RC filter is very well behaved, which is a key reason why sigma-delta conversion circuits are so popular. The limitation, of course, is that sigma-delta circuits can be used only at lower sampling rates.

If a sigma-delta circuit cannot be used for one reason or another, it may be practical to operate a conventional ADC or DAC at a significantly higher sampling rate than would be dictated by the frequency of interest. This allows a less complex filter to be used by increasing the separation between the signal and Nyquist frequencies. Digital signal processing techniques can then be used to filter the digital samples down to a more ideal sampling rate. The difference here is that the complexity of analog filter design is traded off against a faster data conversion system and some computational number crunching. As logic gates have become inexpensive and microprocessors increase in capability, DSP-based filter algorithms are often far superior to their analog implementations, because the nonideal characteristics of phase-delay, amplitude variance, and resonance are overcome by arithmetic manipulations. A major reason behind the proliferation of high-speed ADC and DAC ICs is the emergence of DSP technology in applications in which fully analog circuits formerly existed. DSP algorithms can implement complex and highly stable filters that are extremely costly to implement with analog components.