Digital Signal Processing (DSP)

Digital Signal Processing

- Digital Signal Processing is a powerful technology that is widely used in a many applications, such as automotive, consumer, graphics, imaging, industrial instrumentations, medical, military, and voice/speech.

- DSP converts the signals that are naturally occur in analog form, such as sound, video and information sensors, to digital forms and uses digital technologies to enhance and modify analog signal data for various applications.

Block Diagram of DSP System

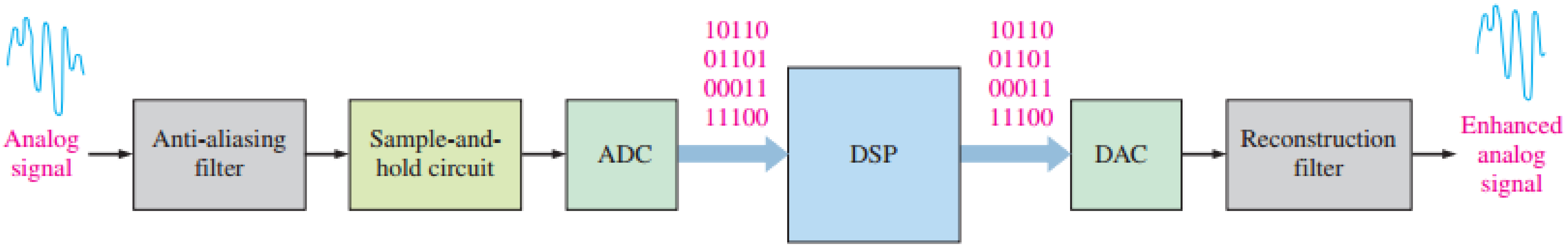


Figure: Basic block diagram of a typical digital signal processing system.

DSP (Digital Signal Processor)

- DSP (Digital Signal Processor) can perform various operations on the incoming data such as
  - removing unwanted interference,
  - increasing the amplitude of some signal frequencies and reducing others,
  - encoding the data for secure transmission and detecting and correcting errors in transmitted codes.

- DSPs make possible, among many others things,
  - the cleanup of sound recording,
  - removal of echos from communications lines,
  - the enhancement of images from CT scans or better medical diagnosis and
  - the scrambling of cellular phone conversations for privacy.

Sampling and Filtering

- An anti-aliasing filter and a sample-and-hold circuit are two functions of DSP.
- The sample-and-hold function does two operations, first of which is sampling.
- **Sampling** is the process of taking a **sufficient number of discrete values at points on a waveform** that will define the shape of the waveform.
- **The more samples you take, the more accurately you can define a waveform**.
- Sampling converts **an analog signal into a series of impulses, each representing the amplitude of the signal** at a given instant in time.
- Figure illustrates the process of sampling.

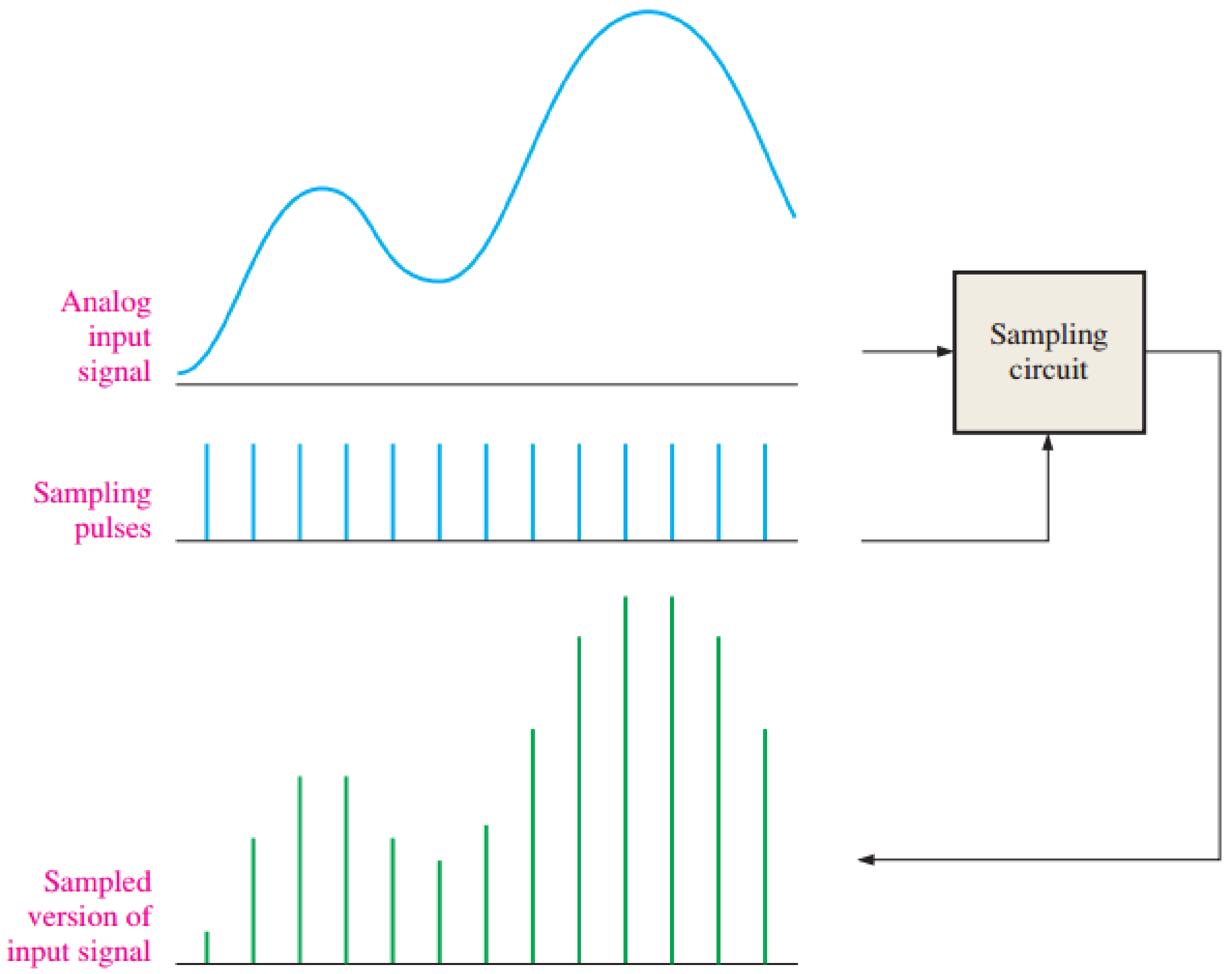
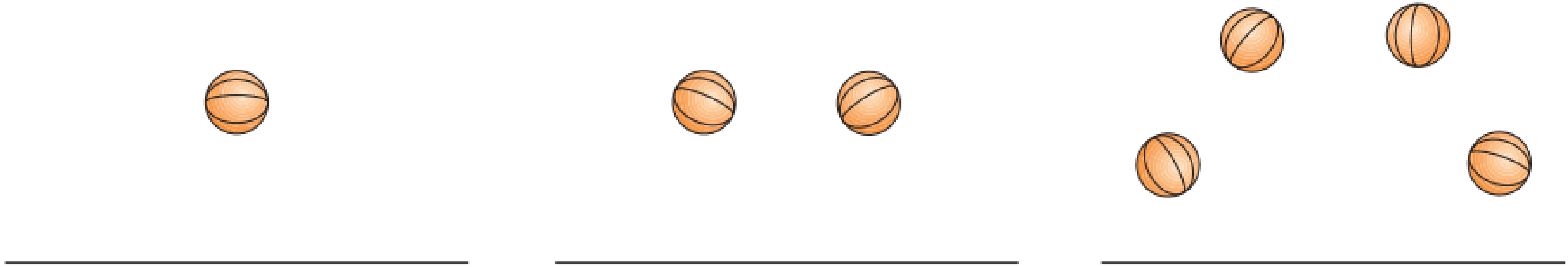


Fig: Illustration of the sampling process

Simple illustration of Sampling Theorem



(a) One sample of a ball during a single bounce

(b) Two samples of a ball during a single bounce. This is the absolute minimum required to tell anything about its movement, but generally insufficient to describe its path.

(c) Four samples of a ball during a single bounce form a rough picture of the path of the ball.

Sampling Theorem

- Before a signal can be sampled, it must be passed through a **low-pass filter (anti-aliasing filter)** to eliminate harmonic frequencies above a certain value as determined by the **Nyquist frequency**.

- Notice in the previous Figure that there are two input waveforms. One is the analog signal and the other is the sampling pulse waveform.

- The sampling theorem states that, in order to represent an analog signal, the sampling frequency, f_{sample}, must be at least twice the highest frequency component $f_{a(max)}$ of the analog signal.

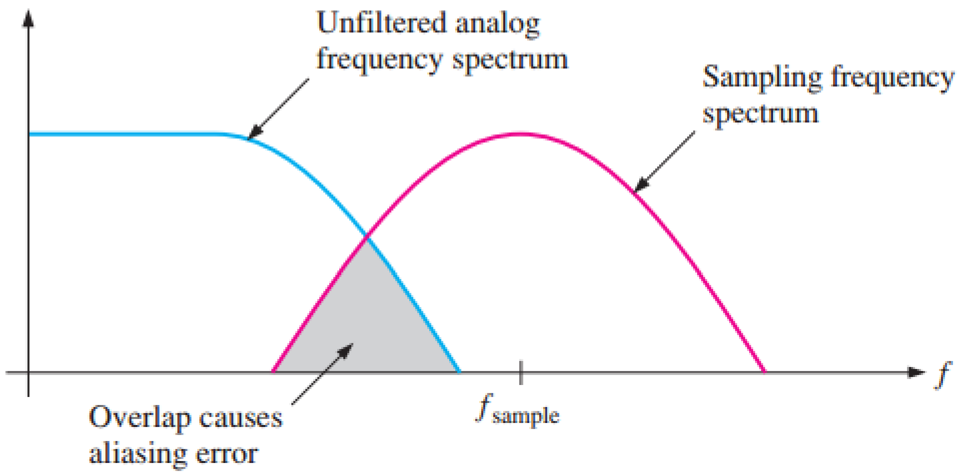
- The frequency $f_{a(max)}$ is known as the **Nyquist frequency**.

$$f_{\text{sample}} > 2f_{\text{a(max)}}$$

The need for Filtering

- Low-pass filtering is necessary to remove all frequency components (harmonics) of the analog signal that exceed the Nyquist frequency.
- If there are any frequency components in the analog signal that exceed the Nyquist frequency, an unwanted condition known as aliasing will occur.
- An alias is a signal produced when the sampling frequency is not at least twice the signal frequency.
- An alias signal has a frequency that is less than the highest frequency in the analog signal being sampled and therefore falls within the spectrum or frequency band of the input analog signal causing distortion.
- Such a signal is actually “posing” as part of the analog signal when it really isn’t, thus the term alias.

The need for Filtering



A basic illustration of the condition $f_{sample} < 2f_{a(max)}$.

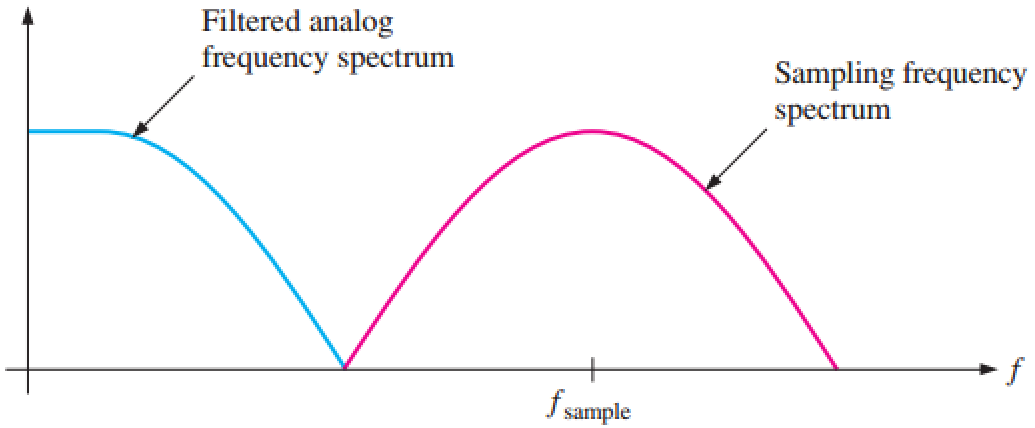


FIGURE 12–4 After low-pass filtering, the frequency spectra of the analog and the sampling signals do not overlap, thus eliminating aliasing error.

Sample and Hold Operation

- The holding operation is the second part of the sample-and-hold function.
- **After filtering and sampling, the sampled level must be held constant until the next sample occurs.**
- **This is necessary for the ADC to have time to process the sampled value.**
- This sample-and-hold operation results in a "stairstep" waveform that approximates the analog input waveform, as shown in Figure 12–5.

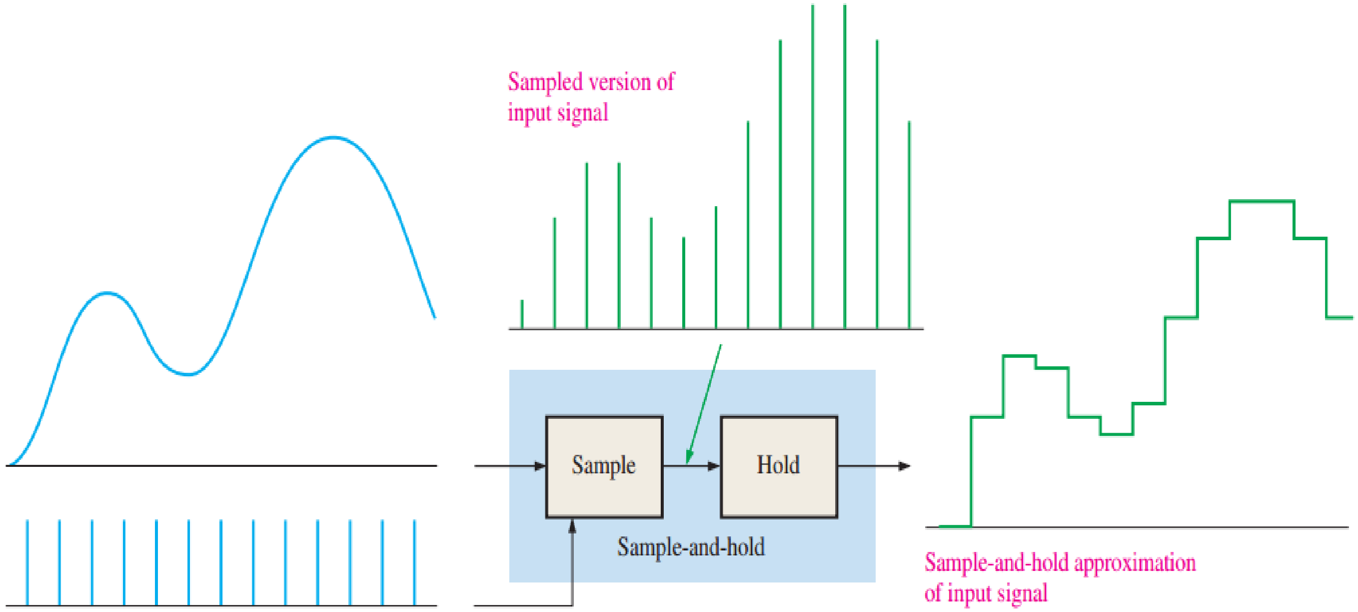


FIGURE 12–5 Illustration of a sample-and-hold operation.

Analog-to-Digital Conversion

- Analog-to-digital conversion is the process of converting **the output** of **the sample and-hold circuit** to a **series of binary codes** that represent the amplitude of the analog input at each of the sample times.

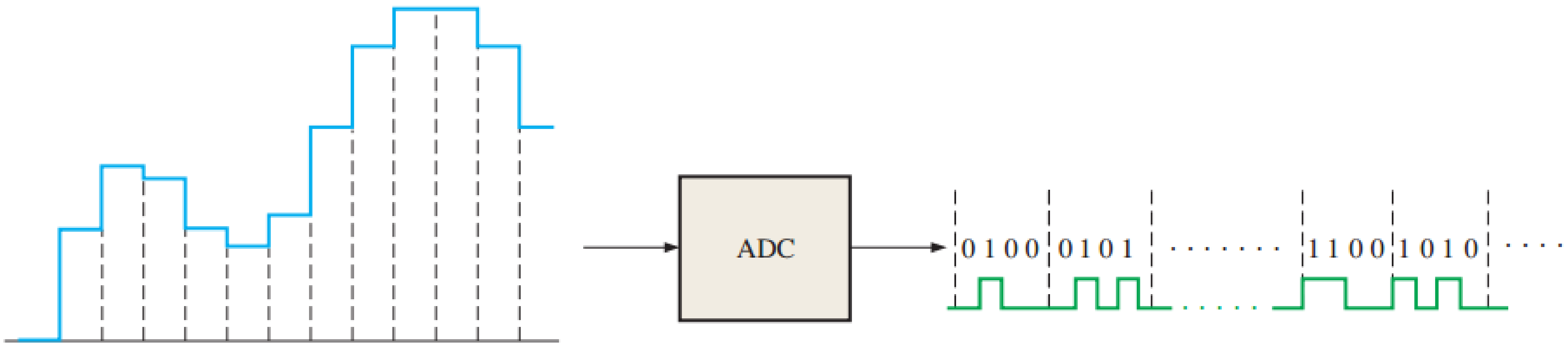


FIGURE 12–6 Basic function of an analog-to-digital converter (ADC) (The binary codes and number of bits are arbitrarily chosen for illustration only). The ADC output waveform that represents the binary codes is also shown.

Quantization

- ❑ The process of converting an analog value to a code is called quantization.
- ❑ During the quantization process, the ADC converts each sampled value of the analog signal to a binary code.
- ❑ The more bits that are used to represent a sampled value, the more accurate is the representation.

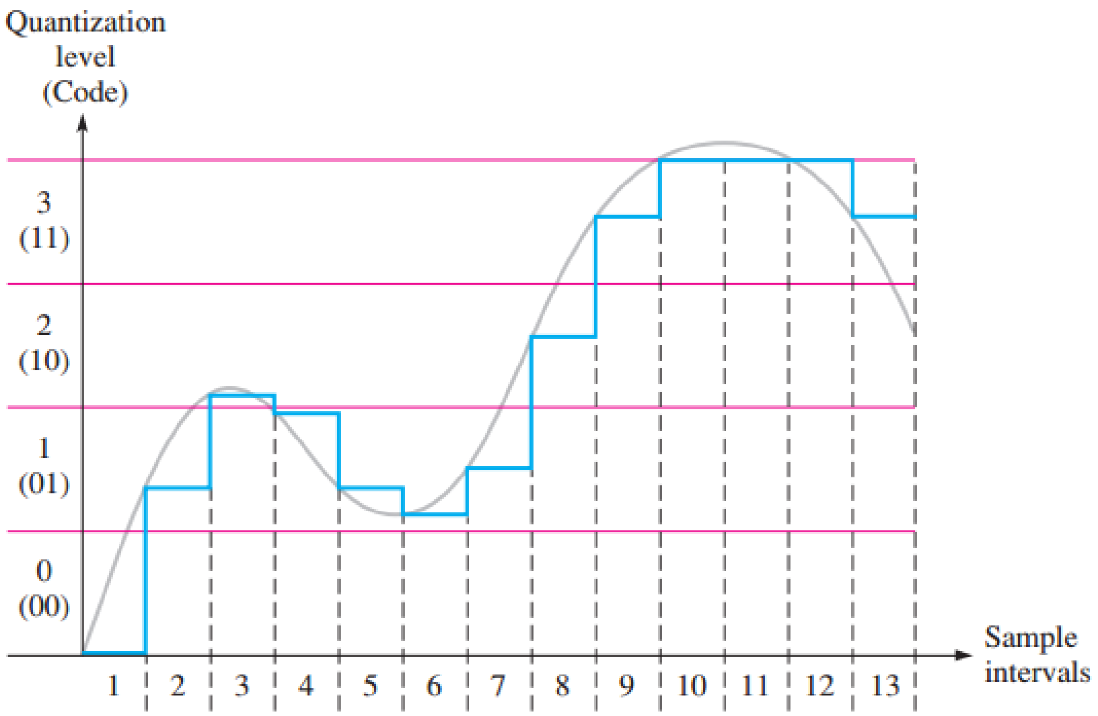


FIGURE 12–7 Sample-and-hold output waveform with four quantization levels. The original analog waveform is shown in light gray for reference.

Quantization

TABLE 12–1

Two-bit quantization for the waveform in Figure 12–7.

Sample Interval	Quantization Level	Code
1	0	00
2	1	01
3	2	10
4	1	01
5	1	01
6	1	01
7	1	01
8	2	10
9	3	11
10	3	11
11	3	11
12	3	11
13	3	11

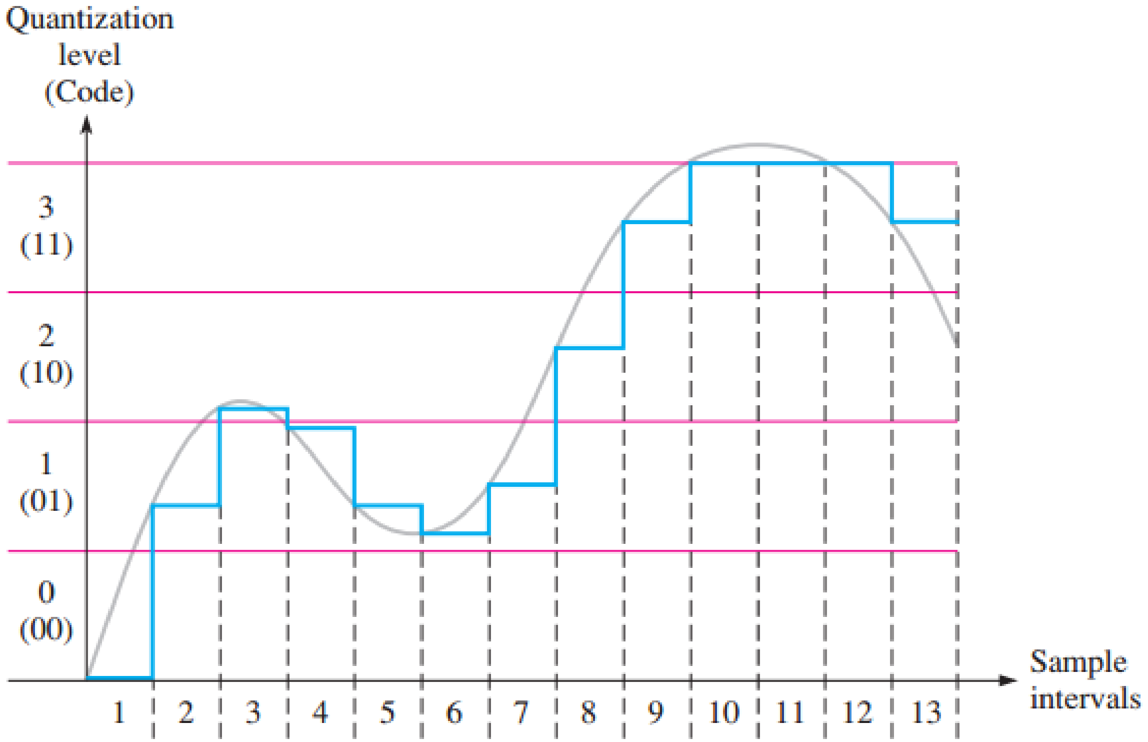


Fig: Quantized with two bits.

Quantization

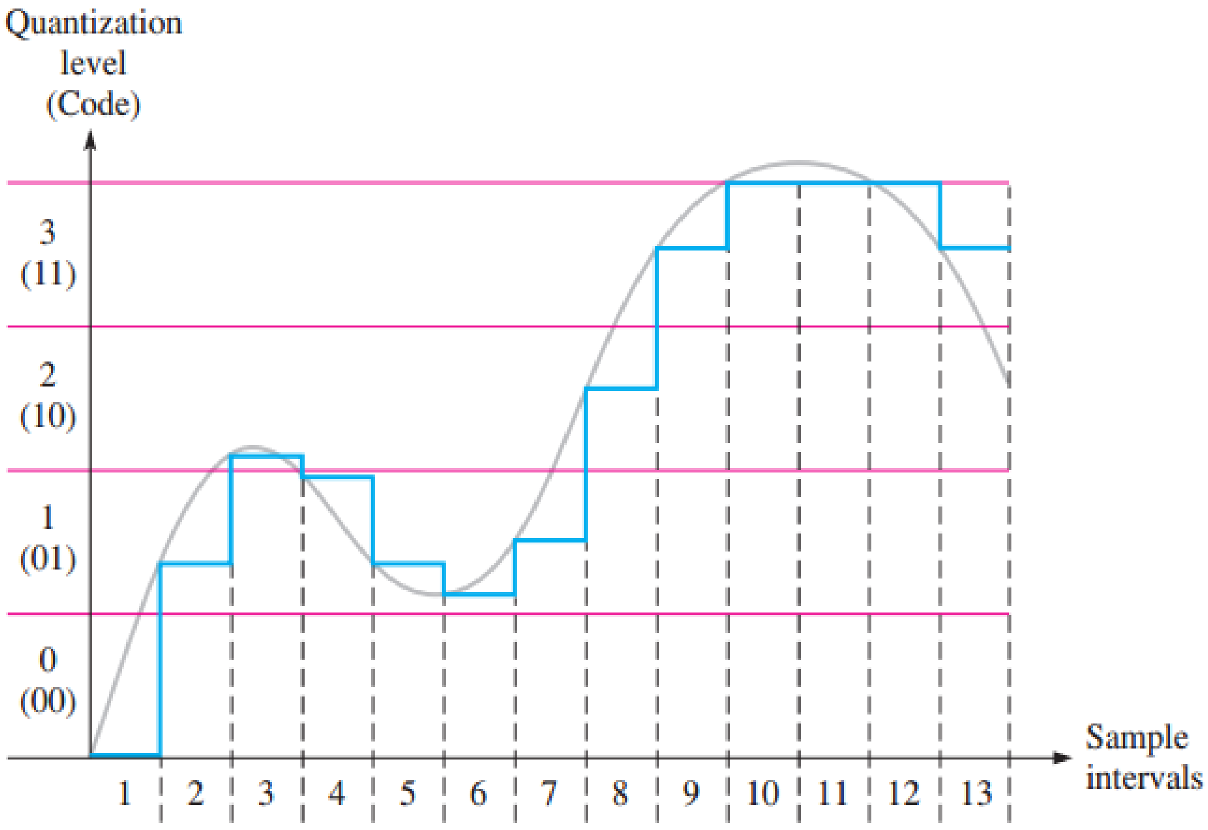


Fig: Original Signal

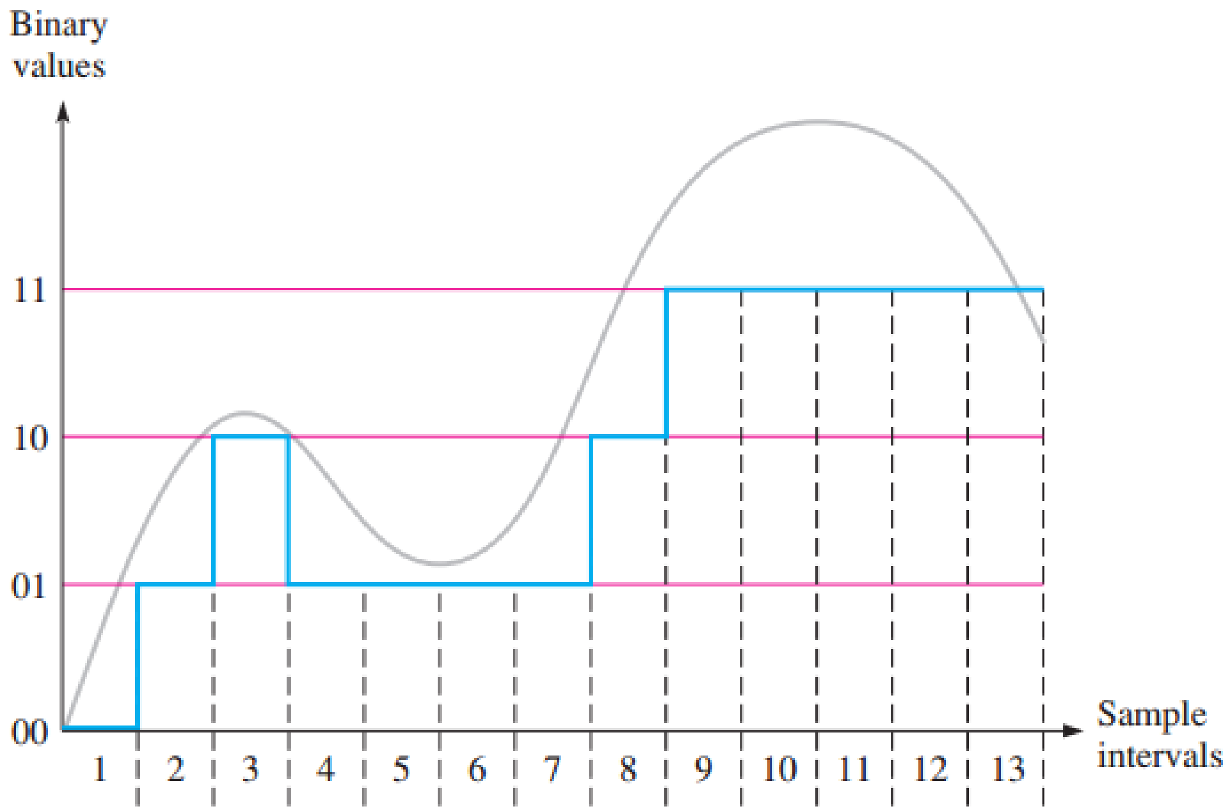


Fig: Quantized with two bits.

Quantization

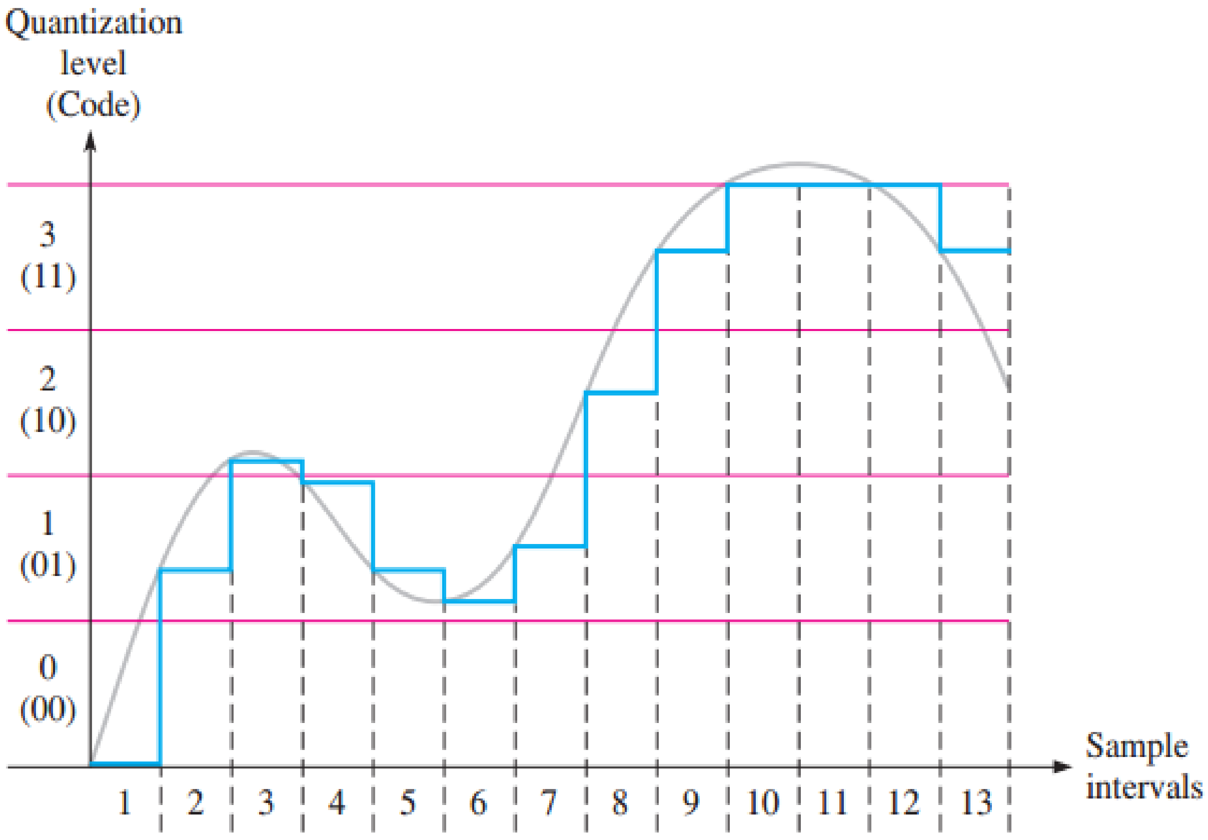


Fig: Original Signal

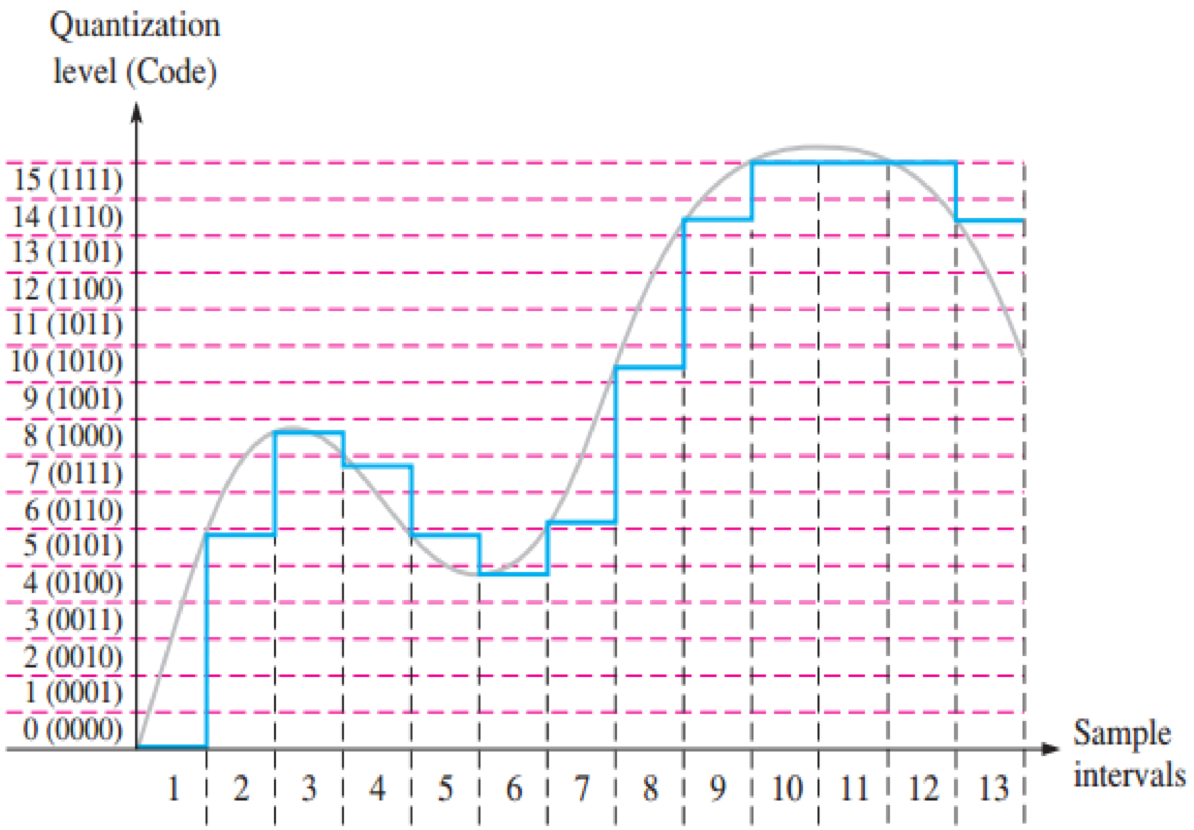


Fig: Quantized with 4-bits.

Checkup

SECTION 12–1 CHECKUP

Answers are at the end of the chapter.

1. What does sampling mean?
2. Why must you hold a sampled value?
3. If the highest frequency component in an analog signal is 20 kHz, what is the minimum sample frequency?
4. What does quantization mean?
5. What determines the accuracy of the quantization process?

Reference

Chapter # 12

Signal Conversion & Processing

"Digital Fundamental" by *Thomas L. Floyd*

11th Edition