Course Code: CSE 3107
Course Title: Communication Engineering

Dr. Foez Ahmed, SMIEEE

Associate Professor | Dept. of ICE

University of Rajshahi, Rajshahi-6205

Email: foez28@ru.ac.bd

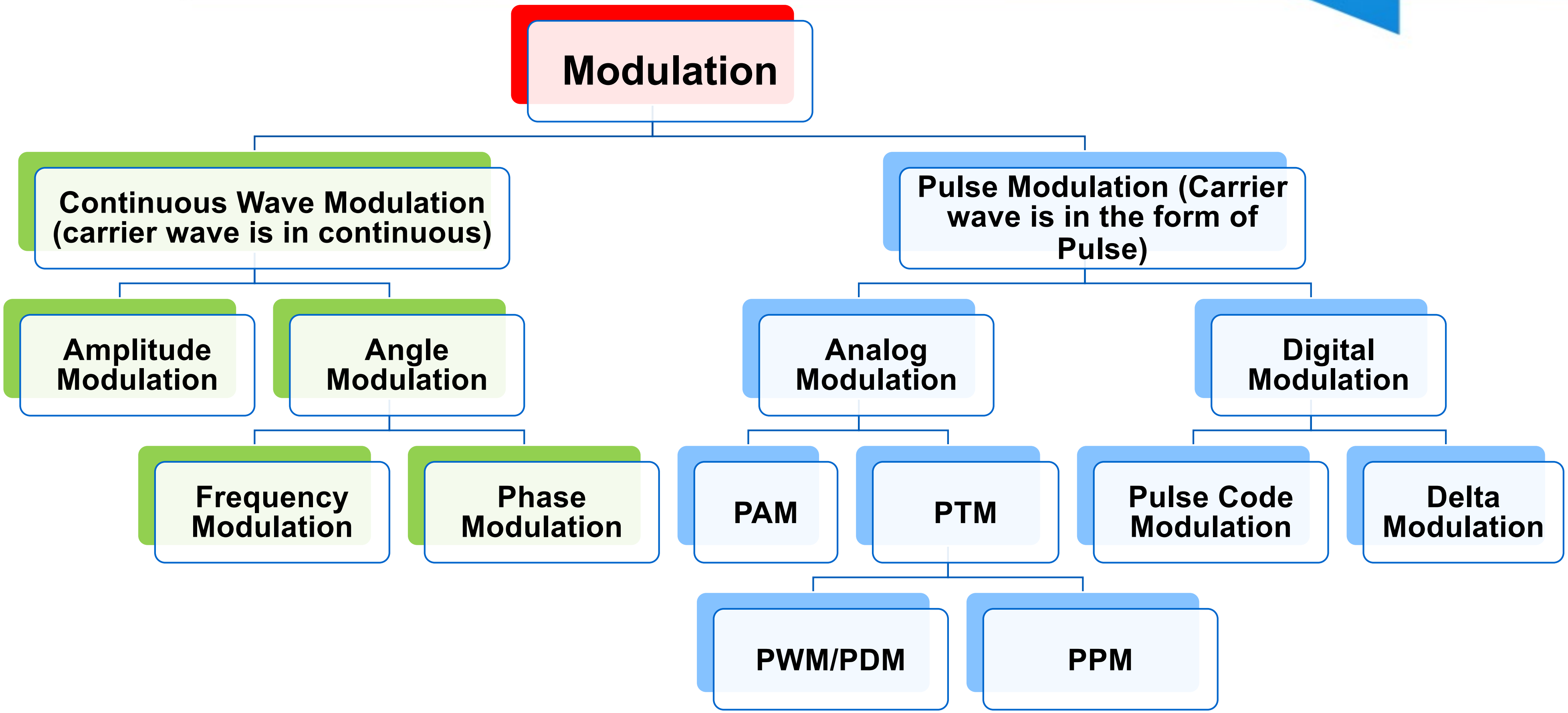
Lecture # 07

Pulse Modulation (PM)

Outline

- **Pulse Modulation**
- **Types of Pulse Modulation**
- **Analog PM: PAM, PTM (PWM/PDM, PPM)**
- **Digital PM: PCM and Delta Modulation**

Pulse Modulation



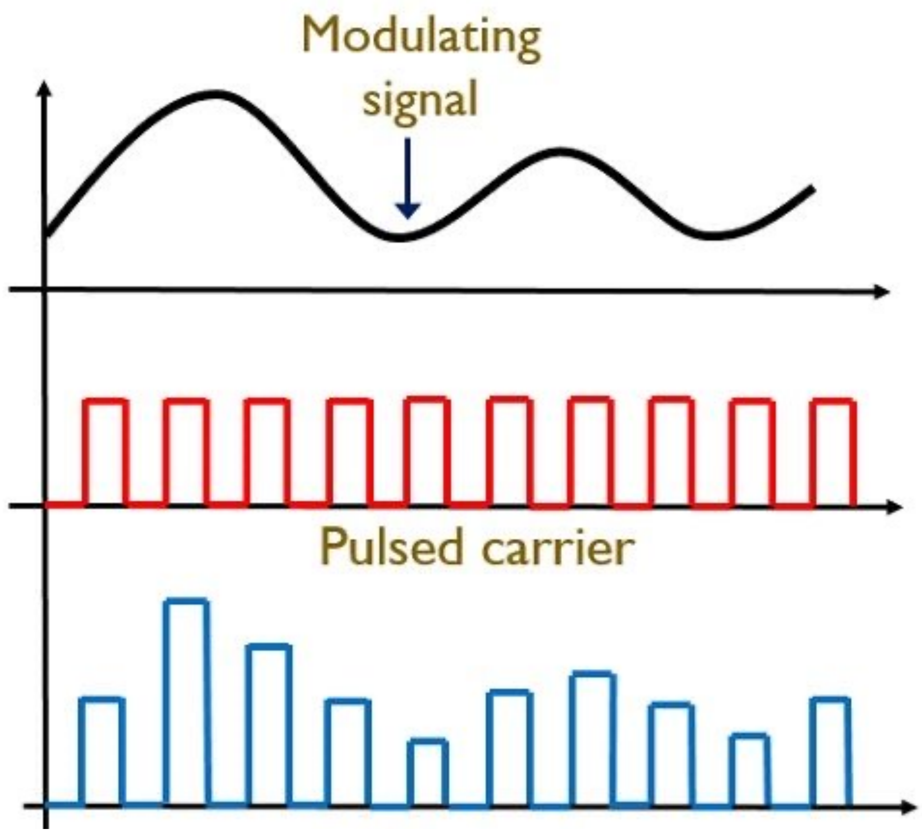
Pulse Modulation

❑ Pulse Modulation:

- ✓ In pulse modulation, some parameters of the pulsed carrier wave is varied as per the instantaneous value of the modulating (message) signal.

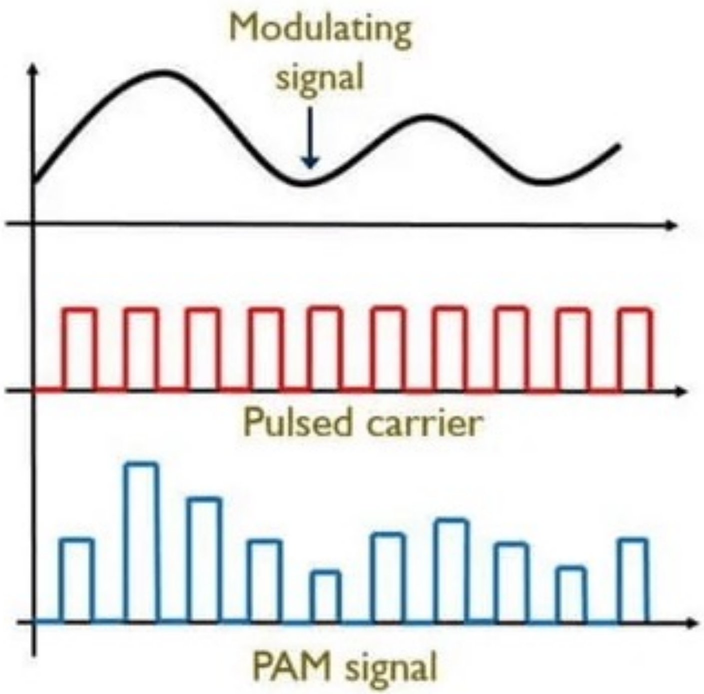
- ✓ These techniques modify pulse characteristics—amplitude, width, or position—to encode information, offering advantages in noise immunity and efficient multiplexing.

- ✓ They are classified into Analog (PAM, PWM/PDM, PPM) and Digital (PCM, DM) types.



Pulse Modulation

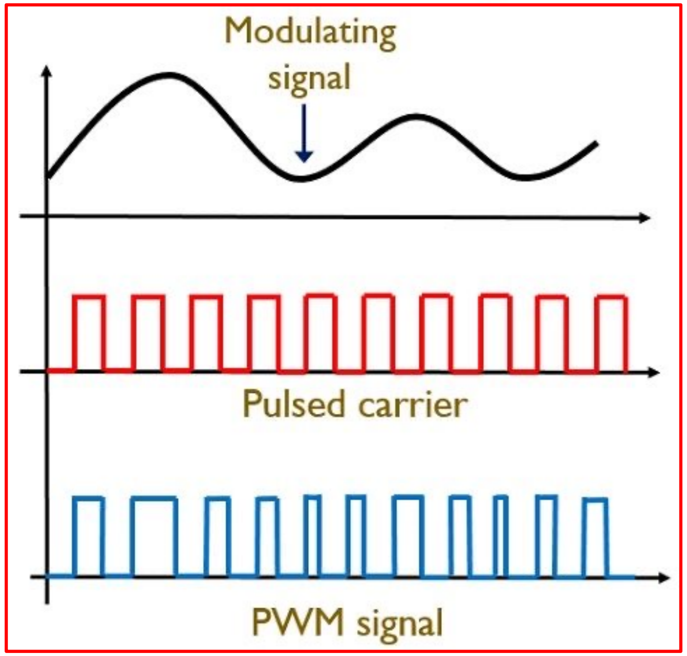
- **Pulse Amplitude Modulation:**
  - ✓ PAM is the modulation process where **amplitude of the pulsed carrier wave** is varied according to the **modulating** (**message**) signal.
  - ✓ It is the simplest form of digital modulation.



- **Advantages:**
  - ✓ System is lowest in complexity to implement
  - ✓ Generation and detection is easy.
- **Disadvantages:**
  - ✓ Noise interference is higher.
  - ✓ It has lowest power efficiency among all three types
  - ✓ Instantaneous power of transmitter varies.
  - ✓ Transmission bandwidth is too large.
  - ✓ It is difficult to remove noise, as this will affect amplitude part which carries information.

Pulse Modulation

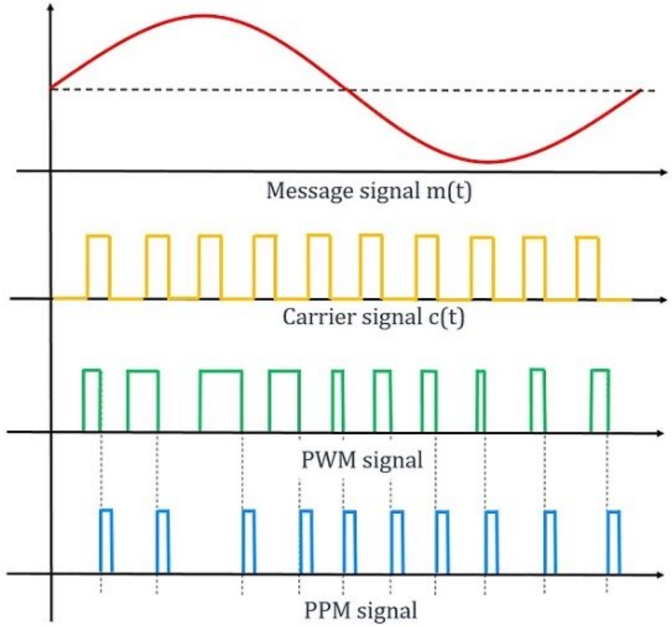
- **Pulse Width (Duration) Modulation (PWM):**
  - ✓ PWM is the modulation process where **width (duration) of the pulsed carrier wave** is varied according to the **modulating** (**message**) signal.



- **Advantages:**
  - ✓ Noise interference is less or minimum.
  - ✓ System is moderate in complexity to implement.
  - ✓ It has moderate power efficiency.
  - ✓ It supports higher power handling capability.
- **Disadvantages:**
  - ✓ Instantaneous power of transmitter varies.
  - ✓ High switching losses due to higher PWM frequency.
  - ✓ The system requires semiconductor devices with low turn-on and turn-off times. Hence, they are very expensive.

Pulse Modulation

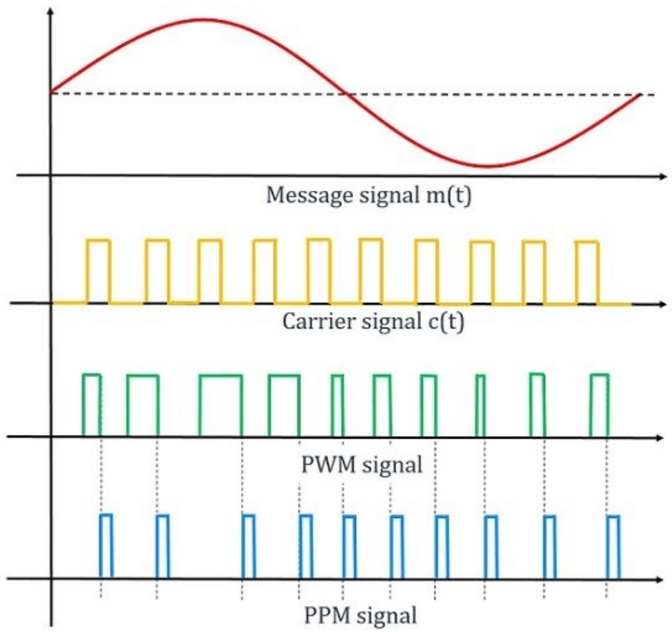
- ❑ **Pulse Position Modulation (PPM):**
  - ✓ PPM is the modulation process where **position of the pulsed carrier wave** is varied according to the **modulating** (**message**) signal.



- ❑ **Advantages:**
  - ✓ Noise interference is less or minimum due to constant amplitude.
  - ✓ It is easy to separate out signal from noisy signal.
  - ✓ It has highest power efficiency among all three types.
  - ✓ It requires less power compared to PAM due to short duration pulses.
  - ✓ Instantaneous power of PPM modulated signal remains constant due to constant pulse widths and pulse amplitudes.

Pulse Modulation

- **Pulse Position Modulation (PPM):**
  - ✓ PPM is the modulation process where **position of the pulsed carrier wave** is varied according to the **modulating** (**message**) signal.

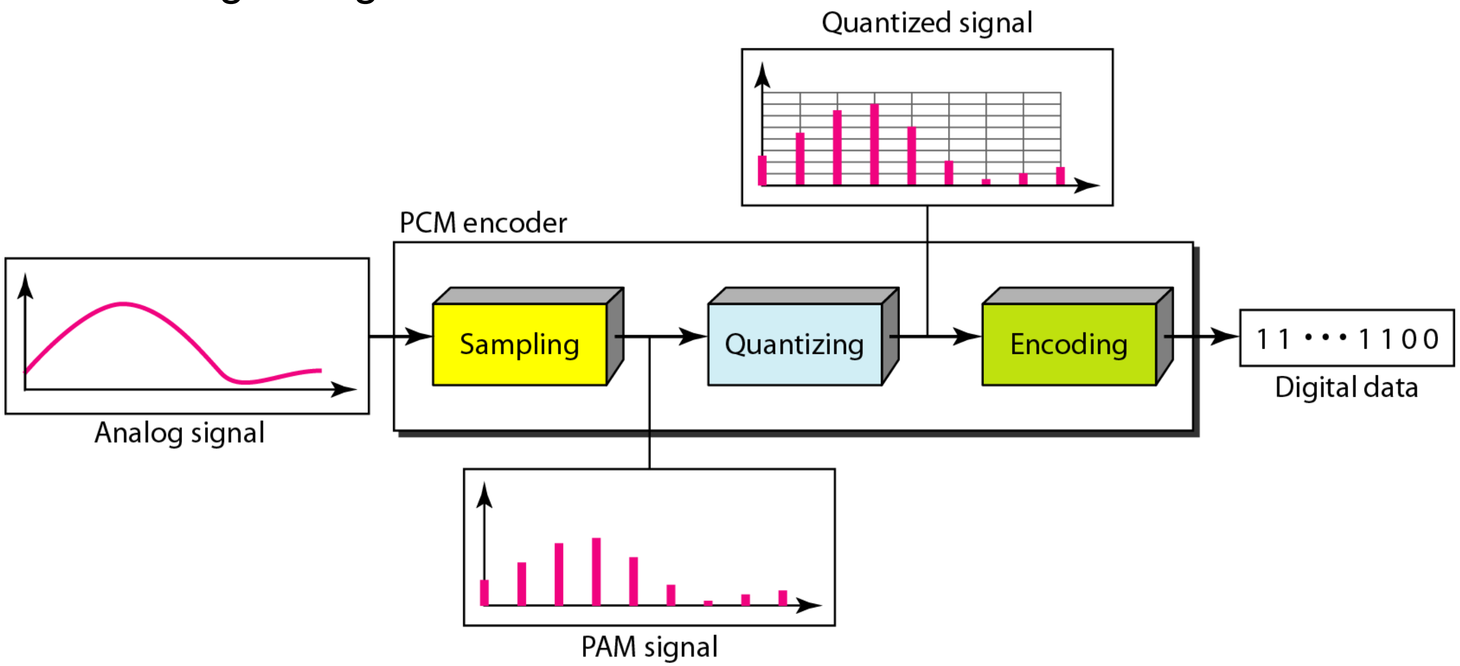


- **Disadvantages:**
  - ✓ System is highest in complexity to implement.
  - ✓ It requires very large bandwidth (BW) compared to PAM.

Pulse Modulation

- **Pulse Code Modulation (PCM):**
  - ✓ The most common technique to change an analog signal to digital data (digitization) is called pulse code modulation (PCM).
  - ✓ PCM > Analog to Digital Conversion

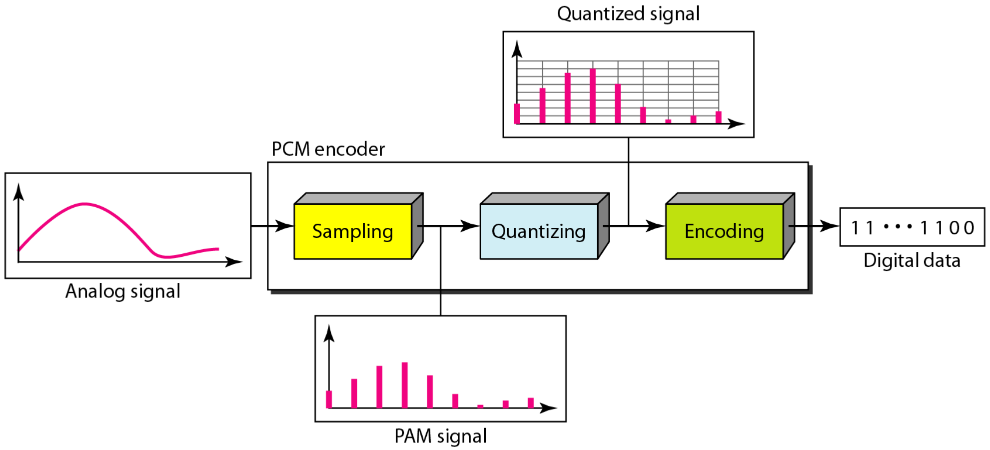
- A **PCM encoder has three processes:**
  1. The analog signal is sampled.
  2. The sampled signal is quantized.
  3. The quantized values are encoded as streams of bits.



Pulse Modulation

- **Pulse Code Modulation (PCM):** Three steps to digitize an analog signal:

1. **Sampling**
2. **Quantization**
3. **Binary encoding**



- Before we sample, we have to filter the signal to limit the **maximum frequency** of the signal as it affects the sampling rate.
- Filtering should ensure that we do not distort the signal, i.e., remove high frequency components that affect the signal shape.

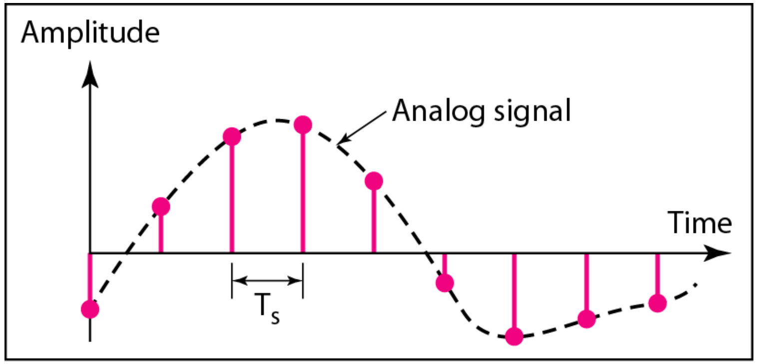
Pulse Modulation

- **Pulse Code Modulation (PCM):** Three steps to digitize an analog signal:

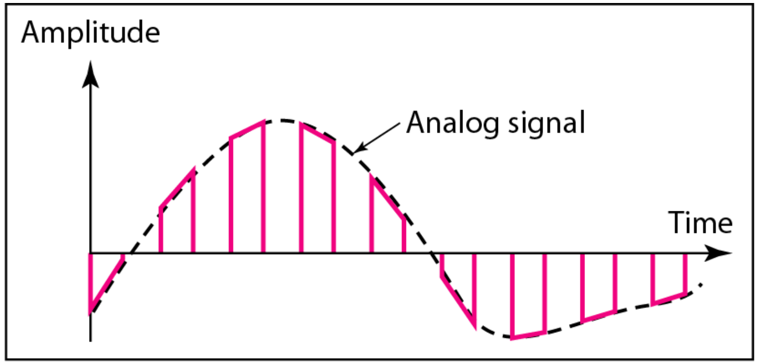
1. **Sampling**

   - Analog signal is sampled every $\mathbf{T_S}$ secs.
   - $\mathbf{T_s}$ is referred to as the sampling interval.
   - $f_s = 1/T_s$ is called the sampling rate or sampling frequency.
   - There are **3 sampling methods**:
     - **Ideal:** an impulse at each sampling instant
     - **Natural:** a pulse of short width with varying amplitude
     - **Flattop:** sample and hold, like natural but with single amplitude value
   - The process is referred to as PAM and the outcome is a signal with analog (non integer) values.

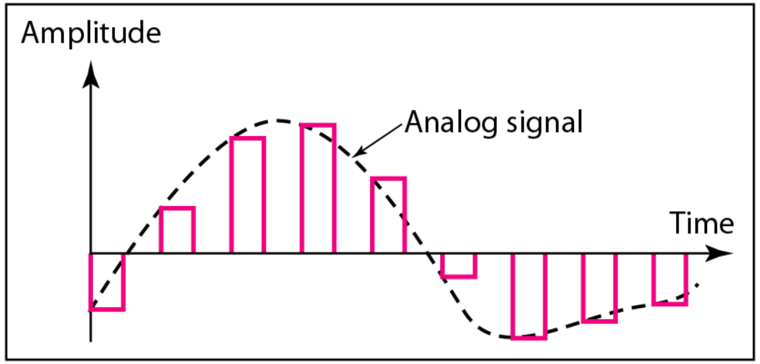
1. Sampling



a. Ideal sampling



b. Natural sampling



c. Flat-top sampling

Pulse Modulation

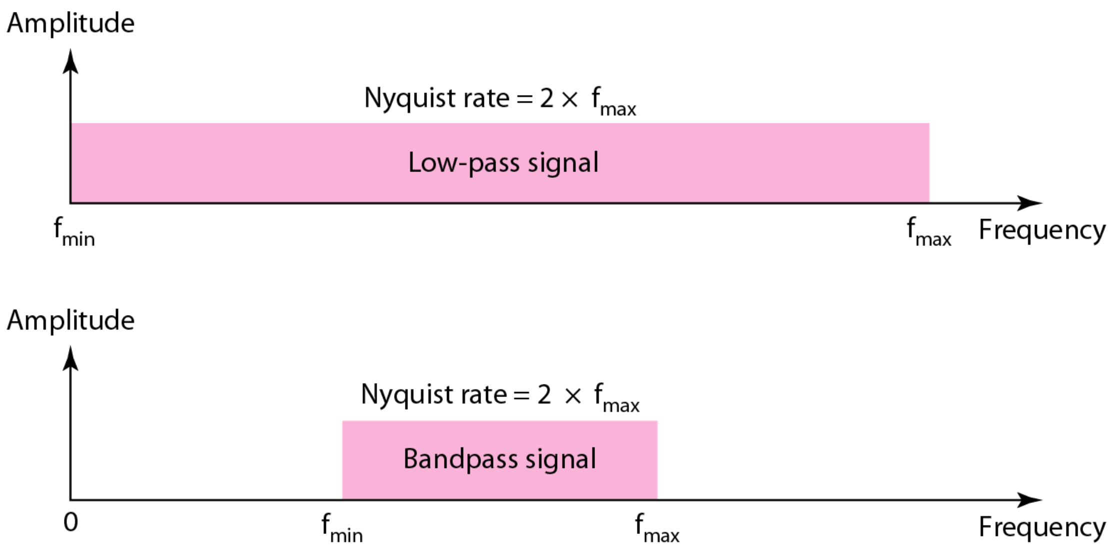
1. **Sampling**

- **Sampling Rate: What are the restrictions on *Ts?***
  - ✓ This question was elegantly answered by Nyquist.
  - ✓ To reproduce the original analog signal,

According to the Nyquist theorem, the sampling rate must be at least 2 times the highest frequency contained in the signal.

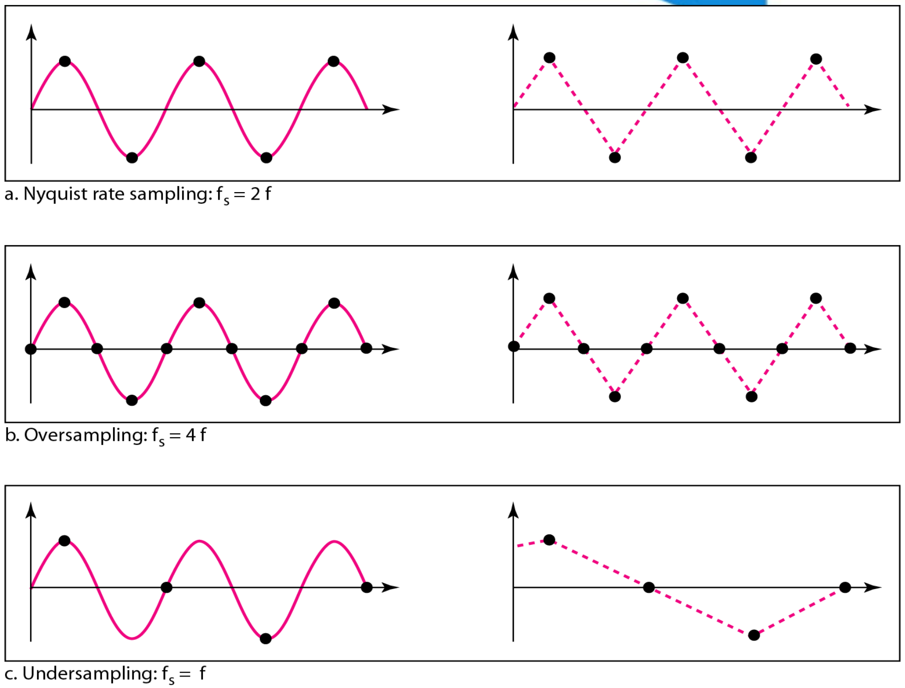
Pulse Modulation

- **Sampling Rate: What are the restrictions on *Ts?***
  - ✓ **First**, we can sample a signal only if the signal is **band-limited**. In other words, a signal with an infinite bandwidth cannot be sampled.
  - ✓ **Second**, the sampling rate must be at least 2 times the **highest frequency**, not the bandwidth.



Pulse Modulation

- **Example**: For an intuitive example of the Nyquist theorem, let us sample a simple sine wave at three sampling rates: $\mathbf{f_s = 4f}$ (2 times the Nyquist rate), $\mathbf{f_s = 2f}$ **(Nyquist rate**), and $\mathbf{f_s = f}$ (one-half the Nyquist rate). Figure shows the sampling and the subsequent recovery of the signal.



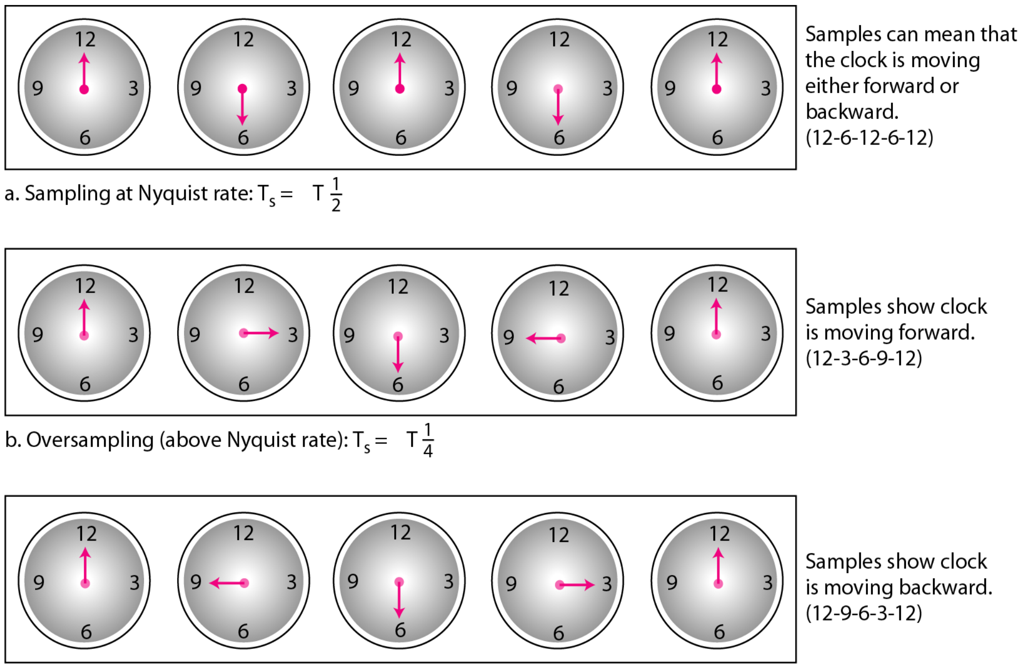
- **Observation:**

✓ It can be seen that sampling at the Nyquist rate can create a good approximation of the original sine wave (part a).

✓ Oversampling in part b can also create the same approximation, but it is redundant and unnecessary. Sampling below the Nyquist rate (part c) does not produce a signal that looks like the original sine wave.

Example: Consider the revolution of a hand of a clock. The second hand of a clock has a **period of 60 s**. According to the Nyquist theorem, we need to sample the hand every **30 s** (T_s = T or f_s = 2f).

- ✓ In Figure 4.25a, the sample points, in order, are 12, 6, 12, 6, 12, and 6. The receiver of the samples cannot tell if the **clock is moving forward or backward**.
- ✓ In part b, we sample at double the Nyquist rate (every 15 s). The sample points are 12, 3, 6, 9, and 12. The clock is **moving forward**.
- ✓ In part c, we sample below the Nyquist rate (T_s = T or f_s = f). The sample points are 12, 9, 6, 3, and 12. Although the clock is moving forward, the receiver thinks that the clock is **moving backward**.



a. Sampling at Nyquist rate: $T_s = T\frac{1}{2}$

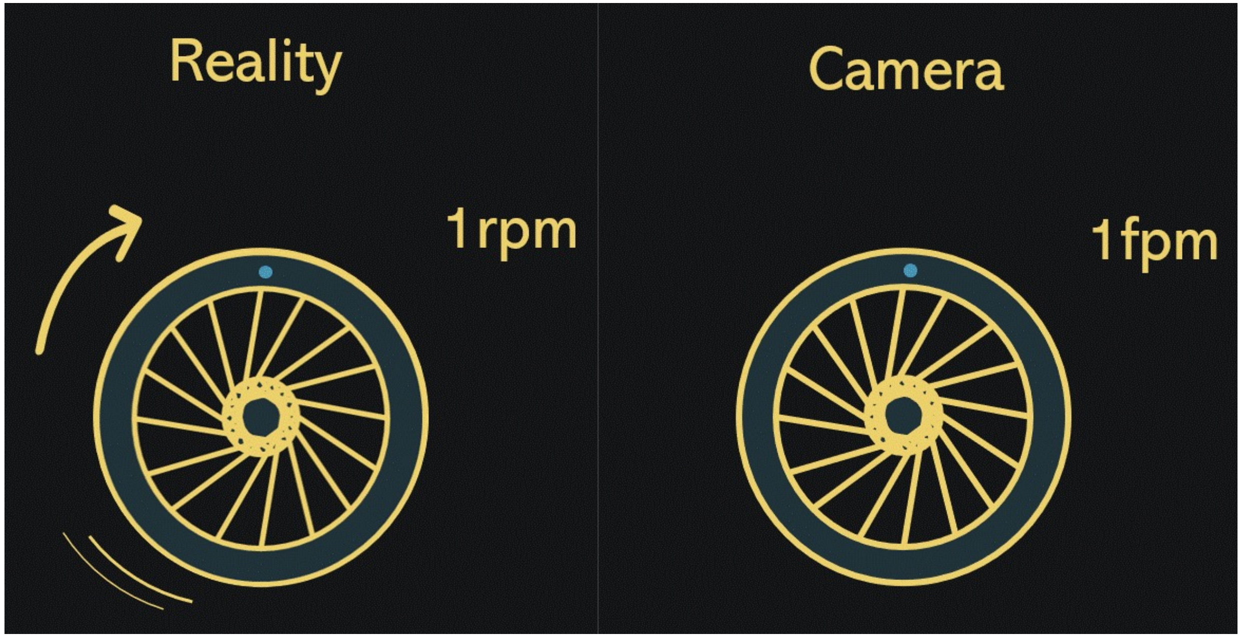
b. Oversampling (above Nyquist rate): $T_s = T\frac{1}{4}$

c. Undersampling (below Nyquist rate): $T_s = T\frac{3}{4}$

Pulse Modulation

- ✓ An example related to Example 4.7 is the seemingly backward rotation of the wheels of a forward-moving car in a movie. This can be explained by **under-sampling**.

- ✓ A movie is filmed at 24 frames per second. If a wheel is rotating more than 12 times per second, the under-sampling creates the impression of a backward rotation.



Pulse Modulation

Example: Telephone companies digitize voice by assuming a maximum frequency of 4000 Hz. The sampling rate therefore is 8000 samples per second.

Problem: A complex low-pass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?

Solution

The bandwidth of a low-pass signal is between 0 and f, where f is the maximum frequency in the signal. Therefore, we can sample this signal at 2 times the highest frequency (200 kHz). The sampling rate is therefore 400,000 samples per second.

Problem: A complex **bandpass signal** has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?

Solution

We cannot find the minimum sampling rate in this case because we do not know where the bandwidth starts or ends. We do not know the maximum frequency in the signal.

2. Quantization

✓ Sampling results in a series of pulses of varying amplitude values ranging between two limits: a ***min*** and a ***max***. The amplitude values are infinite between the two limits. We need to map the *infinite* amplitude values onto a finite set of known values (called quantization). The following are the steps in quantization:

1. We assume that the original analog signal has instantaneous amplitudes between *V*min and *V*max.
2. We divide the range into *L* zones (called quantization level), each of height Δ (delta), i.e.,

$$\Delta = (Vmax - Vmin)/L$$

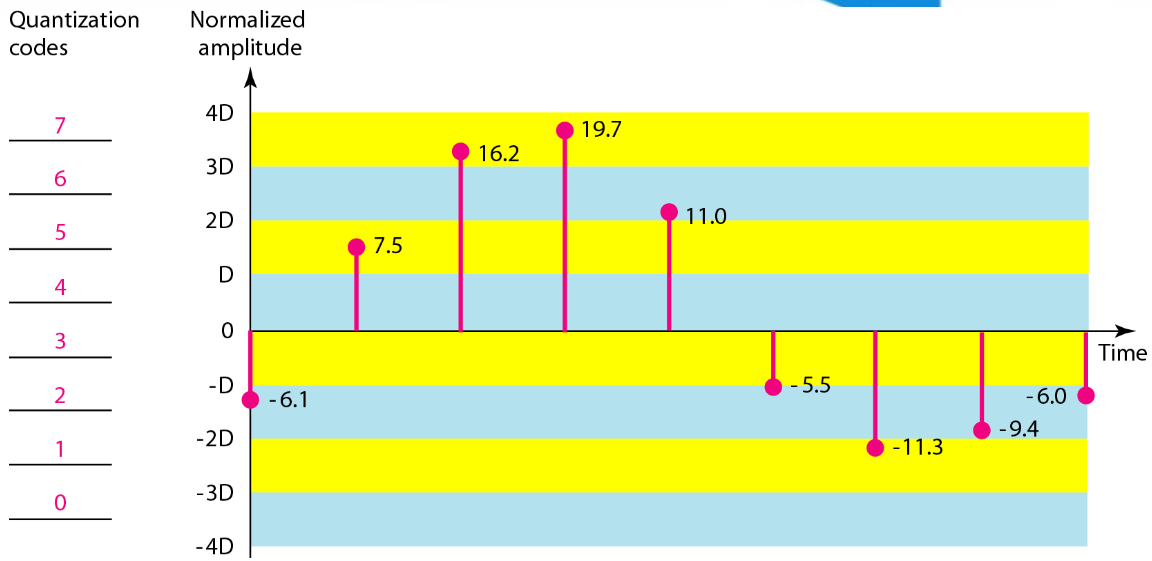
3. We assign quantized values of 0 to *L*- 1 to the midpoint of each zone.
4. We approximate the value of the sample amplitude to the quantized values.

❑ Quantization:

As a simple example, assume that we have a sampled signal and the sample amplitudes are between -20 and +20 V. We decide to have eight levels (*L*= 8). This means that Δ =5 V. Figure 4.26 shows this example.

✓ Number of bits per sample,

$$n_b = \log_2 L$$



Normalized PAM values	-1.22	1.50	3.24	3.94	2.20	-1.10	-2.26	-1.88	-1.20
Normalized quantized values	-1.50	1.50	3.50	3.50	2.50	-1.50	-2.50	-1.50	-1.50
Normalized error	-0.38	0	+0.26	-0.44	+0.30	-0.40	-0.24	+0.38	-0.30
Quantization code	2	5	7	7	6	2	1	2	2
Encoded words	010	101	111	111	110	010	001	010	010

❑ Quantization Error:

- ✓ When a signal is quantized, we introduce an error - the coded signal is an approximation of the actual amplitude value.
- ✓ The difference between actual and coded value (midpoint) is referred to as the quantization error.
- ✓ The more zones, the smaller Δ which results in smaller errors.
- ✓ But, the more zones the more bits required to encode the samples -> higher bit rate
- ✓ The contribution of the quantization error to the SNR_{dB} of the signal depends on the number of quantization levels L, or the bits per sample n_b

$$\mathrm{SNR_{dB}} = 6.02nb + 1.76 \quad \mathbf{dB}$$

Pulse Modulation

- **Quantization Error:** $\mathrm{SNR_{dB}} = 6.02n_b + 1.76 \;\; \mathbf{dB}$

Example 4.12

What is the $\mathrm{SNR_{dB}}$ in the example of Figure 4.26?

Solution

We can use the formula to find the quantization. We have eight levels and 3 bits per sample, so $\mathrm{SNR_{dB}} = 6.02(3) + 1.76 = 19.82$ dB. Increasing the number of levels increases the SNR.

Example 4.13

A telephone subscriber line must have an $\mathrm{SNR_{dB}}$ above 40. What is the minimum number of bits per sample?

❑ Quantization Error and SN_QR:

- Signals with lower amplitude values will suffer more from quantization error as the error range: $\Delta/2$, is fixed for all signal levels.
- Non-linear quantization is used to alleviate this problem. Goal is to keep SN_QR fixed for all sample values.

- Two approaches:
  - The quantization levels follow a logarithmic curve. Smaller Δ's at lower amplitudes and larger Δ's at higher amplitudes.
  - Companding: The sample values are compressed at the sender into logarithmic zones, and then expanded at the receiver. The zones are fixed in height.

❑ Bit Rate and Bandwidth Requirements of PCM:

- The bit rate of a PCM signal can be calculated from the number of bits per sample x the sampling rate

$$\textbf{Bit rate} = n_b \times f_s$$

- The minimum required bandwidth is

$$B_{\min} = n_b \times B_{\text{analog}}$$

- A digitized signal will always need more bandwidth than the original analog signal. This is the price we pay for robustness and other features of digital transmission.

Example 4.14

We want to digitize the human voice. What is the bit rate, assuming 8 bits per sample?

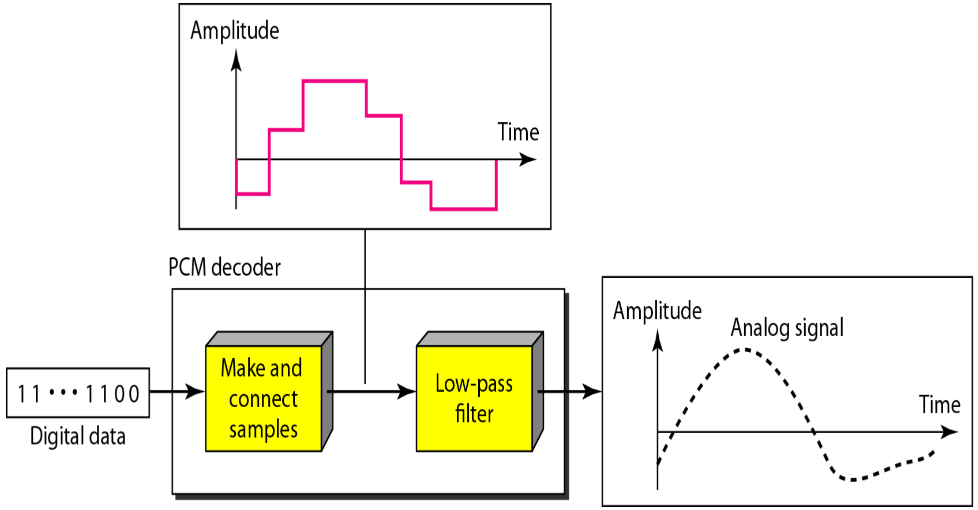
Solution

The human voice normally contains frequencies from 0 to 4000 Hz. So the sampling rate and bit rate are calculated as follows:

Sampling rate ::: 4000 x 2 ::: 8000 samples/s
Bit rate = 8000 x 8 ::: 64,000 bps = 64 kbps

❑ Components of a PCM Decoder:

- To recover an analog signal from a digitized signal, the following steps are followed:
  - We use a **hold circuit** that holds the amplitude value of a pulse till the next pulse arrives.
  - We pass this signal through a **low pass filter** with a cutoff frequency that is equal to the highest frequency in the pre-sampled signal.



✓ The filter has the same cutoff frequency as the original signal at the sender. If the signal has been sampled at (or greater than) the Nyquist sampling rate and if there are enough quantization levels, the original signal will be recreated.

❑ Advantages of PCM:

- ✓ It has a higher noise immunity
- ✓ It has a higher transmitter efficiency
- ✓ Uniform transmission quality
- ✓ Low manufacturing cost
- ✓ Integrated digital circuit
- ✓ Efficient codes are readily available
- ✓ Useful for long-distance communication
- ✓ Increase utilization of the existing circuit
- ✓ Good performance over a very poor transmission path
- ✓ The PCM convenient for long-distance communication

❑ Disadvantages of PCM:

- ✓ It requires large bandwidth
- ✓ Quantization error is produced
- ✓ Encoding, decoding and also have quantizing circuit of PCM is very complex
- ✓ Noise and cross talk is low but rise attenuation

Pulse Modulation

❑ Delta Modulation:

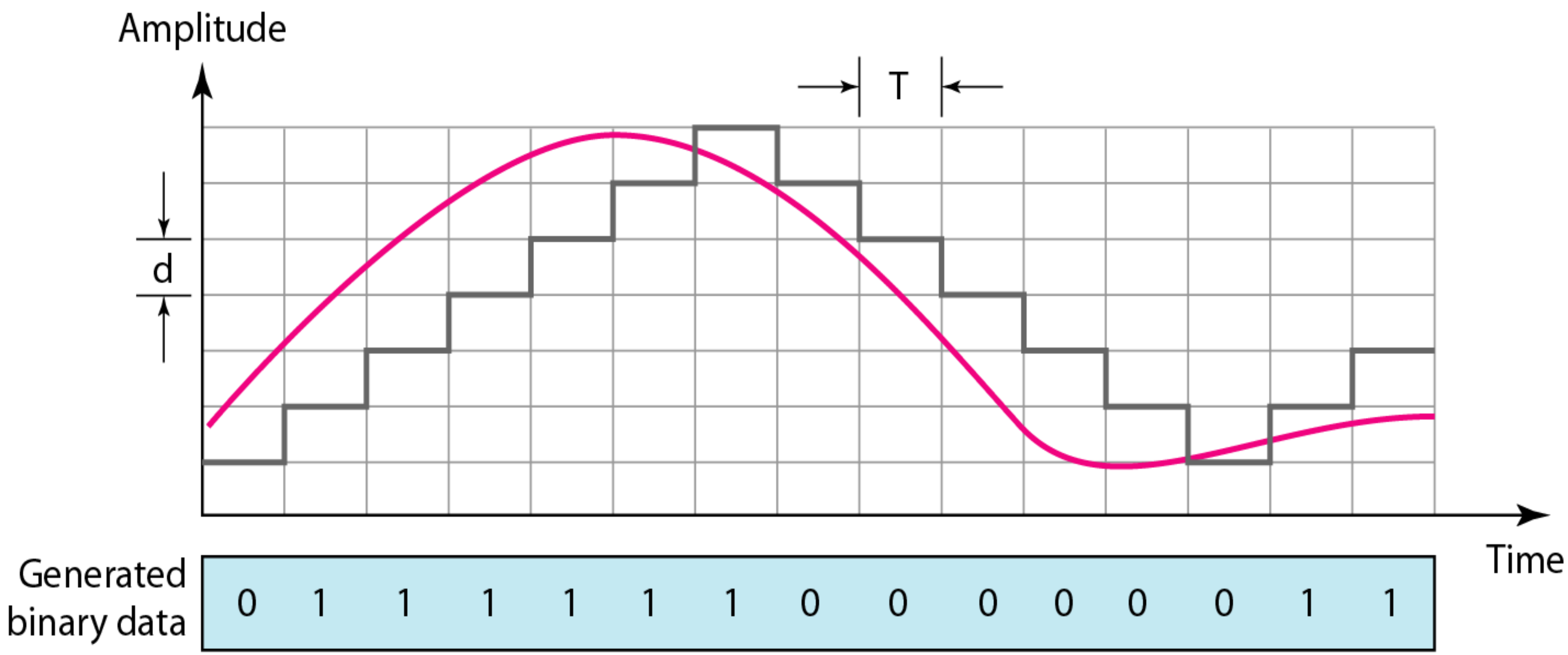
- Delta modulation (DM) is a simplex, but PCM is a very complex technique.

- DM finds the change from the previous sample, whereas PCM finds the value of the signal amplitude for each sample.

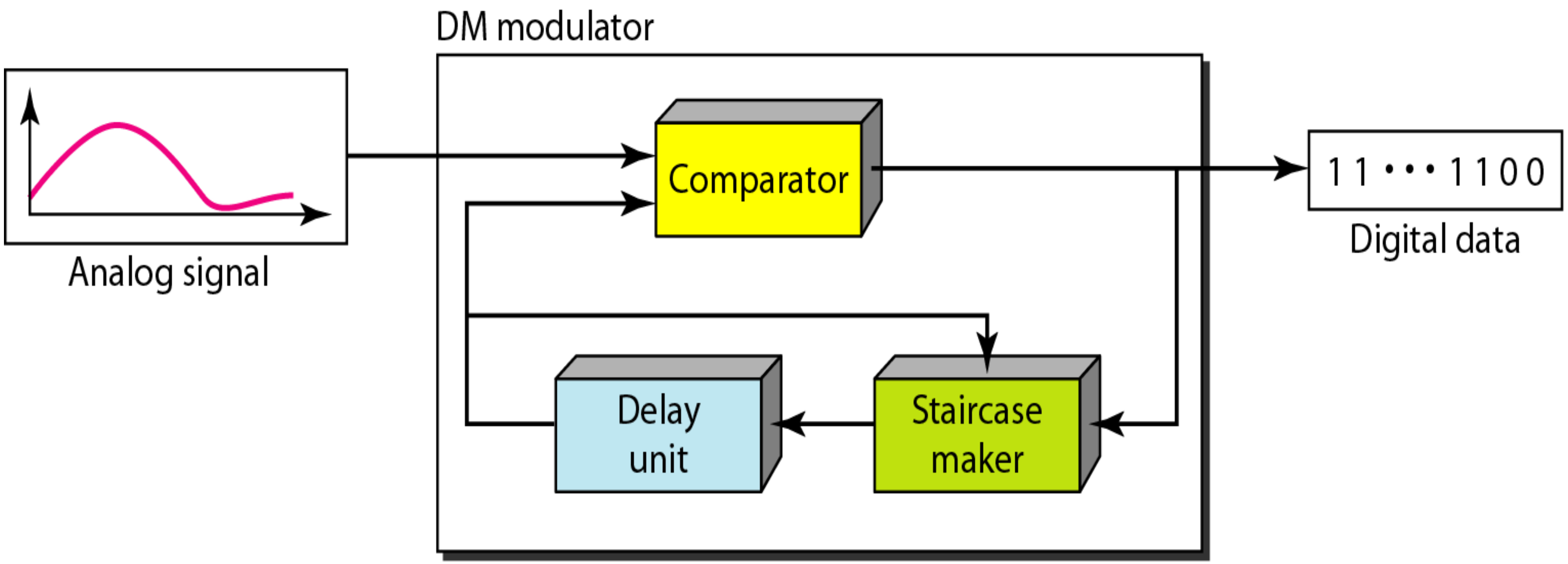
- In DM, the process records the small positive or negative changes, called delta (**δ**).

- If the **δ** positive, the process records a 1; if it is negative, the process records a 0.

❑ Delta Modulation:



❑ Delta Modulator:



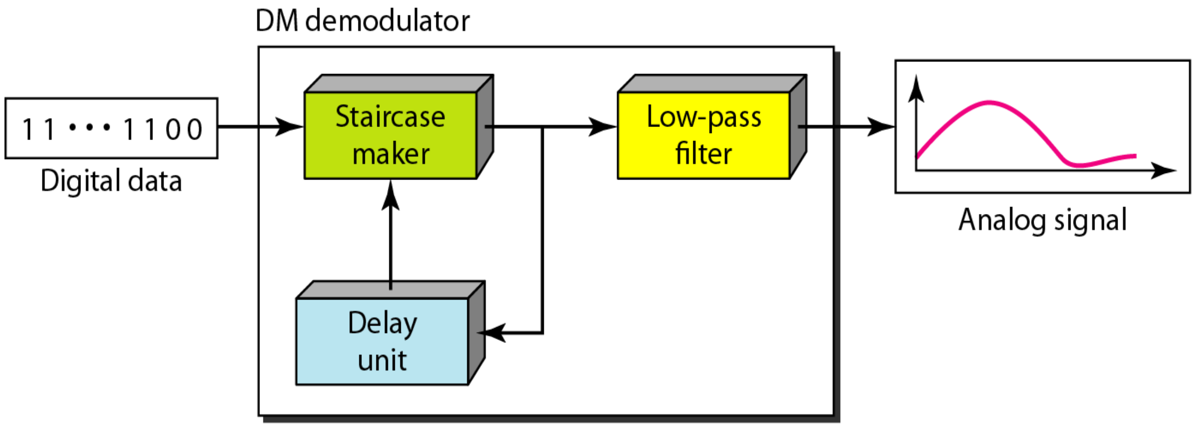
Pulse Modulation

❑ Delta Modulator:

- ✓ This process needs a base against which the analog signal is compared.
- ✓ The modulator builds a second signal that resembles a staircase.
- ✓ The modulator, at each sampling interval, compares the value of the analog signal with the last value of the staircase signal.
- ✓ If the amplitude of the analog signal is larger, the next bit in the digital data is 1; otherwise, it is 0.
- ✓ It is noted that we a delay unit is required to hold the staircase function for a period between two comparisons.

Pulse Modulation

❑ **Delta Demodulator:**



- ✓ The demodulator takes the digital data and, using the staircase maker and the delay unit, creates the analog signal.

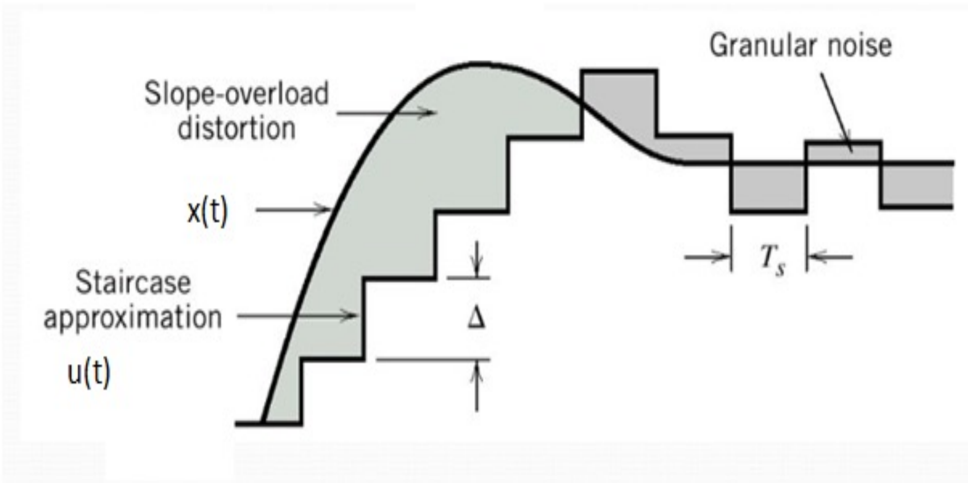
- ✓ The created analog signal, however, needs to pass through a low-pass filter for smoothing.

Pulse Modulation

- **Advantages and Disadvantages of Delta Modulation:**
  - **Advantages:** Delta modulation (DM) offers significant advantages in digital communication by utilizing a simple, cost-effective, one-bit quantization technique. Key benefits include:
    - ✓ Lower bandwidth requirements compared to PCM,
    - ✓ Simplified hardware implementation (no complex ADC),
    - ✓ Low power consumption suitable for battery-powered devices, and
    - ✓ Robustness against channel noise, making it ideal for voice encoding.

❑ Advantages and Disadvantages of Delta Modulation:

❑ Disadvantages:



- ✓ High quantization error.
- ✓ Slope overload distortion (occurs in case of **fast varying** input signal)
- ✓ Granular Noise (occurs in case of **slow varying** input signal)

❑ Delta Pulse Code Modulation (DPCM):

✓ **DPCM** is a digital modulation technique in which we transmit the **difference between the present sample and a predicted value of the previous sample**, instead of sending the actual sample value (as in PCM).

✓ Because adjacent samples of signals (like speech) are very similar, the **difference is small**, so fewer bits are required.

Instead of sending:

$$x(n)$$

We send:

$$e(n) = x(n) - \hat{x}(n)$$

Where:

- $x(n)$ = present sample
- $\hat{x}(n)$ = predicted sample
- $e(n)$ = prediction error (difference signal)

- **Advantages and Disadvantages of Delta PCM:**
  - **Advantages:**
    - ✓ Bandwidth requirement of **DPCM** is less compared to **PCM**.
    - ✓ Quantization error is reduced because of prediction filter.
    - ✓ Numbers of bits used to represent one sample value are also reduced compared to **PCM.**
  - **Disadvantages:**
    - ✓ Requires predictor circuit at both transmitter and receiver, increasing complexity.
    - ✓ Predictor at receiver must match transmitter, otherwise causing distortion.
    - ✓ Hardware design is more complicated than Delta Modulation, limiting practical usage.

Thank You All