Module 2

Physical Layer 1

Analog and Digital

Analog and Digital data

- Analog data – information that is continuous.

Eg: Analog clock

- Digital data – Information that has discrete states.

Eg: Digital clock

- Analog data takes on continuous values.
- Digital data takes on discrete values.

Analog and Digital

Analog and Digital Signals

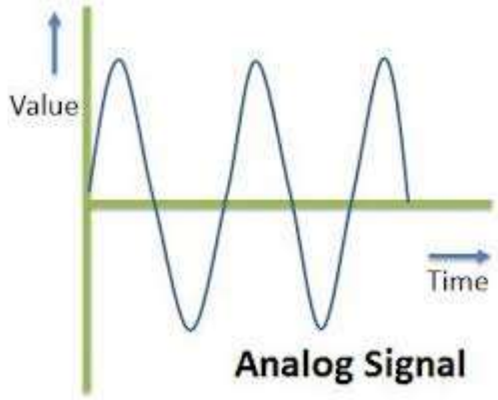
- Analog signal – has infinitely many levels of intensity over a period of time.

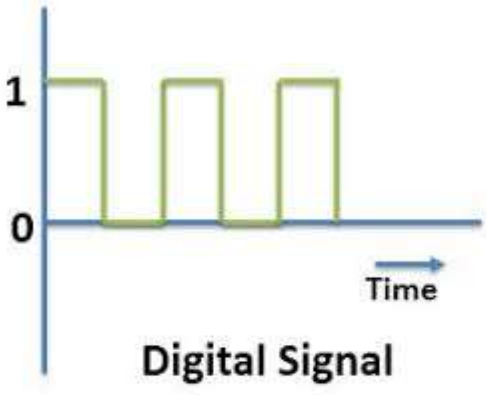
Eg: Waveform

- Digital signal – has limited number of defined values.

Representation of signals

- Plotting signals on a pair of perpendicular axes.
- Vertical axis – represents value or strength of a signal
- Horizontal axis – represents time





Periodic and Non-periodic Signals

- **Periodic signal** – Completes a pattern within a measurable time frame called period and repeats over subsequent identical periods.
- Completion of one full pattern is called a **cycle.**
- **Non-periodic signal –** changes without exhibiting a pattern or cycle that repeats over time.

Periodic analog signals

- Classified as - Simple or composite
- Simple periodic analog signal (Sine wave)- cannot be decomposed into simpler signals.
- Composite periodic analog signal – composed of multiple sine waves

Sine wave

- It is a simple curve, its changes over the course of a cycle is smooth and consistent, continuous , rolling flow.

Sine wave

Sine wave is represented by 3 parameters

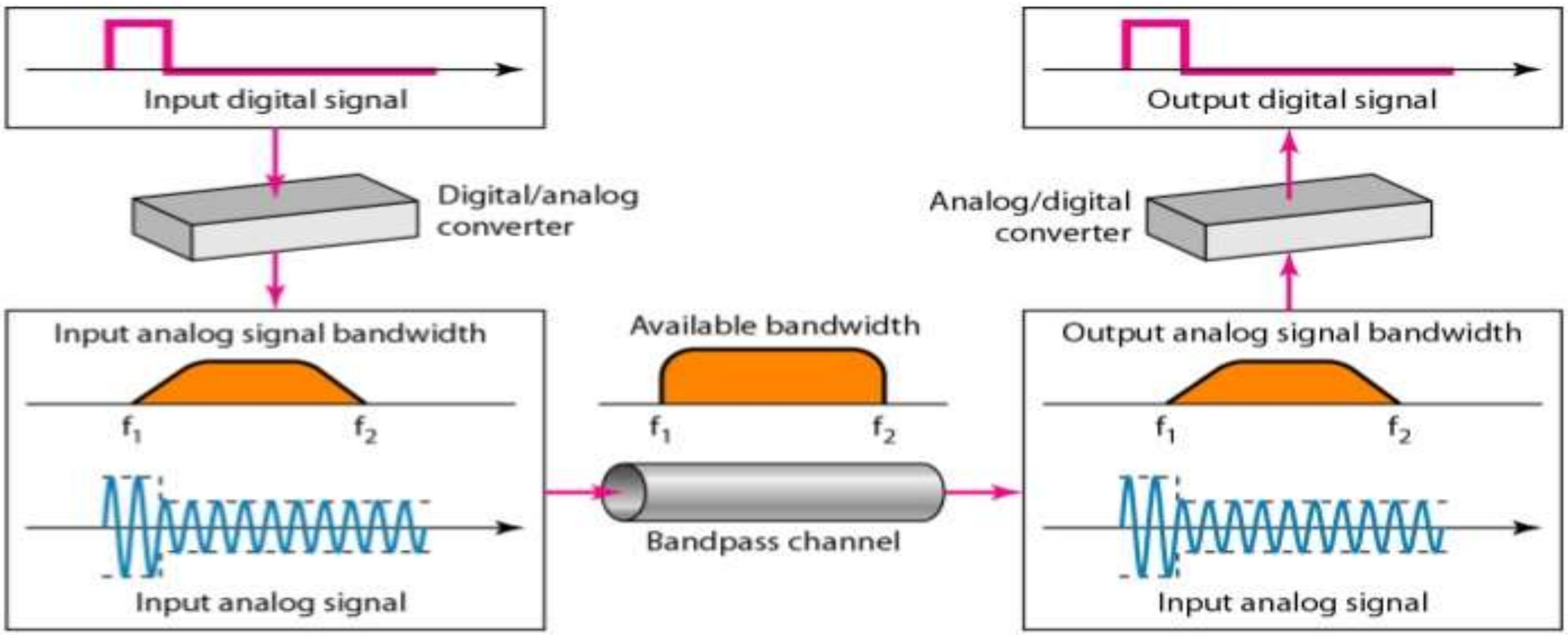
1. **Peak amplitude**

   It is the absolute value of its highest intensity, proportional to the energy it carries.

   For electric signals. Peak amplitude is normally measured in volts.

Sine wave

Figure 3.24 ***Modulation of a digital signal for transmission on a bandpass channel***

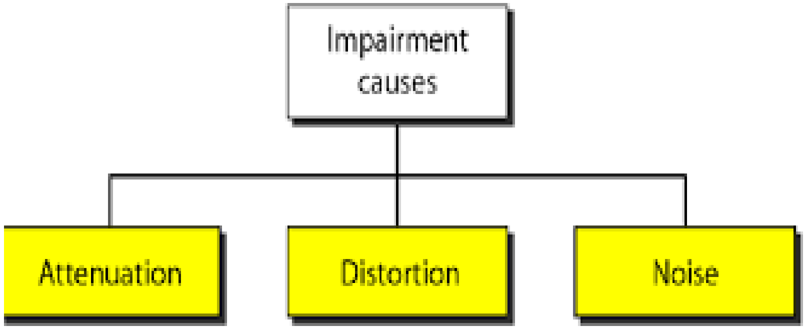


Transmission impairment

- Signal impairment – Signal at the beginning of the medium is not the same as signal at the end of the medium.
- What is sent is not what is received.

3 causes of impairment:

1. Attenuation
2. Distortion
3. Noise



Causes of impairment

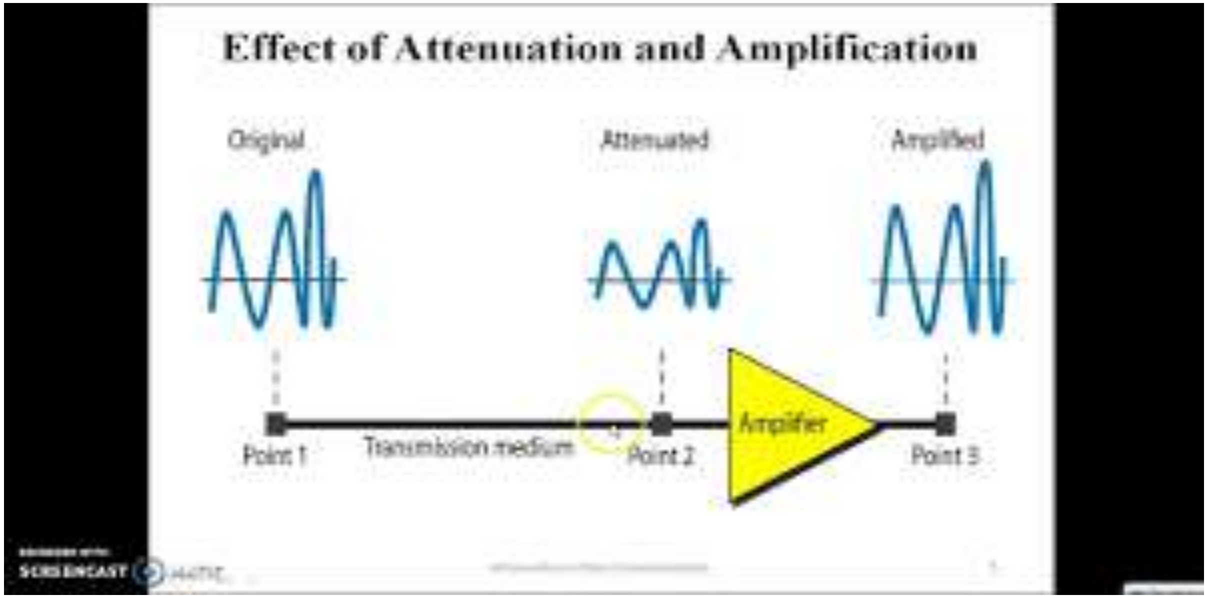
1. Attenuation

- It means loss of energy.
- When a signal travels through a medium, it loses some of its energy in overcoming the resistance of the medium.

Eg: Wire carrying electric signals get warm. Because some electrical energy is converted to heat.

Causes of impairment

Attenuation



Causes of impairment

2. Decibel

- It measures the relative strengths of two signals or one signal at two different points.
- Decibel is **negative** if signal is attenuated and **positive** if a signal is amplified.

$$dB = 10 \text{Log}\left(\frac{P_1}{P_2}\right)$$

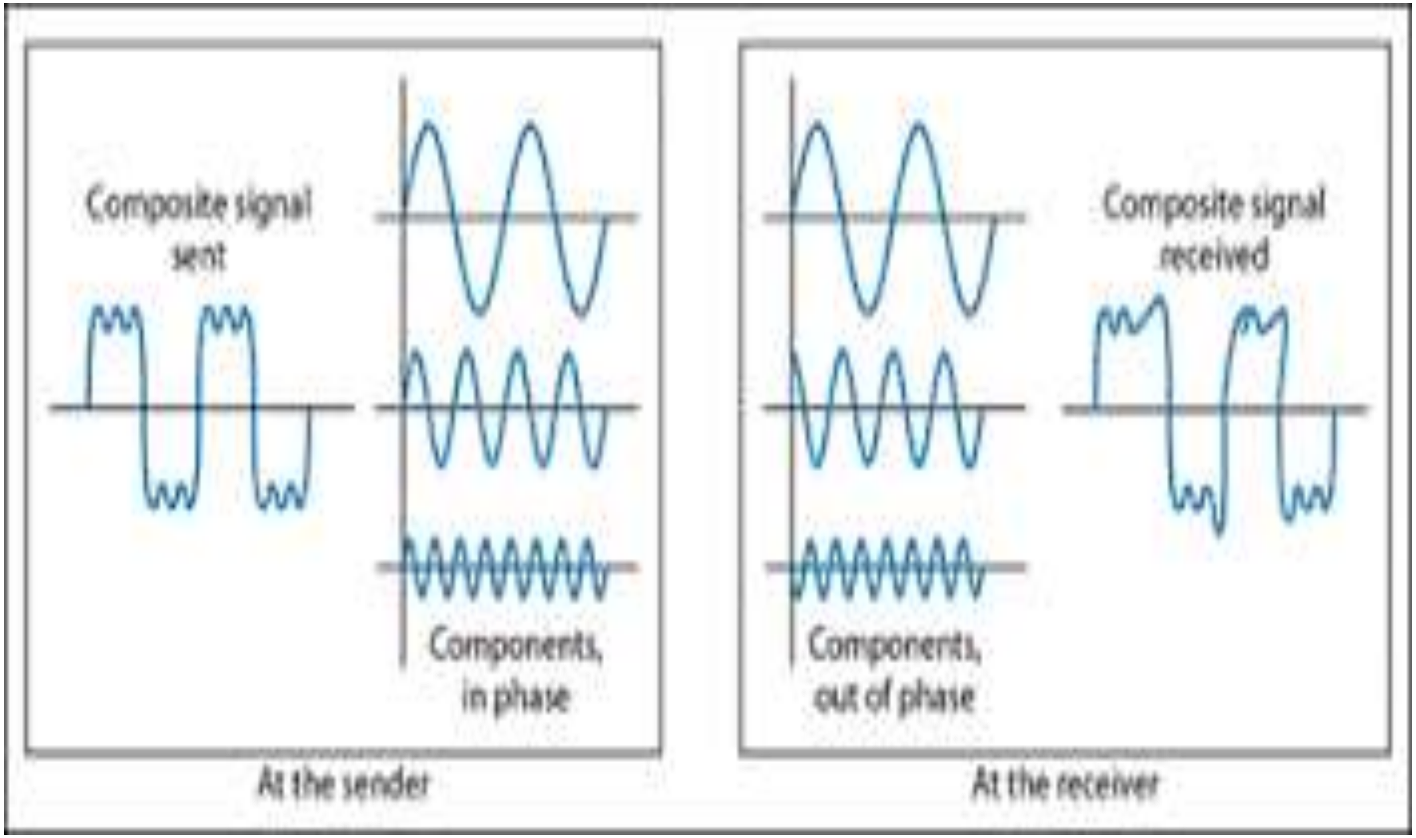
Causes of impairment

3. Distortion

- Change in form or shape of the signal.
- Distortion can occur in composite signals made of different frequencies.
- Each signal component has its own propagation speed and its own delay .
- Differences in delay creates difference in phase.
- Signal components at the receiver has phases different from what they have at sender.

Causes of impairment

Distortion



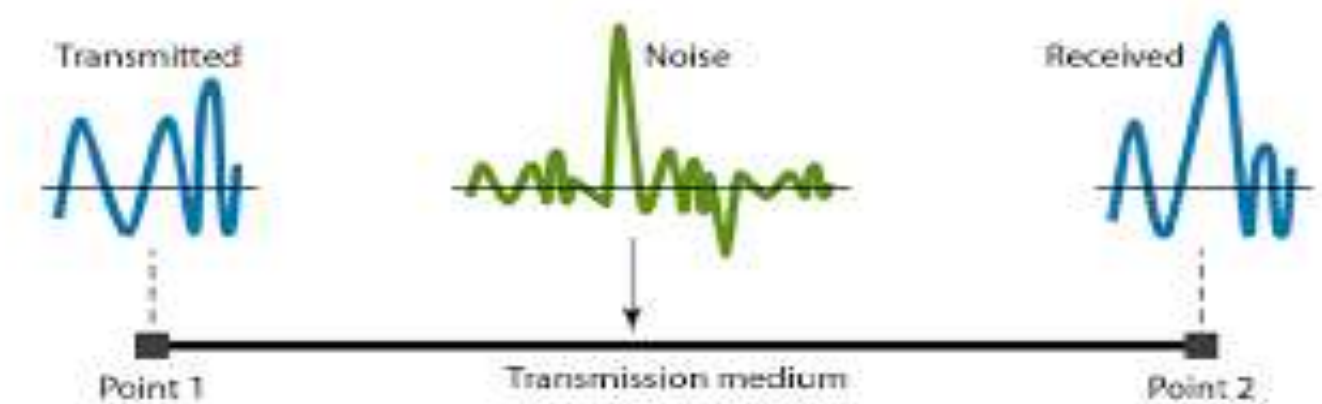
Causes of impairment

4. Noise

- It is another cause for impairment
- Types: thermal noise, induced noise, crosstalk noise, impulse noise
- **Thermal noise** – random motion of electrons in a wire which creates an extra signal not originally sent by transmitter.
- **Induced noise** – comes from sources such as motors and appliances.

Causes of impairment

- **Crosstalk** – It is the effect of one wire on another.
- **Impulse noise** – It is a spike (A signal with high energy in a very short time) that comes from power lines, lightning etc.



Causes of impairment

Signal to noise ratio

- It is the ratio of what is wanted(signal) to what is not wanted(noise)

Wanted component

$$SNR = \frac{P_{signal}}{P_{noise}}$$

Unwanted component

- It is defined in Decibel unit as

Calculation of SNR in db

- *The signal to noise ratio is often given in decibels also.*

  - $SNR_{db} = 10 \log_{10} SNR$

    or
  - $SNR_{db} = 10 \log_{10} S/N$

Module 2

Analog to Digital Conversion

4-2 ANALOG-TO-DIGITAL CONVERSION

A digital signal is superior to an analog signal because it is more robust to noise and can easily be recovered, corrected and amplified. For this reason, the tendency today is to change an analog signal to digital data. In this section we describe two techniques, pulse code modulation and delta modulation.

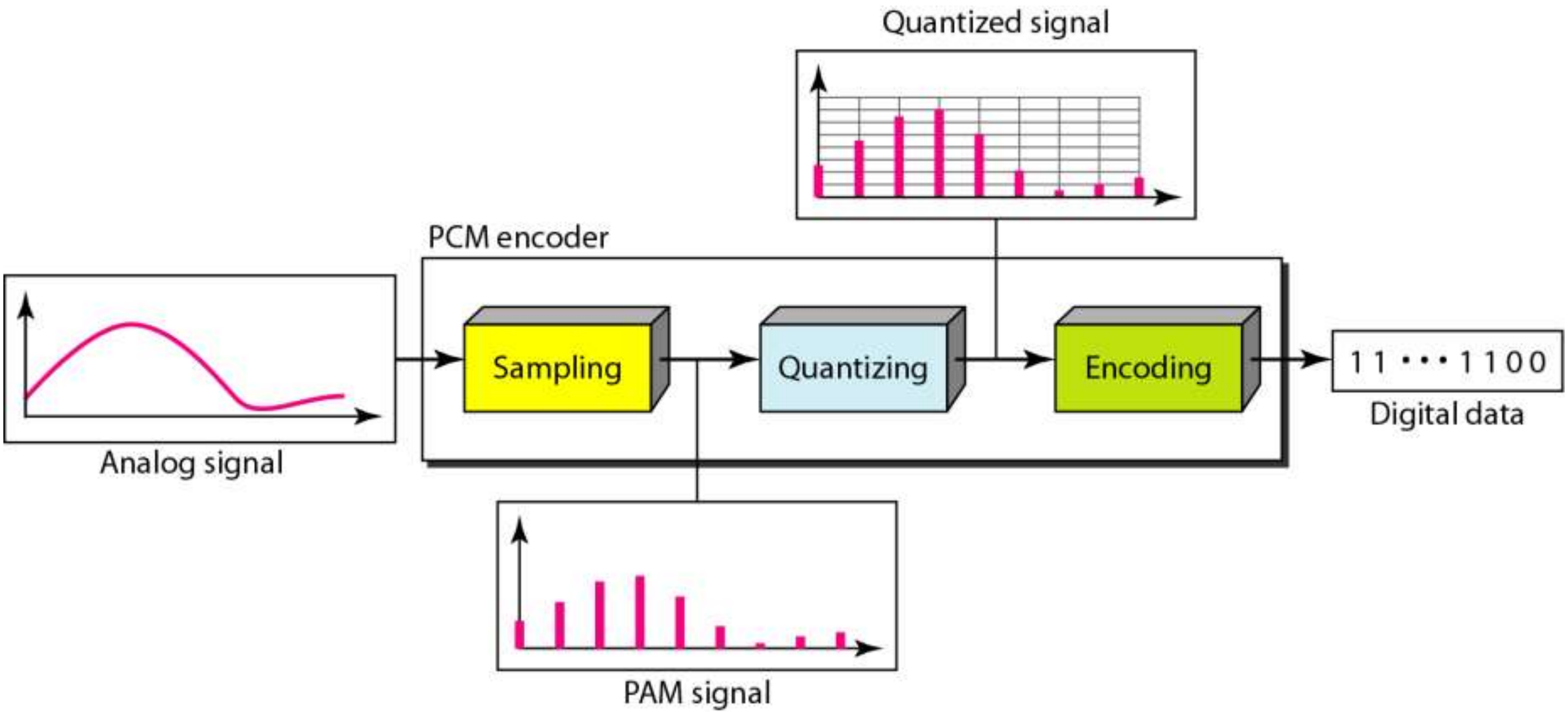
Topics discussed in this section:

- **Pulse Code Modulation (PCM)**
- **Delta Modulation (DM)**

PCM

- PCM consists of 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, ie remove high frequency components that affect the signal shape.

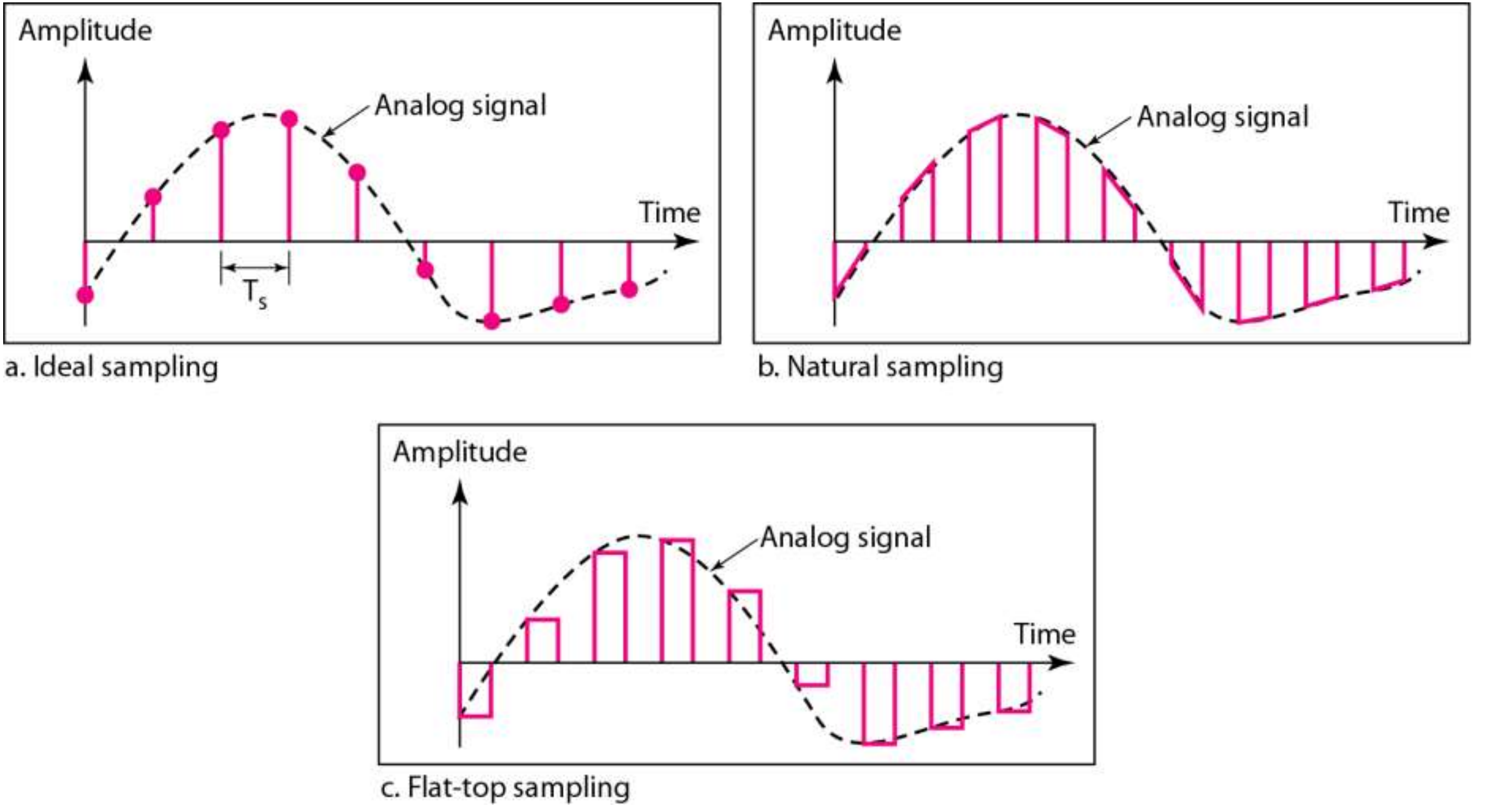
Figure 4.21 *Components of PCM encoder*



Sampling

- Analog signal is sampled every T_S secs.
- 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 pulse amplitude modulation PAM and the outcome is a signal with analog (non integer) values

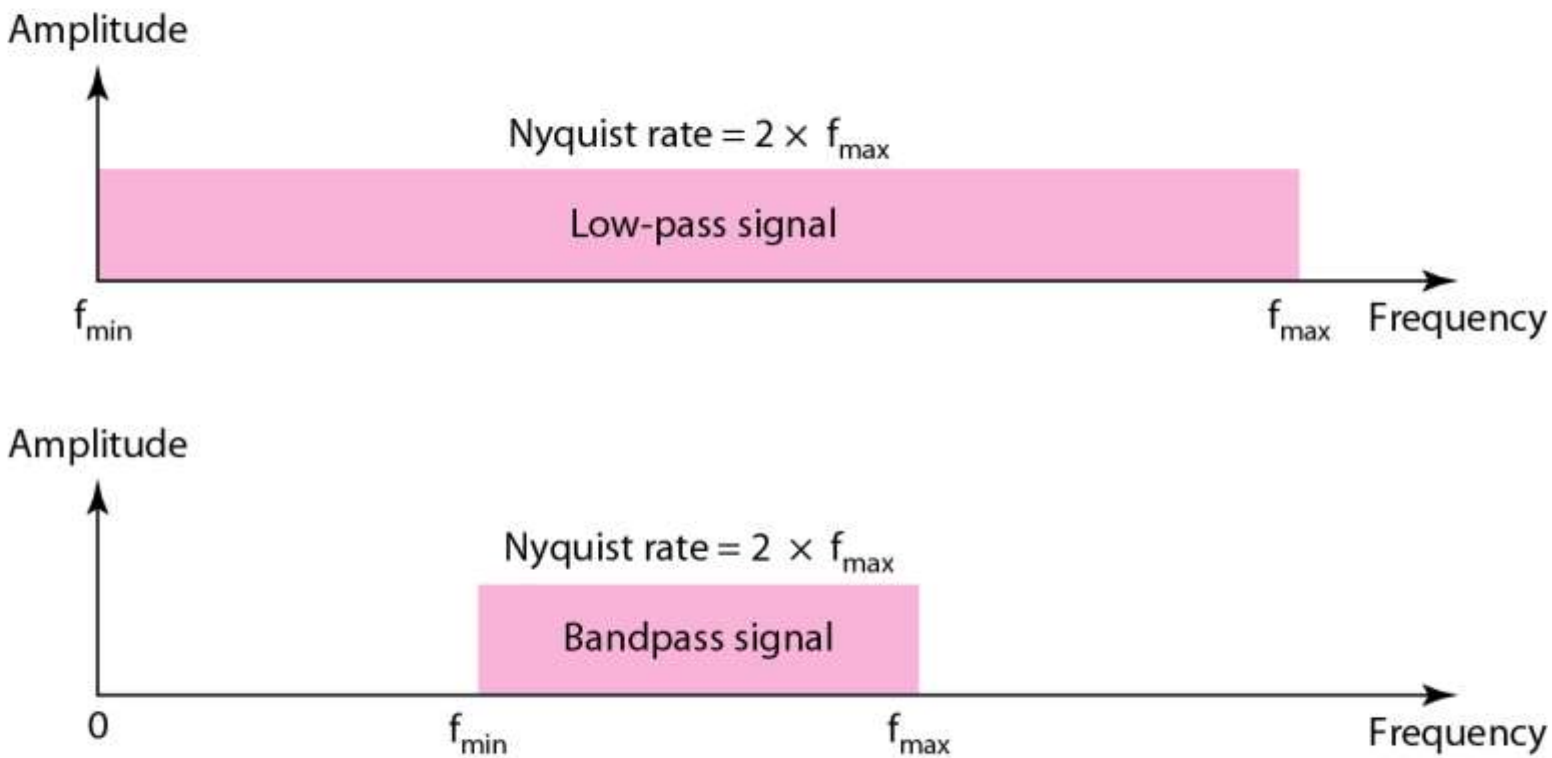
Figure 4.22 *Three different sampling methods for PCM*



Note

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

Figure 4.23 *Nyquist sampling rate for low-pass and bandpass signals*

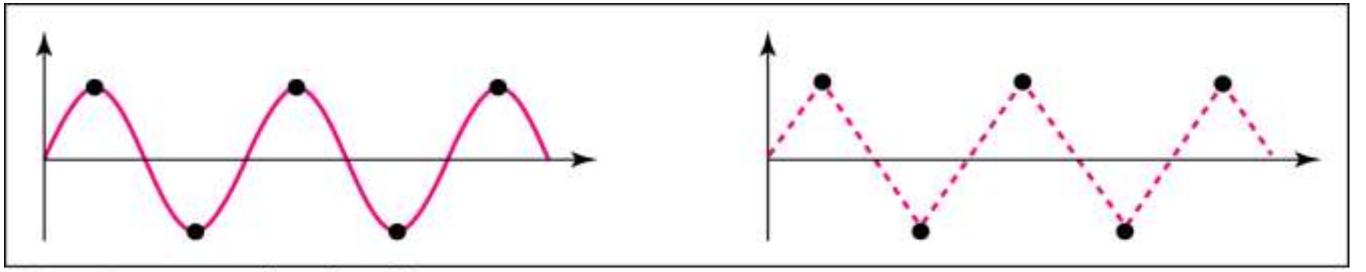


Example 4.6

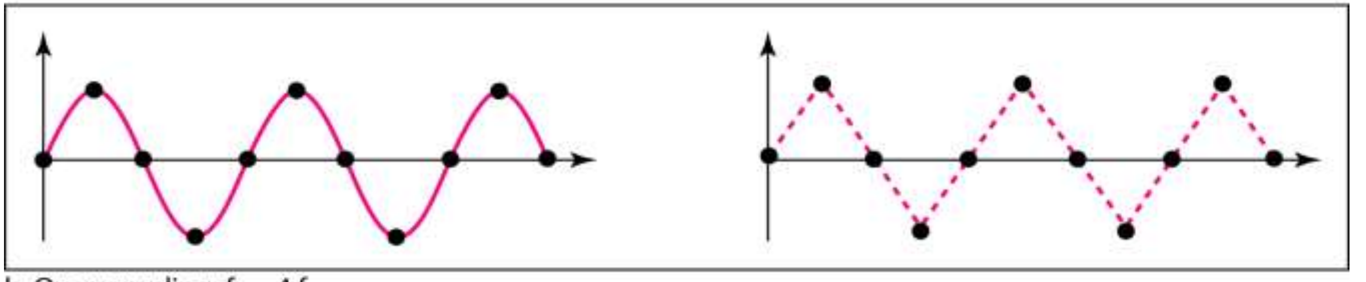
For an intuitive example of the Nyquist theorem, let us sample a simple sine wave at three sampling rates: $f_s = 4f$ (2 times the Nyquist rate), $f_s = 2f$ (Nyquist rate), and $f_s = f$ (one-half the Nyquist rate). Figure 4.24 shows the sampling and the subsequent recovery of the signal.

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.

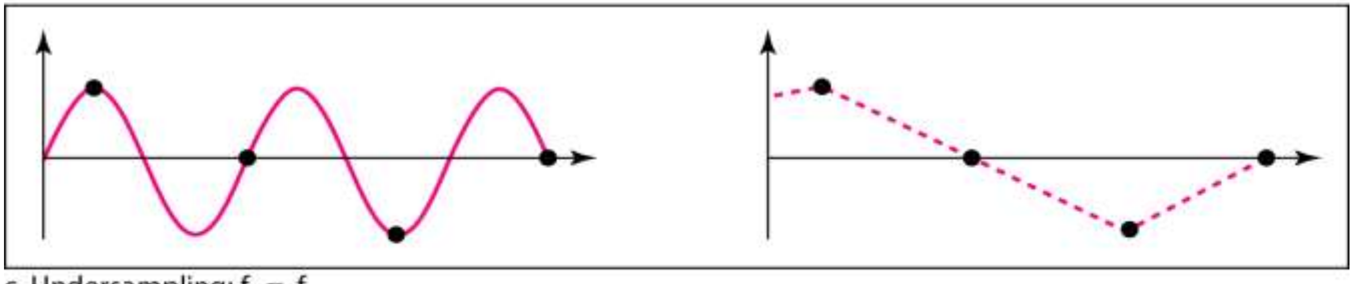
Figure 4.24 *Recovery of a sampled sine wave for different sampling rates*



a. Nyquist rate sampling: $f_s = 2f$



b. Oversampling: $f_s = 4f$

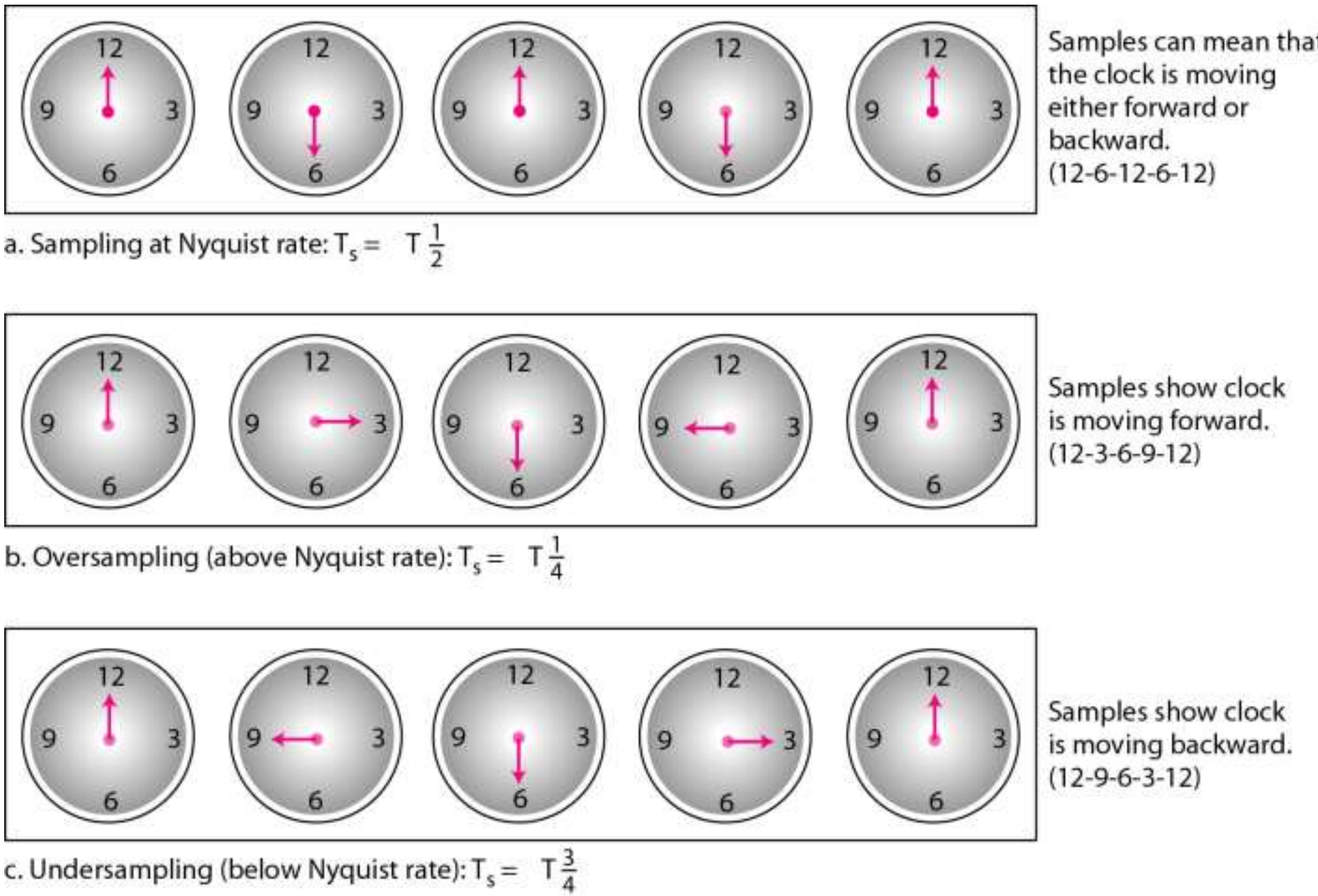


c. Undersampling: $f_s = f$

Example 4.7

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.

Figure 4.25 *Sampling of a clock with only one hand*



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.
- This is achieved by dividing the distance between min and max into L zones, each of height Δ.

$$\Delta = (\max - \min)/L$$

Quantization Levels

- The midpoint of each zone is assigned a value from 0 to L-1 (resulting in L values)
- Each sample falling in a zone is then approximated to the value of the midpoint.

Quantization Zones

- Assume we have a voltage signal with amplitutes V_{min}=-20V and V_{max}=+20V.
- We want to use L=8 quantization levels.
- Zone width Δ = (20 - -20)/8 = 5
- The 8 zones are: -20 to -15, -15 to -10, -10 to -5, -5 to 0, 0 to +5, +5 to +10, +10 to +15, +15 to +20
- The midpoints are: -17.5, -12.5, -7.5, -2.5, 2.5, 7.5, 12.5, 17.5

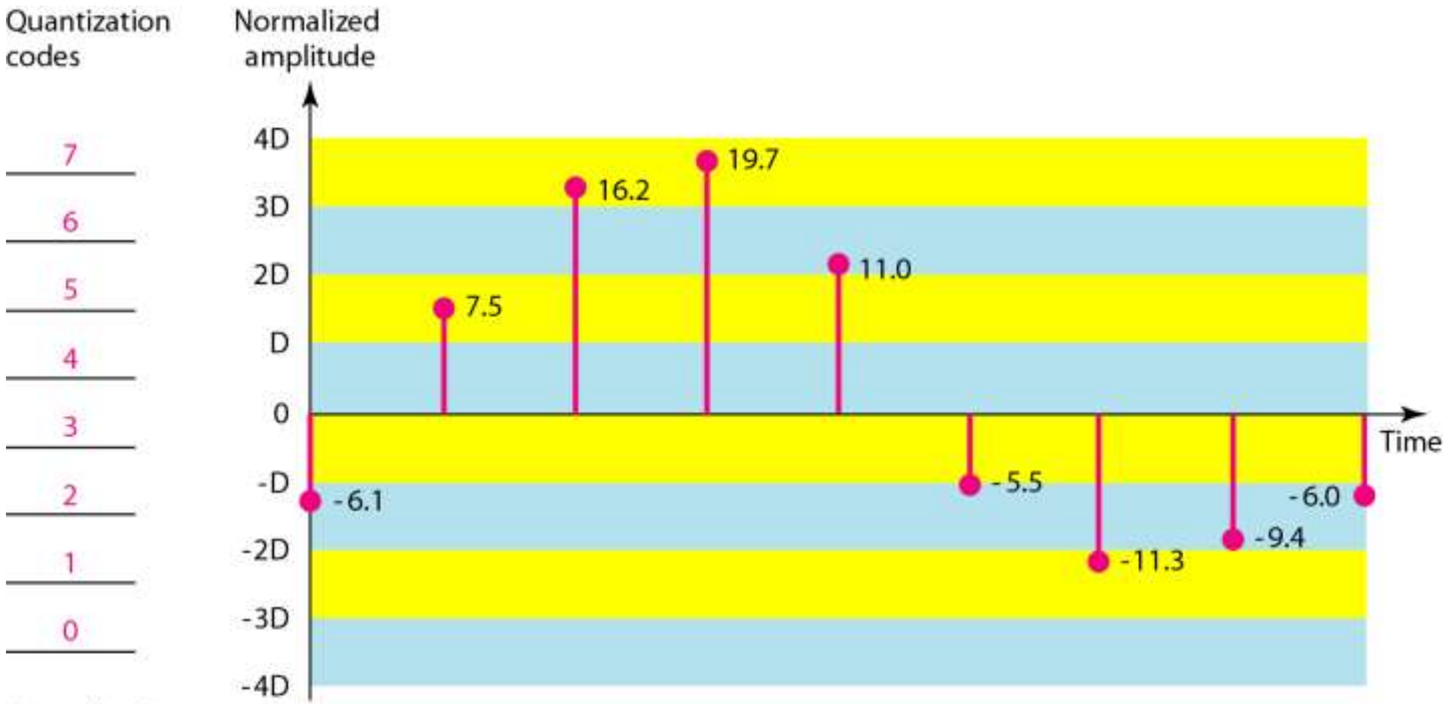
Assigning Codes to Zones

- Each zone is then assigned a binary code.
- The number of bits required to encode the zones, or the number of bits per sample as it is commonly referred to, is obtained as follows:

$$n_b = \log_2 L$$

- Given our example, $n_b = 3$
- The 8 zone (or level) codes are therefore: 000, 001, 010, 011, 100, 101, 110, and 111
- Assigning codes to zones:
  - 000 will refer to zone -20 to -15
  - 001 to zone -15 to -10, etc.

Figure 4.26 *Quantization and encoding of a sampled signal*



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

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 form the number of bits per sample x the sampling rate

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

- The bandwidth required to transmit this signal depends on the type of line encoding used. Refer to previous section for discussion and formulas.
- A digitized signal will always need more bandwidth than the original analog signal. 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

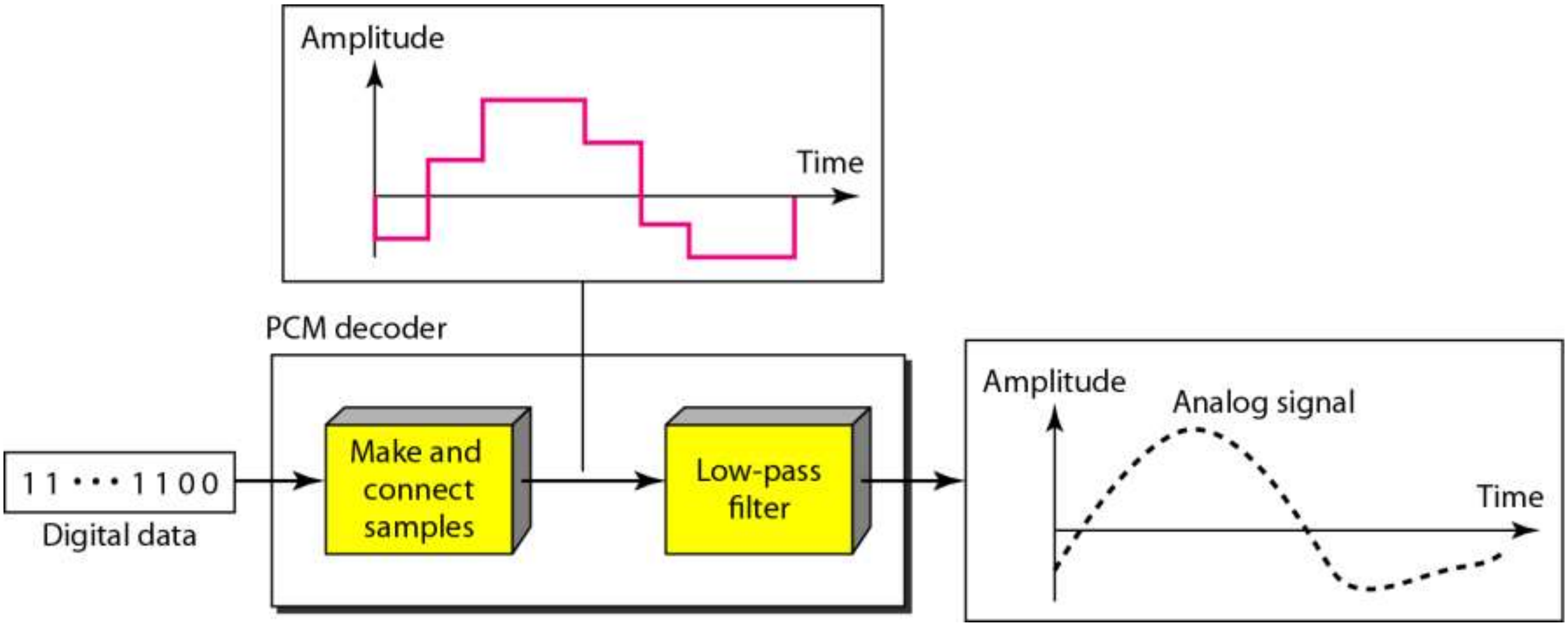
The human voice normally contains frequencies from 0 to 4000 Hz. So the sampling rate and bit rate are calculated as follows:

$$\text{Sampling rate} = 4000 \times 2 = 8000 \text{ samples/s}$$
$$\text{Bit rate} = 8000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}$$

PCM Decoder

- To recover an analog signal from a digitized signal we follow the following steps:
  - 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 higher the value of L, the less distorted a signal is recovered.

Figure 4.27 *Components of a PCM decoder*

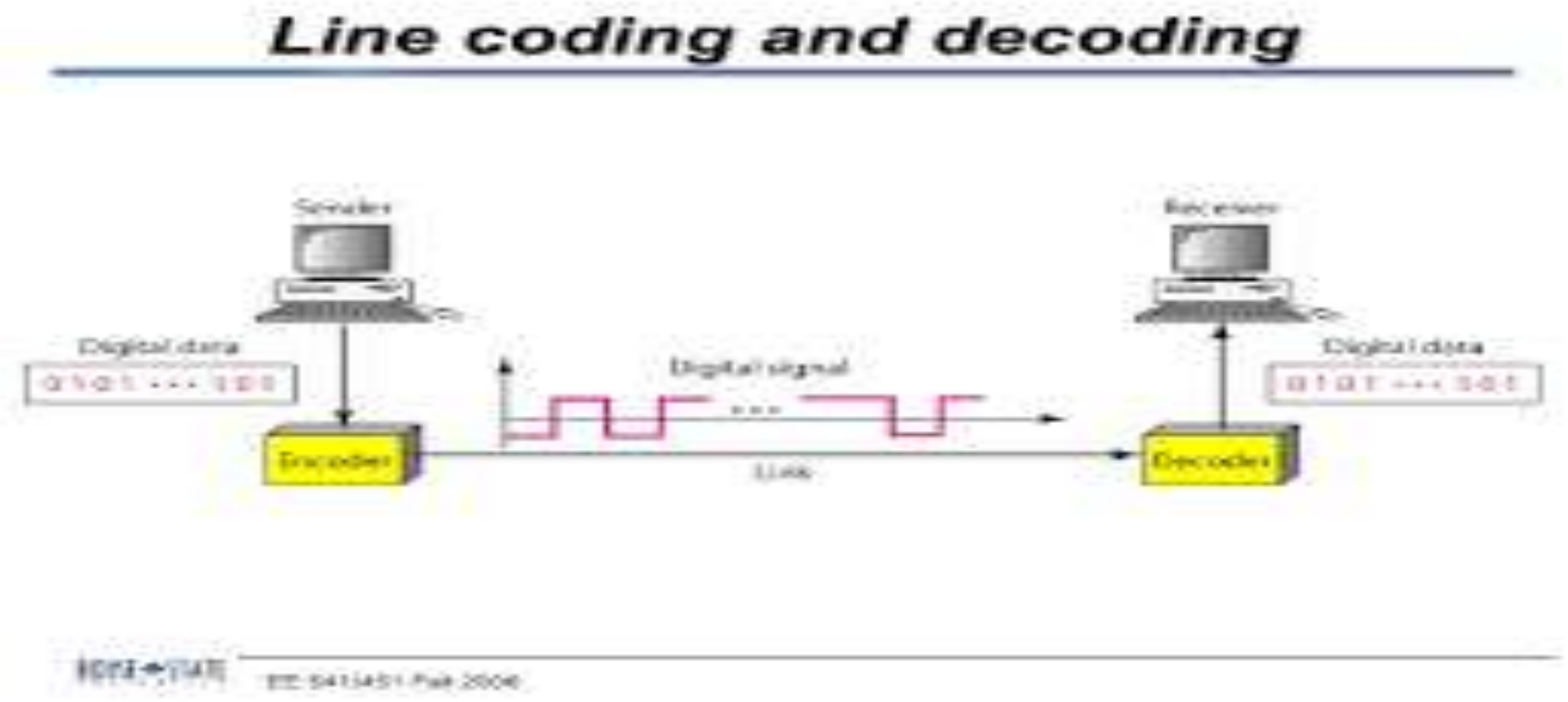


Module 2

Digital to Digital Conversion

Line coding

- It is the process of converting digital data to digital signals.
- Line coding converts a sequence of bits to a digital signal.
- At sender : Digital data are encoded into digital signal.
- At receiver : Digital data are recreated by decoding digital signal.



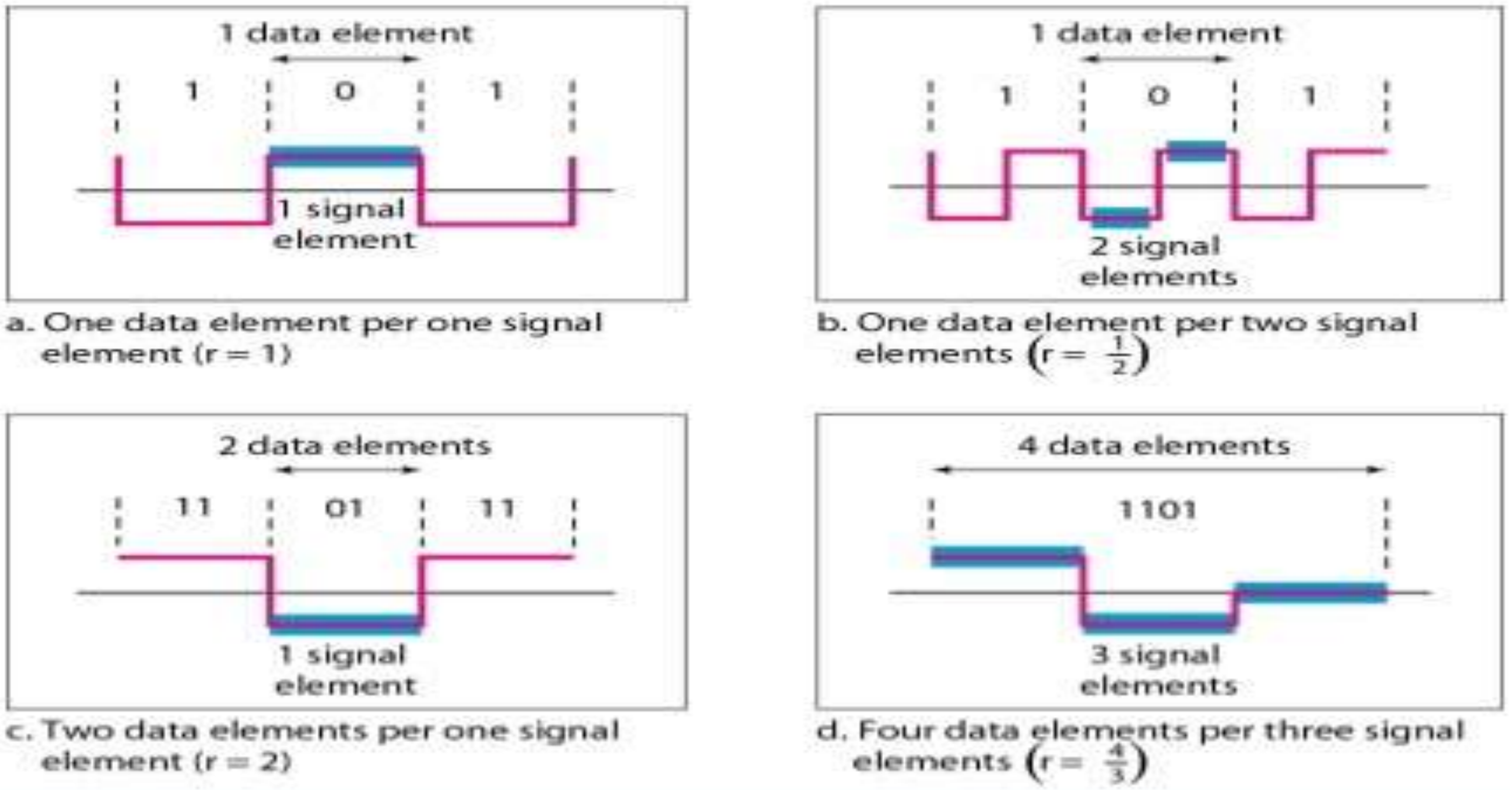
Characteristics

1. Signal Element Vs Data Element

- Data element is the smallest entity that can represent a piece of information i.e, Bit.
- Signal element is the shortest unit(timewise) of a digital signal.
- Data elements are being carried, signal elements are the carriers.

Characteristics

Figure 4.2 ***Signal element versus data element***



Characteristics

2. **Data rate Vs Signal rate**

- Data rate defines the number of data elements(bits) sent in 1s.
- Signal rate is the number of signal elements sent in 1s.
- The unit is **BAUD.**
- Data rate is also called **bit rate.**
- Signal rate is also called **pulse rate, modulation rate or baud rate.**

Goal : Increase data rate and decrease signal rate

Characteristics

- Relationship between data rate and signal rate depends on value of r.
- To derive a formula for relationship, three cases are defined.
  1. Worst case : when we need maximum signal rate.
  2. Best case : When we need minimum signal rate.
  3. Average case : Intermediate signal rate.

Relationship between signal rate and data rate

S= c* N * 1/r baud

N = data rate (bps)

C = Case factor (varies for each case)

S = number of signal elements

Characteristics

3. Bandwidth

- Actual bandwidth of a digital signal is infinite and the effective bandwidth is finite.
- Bandwidth reflects the range of frequencies.
- Bandwidth(range of frequencies) is proportional to the signal rate(baud rate)

Minimum bandwidth is given as

$$B_{min} = c * N * 1/r$$

Characteristics

4. Baseline Wandering

- In decoding digital signal, the receiver calculates a running average f the received signal power. This average is called baseline.
- The incoming power is evaluated against this baseline to determine the value of data element.
- A long string of 0s and 1s can cause a drift in the baseline (baseline wandering).
- Good line coding scheme must prevent baseline wandering.

Characteristics

5. DC Components

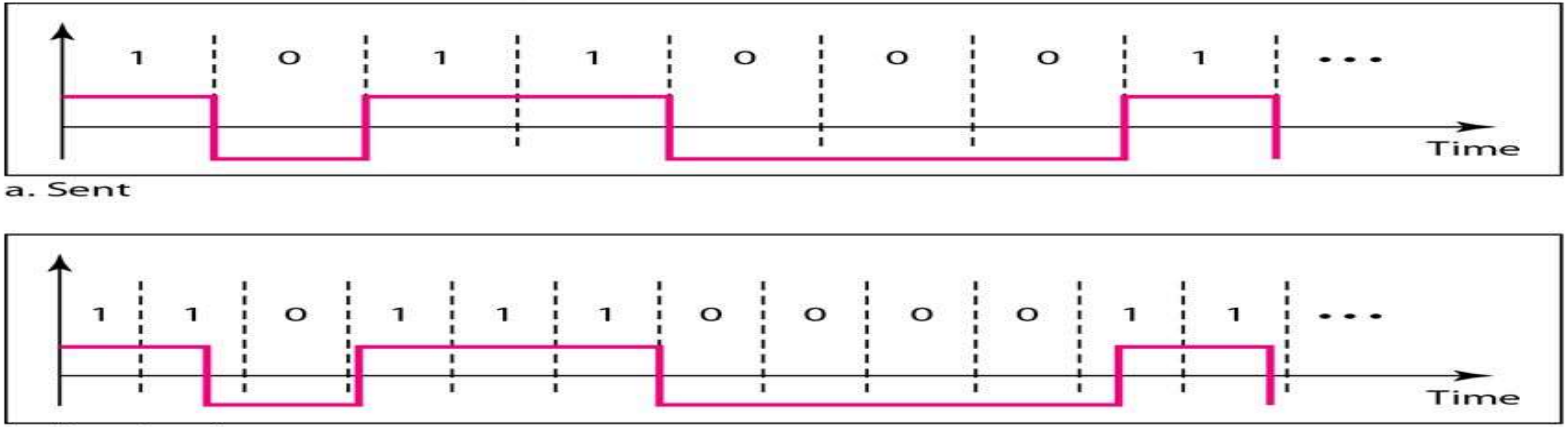
- When voltage level in a digital system is constant for a while, the spectrum creates very low frequencies. These frequencies around zero are called DC (direct current) components.

6. Self Synchronization

- To correctly interpret the signals received from the sender, the receiver bit intervals must correspond exactly to the sender bit intervals.
- If the receiver clock is faster or slower, bit intervals are not matched and receiver might misinterpret the signals.

Characteristics

Effect of lack of Synchronisation



a. Sent

b. Received

A self-synchronizing digital signal includes timing information in the data being transmitted. This can be achieved if there are transitions in the signal that alert the receiver to the beginning, middle, or end of the pulse. If the receiver' s clock is out of synchronization, these points can reset the clock.

Characteristics

7. Built in error Detection

- Better to have mechanisms to detect some f or all the errors that occurred during transmission.

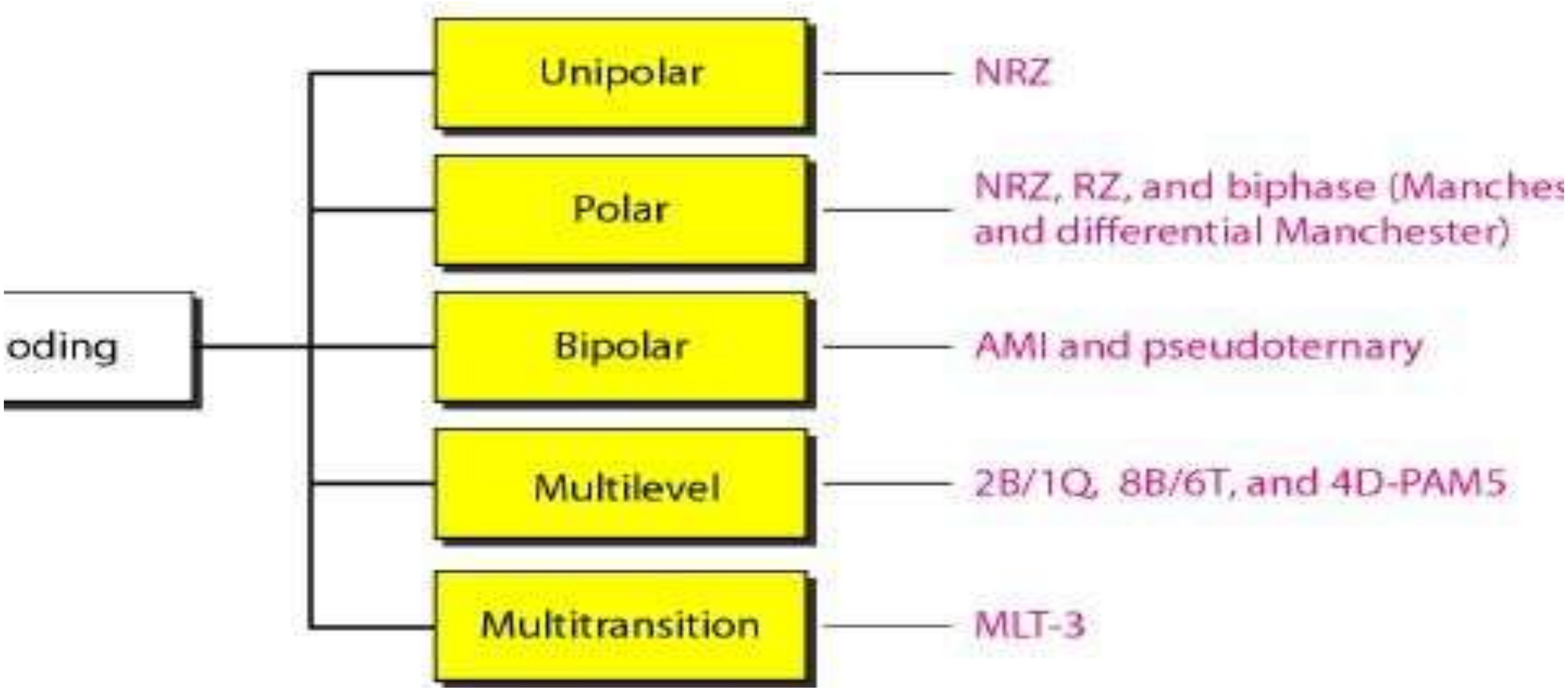
8. Immunity to noise and Interference

- Desirable code characteristic is a code that is immune to noise and other interferences.

9. Complexity

- A complex scheme is more costly to implement than a simple one.

Line Coding Schemes



Line Coding Schemes

1. Unipolar Scheme

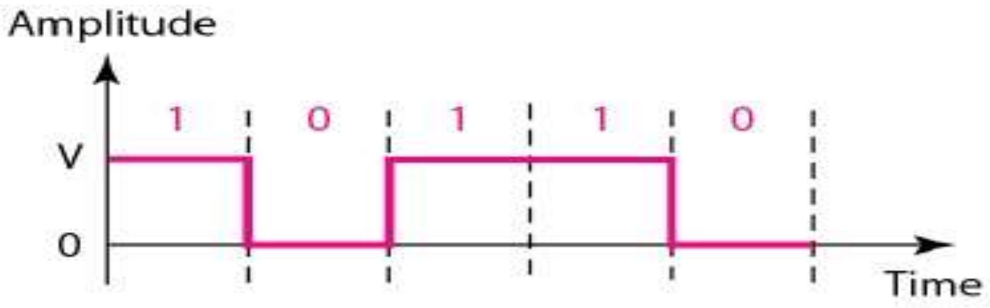
- All the signal levels are on one side of the time axis, either above or below.

NRZ (Non Return to Zero)

- Here, the positive voltage defines bit 1 and zero voltage defines bit 0.
- It is called NRZ because the signal does not return to zero at the middle of the bit.
- It is costly and normally not used in data communications today.

Line Coding Schemes

Figure 4.5 ***Unipolar NRZ scheme***



$$\boxed{\frac{1}{2}V^2 + \frac{1}{2}(0)^2 = \frac{1}{2}V^2}$$

Normalized power

Line Coding Schemes

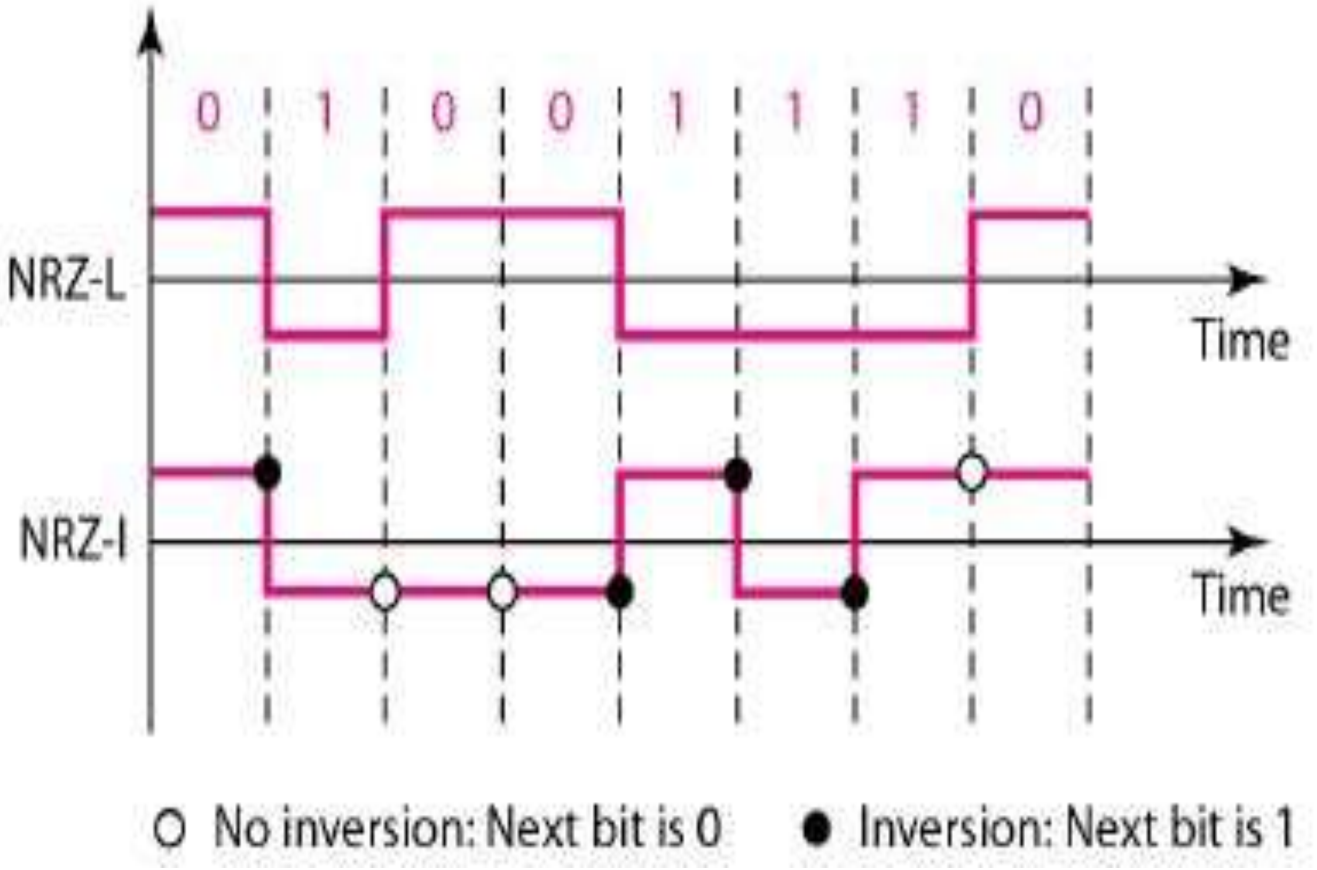
2. Polar Schemes

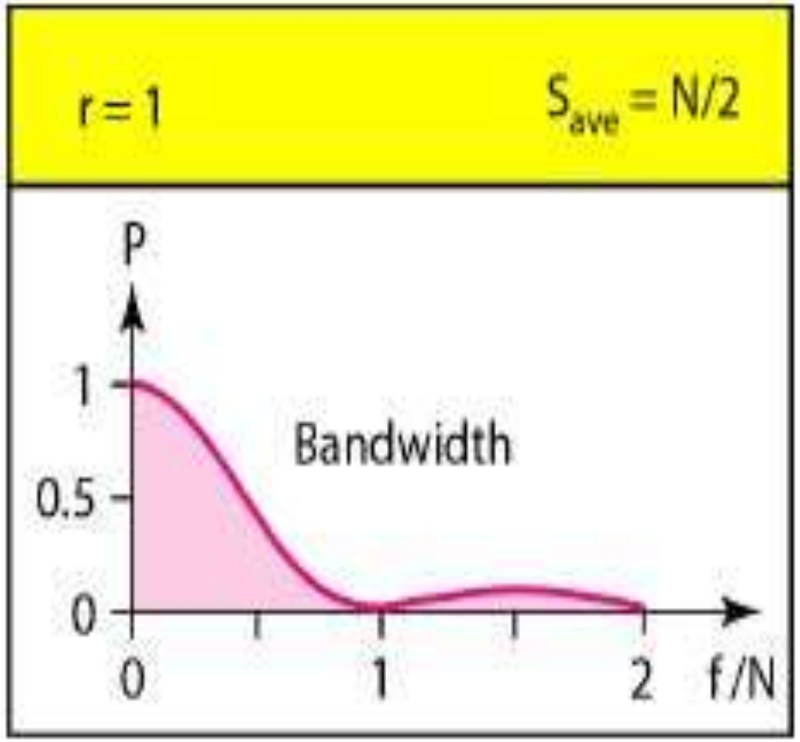
- The voltages are on both the sides of the time axis.

1. Non Return to Zero

- In polar NRZ encoding we use two levels of voltage amplitude.
- Two versions of polar NRZ
  - NRZ-L and NRZ-I
- NRZ-L(NRZ Level) : level of voltage determines value of the bit.
- NRZ-I(NRZ Invert) : change or lack of change in the voltage determines value of the bit. **<u>If no change bit is 0, if there is change bit is 1.</u>**

Line Coding Schemes





Line Coding Schemes

NRZ-L and NRZ-I Schemes

1. Baseline wandering is a problem in both variations. It is severe in NRZ-L.

2. Synchronization problem also exists in both. It is more serious in NRZ-L.

3. NRZ-I and NRZ-L both have an average signal rate of N/2 baud.

4. They both have DC component problem.

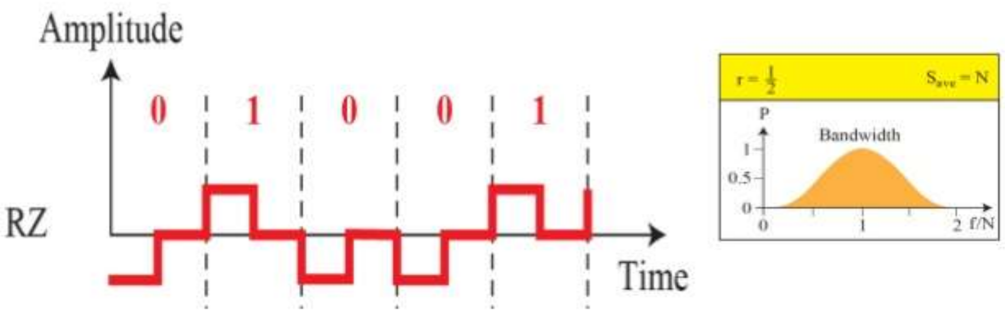
Line Coding Schemes

2. Return to Zero

- It is the solution to synchronization
- problem that occurs in NRZ scheme.
- It uses three values:
  - **Positive, Negative and Zero**

  In RZ, the signal changes not between bits but during the bits.

Figure: Polar schemes (RZ)



RZ scheme uses self clocking

4.21

- Here the signal goes to 0 in the middle of each bit. It remains there until the beginning of next bit.
- Disadvantage: It requires two signal changes to encode a bit and therefore occupies greater bandwidth.

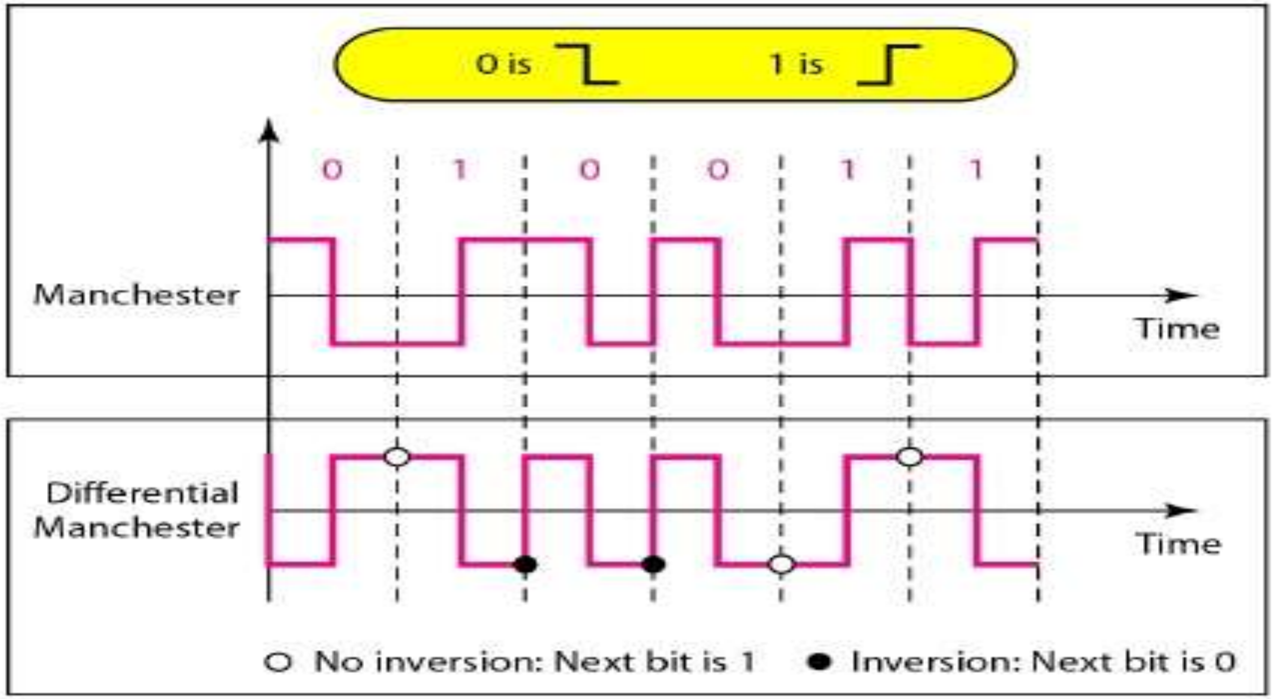
Line Coding Schemes

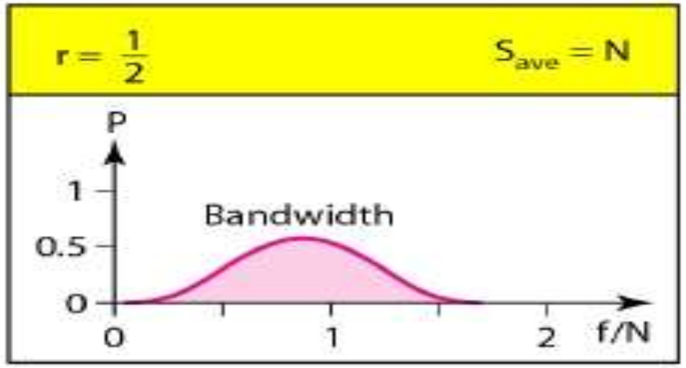
3. Biphase : Manchester and Differential Manchester

- Idea of RZ and NRZ-L are combined into Manchester scheme.
- The duration of the bit is divided into two halves.
- Voltage remains at one level during first half and moves to other level in the second half.
- Idea f RZ and NRZ-I are combined into Differential Manchester.
- If the next bit is 0, there is a transition. If the next bit is 1, there is o transition.

Line Coding Schemes

Figure 4.8 ***Polar biphase: Manchester and differential Manchester schemes***





Line Coding Schemes

Bipolar Schemes

- It is also called multilevel binary.
- There are three voltage levels: Positive, Negative and Zero.

1. AMI and Pseudoternary

- Alternate Mark Inversion, in which mark means 1.
- Zero voltage represents binary 0. Binary 1s are represented by alternating positive and negative voltages.
- An variation of AMI is called Pseudoternary in which 1 bit is encoded as zero voltage and 0 bit is encoded as alternating positive and negative voltages.

Line Coding Schemes