Analog and Digital Signals 2

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2.1 Chapter Objectives

"Digital" has been the buzz word for the last couple of decades. The latest and greatest electronic devices have been marketed as digital and cardiology equipment has been no exception. "Analog" has the connotation of being old and outdated, while "digital" has been associated with new and advanced. What do these terms actually mean and is one really better than the other? By the end of the chapter, the reader should know what analog and digital signals are, their respective characteristics, and the advantages and disadvantages of both signal types. The reader will also understand the fundamentals of sampling, including the trade-offs of high sampling rates and high amplitude resolution, and the distortion that is possible with low sampling rates and low amplitude resolution.

2.2 Analog Signals

To say a signal is analog simply means that the signal is continuous in time and amplitude. Take, for example, your standard mercury glass thermometer. This device is analog because the temperature reading is updated constantly and changes at any time interval. A new value of temperature can be obtained whether you look at the thermometer 1 s later, half a second later, or a millionth of a second later, assuming temperature can change that fast. The readings from the thermometer are also continuous in amplitude. This means that assuming your eyes are sensitive enough to read the mercury level, readings of 37, 37.4, or 37.440183432°C are possible. In actuality, most cardiac signals of interest are analog by nature. For example, voltages recorded on the body surface and cardiac motion are continuous functions in time and amplitude.

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If the description of analog instrumentation and signals stopped here, it would seem like this would be the ideal method to record signals. Why then have analog tape players and VCRs been replaced by digital CD players and DVD players, if tape players can reproduce continuous time and amplitude signals with near infinite resolution? The reason is that analog recording and signals suffer one major drawback – their susceptibility to noise and distortion. Consider an audio tape with your favorite classical music performance that you bought in the 1980s. Chances are that the audio quality has degraded since the tape was purchased. Also consider the situation where a duplicate of the tape was made. The copy of the tape would not have the same quality as the original. If a duplicate of the duplicate of the duplicate was made, the imperfections of the duplication process would add up. In an analog system, noise cannot be easily removed once it has entered the system.

2.3 Digital Signals

Digital systems attempt to overcome the analog system's susceptibility to noise by sacrificing the infinite time and amplitude resolution to obtain perfect reproduction of the signal, no matter how long it has been stored or how many times it has been duplicated. That is why your audio CD purchased in the 1990s (assuming it is not too scratched up) will sound the same as when you first purchased it. The advantages can also be readily seen in the cardiology field. For example, making photocopies of an ECG tracing will result in loss of quality. However, printing a new copy from the saved digital file will give you a perfect reproduction every time. The discrete time and discrete amplitude nature of the digital signal provide a buffer to noise that may enter the system through transmission or otherwise. Digital signals are usually stored and transmitted in the form of ones and zeros. If a digital receiver knows that only zeros or ones are being transmitted and when approximately to expect them, there is a certain acceptable level of noise that the receiver can handle. Consider the example in the Fig. 2.1. The top plot shows a digital series of eight ones and zeros. This series could represent some analog value. The transmission or reproduction of the digital series results in noise being added to the series, such that the values now vary around one and zero as shown in the middle plot. If a digital receiver of the transmitted series uses the 0.5 level as the detection threshold, any value above 0.5 would be considered a one and any value below 0.5 would be considered a zero. With this criterion, all the ones and zeros would be detected correctly (bottom plot), despite the presence of noise. Thus the received digital signal provides a more accurate representation of the true signal of interest than would be if the analog signal itself was transmitted through the noisy channel.

Beyond the advantages of noise robustness during reproduction and transmission, digital signals have many other advantages. These include the ability to use computer algorithms to filter the signal, data compression to save storage space, and signal processing to extract information that may not be possible through manual human analysis. Thus there can be a large benefit in converting many of the signals that are used in cardiology to digital form.

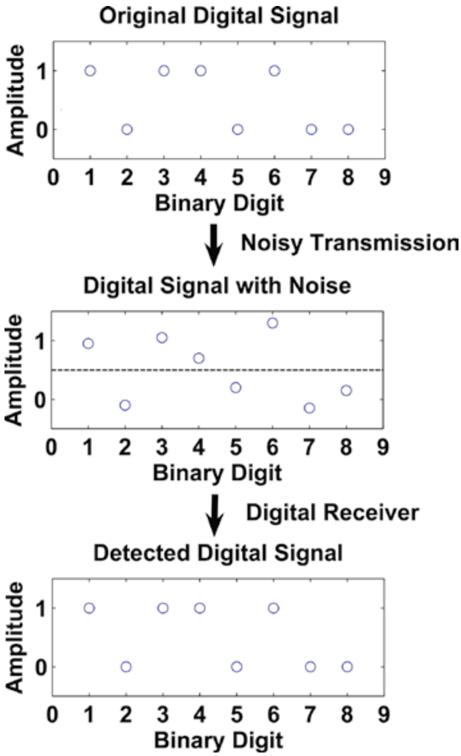

Fig. 2.1 Illustration of a digital signal transmitted through a noisy channel. The top panel shows a plot of the eight binary digits (amplitude values 0 or 1). The middle panel shows the same eight digits after transmission through a noisy channel causing deviation from 0 and 1. The 0.5 level represents the threshold, above or below which a 0 or 1 is decided by the digital receiver. The bottom panel shows the results of the decisions, which are equivalent to the original digital signal. Transmitting a digitized signal through a noisy channel usually results in a more accurate representation of the true signal than transmitting the original analog signal

2.4 Analog-to-Digital Conversion

2.4.1 Sampling

The process of converting an analog signal to a digital signal has two parts: sampling and quantization. The sampling process converts a continuous time signal to a discrete time signal with a defined time resolution. The time resolution is determined by what is known as the sampling rate, usually expressing in Hertz (Hz) or samples per second. Thus if the sampling rate is 1,000 Hz, or 1,000 times/s, this means that the signal is being sampled every 1 ms. The sampling rate needed for a faithful reproduction of the signal depends on the sharpness of the fluctuations of the signal being sampled. An illustration of a sine wave with the frequency of 8 Hz or 8 cycles/s that is sampled 100 times a second (100 Hz) is shown in the top example of Fig. 2.2. As shown in the second panel, the 100 Hz sampling takes points from the sine wave every 10 ms. Connecting the points as shown in the third panel produces a good reproduction of the original sine wave.

The middle example of Fig. 2.2 shows the same sine wave with 25 Hz sampling. With 25 Hz sampling, the reconstructed signal is clearly not as good as when the sine wave was sampled at 100 Hz. However, the oscillations at 8 cycles/s can still be recognized. Decreasing

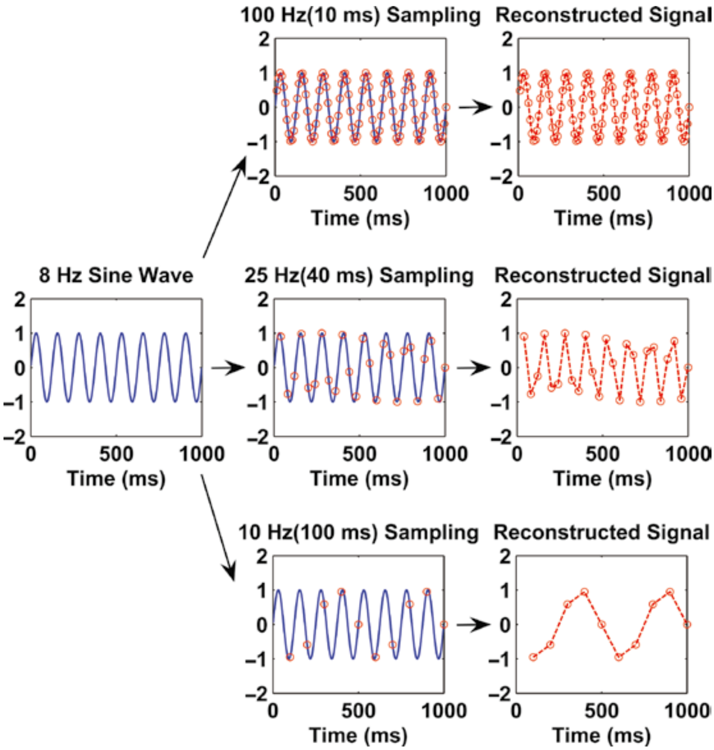

Fig. 2.2 Plots showing a sine wave with a frequency of 8 Hz sampled at rates of 100, 25, and 10 Hz. The reconstructed signal with 10 Hz sampling shows aliasing, the distortion resulting from undersampling

the sampling rate even further can result in a completely distorted reconstruction of the signal. The bottom example of Fig. 2.2 shows the sine wave sampled at 10 Hz. The reconstructed signal in this case resembles a 2-Hz sine wave, rather than an 8-Hz sine wave.

Distortion to the point where the original sine wave is unrecognizable because of undersampling is known as aliasing. Stated differently, if the signal is changing at a frequency that is faster than the sampling rate, important information about the signal will be lost. Consider the three signals in Fig. 2.3 – all three have the same sampled signal, but the original signals are not identical. This is because the signal contains high frequency components that are not detected by the relatively low sampling rate. To prevent aliasing, the Nyquist sampling rule states that the sampling rate must be at least twice the frequency of the sine wave. For our example of an 8-Hz sine wave, a sampling rate of at least 16 Hz is needed to prevent aliasing. Nonsinusoidal signals must be sampled at least 2 times the highest frequency component of the signal to avoid aliasing. Higher sampling rates are preferable in terms of the fidelity of the sampling. However, higher sampling rates come at the cost of additional size of the data. Thus, storage space is a consideration while determining an appropriate sampling rate.

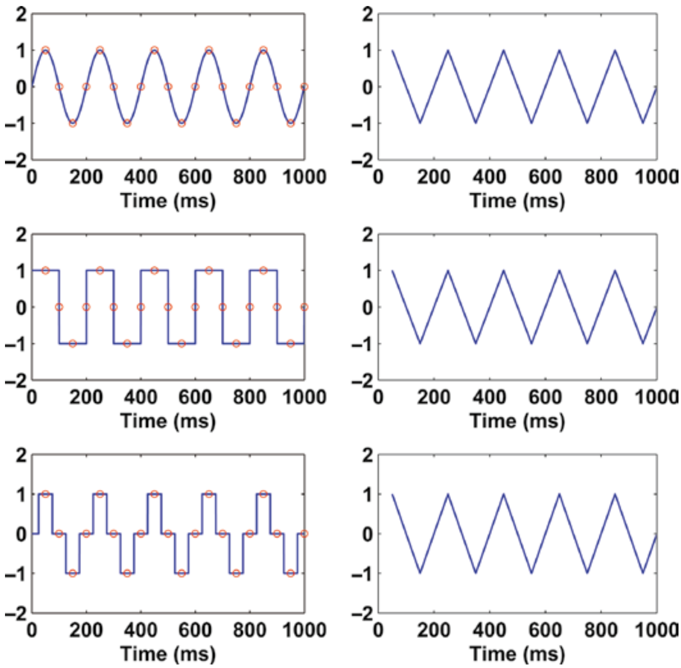

Fig. 2.3 Illustration of aliasing due to undersampling. A sine wave, square wave, and an alternating positive and negative pulse function produce the identical triangle wave when sampled at the same rate. Sampling occurs at 20 Hz and is indicated by the open *red circles*

2.4.2 Quantization

The second aspect of analog-to-digital conversion is quantization. Quantization converts continuous amplitude signals to a signal with a finite number of possible amplitude values. Quantizing with a high amplitude resolution will allow representation of the original signal with the least amount of error. However, higher resolution also comes with the trade-off of requiring more storage space. Quantization occurs with a fixed range of voltage. Therefore, proper amplification is important to get the best resolution possible. Figure 2.4 shows a sine wave that is sampled by a nine-level quantizer with a sample rate of 100 Hz and an amplitude resolution of 0.25 units or one eighth of the peak-to-peak amplitude of the signal. The amplitude of the sine wave in this example is perfectly fit over the quantization range. Reconstruction of the sine wave after quantization shows a decent approximation of the original sine wave.

Quantization can be poor if the amplification of the signal is less than ideal. In Fig. 2.5, the same nine levels are used to quantize a signal with a peak-to-peak amplitude of 0.5 units. The amplitude resolution of 0.25 is now only half of the peak-to-peak amplitude.

The poor relative amplitude resolution in this example results in a reconstructed signal that resembles more like a trapezoidal wave than a sine wave since only three of the nine levels are being utilized.

Figure 2.6 shows an example of a sine wave that is amplified beyond the range of the quantizer (peak-to-peak amplitude of two). In this situation, the signal is clipped at −1 and 1.

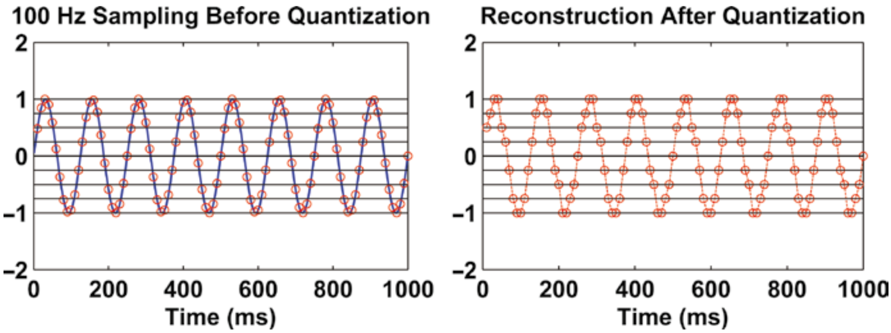

Fig. 2.4 Eight Hertz sine wave with 100 Hz sampling and nine-level quantization. The quantization levels are equally distributed over the amplitude range of the sine wave resulting in a good reproduction of the original signal. Note that the quantized signal takes on only the nine values indicated by the horizontal lines while the original signal spans over a range of values at the times sampled (indicated by the open *red circles*)

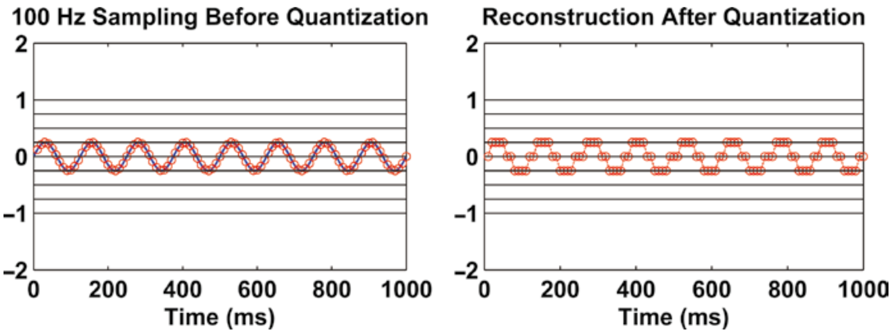

Fig. 2.5 Example of the sine wave that is underquantized leading to poor reproduction of the signal

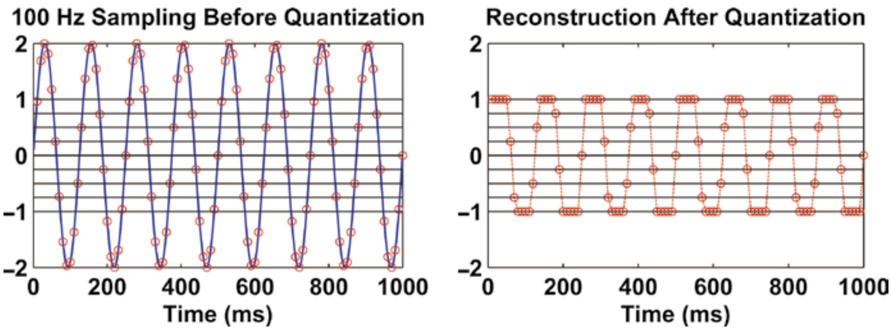

Fig. 2.6 Example of sine wave whose amplitude is beyond the range of the quantization. The result is a signal that is clipped at the upper and lower parts of the signal (also known as saturation)

This distortion is also known as "saturation" or "clipping." Thus even though the relative amplitude resolution of the quantization is high, the clipping of the signal cannot be undone.

2.4.3 Summary

In summary, the sampling rate and amplitude resolution of the sampling process are important qualities for digital signals to be a good representation of an analog signal. High sampling rates and amplitude resolution require more hardware complexity and storage space to store the signals. However, low sampling rates and amplitude resolution can result in distorted signals with loss of information. Proper amplification of the signal is also required to optimize the quantization process.

Summary of Key Terms

- Analog signal – Continuous time and continuous amplitude signal.
- Digital signal – Signal characterized by discrete time points and discrete amplitude values having a defined sampling rate and amplitude resolution.
- Sampling – The process of obtaining data points from an analog signal at defined intervals. The frequency of the sampling is known as the sampling rate.
- Quantization – The process of assigning sampled data points to discrete amplitude values of a defined resolution.
- Undersampling – Sampling at too slow of a rate which results in loss of information.
- Nyquist sampling rate – The rate of twice the highest frequency of a signal that is the minimum rate required to sample without loss of information.
- Aliasing – The distortion that occurs when a signal is undersampled (i.e., the Nyquist sampling criterion is not met).
- Saturation – The distortion of a signal that occurs when the amplitude of the signal is beyond the limits of the highest or lowest quantization amplitude. This is also known as clipping.

References

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