Chapter 2 Analog-to-Digital Conversion

This chapter examines general considerations for analog-to-digital converter (ADC) measurements. Discussed are the four basic ADC types, providing a general description of each while comparing their speed and resolution. Issues such as calibration, linearity, missing codes, and noise are discussed, as are their effects on ADC accuracy.

This chapter also includes information on simultaneous sample and hold (SS&H) and selectable input ranges. Finally, this chapter contains a section on discrete sampling, which includes Fourier theory, aliasing, windowing, fast Fourier transforms (FFTs), standard Fourier transforms, and digital filtering.

ADC Types

An ADC converts an analog voltage to a digital number. The digital number represents the input voltage in discrete steps with finite resolution. ADC resolution is determined by the number of bits that represent the digital number. An \( n \)-bit ADC has a resolution of 1 part in \( 2^n \). For example, a 12-bit ADC has a resolution of 1 part in 4096 (\( 2^{12}=4,096 \)). Twelve-bit ADC resolution corresponds to 2.44 mV for a 10V range. Similarly, a 16-bit ADC’s resolution is 1 part in 65,536 (\( 2^{16}=65,536 \)), which corresponds to 0.153 mV for a 10V range.

Many different types of analog-to-digital converters are available. Differing ADC types offer varying resolution, accuracy, and speed specifications. The most popular ADC types are the parallel (flash) converter, the successive approximation ADC, the voltage-to-frequency ADC, and the integrating ADC. Descriptions of each follow.

Parallel (Flash) Converter

The parallel converter is the simplest ADC implementation. It uses a reference voltage at the full scale of the input range and a voltage divider composed of \( 2^n + 1 \) resistors in series, where \( n \) is the ADC resolution in bits. The value of the input voltage is determined by using a comparator at each of the \( 2^n \) reference voltages created in the voltage divider. Figure 2.01 depicts a 2-bit parallel converter.

Flash converters are very fast (up to 500 MHz) because the bits are determined in parallel. This method requires a large number of comparators, thereby limiting the resolution of most parallel converters to 8 bits (256 comparators). Flash converters are commonly found in transient digitizers and digital oscilloscopes.
**Successive Approximation ADC**

A successive approximation ADC employs a digital-to-analog converter (DAC) and a single comparator. It effectively makes a bisection or binomial search by beginning with an output of zero. It provisionally sets each bit of the DAC, beginning with the most significant bit. The search compares the output of the DAC to the voltage being measured. If setting a bit to one causes the DAC output to rise above the input voltage, that bit is set to zero. A diagram of a successive approximation ADC is shown in Figure 2.02.

Successive approximation is slower than flash conversion because the comparisons must be performed in a series, and the ADC must pause at each step to set the DAC and wait for it to settle. However, conversion rates over 200 kHz are common. Successive approximation is relatively inexpensive to implement for 12- and 16-bit resolution. Consequently, they are the most commonly used ADCs, and can be found in many PC-based data acquisition products.

**Voltage-to-Frequency ADC**

Figure 2.03 depicts the voltage-to-frequency technique. Voltage-to-frequency ADCs convert an input voltage to an output pulse train with a frequency proportional to the input voltage. Output frequency is determined by counting pulses over a fixed time interval, and the voltage is inferred from the known relationship.

Voltage-to-frequency conversion has a high degree of noise rejection, because the input signal is effectively integrated over the counting interval. Voltage-to-frequency conversion is commonly used to convert slow and often noisy signals.
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It is also useful for remote sensing applications in noisy environments. The input voltage is converted to a frequency at the remote location, and the digital pulse train is transmitted over a pair of wires to the counter. This eliminates the noise that can be introduced in the transmission of an analog signal over a long distance.

**Integrating ADC**

A number of ADCs use integrating techniques, which measure the time to charge or discharge a capacitor to determine input voltage. Figure 2.04 shows “Dual-slope” integration, a common integration technique. Using a current that is proportional to the input voltage, a capacitor is charged for a fixed time period. The average input voltage is determined by measuring the time required to discharge the capacitor using a constant current.

Integrating the ADC input over an interval reduces the effect of noise pickup at the AC line frequency if the integration time is matched to a multiple of the AC period. For this reason, it is commonly used in precision digital multimeters and panel meters. Twenty-bit accuracy is not uncommon. The disadvantage is a relatively slow conversion rate (60 Hz maximum, slower for ADCs that integrate over multiple line cycles).

**Summary of ADC Types**

Figure 2.05 summarizes the previously discussed ADC types and their resolution and speed ranges.

<table>
<thead>
<tr>
<th>ADC Type</th>
<th>Typical Resolution</th>
<th>Typical Speed</th>
</tr>
</thead>
<tbody>
<tr>
<td>Parallel Converter</td>
<td>4-8 bit</td>
<td>100 kHz-500 MHz</td>
</tr>
<tr>
<td>Successive Approximation</td>
<td>8-16 bit</td>
<td>10 kHz-1 MHz</td>
</tr>
<tr>
<td>Voltage-to-Frequency</td>
<td>8-12 bit</td>
<td>1-60 Hz*</td>
</tr>
<tr>
<td>Integrating</td>
<td>12-24 bit</td>
<td>1-60 Hz*</td>
</tr>
</tbody>
</table>

* With line cycle rejection

**Fig 2.05:** Summary of ADC types
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Accuracy

Accuracy is an important consideration when selecting an ADC for use in test and measurement applications. The following section provides an in-depth discussion of accuracy considerations, pertaining to resolution, calibration, linearity, missing codes, and noise.

Accuracy vs. Resolution

The accuracy of a measurement is influenced by a variety of factors. If each independent error is $\sigma_i$, the total error is

$$\sigma_{\text{total}} = \sqrt{\sum \sigma_i^2}$$

This calculation includes errors resulting from the transducer, noise pickup, ADC quantization, gain, offset, and other factors.

ADC resolution error is termed quantization error. In an ideal ADC, any voltage in the range that corresponds to a unique digital code is represented by that code. The error in this case is half of the least significant bit (LSB) at most. For a 12-bit ADC with a 10V range, this error is 2.44 mV (0.0244%). There are three common methods of specifying a contribution to ADC error: the error in least significant bits (LSBs), the voltage error for a specified range, and the percent-of-reading error. It is important to recognize that most ADCs are not as accurate as their specified resolution, because quantization error is only one potential source of error. Nonetheless, the accuracy of a good ADC should approach its specified resolution.

For more information concerning accuracy, refer to the Calibration section, which follows. For an in-depth discussion of errors arising in particular transducers, refer to Chapters 3 through 8.

Figure 2.06 illustrates common error types encountered when using a 3-bit ADC. If the manufacturer provides calibration procedures, offset and gain errors can usually be reduced to negligible levels, as discussed below. However, errors in linearity and missing codes will contribute to the overall error.

Calibration

There are several common methods for calibrating an ADC. In hardware calibration, the offset and gain of the instrumentation amplifier that serves as the ADC front end is adjusted with trim pots. (The gain of the ADC can also be adjusted by changing the reference voltage.) In hardware/software calibration, digital-to-analog converters that null the offset and set the full scale voltages are programmed via software. In software calibration, there is no hardware adjustment. Calibration correction factors are stored in the nonvolatile memory of the data acquisition system or in the computer and used to convert the reading from the ADC.
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Even if an ADC is calibrated at the factory, it will need to be calibrated again after a period of time (typically six months to a year, but possibly more frequently for ADCs of greater resolution than 16 bits). Variations in the operating temperature can also affect instrument calibration. Calibration procedures vary but usually require either a known reference source or a meter of greater accuracy than the device being calibrated. Typically, offset is set via a 0V input, and gain is set via a full-scale input.

In many measurements, the voltage is not the physical quantity under test. Consequently, it may be preferable to calibrate the complete measurement system rather than its individual parts. For example, consider a load cell for which the manufacturer specifies the output for a given load and excitation voltage. One could calibrate the ADC and combine this with the manufacturer’s specification and a measurement of the excitation voltage; however, this technique is open to error. Specifically, three distinct error sources are possible in this technique: error in the ADC calibration, error in the manufacturer’s specifications, and error in the measurement of the excitation voltage. To circumvent these error sources, one can calibrate the measurement system using known loads and obtain a direct relationship between load and ADC output.

Fig 2.06: Common ADC error sources. The straight line is ideal output from an ADC with infinite-bit resolution. The step function shows the indicated error for a 3-bit ADC.
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Linearity
If input voltage and ADC output deviate from the diagonal lines in Figure 2.06 more than the ideal step function, the result is ADC error that is nearly impossible to eliminate by calibration. This type of ADC error is referred to as nonlinearity error. If nonlinearity is present in a calibrated ADC, the error is often largest near the middle of the input range, as shown in Figure 2.06. The nonlinearity in a good ADC should be 1 LSB or less.

Missing Codes
Some ADCs have missing codes. In Figure 2.06, the ADC does not provide an output of four for any input voltage. This error can result in a significant loss in resolution and accuracy. A quality ADC should have no missing codes.

Noise
Many users are surprised by noise encountered when measuring millivolt signals or attempting accurate measurements on larger signals. Investing in an accurate ADC is only the first step in accurately measuring analog input signals. Controlling noise is imperative.

Many ADCs reside on cards that plug into a PC expansion bus, where electrical noise can present serious problems. Expansion bus noise often far exceeds the ADC’s sensitivity resulting in significant loss of measurement accuracy. Placing the ADC outside the PC is often a better solution. An ADC in an external enclosure can communicate with the computer over an IEEE 488 bus, serial port, or parallel port. If an application requires placement of the ADC within the computer, the noise level should be tested by connecting the ADC input to signal common and observing deviations in ADC output. (Connecting the ADC input to signal common isolates the cause of the noise to the circuit card. More careful diagnostics are necessary when using an external voltage source because noise can arise from the external source and from the input leads.)

Noise Reduction and Measurement Accuracy
One technique for reducing noise and ensuring measurement accuracy is with isolation, which also eliminates ground loops. Ground loops occur when two or more devices in a system, such as a measurement instrument and a transducer, are connected to ground at different physical locations. Slight differences in the actual potential of each ground results in a current flow from one device to the other. This current, which often flows through the low lead of a pair of measurement wires, generates a voltage drop which can directly lead to measurement inaccuracies and noise. If at least one device is isolated, such as the measurement device, then there is no path for the current flow, and thereby no contribution to noise or inaccuracy.
Protection

Many data acquisition systems utilize solid-state multiplexing circuitry in order to very quickly scan multiple input channels. These solid-state multiplexers are among the most susceptible circuitry to overload voltages, which can commonly occur in a system. Typically, multiplexers can only accept up to 20 or 30 volts before damage occurs. Other solid-state devices in a measurement system include input amplifiers and bias sources, both of which are also susceptible to damage from over voltage. Isolation is one of a number of techniques used to protect sensitive solid-state circuitry in a data acquisition system.

Although isolation does not protect against excessive normal-mode input voltage (voltage across a pair of inputs), it does protect against excessive common-mode voltage. It accomplishes this by eliminating the potentially large current that would otherwise flow from the signal source to the data acquisition system, as a result of the common mode voltage. By eliminating this current flow, the possibility of damage is eliminated.

High common-mode voltage measurements

It is often necessary to measure a small voltage which is residing on another, much larger voltage. For example, if a thermocouple is mounted to one terminal of a battery, then the measurement device must be capable of measuring the microvolt output of the thermocouple while rejecting the battery voltage. If the common-mode voltage is less than 10-15V, a differential measurement via an instrumentation amplifier will read the thermocouple voltage while ignoring the battery voltage. If the common-mode voltage is higher than 10-15V, an isolation method is generally required.

There are several isolation methods with the common characteristic of a high common-mode voltage from input to output. Each channel can have an isolation amplifier, or, a group of channels that are not isolated from each other can be multiplexed and digitized by an analog-to-digital converter before the digital data is isolated from the remainder of the system.

Actual isolation barriers can be optical, magnetic, or capacitive. The most common are optical schemes in which
infrared emission from an LED is detected by a photodiode on the opposite side of a quartz barrier. Figure 2.07 illustrates an optically-coupled isolation amplifier. Optocouplers can be used to transmit pulse trains in which frequency or pulse width vary with analog signal magnitude or which contain numerical data in serial pulse trains. It is even possible to transmit analog information by varying LED current as in Figure 2.07. Magnetic barriers using transformers and capacitive barriers are generally internal and used in monolithic or hybrid isolation amplifiers.

**Frequency Coupled Isolation.** In frequency coupled isolation, a high frequency carrier signal is inductively or capacitively coupled across the isolation barrier. The signal is modulated on the input side and demodulated on the output side to reproduce the original input signal.

**Isolated ADC.** When using an isolated ADC, the ADC and accompanying signal conditioning are floated. The input signal is converted to a digital signal by the ADC and the interface for transferring the digital code is digitally isolated. See Chapter 6 for a detailed discussion.

**Discrete Sampling Considerations**

The Nyquist sampling theorem says that if a signal only contains frequencies less than cutoff frequency \( f_c \), all the information in the signal can be captured by sampling at \( 2f_c \). The upshot of this is that capturing a signal with maximum frequency component \( f_{\text{max}} \) requires sampling at a rate of at least \( 2f_{\text{max}} \). In practice, for working in the frequency domain, it is best to set the sampling rate between five and ten times the signal’s highest frequency component. However, for viewing waveforms in the time domain, it is not uncommon to sample 10 times the frequency of interest. One reason is to retain accuracy at the signal’s higher frequency components.

**Aliasing**

Aliasing can also be seen in the time domain. Figure 2.09 shows a 1-kHz sine wave sampled at 800 Hz. The apparent frequency of the sine wave is much too low. Figure 2.08 shows the result of sampling the same 1-kHz sine wave at 5 kHz. The sampled wave appears to have the correct frequency.

Aliasing is the main reason to sample at a rate higher than the Nyquist frequency. Aliasing—the generation of false, low-frequency signals—occurs when an ADC’s sampling rate is too low. Input signals are seldom bandwidth limited with zero amplitude higher than \( f_{\text{max}} \). A signal with frequency components higher than one-half the sampling frequency will cause the amplitude to appear below one-half the sampling frequency in the Fourier transform. This is called aliasing, and it can cause inaccuracies in sampled signal and also in the Fourier transform.
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Aliasing is illustrated in Figure 2.10, which shows a square wave’s Fourier transform. For the purpose of illustration, assume that the experiment has been designed to provide only frequencies of under 2 kHz. Ideally, a Fourier transform of a 500-Hz square wave contains one peak at 500 Hz, and another at 1500 Hz, which is one third the height of the first peak. In Figure 2.10, however, higher frequency peaks are aliased into the Fourier transform’s low-frequency range.

The use of a low-pass filter at 2 kHz, as shown in Figure 2.11, removes most of the aliased peaks. Low-pass filters used for this purpose are often called “anti-aliasing” filters.

When the sampling rate is increased to four times the highest frequency of interest, the Fourier transform in the range of interest looks even better. Although a small peak remains at 1,000 Hz, it is probably the result of an imperfect square wave rather than an effect of aliasing. See Figure 2.12.
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Windowing

Windowing is the multiplication of the input signal with a weighting function to reduce spurious oscillations in a Fourier transform. Real measurements are performed over finite time intervals. In contrast, Fourier transforms are defined over infinite time intervals. As such, the Fourier transform of sampled data is an approximation. Consequently, the resolution of the Fourier transform is limited to roughly 1/T, where T is the finite time interval over which the measurement was made. Fourier transform resolution can only be improved by sampling for a longer interval.

Using a finite time interval also causes spurious oscillations in the Fourier transform. From a mathematical viewpoint, spurious oscillations are caused by the signal being instantaneously turned on at the beginning of the measurement and then suddenly turned off at the end of the measurement. Figure 2.13 illustrates an example of spurious oscillations.

Implementing window functions can help minimize spurious oscillations of a signal. Multiplying the sampled data by a window function that rises gradually from zero decreases the spurious oscillations at the expense of a slight loss in triggering resolution. There are many possible window functions, all of which involve trade-offs between amplitude estimation and frequency resolution.

Fig. 2.11: Fourier transform of 500-Hz square wave sampled at a 4 kHz with low-pass filter cutoff at 2 kHz

Fig. 2.12: Fourier transform of 500-Hz square wave sampled at 8 kHz with low-pass filter cutoff at 2 kHz
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Fast Fourier Transforms

The fast Fourier transform (FFT) is so common today that “FFT” has become an imprecise synonym for Fourier transforms in general. The FFT is a digital algorithm for computing Fourier transforms of data discretely sampled at a constant interval. The FFT’s simplest implementation requires $2^n$ samples. Other implementations accept other special numbers of samples. If the data set to be transformed has a different number of samples than required by the FFT algorithm, the data is often padded with zeros to achieve the required number. This leads to inaccuracies, but they are often tolerable.

Fig. 2.13: A Fourier transform with window function and without window function

Fig. 2.14: Without window function, abrupt beginning and end points of waveform produce erroneous frequency information

Fig. 2.15: When multiplied by a window function, the frequency information is preserved while minimizing the affect of irregularities at the beginning and end of the sample segment

Fig. 2.16: Three common window types
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Standard Fourier Transforms

A standard Fourier transform (SFT) can be used in applications where the number of samples cannot be arranged to fall on one of the special numbers required by an FFT. The SFT is also useful for applications that cannot tolerate the inaccuracies introduced by padding with zeros, a handicap of the FFT.

The SFT is also suited to applications where the data is not sampled at evenly spaced intervals or where sample points are missing. Finally, the SFT can be used to provide more closely spaced points in the frequency domain than can be obtained with an FFT. (In an FFT, adjacent points are separated by half the sampling frequency. Points arbitrarily close in frequency can be obtained using an SFT.)

There are many standard numerical integration techniques available for computing SFTs from sampled data. Whichever technique is selected for the problem at hand, it will probably be much slower than an FFT of a similar number of points. This is becoming less of an issue, however, as the speed of modern computers increases.

Digital Filtering

Digital filtering is accomplished in three steps. First, the signal must be subjected to a Fourier transform. Then, the signal’s amplitude in the frequency domain must be multiplied by the desired frequency response. Finally, the transferred signal must be inverse Fourier transformed back into the time domain. Figure 2.17 shows the effect of digital filtering on the noisy signal. Note that the solid line represents the unfiltered signal, while the two dashed lines represent different digital filters.

![Figure 2.17: The effect of digital filtering on the noisy signal](image)

Digital filtering is advantageous because the filter itself can be easily tailored to any frequency response without introducing phase error. However, one disadvantage of digital filtering is that it cannot be used for anti-aliasing.

Analog Filtering

In contrast to digital filtering, analog filtering can be used for anti-aliasing, but it is more difficult to change the frequency response curves, since all analog filters introduce some element of phase error.
Sampling Hints

There are several important “sampling hints” to observe when designing an application. These hints are not absolutes nor do they guarantee optimal results. However, they do provide a useful starting point for planning frequency analysis of a physical process. These sampling hints include:

- A Fourier transform’s highest meaningful frequency is one-half of the sampling frequency
- The sampling rate should be at least three to five times the highest frequency of interest
- An anti-aliasing low-pass filter is typically required; the cutoff frequency should be close to the signal’s highest frequency of interest
- Digital filtering can be used to smooth the data or to remove noise in a specified range after acquisition; however, aliasing can only be prevented with an analog low-pass filter
- If the phase relationship between multiple signals is important, a simultaneous sample and hold circuit should be used (see multiplexing in Chapter 3)
- Fourier transform resolution is inversely proportional to measurement time; acquiring data over a longer period of time results in narrower peaks in the Fourier transform.