

3

Signal Conversion

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The power of the computer to analyze and visually represent biomedical signals is of little use if the analog biomedical signal cannot be accurately captured and converted to a digital representation. This chapter discusses basic sampling theory and the fundamental hardware required in a typical signal conversion system. Section 3.1 discusses sampling basics in a theoretical way, and section 3.2 describes the circuits required to implement a real signal conversion system. We examine the overall system requirements for an ECG signal conversion system and discuss the possible errors involved in the conversion. We review digital-to-analog and analog-to-digital converters and other related circuits including amplifiers, sample-and-hold circuits, and analog multiplexers.

3.1 SAMPLING BASICS*

The whole concept of converting a continuous time signal to a discrete representation usable by a microprocessor, lies in the fact that we can represent a continuous time signal by its instantaneous amplitude values taken at periodic points in time. More important, we are able to reconstruct the original signal perfectly with just these sampled points. Such a concept is exploited in movies, where individual frames are snapshots of a continuously changing scene. When these individual frames are played back at a sufficiently fast rate, we are able to get an accurate representation of the original scene (Oppenheim and Willsky, 1983).

3.1.1 Sampling theorem

The *sampling theorem* initially developed by Shannon, when obeyed, guarantees that the original signal can be reconstructed from its samples without any loss of information. It states that, for a continuous bandlimited signal that contains no frequency components higher than f_c , the original signal can be completely recovered

* Section 3.1 was written by Annie Foong.

without distortion if it is sampled at a rate of at least $2 \times f_c$ samples/s. A sampling frequency f_s of twice the highest frequency *present* in a signal is called the Nyquist frequency.

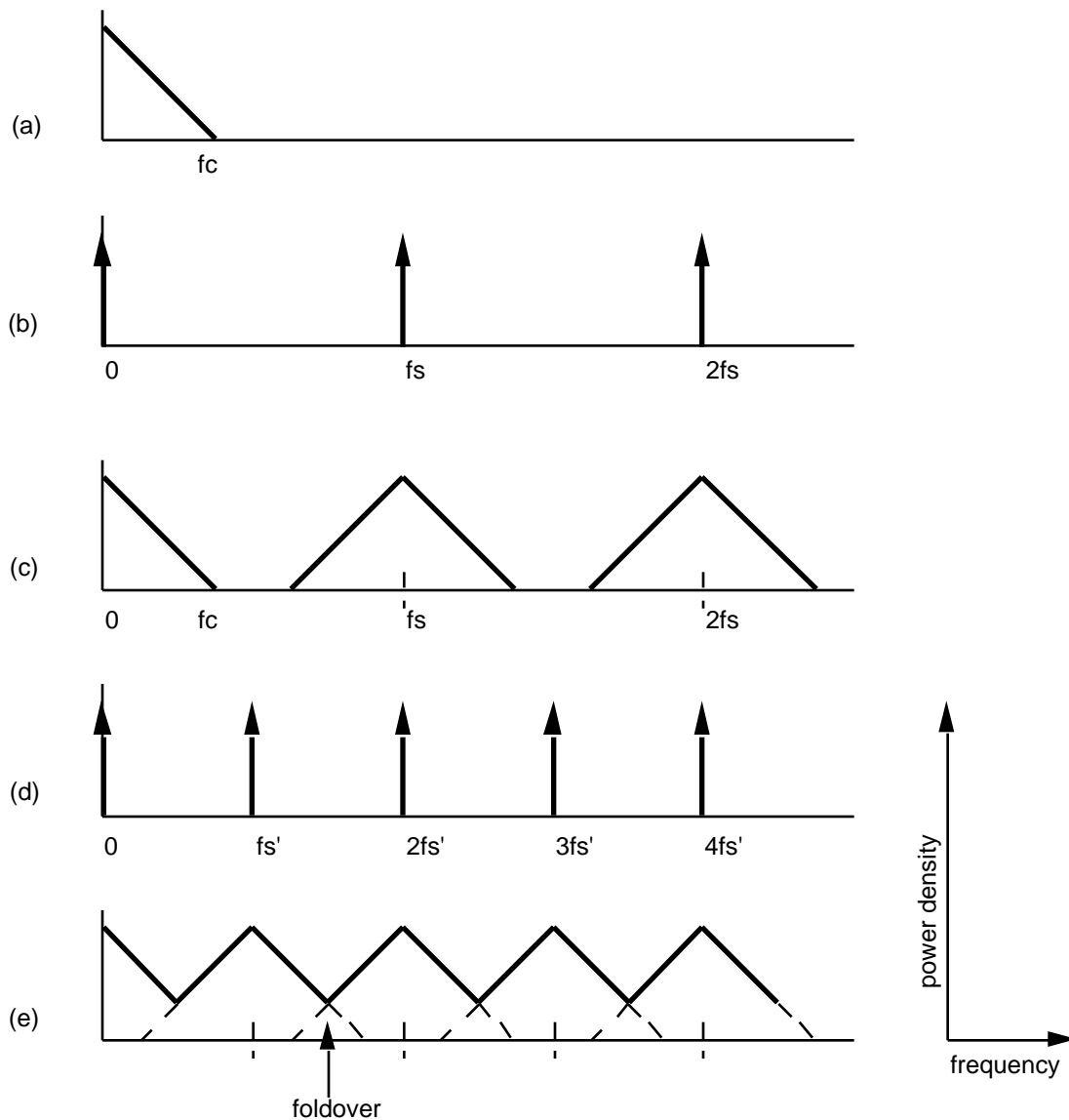


Figure 3.1 Effect in the frequency domain of sampling in the time domain. (a) Spectrum of original signal. (b) Spectrum of sampling function. (c) Spectrum of sampled signal with $f_s > 2f_c$. (d) Spectrum of sampling function with $f_s' < 2f_c$. (e) Spectrum of sampled signal with $f_s' < 2f_c$.

3.1.2 Aliasing, foldover, and other practical considerations

To gain more insight into the mechanics of sampling, we shall work in the frequency domain and deal with the spectra of signals. As illustrated in Figure 3.1(c), if we set the sampling rate larger than $2 \times f_c$, the original signal can be recovered

by placing a low-pass filter at the output of the sampler. If the sampling rate is too low, the situation in Figure 3.1(e) arises. Signals in the overlap areas are *dirtied* and cannot be recovered. This is known as *aliasing* where the higher frequencies are reflected into the lower frequency range (Oppenheim and Willsky, 1983).

Therefore, if we know in advance the highest frequency in the signal, we can theoretically establish the sampling rate at twice the highest frequency present. However, real-world signals are corrupted by noise that frequently contains higher frequency components than the signal itself. For example, if an ECG electrode is placed over a muscle, an unwanted electromyographic (EMG) signal may also be picked up (Cromwell et al., 1976). This problem is usually minimized by placing a low-pass filter at the sampler's input to keep out the unwanted frequencies.

However, nonideal filters can also prevent us from perfectly recovering the original signal. If the input filter is not ideal as in Figure 3.2(a), which is the case in practice, high frequencies may still slip through, and aliasing may still be present.

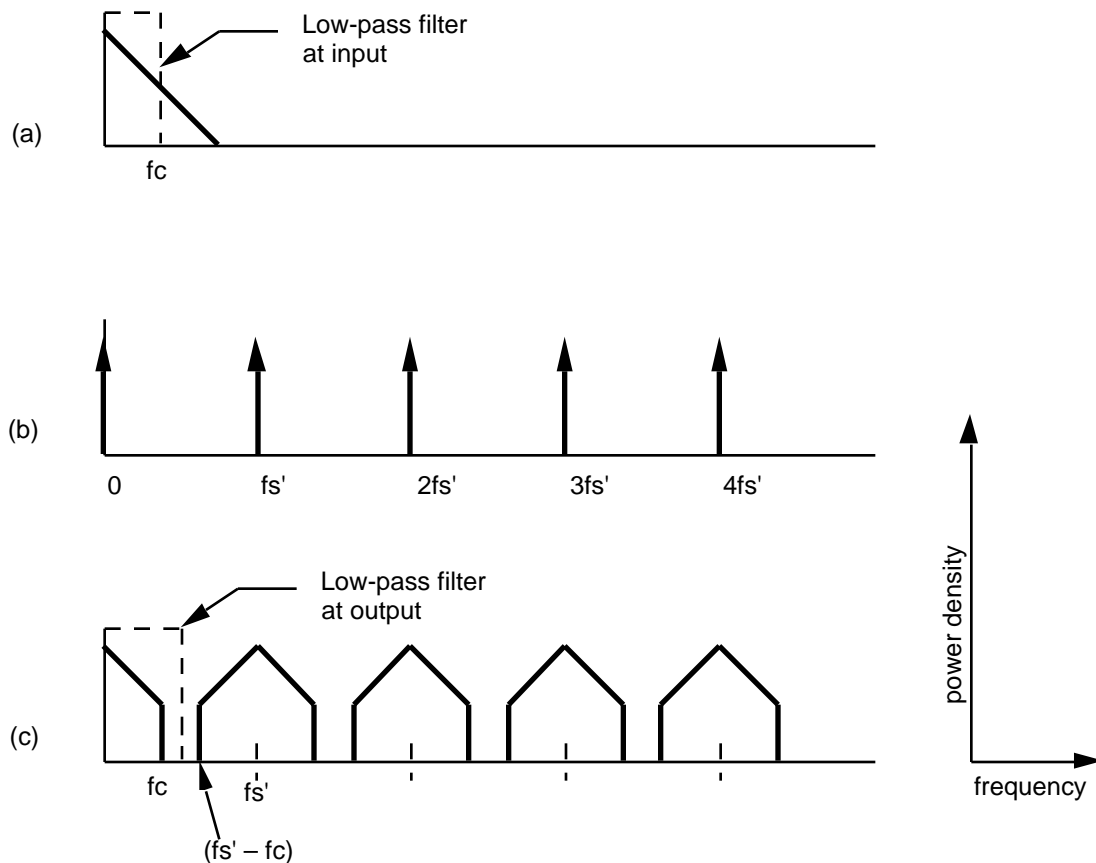


Figure 3.2 Effects of input and output filters. (a) Low-pass filter at input to remove high frequencies of signal. (b) Spectrum of sampling function. (c) No foldover present, low-pass filter at output to recover original signal.

Often ignored is the effect of the output filter. It can be seen in Figure 3.2(c) that, if the output filter is nonideal, the reconstructed signal may not be correct. In

particular, the cutoff frequency of the output filter must be larger than f_c , but smaller than $(f_s' - f_c)$ so as not to include undesired components from the next sequence of spectra.

Finally, we may be limited by practical considerations not to set the sampling rate at Nyquist frequency even if we do know the highest frequency present in a signal. Most biomedical signals are in the low-frequency range. Higher sampling rates require more expensive hardware and larger storage space. Therefore, we usually tolerate an acceptable level of error in exchange for more practical sampling rates.

3.1.3 Examples of biomedical signals

Figure 3.3 gives the amplitudes and frequency bands of several human physiological signals.

Electroencephalogram (EEG)	Frequency range: dc–100 Hz (0.5–60 Hz)	Signal range: 15–100 mV
Electromyogram (EMG)	Frequency range: 10–200 Hz	Signal range: function of muscle activity and electrode placement
Electrocardiogram (ECG)	Frequency range: 0.05–100 Hz	Signal range: 10 μ V(fetal), 5 mV(adult)
Heart rate	Frequency range: 45–200 beats/min	
Blood pressure	Frequency range: dc–200 Hz (dc–60 Hz)	Signal range: 40–300 mm Hg (arterial); 0–15 mm Hg (venous)
Breathing rate	Frequency range: 12–40 breaths/min	

Figure 3.3 Biomedical signals and ranges; major diagnostic range shown in brackets.

3.2 SIMPLE SIGNAL CONVERSION SYSTEMS

Biomedical signals come in all shapes and sizes. However, to capture and analyze these signals, the same general processing steps are required for all the signals. Figure 3.4 illustrates a general analog-to-digital (A/D) signal conversion system.

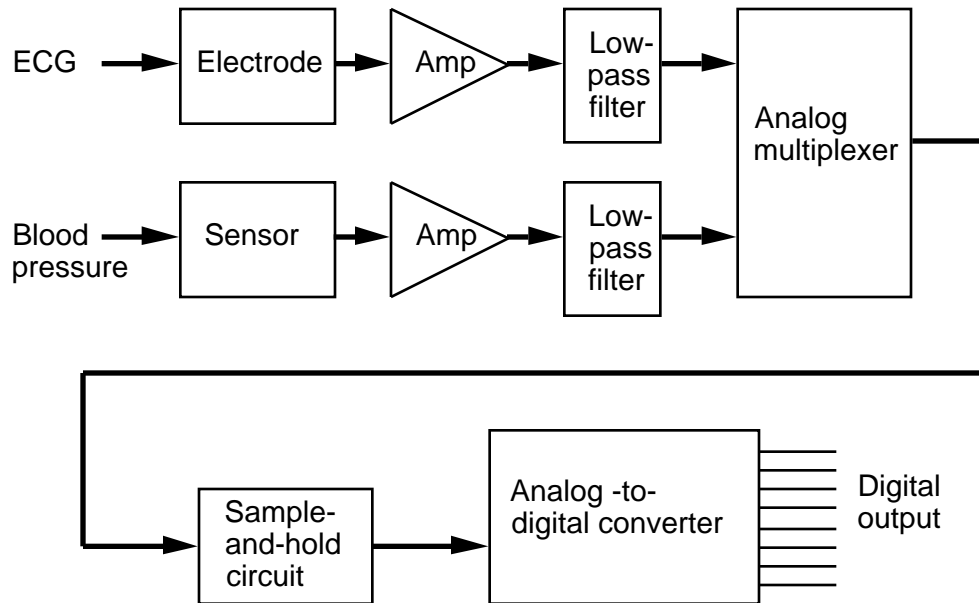


Figure 3.4 A typical analog-to-digital signal conversion system consists of sensors, amplifiers, multiplexers, a low-pass filter, a sample-and-hold circuit, and the A/D converter.

First the signal must be captured. If it is electrical in nature, a simple electrode can be used to pass the signal from the body to the signal conversion system. For other signals, a sensor is required to convert the biomedical signal into a voltage. The signal from the electrode or sensor is usually quite small in amplitude (e.g., the ECG ranges from 10 μV to 5 mV). Amplification is necessary to bring the amplitude of the signal into the range of the A/D converter. The amplification should be done as close to the signal source as possible to prevent any degradation of the signal. If there are several input signals to be converted, an analog multiplexer is needed to route each signal to the A/D converter. In order to minimize aliasing, a low-pass filter is often used to bandlimit the signal prior to sampling. A sample-and-hold circuit is required (except for very slowly changing signals) at the input to the A/D converter to hold the analog signal at a constant value during the conversion process. Finally, the A/D converter changes the analog voltage stored by the sample-and-hold circuit to a digital representation.

Now that a digital version of the biomedical signal has been obtained, what can it be used for? Often the digital information is stored in a memory device for later processing by a computer. The remaining chapters discuss in detail a variety of digital signal processing algorithms commonly used for processing biomedical signals. In some cases this processing can be done in real time. Another possible use for the digital signal is in a control system. In this case the signal is processed by a computer and then fed back to the device to be controlled. Often the controller requires an analog signal, so a digital-to-analog (D/A) converter is needed. Figure 3.5 illustrates a general D/A conversion system. The analog output might be used to control the flow of gases in an anesthesia machine or the temperature in an incubator.

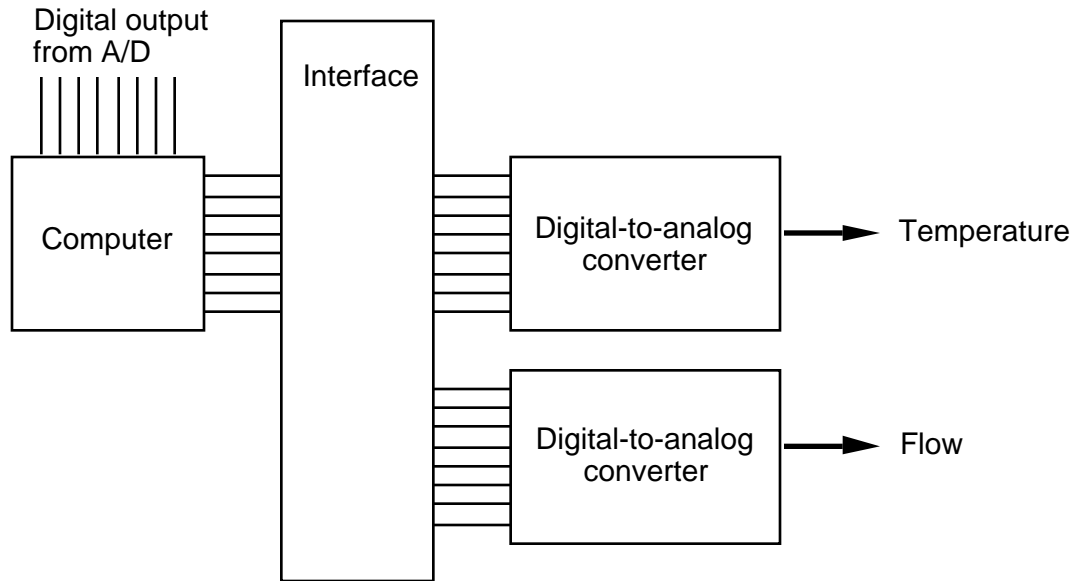


Figure 3.5 A typical digital-to-analog signal conversion system with a computer for processing the digital signal prior to conversion.

3.3 CONVERSION REQUIREMENTS FOR BIOMEDICAL SIGNALS

As discussed in section 3.1.3, biomedical signals have a variety of characteristics. The ultimate goal of any conversion system is to convert the biomedical signal to a digital representation with a minimal loss of information. The specifications for any conversion system are dependent on the signal characteristics and the application. In general, from section 3.1.3, one can see that biomedical signals are typically low frequency and low amplitude in nature. The following attributes should be considered when designing a conversion system: (1) accuracy, (2) sampling rate, (3) gain, (4) processing speed, (5) power consumption, and (6) size.

3.4 SIGNAL CONVERSION CIRCUITS

The digital representation of a continuous analog signal is discrete in both time (determined by the sampling rate) and amplitude (determined by the number of bits in a sampled data word). A variety of circuit configurations are available for converting signals between the analog and digital domains. Many of these are discussed in this chapter. Each method has its own advantages and shortcomings. The discussion here is limited to those techniques most commonly used in the conversion of biomedical signals. The D/A converter is discussed first since it often forms part of an A/D converter.

3.4.1 Converter characteristics

Before describing the details of the converter hardware, it is important to gain some knowledge of the basic terminology used in characterizing a converter's performance. One common method of examining the characteristics of a D/A converter (or an A/D converter) is by looking at its static and dynamic properties as described by Allen and Holberg (1987).

Static

For illustrative purposes a D/A converter is used, but the static errors discussed also apply to A/D converters. The ideal static behavior of a 3-bit D/A converter is shown in Figure 3.6. All the combinations of the digital input word are on the horizontal axis, while the analog output is on the vertical axis. The maximum analog output signal is $7/8$ of the full scale reference voltage V_{ref} . For each unique digital input there should be a unique analog output. Any deviations from Figure 3.6 are known as static conversion errors. Static conversion errors can be divided into integral linearity, differential linearity, monotonicity, and resolution.

Integral linearity is the maximum deviation of the output of the converter from a straight line drawn from its ideal minimum to its ideal maximum. Integral linearity is often expressed in terms of a percentage of the full scale range or in terms of the least significant bit (LSB). Integral linearity can be further divided into absolute linearity, offset or zero error, full scale error, gain error, and monotonicity errors. Absolute linearity emphasizes the zero and full scale errors. The zero or offset error is the difference between the actual output and zero when the digital word for a zero output is applied. The full scale error is the difference between the actual and the ideal voltage when the digital word for a full scale output is applied. A gain error exists when the slope of the actual output is different from the slope of the ideal output. Figure 3.7(a) illustrates offset and gain errors. Monotonicity in a D/A converter means that as the digital input to the converter increases over its full scale range, the analog output never exhibits a decrease between one conversion step and the next. In other words, the slope of the output is never negative. Figure 3.7(b) shows the output of a converter that is nonmonotonic.

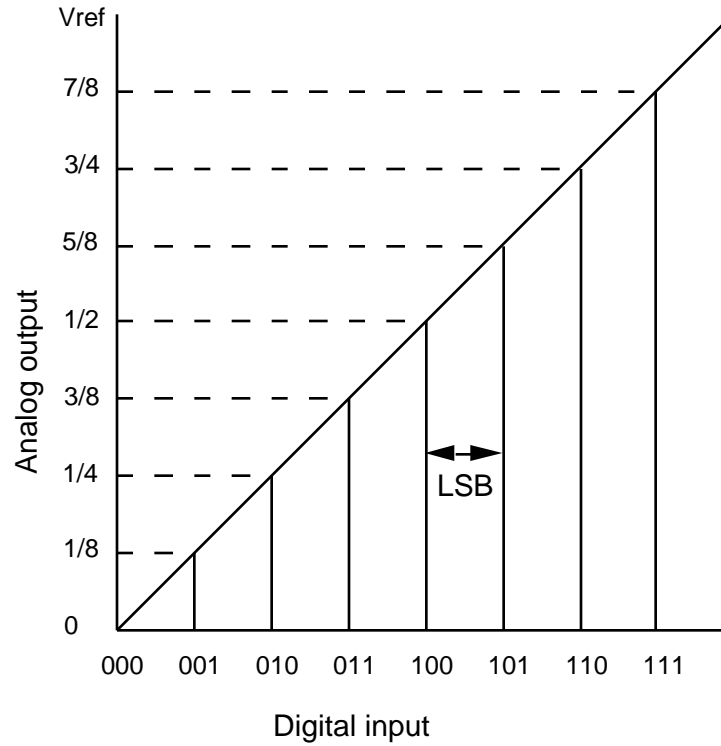


Figure 3.6 The ideal static behavior for a three-bit D/A converter. For each digital word there should be a unique analog signal.

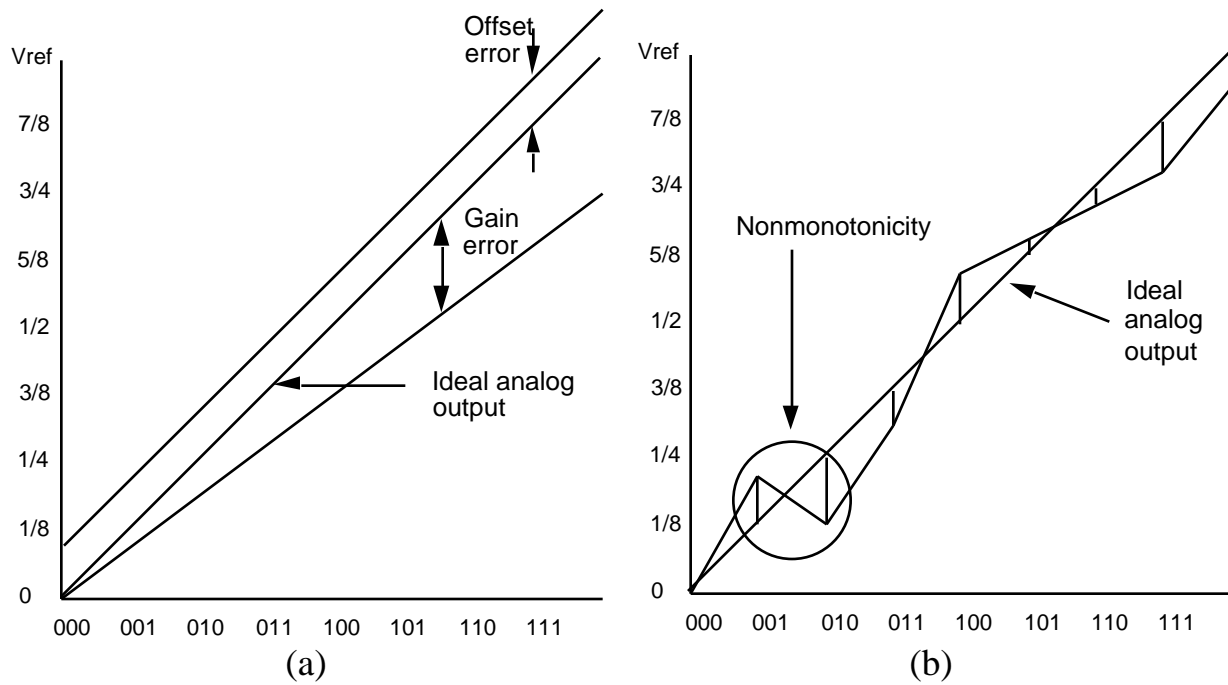


Figure 3.7 Digital-to-analog converter characteristics. (a) Gain and offset errors. (b) Monotonicity errors.

Differential linearity differs from integral linearity in that it is a measure of the separation between adjacent levels (Allen and Holberg, 1987). In other words, differential linearity measures bit-to-bit deviations from the ideal output step size of 1 LSB. Figure 3.8 illustrates the differences between integral and differential linearity.

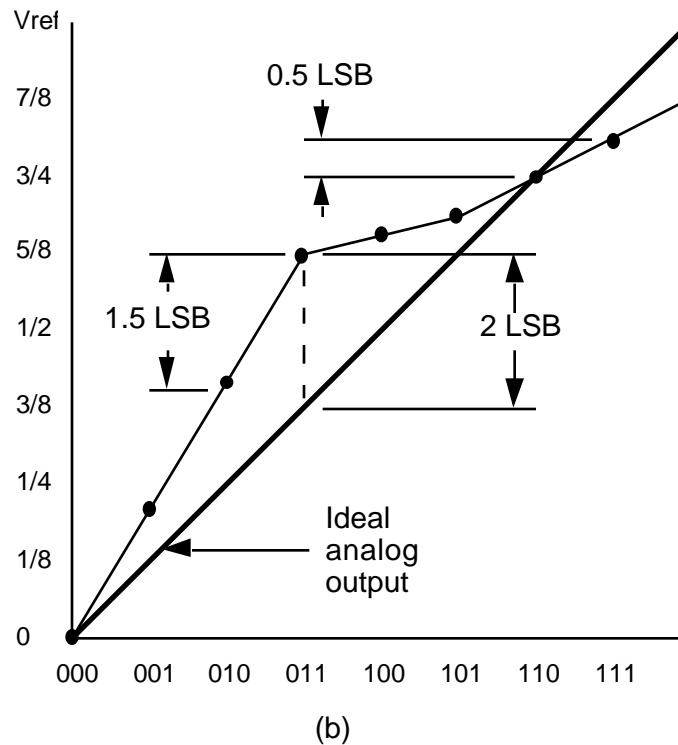


Figure 3.8 A D/A converter with ± 2 LSB integral linearity and ± 0.5 LSB differential linearity.

Resolution is defined as the smallest input digital code for which a unique analog output level is produced. Theoretically the resolution of an n -bit D/A converter is 2^n discrete analog output levels. In reality, the resolution is often less due to noise and component drift.

Dynamic

Settling time, for a D/A converter, is defined as the time it takes for the converter to respond to an input change. More explicitly, settling time is the time between the time a new digital signal is received at the converter input and the time when the output has reached its final value (within some specified tolerance). Many factors affect the settling time, so the conditions under which the settling time was found must be clearly stated if comparisons are to be made. Factors to note include the magnitude of the input change applied and the load on the output. Settling time in D/A converters is important as it relates directly to the speed of the conversion. The output must settle sufficiently before it can be used in any further processing.

3.4.2 Digital-to-analog converters

The objective of the D/A converter is to construct an analog output signal for a given digital input. The first requirement for a D/A converter is an accurate voltage reference. Next the reference must be scaled to provide analog outputs at levels corresponding to each possible digital input. This is usually implemented using either voltage or charge scaling. Finally, the output can be interpolated to provide a smooth analog output.

Voltage reference

An accurate voltage reference is essential to the operation of a D/A converter. The analog output is derived from this reference voltage. Common reference voltage errors are either due to initial adjustments or generated by drifts with time and temperature. Two types of voltage references are used. One type uses the reverse breakdown voltage of a zener diode, while the other type derives its reference voltage from the extrapolated band-gap voltage of silicon. Temperature compensation is used in both cases. Scaling of this reference voltage is usually accomplished with passive components (resistors for voltage scaling and capacitors for charge scaling).

Voltage scaling

Voltage scaling of the reference voltage uses series resistors connected between the reference voltage and ground to selectively obtain discrete voltages between these limits. Figure 3.9 shows a simple voltage scaling 3-bit D/A converter. The digital input is decoded to select the corresponding output voltage. The voltage scaling structure is very regular and uses a small range of resistances. This is well suited to integrated circuit technology. Integrated circuit fabrication is best at making the same structure over and over. So while control over the absolute value of a resistor might be as high as 50 percent, the relative accuracy can be as low as one percent (Allen and Holberg, 1987). That is, if we fabricate a set of resistors on one piece of silicon, each with a nominal value of $10\text{ k}\Omega$, the value of each might actually be $15\text{ k}\Omega$, but they will all be within one percent of each other in value.

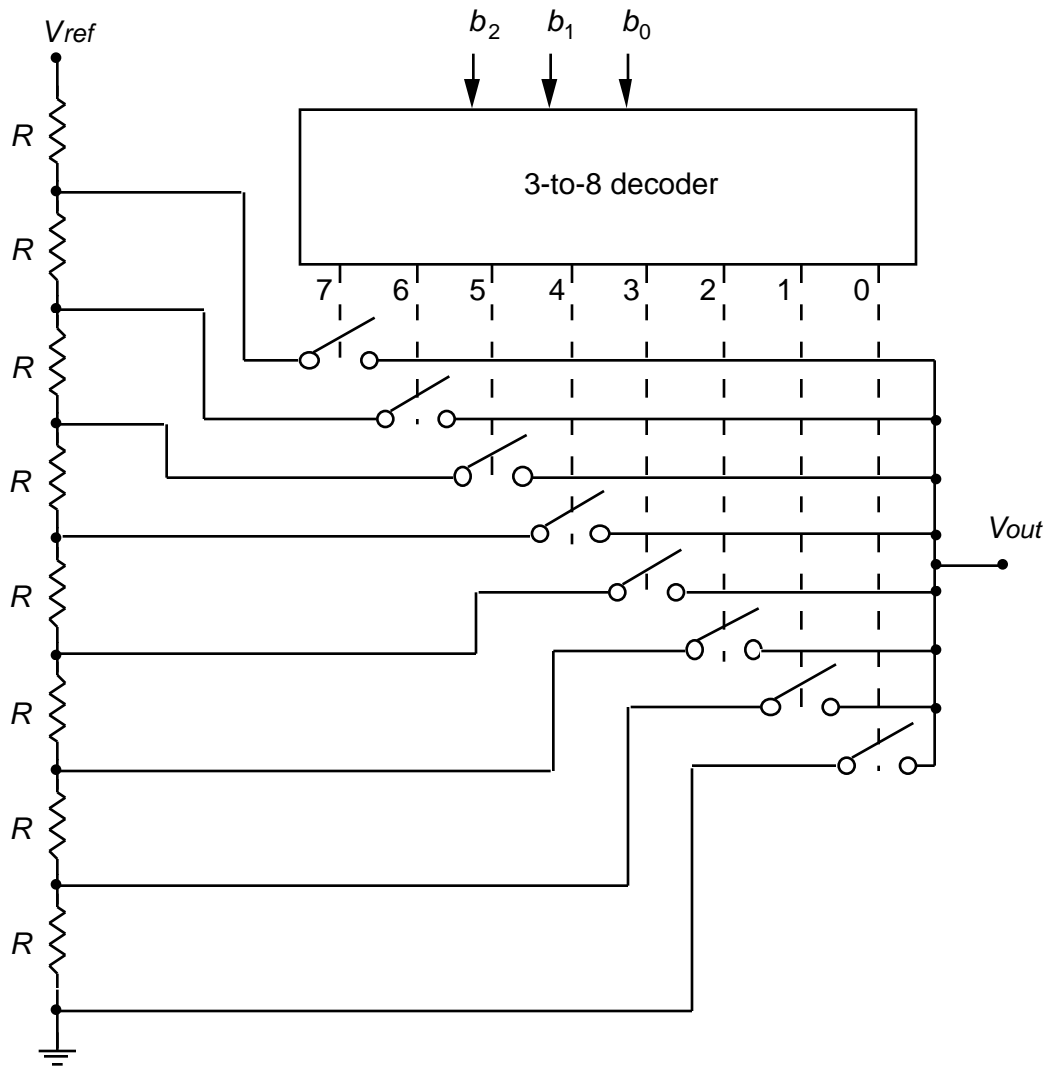


Figure 3.9 A simple voltage scaling D/A converter.

Charge scaling

Charge scaling D/A converters operate by doing binary division of the total charge applied to a capacitor array. Figure 3.10 shows a 3-bit charge scaling D/A converter. A two-phase clock is used. During phase 1, S_0 is closed and switches b_2 , b_1 , b_0 are closed shorting all the capacitors to ground. During phase 2 the capacitors associated with bits that are “1” are connected to V_{ref} and those with bits that are “0” are connected to ground. The output is valid only during phase 2. Equations (3.1) and (3.2) describe this operation. Note that the total capacitance is always $2C$ regardless of the number of bits in the word to be converted. The accuracy of charge scaling converters depends on the capacitor ratios. The error ratio for integrated circuit capacitors is frequently as low as 0.1 percent (Allen and Holberg, 1987).

$$C_{eq} = b_0 C + b_1 \frac{C}{2} + b_2 \frac{C}{4} \quad (3.1)$$

$$V_{out} = \frac{C_{eq}}{2C} \quad (3.2)$$

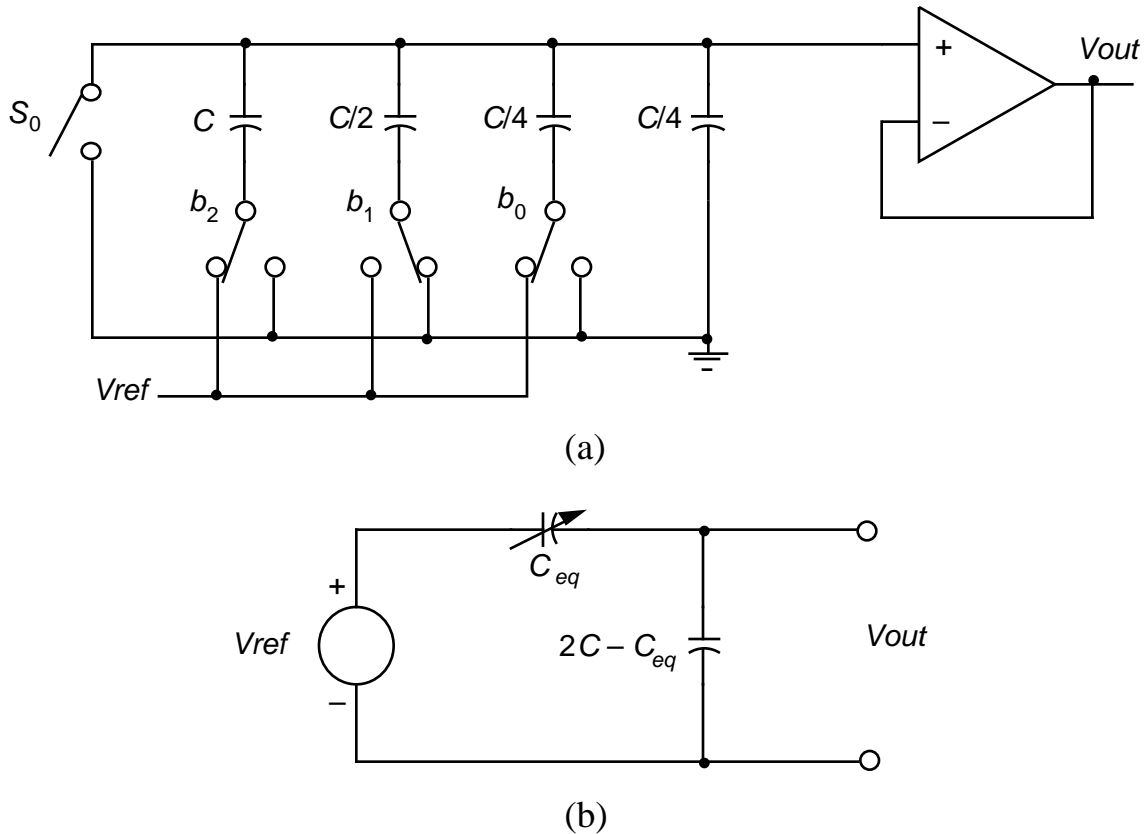


Figure 3.10 A 3-bit charge scaling D/A converter. a) Circuit with a binary 101 digital input. b) Equivalent circuit with any digital input.

Output interpolation

The outputs of simple D/A converters, such as those shown in Figures 3.9 and 3.10, are limited to discrete values. Interpolation techniques are often used to reconstruct an analog signal. Interpolation methods that can be easily implemented with electronic circuits include the following techniques: (1) zero-order hold or one-point, (2) linear or two-point, (3) bandlimited or low-pass (Tompkins and Webster, 1981).

3.4.3 Analog-to-digital converters

The objective of an A/D converter is to determine the output digital word for a given analog input. As mentioned previously, A/D converters often make use of D/A converters. Another method commonly used involves some form of integration or ramping. Finally, for high-speed sampling, a parallel or flash converter is used. In most converters a sample-and-hold circuit is needed at the input since it is not possible to convert a changing input signal. For very slowly changing signals, a sample-and-hold circuit is not always required. The errors associated with A/D converters are similar to those described in section 3.4.1 if the input and output definitions are interchanged.

Counter

The counter A/D converter increments a counter to build up the internal output one LSB at a time until it equals the analog input signal. A comparator stops the counter when the internal output has built up to the input signal level. At this point the count equals the digital output. The disadvantage of this scheme is a conversion time that varies with the level of the input signal. So for low-amplitude signals the conversion time can be fast, but if the signal amplitude doubles the conversion time will also double. Also, the accuracy of the conversion is subject to the error in the ramp generation.

Tracking

A variation of the counter A/D converter is the tracking A/D converter. While the counter converter resets its internal output to zero after each conversion, the internal output in the tracking converter continues to follow the analog input. Figure 3.11 illustrates this difference. By externally stopping the tracking A/D converter, it can be used as a sample-and-hold circuit with a digital output. Also by disabling the up or the down control, the tracking converter can be used to find the maximum or minimum value reached by the input signal over a given time period (Tompkins and Webster, 1988).

Dual slope

In a dual-slope converter, the analog input is integrated for a fixed interval of time (T_1). The length of this time is equal to the maximum count of the internal counter. The charge accumulated on the integrator's capacitor during this integration time is proportional to input voltage according to

$$Q = CV \quad (3.3)$$

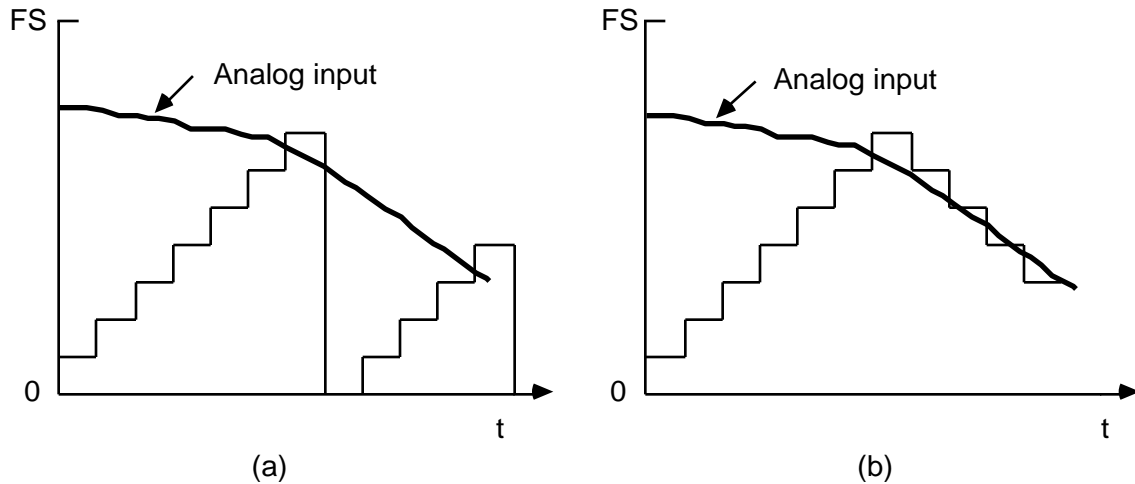


Figure 3.11 Internal outputs of the counter and tracking converters. a) The internal output of a counter A/D converter resets after each conversion. b) The internal output of the tracking A/D converter follows the analog input.

The slope of the integrator output is proportional to the amplitude of the analog input. After time T_1 the input to the integrator is switched to a negative reference voltage V_{ref} , thus the integrator integrates negatively with a constant slope. A counter counts the time t_2 that it takes for the integrator to reach zero. The charge gained by the integrator capacitor during T_1 must equal the charge lost during t_2

$$T_1 V_{in(avg)} = t_2 V_{ref} \quad (3.4)$$

Note that the ratio of t_2 to T_1 is also the ratio of the counter values

$$\frac{t_2}{T_1} = \frac{V_{in}}{V_{ref}} = \frac{\text{counter}}{\text{fixed count}} \quad (3.5)$$

So the count at the end of t_2 is equal to the digital output of the A/D converter. Figure 3.12 shows a block diagram of a dual-slope A/D converter and its associated waveforms. Note that the output of the dual-slope A/D converter is not a function of the slope of the integrator nor of the clock rate. As a result, this method of conversion is very accurate.

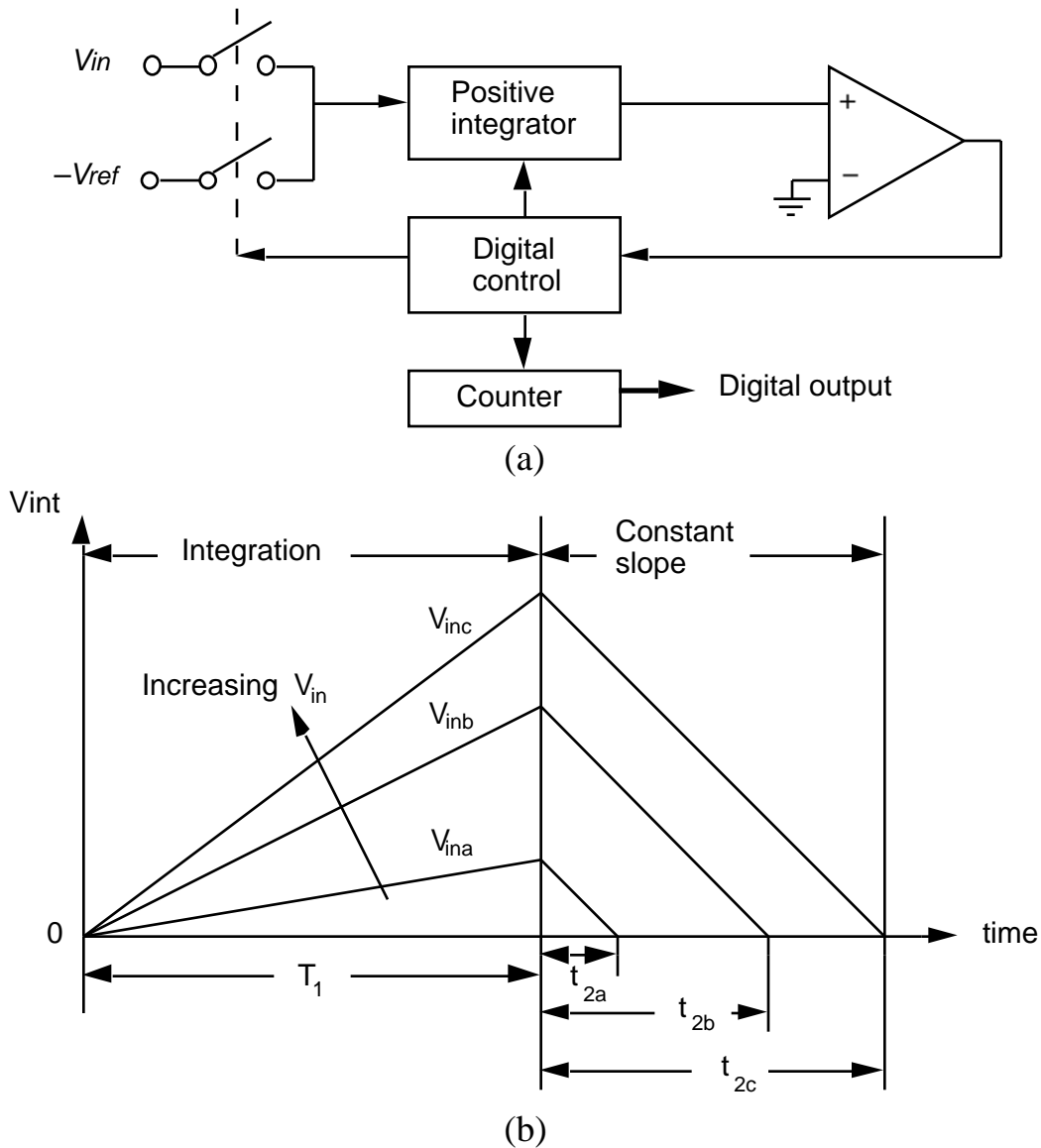


Figure 3.12 Dual-slope A/D converter. a) Block diagram. b) Waveforms illustrate the operation of the converter.

Successive approximation converter

The successive approximation converter uses a combination of voltage-scaling and charge-scaling D/A converters. Figure 3.13 shows a block diagram of a typical successive approximation A/D converter. It consists of a comparator, a D/A converter, and digital control logic. The conversion begins by sampling the analog signal to be converted. Next, the control logic assumes that the MSB is "1" and all other bits are "0". This digital word is applied to the D/A converter and an internal analog signal of $0.5 V_{ref}$ is generated. The comparator is now used to compare this generated analog signal to the analog input signal. If the comparator output is high, then the MSB is indeed "1". If the comparator output is "0", then the MSB is

changed to “0”. At this point the MSB has been determined. This process is repeated for each remaining bit in order.

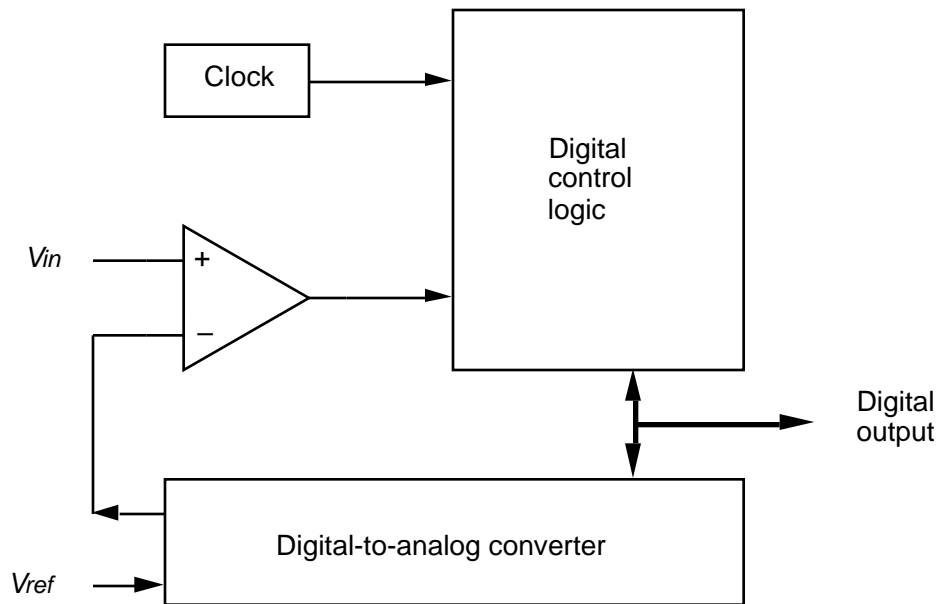


Figure 3.13 Block diagram of a typical successive approximation A/D converter.

Figure 3.14 shows the possible conversion paths for a 3-bit converter. Note that the number of clock cycles required to convert an n -bit word is n .

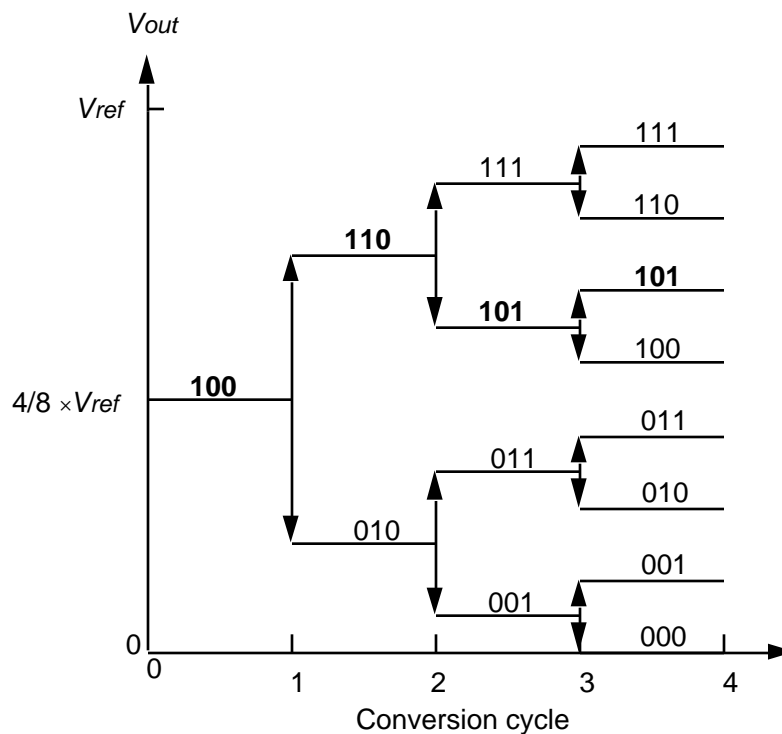


Figure 3.14 The successive approximation process. The path for an analog input equal to $5/8 \times V_{ref}$ is shown in bold.

Parallel or flash

For very high speed conversions a parallel or flash type converter is used. The ultimate conversion speed is one clock cycle, which would consist of a setup and convert phase. In this type of converter the sample time is often the limiting factor for speed. The operation is straightforward and it is illustrated in Figure 3.15. To convert an n -bit word, $2^n - 1$ comparators are required.

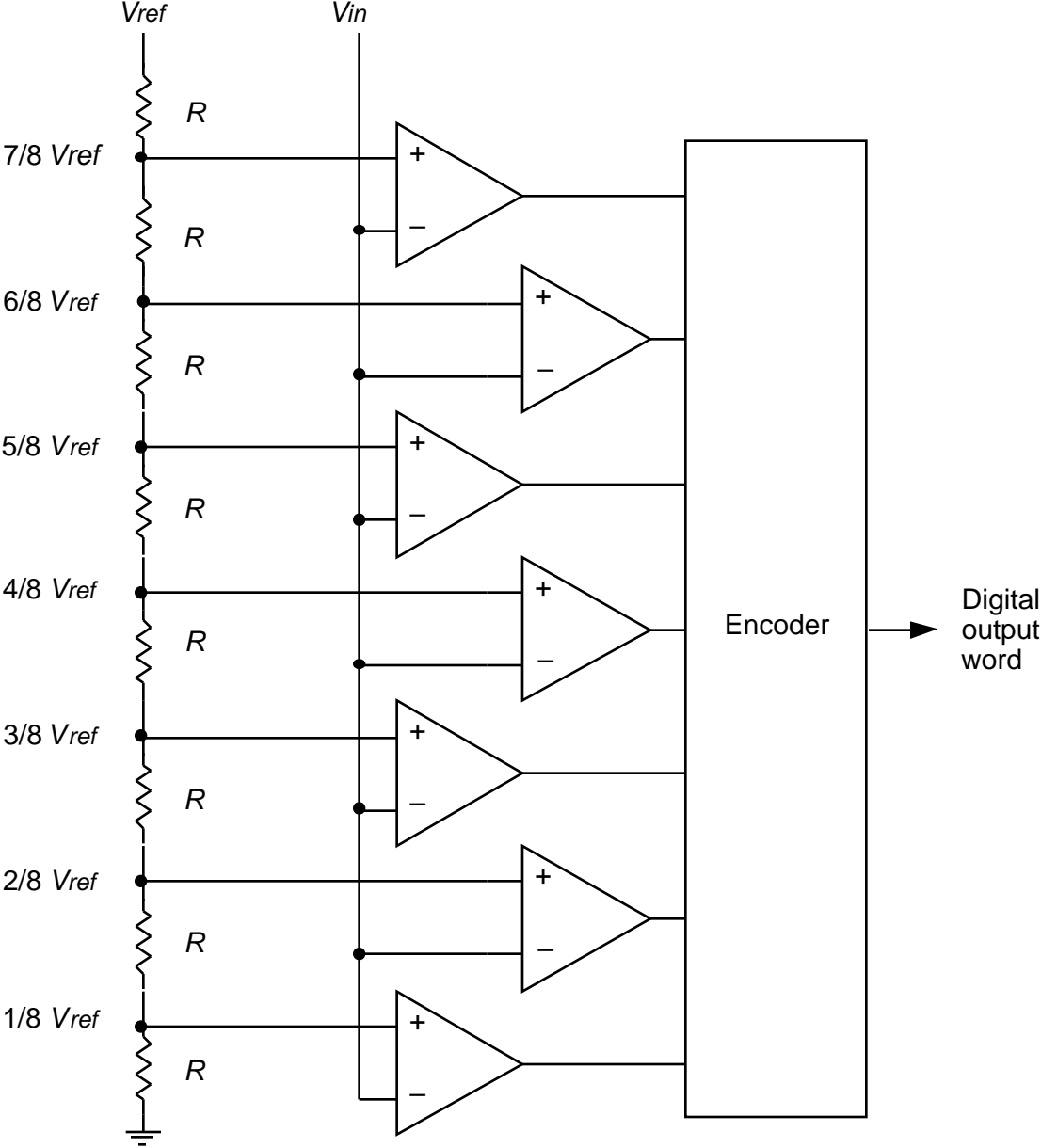


Figure 3.15 A 3-bit flash A/D converter.

3.4.4 Sample-and-hold circuit

Since the conversion from an analog signal to a digital signal takes some finite amount of time, it is advantageous to hold the analog signal at a constant value during this conversion time. Figure 3.16 shows a simple sample-and-hold circuit that can be used to sample the analog signal and freeze its value while the A/D conversion takes place.

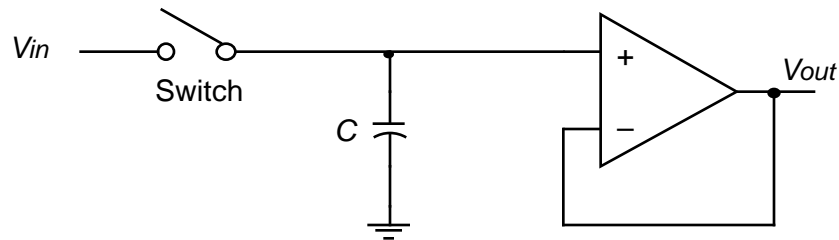


Figure 3.16 A simple implementation of a sample-and-hold circuit.

Errors introduced by the sample-and-hold circuit include offset in the initial voltage storage, amplifier drift, and the slow discharge of the stored voltage. The dynamic properties of the sample-and-hold circuit are important in the overall performance of an A/D converter. The time required to complete one sample determines the minimum conversion time for the A/D. Figure 3.17 illustrates the acquisition time (t_a) and the settling time (t_s).

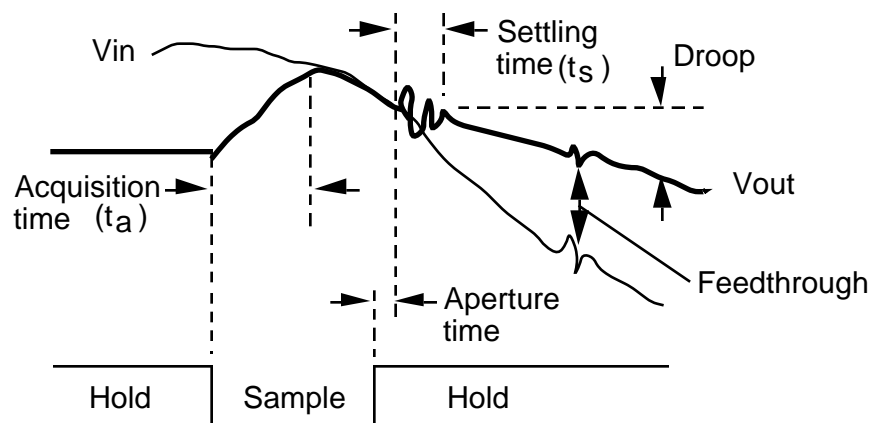


Figure 3.17 Sample-and-hold input and output voltages illustrate specifications.

3.4.5 Analog multiplexer

When several signals need to be converted, it is necessary to either provide an A/D converter for each signal or use an analog multiplexer to direct the various signals to a single converter. For most biomedical signals, the required conversion rates are low enough that multiplexing the signals is the appropriate choice.

Common analog multiplexers utilize either JFET or CMOS transistors. Figure 3.18 shows a simple CMOS analog switch circuit. A number of these switches are connected to a single V_{out} to make a multiplexer. The switches should operate in a break-before-make fashion to ensure that two input lines are not shorted together. Other attributes to be considered include on-resistance, leakage currents, crosstalk, and settling time.

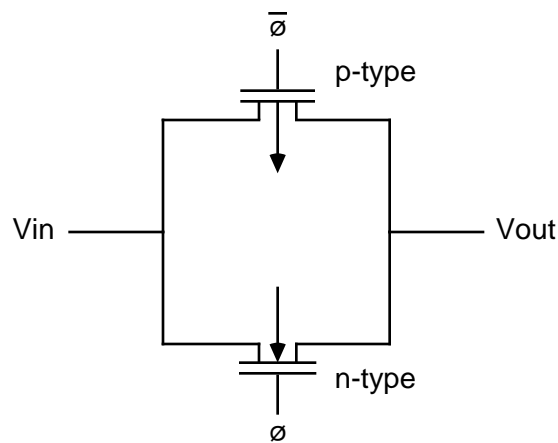


Figure 3.18 A simple CMOS analog switch. The basic functional block of a CMOS analog multiplexer.

3.4.6 Amplifiers

The biomedical signal produced from the sensor or electrode is typically quite small. As a result, the first step before the A/D conversion process is often amplification. Analog amplification circuits can also provide filtering. Analog filters are often used to bandlimit the signal prior to sampling to reduce the sampling rate required to satisfy the sampling theorem and to eliminate noise.

General

For common biomedical signals such as the ECG and the EEG, a simple instrumentation amplifier is used. It provides high input impedance and high CMRR (Common Mode Rejection Ratio). Section 2.4 discusses the instrumentation amplifier and analog filtering in more detail.

Micropower amplifiers

The need for low-power devices in battery-operated, portable and implantable biomedical devices has given rise to a class of CMOS amplifiers known as micropower devices. Micropower amplifiers operate in the weak-inversion region of transistor operation. This operation greatly reduces the power-supply currents required and also allows for operation on very low supply voltages (1.5 V or even lower). Obviously at such low supply voltages the signal swings must be kept small.

3.5 LAB: SIGNAL CONVERSION

This lab demonstrates the effect of the sampling rate on the frequency spectrum of the signal and illustrates the effects of aliasing in the frequency domain.

3.5.1 Using the Sample utility

1. Load the UW DigiScope program using the directions in Appendix D, select **ad(v) Ops**, then **(s)ample**. The Sample menu allows you to read and display a waveform data file, sample the waveform at various rates, display the sampled waveform, recreate a reconstructed version of the original waveform by interpolation, and find the power spectrum of this waveform. The following steps illustrate these functions. You should use this as a tutorial. After completing the tutorial, use the sample functions and go through the lab procedure. Three data files are available for study: a sine wave, sum of sine waves of different frequencies, and a square wave. The program defaults to the single sine wave.

2. The same waveform is displayed on both the upper and lower channels. Since a continuous waveform cannot be displayed, a high sampling rate of 5000 Hz is used. Select **(P)wr Spect** from the menu to find the spectrum of this waveform. Use the **(M)easure** option to determine the frequency of the sine wave.

3. Select **(s)ample**. Type the desired sampling rate in samples/s at the prompt (try **1000**) and hit **RETURN**. The sampled waveform is displayed in the time domain on the bottom channel.

4. Reconstruct the sampled signal using **(R)ecreate** followed by the zero-order hold **(z)oh** option. This shows the waveform as it would appear if you directly displayed the sampled data using a D/A converter.

5. Select **(P)wr Spect** from the menu to find the spectrum of this waveform. Note that the display runs from 0 to one-half the sampling frequency selected. Use the **(M)easure** option to determine the predominant frequencies in this waveform. What is the difference in the spectrum of the signal before and after sampling and reconstruction?

6. These steps may be repeated for three other waveforms with the **(D)ata Select** option.

3.5.2 Procedure

1. Load the sine wave and measure its period. Sample this wave at a frequency much greater than the Nyquist frequency (e.g., 550 samples per second) and reconstruct the waveform using the zero-order hold command. What do you expect the power spectrum of the sampled wave to look like? Perform `(P)wr spect` on the sampled data and explain any differences from your expectations. Measure the frequencies at which the peak magnitudes occur.

2. Sample at sampling rates 5–10 percent above, 5–10 percent below, and just at the Nyquist frequency. Describe the appearance of the sampled data and its power spectrum. Measure the frequencies at which peaks in the response occur. Sample the sine wave at the Nyquist frequency several times. What do you notice? Can the signal be perfectly reproduced?

3. Recreate the original data by interpolation using zero-order hold, linear and sinusoidal interpolation. What are the differences between these methods? What do you expect the power spectrum of the recreated data will look like?

4. Repeat the above steps 1–3 using the data of the sum of two sinusoids.

5. Repeat the above steps 1–3 using the square wave data.

3.6 REFERENCES

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3.7 STUDY QUESTIONS

- 3.1 What is the purpose of using a low-pass filter prior to sampling an analog signal?
- 3.2 Draw D/A converter characteristics that illustrate the following errors: (1) offset error of one LSB, (2) integral linearity of ± 1.5 LSB, (3) differential linearity of ± 1 LSB.
- 3.3 Design a 4-bit charge scaling D/A converter. For $V_{ref} = 5$ V, what is V_{out} for a digital input of 1010?
- 3.4 Why is a dual-slope A/D converter considered an accurate method of conversion? What will happen to the output if the integrator drifts over time?
- 3.5 Show the path a 4-bit successive approximation A/D converter will make to converge given an analog input of $9/16 V_{ref}$.
- 3.6 Discuss reasons why each attribute listed in section 3.3 is important to consider when designing a biomedical conversion system. For example, size would be important if the device was to be portable.
- 3.7 List the specifications for an A/D conversion system that is to be used for an EEG device.
- 3.8 Draw a block diagram of a counter-type A/D converter.

- 3.9 Explain Shannon's sampling theorem. If only two samples per cycle of the highest frequency in a signal is obtained, what sort of interpolation strategy is needed to reconstruct the signal?
- 3.10 A 100-Hz-bandwidth ECG signal is sampled at a rate of 500 samples per second. (a) Draw the approximate frequency spectrum of the new digital signal obtained after sampling, and label important points on the axes. (b) What is the bandwidth of the new digital signal obtained after sampling this analog signal? Explain.
- 3.11 In order to minimize aliasing, what sampling rate should be used to sample a 400-Hz triangular wave? Explain.
- 3.12 A 100-Hz full-wave-rectified sine wave is sampled at 200 samples/s. The samples are used to directly reconstruct the waveform using a digital-to-analog converter. Will the resulting waveform be a good representation of the original signal? Explain.
- 3.13 An A/D converter has an input signal range of 10 V. What is the minimum signal that it can resolve (in mV) if it is (a) a 10-bit converter, (b) an 11-bit converter?
- 3.14 A 10-bit analog-to-digital converter can resolve a minimum signal level of 10 mV. What is the approximate full-scale voltage range of this converter (in volts)?
- 3.15 A 12-bit D/A converter has an output signal range of ± 5 V. What is the approximate minimal step size that it produces at its output (in mV)?
- 3.16 For an analog-to-digital converter with a full-scale input range of +5 V, how many bits are required to assure resolution of 0.5-mV signal levels?
- 3.17 A normal QRS complex is about 100 ms wide. (a) What is the American Heart Association's (AHA) specified sampling rate for clinical electrocardiography? (b) If you sample an ECG at the AHA standard sampling rate, about how many sampled data points will define a normal QRS complex?
- 3.18 An ECG with a 1-mV peak-to-peak QRS amplitude is passed through a filter with a very sharp cutoff, 100-Hz passband, and sampled at 200 samples/s. The ECG is immediately reconstructed with a digital-to-analog converter (DAC) followed by a low-pass reconstruction filter. Comparing the DAC output with the original signal, comment on any differences in appearance due to (a) aliasing, (b) the sampling process itself, (c) the peak-to-peak amplitude, and (d) the clinical acceptability of such a signal.
- 3.19 An ECG with a 1-mV peak-to-peak QRS amplitude and a 100-ms duration is passed through an ideal low-pass filter with a 100-Hz cutoff. The ECG is then sampled at 200 samples/s. Due to a lack of memory, every other data point is thrown away after the sampling process, so that 100 data points per second are stored. The ECG is immediately reconstructed with a digital-to-analog converter followed by a low-pass reconstruction filter. Comparing the reconstruction filter output with the original signal, comment on any differences in appearance due to (a) aliasing, (b) the sampling process itself, (c) the peak-to-peak amplitude, and (d) the clinical acceptability of such a signal.
- 3.20 An IBM PC signal acquisition board with an 8-bit A/D converter is used to sample an ECG. An ECG amplifier provides a peak-to-peak signal of 1 V centered in the 0-to-5-V input range of the converter. How many bits of the A/D converter are used to represent the signal?
- 3.21 A commercial 12-bit signal acquisition board with a ± 10 -V input range is used to sample an ECG. An ECG amplifier provides a peak-to-peak signal of ± 1 V. How many discrete amplitude steps are used to represent the ECG signal?
- 3.22 Explain the relationship between the frequencies present in a signal and the sampling theorem.
- 3.23 Describe the effects of having a nonideal (a) input filter; (b) output filter.
- 3.24 What sampling rate and filter characteristics (e.g., cutoff frequency) would you use to sample an ECG signal?
- 3.25 What type of A/D converter circuit provides the fastest sampling speed?
- 3.26 What type of A/D converter circuit tends to average out high-frequency noise?
- 3.27 In an 8-bit successive-approximation A/D converter, what is the initial digital approximation to a signal?

- 3.28 A 4-bit successive-approximation A/D converter gets a final approximation to a signal of 0110. What approximation did it make just prior to this final result?
- 3.29 In a D/A converter design, what are the advantages of an R - $2R$ resistor network over a binary-weighted resistor network?
- 3.30 For an 8-bit successive approximation analog-to-digital converter, what will be the next approximation made by the converter (in hexadecimal) if the approximation of (a) 0x90 to the input signal is found to be too low, (b) 0x80 to the input signal is found to be too high?
- 3.31 For an 8-bit successive-approximation analog-to-digital converter, what are the possible results of the next approximation step (in hexadecimal) if the approximation at a certain step is a) 0x10, b) 0x20?
- 3.32 An 8-bit analog-to-digital converter has a clock that drives the internal successive approximation circuitry at 80 kHz. (a) What is the fastest possible sampling rate that could be achieved by this converter? (b) If 0x80 represents a signal level of 1 V, what is the minimum signal that this converter can resolve (in mV)?
- 3.33 What circuit is used in a signal conversion system to store analog voltage levels? Draw a schematic of such a circuit and explain how it works.
- 3.34 The internal IBM PC signal acquisition board described in Appendix A is used to sample an ECG. An amplifier amplifies the ECG so that a 1-mV level uses all 12 bits of the converter. What is the smallest ECG amplitude that can be resolved (in μ V)?
- 3.35 An 8-bit successive-approximation analog-to-digital converter is used to sample an ECG. An amplifier amplifies the ECG so that a 1-mV level uses all 8 bits of the converter. What is the smallest ECG amplitude that can be resolved (in μ V)?
- 3.36 The Computers of Wisconsin (COW) A/D converter chip made with CMOS technology includes an 8-bit successive-approximation converter with a 100- μ s sampling period. An on-chip analog multiplexer provides for sampling up to 8 channels. (a) With this COW chip, how fast could you sample a single channel (in samples per s)? (b) How fast could you sample each channel if you wanted to use all eight channels? (c) What is the minimal external clock frequency necessary to drive the successive-approximation circuitry for the maximal sampling rate? (d) List two advantages that this chip has over an equivalent one made with TTL technology.