# MODULE 1(CONT.): DATA AND SIGNALS

## 1.8 DATA AND SIGNALS

### 1.8.1 Analog & Digital Data

• To be transmitted, data must be transformed to electromagnetic-signals.
• Data can be either analog or digital.

1) **Analog Data** refers to information that is continuous.  For example:
   The sounds made by a human voice.
2) **Digital Data** refers to information that has discrete states.  For example:
   Data are stored in computer-memory in the form of 0s and 1s.

### 1.8.2 Analog & Digital Signals

• Signals can be either analog or digital (Figure 3.2).

**1) Analog Signal** has infinitely many levels of intensity over a period of time. **2) Digital Signal** can have only a limited number of defined values.

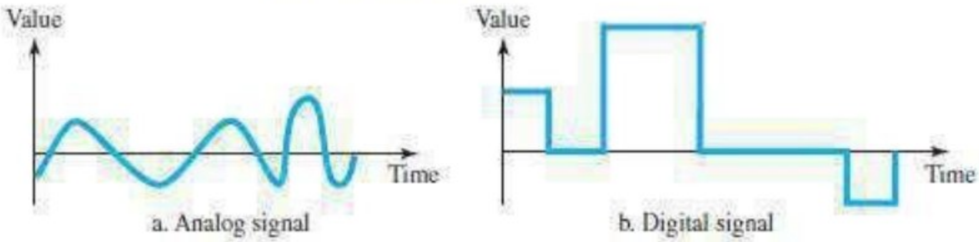



Figure 3.2 *Comparison of analog and digital signals*

### 1.8.3 Periodic & Non-Periodic Signals

• The signals can take one of 2 forms: periodic or non-periodic. **1)**

**Periodic Signal**

 Signals which repeat itself after a fixed time period are called Periodic Signals. The completion of one full pattern is called a cycle.

**2) Non-Periodic Signal**

 Signals which do not repeat itself after a fixed time period are called Non-Periodic Signals.

### 1.9 DIGITAL SIGNALS

• Information can be represented by a digital signal.
• For example:
    1) 1 can be encoded as a positive voltage.
        0 can be encoded as a zero voltage (Figure 3.17a).
    2) A digital signal can have more than 2 levels (Figure 3.17b).

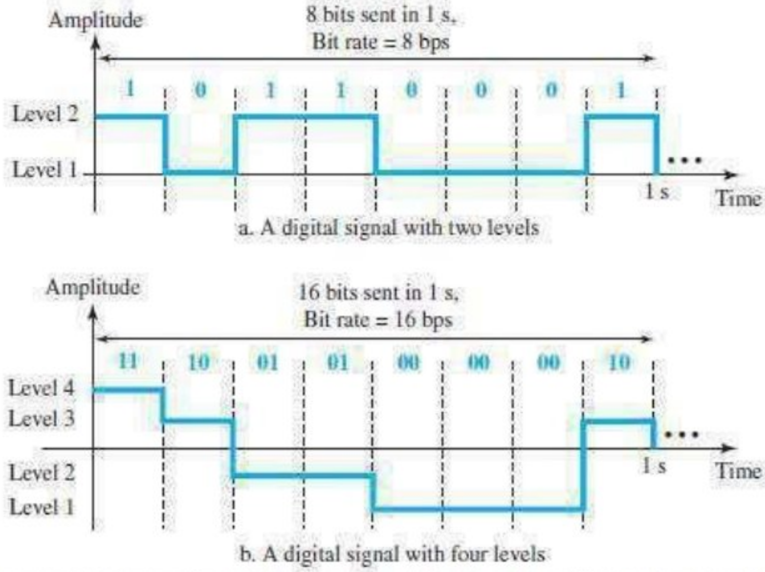



**Figure 3.17** *Two digital signals: one with two signal levels and the other with four signal levels*

**Example 1.1**

A digital signal has eight levels. How many bits are needed per level? We calculate the number of bits from the following formula. Each signal level is represented by 3 bits.

$$\text{Number of bits per level} = \log_2 8 = 3$$

### 1.9.1 Bit Rate

• The bit rate is the number of bits sent in 1s.
• The bit rate is expressed in bits per second (bps).

**Example 1.2**

Assume we need to download text documents at the rate of 100 pages per second. What is the required bit rate of the channel?

**Solution**

A page is an average of 24 lines with 80 characters in each line. If we assume that one character requires 8 bits, the bit rate is

$$100 \times 24 \times 80 \times 8 = 1{,}536{,}000 \text{ bps} = 1.536 \text{ Mbps}$$

**Example 1.3**

A digitized voice channel, as we will see in Chapter 4, is made by digitizing a 4-kHz bandwidth analog voice signal. We need to sample the signal at twice the highest frequency (two samples per hertz). We assume that each sample requires 8 bits. What is the required bit rate?

**Solution**

The bit rate can be calculated as

$$2 \times 4000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}$$

**Example 1.4**

What is the bit rate for high-definition TV (HDTV)?

**Solution**

HDTV uses digital signals to broadcast high quality video signals. The HDTV screen is normally a ratio of 16 : 9 (in contrast to 4 : 3 for regular TV), which means the screen is wider. There are 1920 by 1080 pixels per screen, and the screen is renewed 30 times per second. Twenty-four bits represents one color pixel. We can calculate the bit rate as

$$1920 \times 1080 \times 30 \times 24 = 1{,}492{,}992{,}000 \approx 1.5 \text{ Gbps}$$

### 1.9.2 Bit Length

- The bit length is the distance one bit occupies on the transmission medium.

**Bit length = propagation speed × bit duration**

### 1.9.3 Digital Signal as a Composite Analog Signal

- A digital signal is a composite analog signal.
- A digital signal, in the time domain, comprises connected vertical and horizontal line segments.
  1) A vertical line in the time domain means a frequency of infinity (sudden change in time); 2) A horizontal line in the time domain means a frequency of zero (no change in time).
- Fourier analysis can be used to decompose a digital signal.
  1) If the digital signal is periodic, the decomposed signal has a frequency domain representation with an infinite bandwidth and discrete frequencies (Figure 3.18a).
  2) If the digital signal is non-periodic, the decomposed signal has a frequency domain representation with an infinite bandwidth and continuous frequencies (Figure 3.18b).

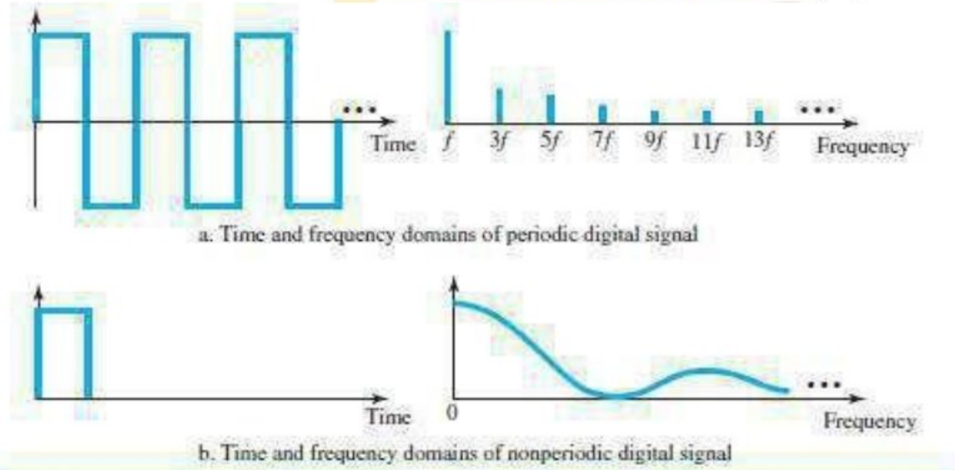



**Figure 3.18** *The time and frequency domains of periodic and nonperiodic digital signals*

### 1.9.4 Transmission of Digital Signals

- Two methods for transmitting a digital signal:
  1) Baseband transmission 2) Broadband transmission (using modulation).

### 1.9.4.1 Baseband Transmission

• Baseband transmission means sending a digital signal over a channel without changing the digital signal to an analog signal (Figure 3.19).

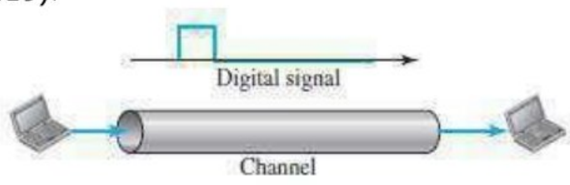



Figure 3.19 *Baseband transmission*

• Baseband transmission requires that we have a low-pass channel.
• Low-pass channel means a channel with a bandwidth that starts from zero.
• For example, we can have a dedicated medium with a bandwidth constituting only one channel.

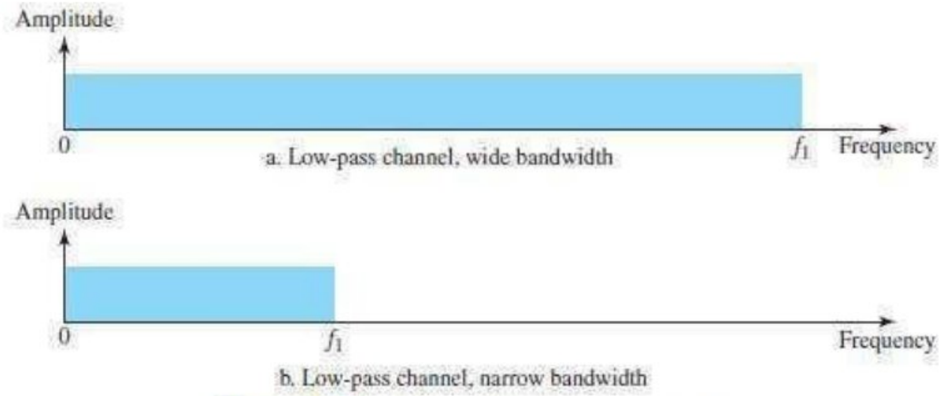



Figure 3.20 *Bandwidths of two low-pass channels*

• Two cases of a baseband communication:
Case 1: Low-pass channel with a wide bandwidth (Figure 3.20a)
Case 2: Low-pass channel with a limited bandwidth (Figure 3.20b)

**Case 1: Low-Pass Channel with Wide Bandwidth**

- To preserve the shape of a digital signal, we need to send the entire spectrum i.e. the continuous range of frequencies between zero and infinity.
- This is possible if we have a dedicated medium with an infinite bandwidth between the sender and receiver.
- If we have a medium with a very wide bandwidth, 2 stations can communicate by using digital signals with very good accuracy (Figure 3.21).
- Although the output signal is not an exact replica of the original signal, the data can still be deduced from the received signal.

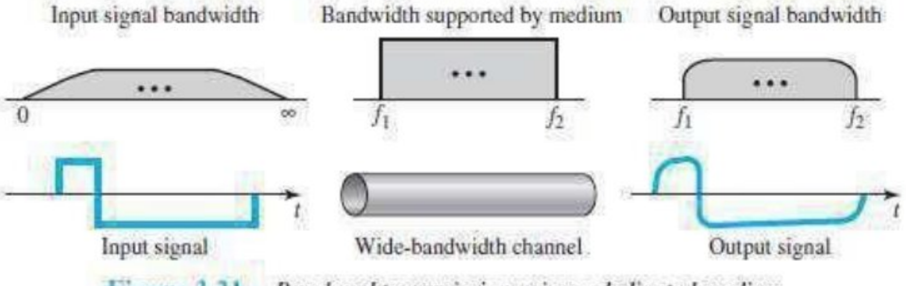



Figure 3.21 *Baseband transmission using a dedicated medium*

**Case 2: Low-Pass Channel with Limited Bandwidth**

- In a low-pass channel with limited bandwidth, we approximate the digital signal with an analog signal.

- The level of approximation depends on the bandwidth available.

**A) Rough Approximation**

¤ Assume that we have a digital signal of bit rate N (Figure 3.22).

¤ If we want to send analog signals to roughly simulate this signal, we need to consider the worst case, a maximum number of changes in the digital signal.

¤ This happens when the signal carries the sequence 01010101 . . or 10101010 .............

¤ To simulate these two cases, we need an analog signal of frequency f = N/2.

¤ Let 1 be the positive peak value and 0 be the negative peak value.

¤ We send 2 bits in each cycle; the frequency of the analog signal is one-half of the bit rate, or N/2.

¤ This rough approximation is referred to as using the first harmonic (N/2) frequency. The required bandwidth is

$$\textbf{Bandwidth} = \frac{N}{2} - 0 = \frac{N}{2}$$

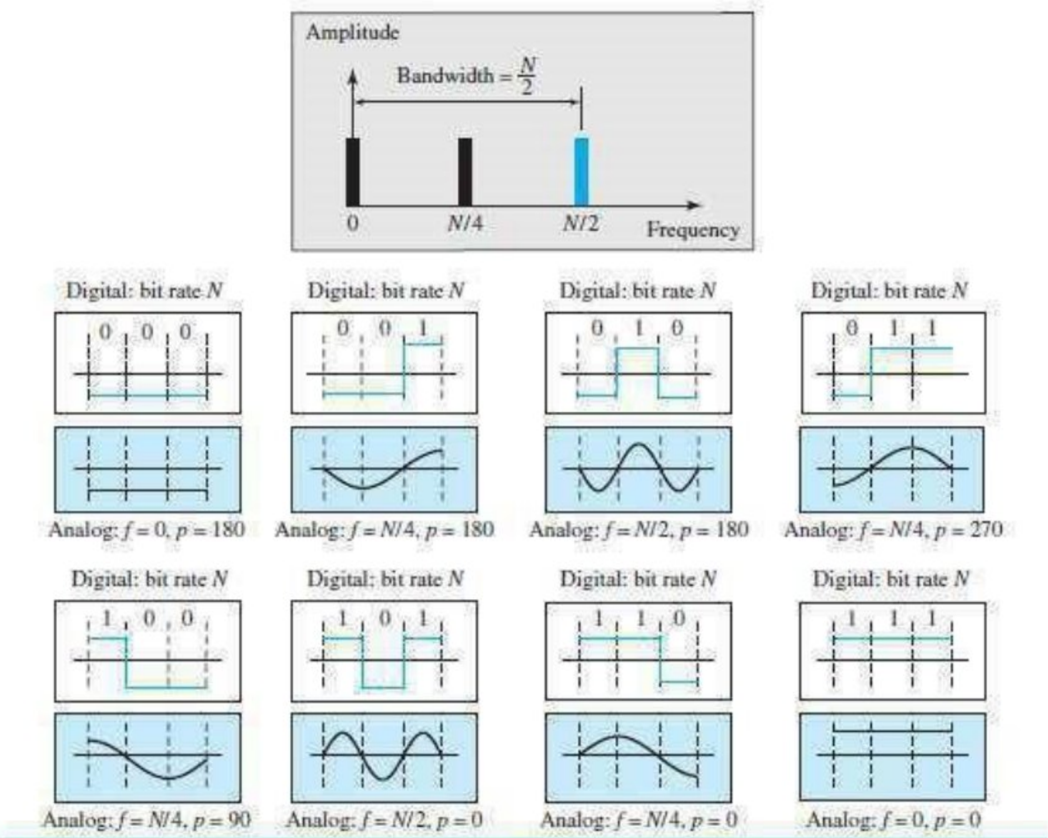



**Figure 3.22** *Rough approximation of a digital signal using the first harmonic for worst case*

**B) Better Approximation**

¤ To make the shape of the analog signal look more like that of a digital signal, we need to add more harmonics of the frequencies (Figure 3.23).

¤ We can increase the bandwidth to 3N/2, 5N/2, 7N/2, and so on.

¤ In baseband transmission, the required bandwidth is proportional to the bit rate;
If we need to send bits faster, we need more bandwidth.

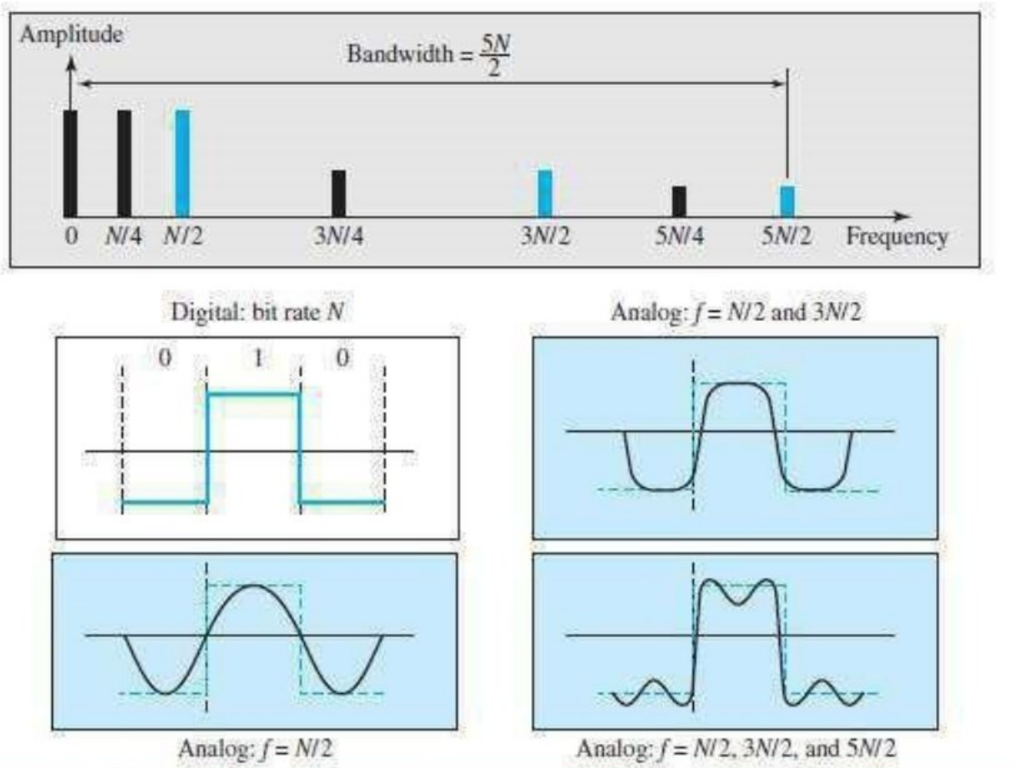



**Figure 3.23** *Simulating a digital signal with first three harmonics*

**Table 3.2** *Bandwidth requirements*

| *Bit Rate* | *Harmonic 1* | *Harmonics 1, 3* | *Harmonics 1, 3, 5* |
|---|---|---|---|
| $n$ = 1 kbps | $B$ = 500 Hz | $B$ = 1.5 kHz | $B$ = 2.5 kHz |
| $n$ = 10 kbps | $B$ = 5 kHz | $B$ = 15 kHz | $B$ = 25 kHz |
| $n$ = 100 kbps | $B$ = 50 kHz | $B$ = 150 kHz | $B$ = 250 kHz |

**Example 1.5**

What is the required bandwidth of a low-pass channel if we need to send 1 Mbps by using baseband transmission?

**Solution**

The answer depends on the accuracy desired.

a. The minimum bandwidth, a rough approximation, is $B$ = bit rate /2, or 500 kHz. We need a low-pass channel with frequencies between 0 and 500 kHz.

b. A better result can be achieved by using the first and the third harmonics with the required bandwidth $B = 3 \times 500$ kHz = 1.5 MHz.

c. A still better result can be achieved by using the first, third, and fifth harmonics with $B = 5 \times 500$ kHz = 2.5 MHz.

**Example 1.6**

We have a low-pass channel with bandwidth 100 kHz. What is the maximum bit rate of this channel?

**Solution**

The maximum bit rate can be achieved if we use the first harmonic. The bit rate is 2 times the available bandwidth, or 200 kbps.

### 1.9.4.2 Broadband Transmission (Using Modulation)

- Broadband transmission or modulation means changing the digital signal to an analog signal for transmission.
- Modulation allows us to use a bandpass channel (Figure 3.24).
- Bandpass channel means a channel with a bandwidth that does not start from zero.
- This type of channel is more available than a low-pass channel.


Figure 3.24 *Bandwidth of a bandpass channel*

- If the available channel is a bandpass channel,
  We cannot send the digital signal directly to the channel;
  We need to convert the digital signal to an analog signal before transmission (Figure 3.25).

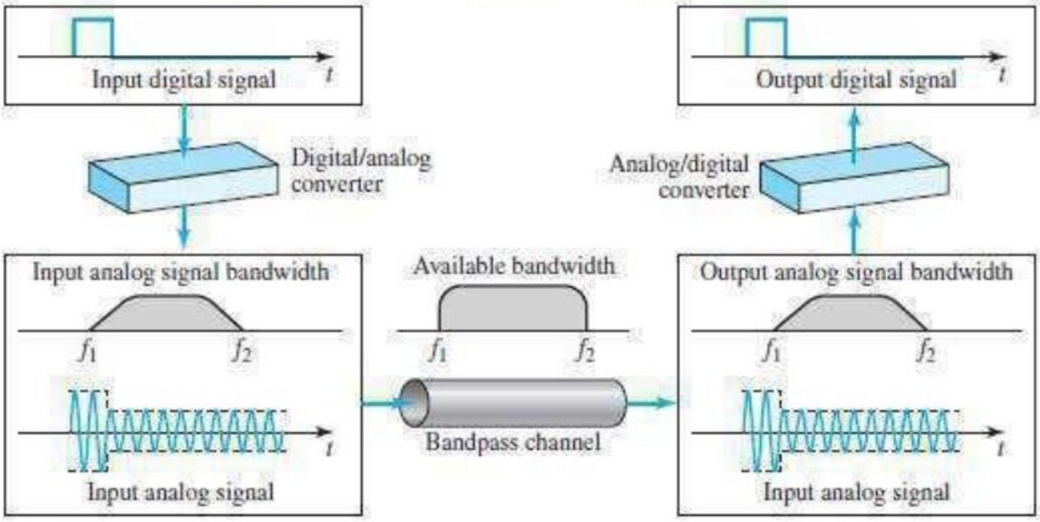



Figure 3.25 *Modulation of a digital signal for transmission on a bandpass channel*

### 1.10 TRANSMISSION IMPAIRMENT

- Signals travel through transmission media, which are not perfect.
- The imperfection causes signal-impairment.
- This means that signal at beginning of the medium is not the same as the signal at end of medium.
- What is sent is not what is received.
- Three causes of impairment are (Figure 3.26):

1) Attenuation
2) Distortion &
3) Noise.

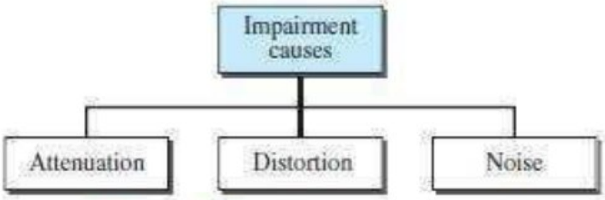



Figure 3.26 *Causes of impairment*

**1.10.1 Attenuation**

- As signal travels through the medium, its strength decreases as distance increases. This is called attenuation (Figure 3.27).
- As the distance increases, attenuation also increases.
- For example:
  Voice-data becomes weak over the distance & loses its contents beyond a certain distance.
- To compensate for this loss, amplifiers are used to amplify the signal.

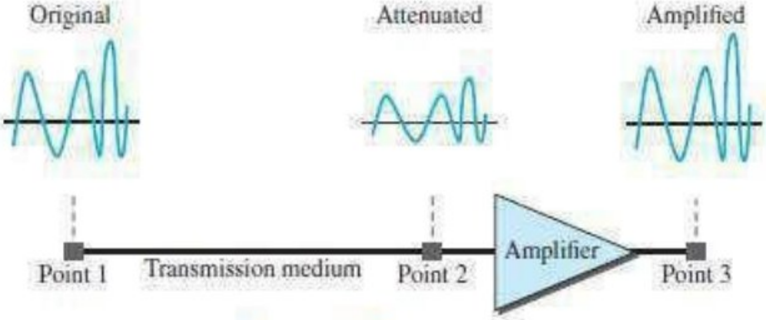



Figure 3.27 *Attenuation*

**1.10.1.1 Decibel**

- The decibel (dB) measures the relative strengths of
  → 2 signals or
  → one signal at 2 different points.
- The decibel is negative if a signal is attenuated.
  The decibel is positive if a signal is amplified.

$$dB = 10 \log_{10} \frac{P_2}{P_1}$$

- Variables $P_1$ and $P_2$ are the powers of a signal at points 1 and 2, respectively.
- To show that a signal has lost or gained strength, engineers use the unit of decibel.

**Example 1.7**

Suppose a signal travels through a transmission medium and its power is reduced to one-half. This means that $P_2 = \frac{1}{2} P_1$. In this case, the attenuation (loss of power) can be calculated as

$$10 \log_{10} \frac{P_2}{P_1} = 10 \log_{10} \frac{0.5P_1}{P_1} = 10 \log_{10} 0.5 = 10(-0.3) = -3 \text{ dB}$$

**Example 1.8**

A signal travels through an amplifier, and its power is increased 10 times. This means that $P_2 = 10P_1$. In this case, the amplification (gain of power) can be calculated as

$$10 \log_{10} \frac{P_2}{P_1} = 10 \log_{10} \frac{10P_1}{P_1} = 10 \log_{10} 10 = 10(1) = 10 \text{ dB}$$

**Example 1.9**

Sometimes the decibel is used to measure signal power in milliwatts. In this case, it is referred to as $dB_m$ and is calculated as $dB_m = 10 \log_{10} P_m$, where $P_m$ is the power in milliwatts. Calculate the power of a signal if its $dB_m = -30$.

**Solution**

We can calculate the power in the signal as

$$dB_m = 10 \log_{10} \longrightarrow dB_m = -30 \longrightarrow \log_{10} P_m = -3 \longrightarrow P_m = 10^{-3} \text{ mW}$$

**Example 1.10**

The loss in a cable is usually defined in decibels per kilometer (dB/km). If the signal at the beginning of a cable with –0.3 dB/km has a power of 2 mW, what is the power of the signal at 5 km?

**Solution**

The loss in the cable in decibels is $5 \times (-0.3) = -1.5$ dB. We can calculate the power as

$$\text{dB} = 10 \log_{10} (P_2 / P_1) = -1.5 \longrightarrow (P_2 / P_1) = 10^{-0.15} = 0.71$$

$$P_2 = 0.71P_1 = 0.7 \times 2 \text{ mW} = 1.4 \text{ mW}$$

### 1.10.2 Distortion

- Distortion means that the signal changes its form or shape (Figure 3.29).
- Distortion can occur in a composite signal made of different frequencies.
- Different signal-components
  - → have different propagation speed through a medium.
  - → have different delays in arriving at the final destination.
- Differences in delay create a difference in phase if delay is not same as the period-duration.
- Signal-components at the receiver have phases different from what they had at the sender.
- The shape of the composite signal is therefore not the same.

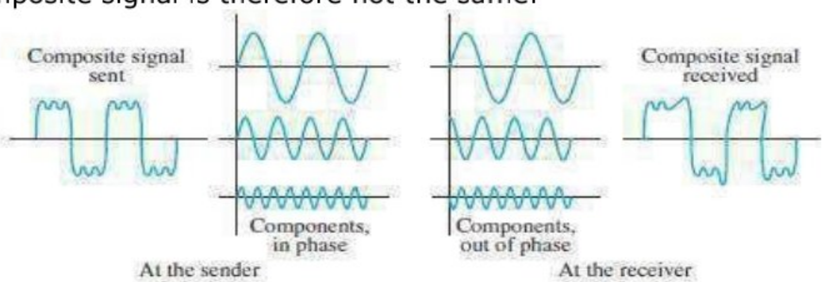



Figure 3.29 Distortion

### 1.10.3 Noise

- Noise is defined as an unwanted data (Figure 3.30).
- In other words, noise is the external energy that corrupts a signal.
- Due to noise, it is difficult to retrieve the original data/information.
- Four types of noise:

**i) Thermal Noise**

- It is random motion of electrons in wire which creates extra signal not originally sent by transmitter.

**ii) Induced Noise**

- Induced noise comes from sources such as motors & appliances.
- These devices act as a sending-antenna. The transmission-medium acts as the receiving-antenna.

**iii) Crosstalk**

- Crosstalk is the effect of one wire on the other.
- One wire acts as a sending-antenna and the other as the receiving-antenna.

**iv) Impulse Noise**

- Impulse Noise is a spike that comes from power-lines, lightning, and so on. (spike a signal with high energy in a very short time)

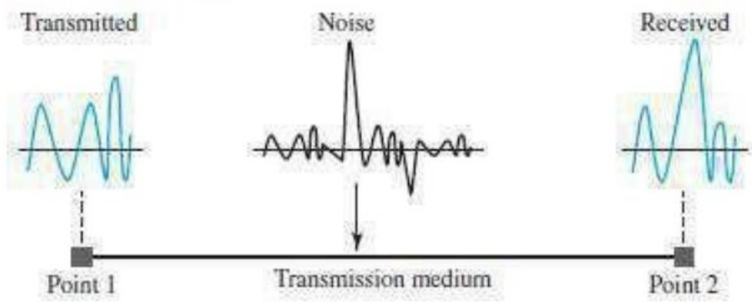



Figure 3.30 *Noise*

#### 1.10.3.1 Signal-to-Noise Ratio (SNR)

- SNR is used to find the theoretical bit-rate limit.
- SNR is defined as

$$\mathrm{SNR} = \frac{\text{average signal power}}{\text{average noise power}}$$

- SNR is actually the ratio of what is wanted (signal) to what is not wanted (noise).
- A high-SNR means the signal is less corrupted by noise. A low-SNR means the signal is more corrupted by noise.
- Because SNR is the ratio of 2 powers, it is often described in decibel units, $\mathrm{SNR_{dB}}$, defined as

$$\mathrm{SNR_{dB}} = 10\log_{10} \mathrm{SNR}$$

**Example 1.11**

The power of a signal is 10 mW and the power of the noise is 1 μW; what are the values of SNR and $\mathrm{SNR_{dB}}$?

**Solution**

The values of SNR and $\mathrm{SNR_{dB}}$ can be calculated as follows:

$$\mathrm{SNR} = (10{,}000\ \mu\mathrm{w}) / (1\ \mu\mathrm{w}) = 10{,}000 \qquad \mathrm{SNR_{dB}} = 10 \log_{10} 10{,}000 = 10 \log_{10} 10^4 = 40$$

***DATA COMMUNICATION***

## 1.11 DATA RATE LIMITS

• Data-rate depends on 3 factors:
1) Bandwidth available
2) Level of the signals
3) Quality of channel (the level of noise)

• Two theoretical formulas can be used to calculate the data-rate:
1) Nyquist for a noiseless channel and 2) Shannon for a noisy channel.

### 1.11.1 Noiseless Channel: Nyquist Bit Rate

- For a noiseless channel, the Nyquist bit-rate formula defines the theoretical maximum bit-rate

$$\text{Bitrate} = 2 \times \text{Bandwidth} \times \log_2 L$$

where bandwidth = bandwidth of the channel
L = number of signal-levels used to represent data
BitRate = bitrate of channel in bps

- According to the formula,
  - ¤ By increasing number of signal-levels, we can increase the bit-rate.
  - ¤ Although the idea is theoretically correct, practically there is a limit.
  - ¤ When we increase the number of signal-levels, we impose a burden on the receiver. ¤ If no. of levels in a signal is 2, the receiver can easily distinguish b/w 0 and 1.
  - ¤ If no. of levels is 64, the receiver must be very sophisticated to distinguish b/w 64 different levels.
  - ¤ In other words, increasing the levels of a signal reduces the reliability of the system.

**Example 1.12**

Consider a noiseless channel with a bandwidth of 3000 Hz transmitting a signal with two signal levels. The maximum bit rate can be calculated as

$$\text{BitRate} = 2 \times 3000 \times \log_2 2 = 6000 \text{ bps}$$

**Example 1.13**

Consider the same noiseless channel transmitting a signal with four signal levels (for each level, we send 2 bits). The maximum bit rate can be calculated as

$$\text{BitRate} = 2 \times 3000 \times \log_2 4 = 12{,}000 \text{ bps}$$

**Example 1.14**

We need to send 265 kbps over a noiseless channel with a bandwidth of 20 kHz. How many signal levels do we need?

**Solution**

We can use the Nyquist formula as shown:

$$265{,}000 = 2 \times 20{,}000 \times \log_2 L \longrightarrow \log_2 L = 6.625 \longrightarrow L = 2^{6.625} = 98.7 \text{ levels}$$

### 1.11.2 Noisy Channel: Shannon Capacity

- In reality, we cannot have a noiseless channel; the channel is always noisy.
- For a noisy channel, the Shannon capacity formula defines the theoretical maximum bit-rate.

$$\text{Capacity} = \text{bandwidth} \times \log_2 (1 + \text{SNR})$$

where bandwidth = bandwidth of channel in bps.
SNR = signal-to-noise ratio and
Capacity = capacity of channel in bps.

- This formula does not consider the no. of levels of signals being transmitted (as done in the Nyquist bit rate).
  This means that no matter how many levels we have, we cannot achieve a data-rate higher than the capacity of the channel.
- In other words, the formula defines a characteristic of the channel, not the method of transmission.

**Example 1.15**

We can calculate the theoretical highest bit rate of a regular telephone line. A telephone line normally has a bandwidth of 3000 Hz (300 to 3300 Hz) assigned for data communications. The signal-to-noise ratio is usually 3162. For this channel the capacity is calculated as

$$C = B \log_2 (1 + \text{SNR}) = 3000 \log_2(1 + 3162) = 3000 \times 11.62 = 34{,}860 \text{ bps}$$

**Example 1.16**

The signal-to-noise ratio is often given in decibels. Assume that $\text{SNR}_{\text{dB}} = 36$ and the channel bandwidth is 2 MHz. The theoretical channel capacity can be calculated as

$$\text{SNR}_{\text{dB}} = 10 \log_{10}\text{SNR} \longrightarrow \text{SNR} = 10^{\text{SNR}_{\text{dB}}/10} \longrightarrow \text{SNR} = 10^{3.6} = 3981$$

$$C = B \log_2(1 + \text{SNR}) = 2 \times 10^6 \times \log_2 3982 = 24 \text{ Mbps}$$

**Example 1.17**

We have a channel with a 1-MHz bandwidth. The SNR for this channel is 63. What are the appropriate bit rate and signal level?

**Solution**

First, we use the Shannon formula to find the upper limit.

$$C = B \log_2(1 + \text{SNR}) = 10^6 \log_2(1 + 63) = 10^6 \log_2 64 = 6 \text{ Mbps}$$

The Shannon formula gives us 6 Mbps, the upper limit. For better performance we choose something lower, 4 Mbps, for example. Then we use the Nyquist formula to find the number of signal levels.

$$4 \text{ Mbps} = 2 \times 1 \text{ MHz} \times \log_2 L \longrightarrow L = 4$$

## 1.12 PERFORMANCE

### 1.12.1 Bandwidth

- One characteristic that measures network-performance is bandwidth.
- Bandwidth of analog and digital signals is calculated in separate ways:

**(1) Bandwidth of an Analog Signal (in hz)**

- Bandwidth of an analog signal is expressed in terms of its frequencies.
- Bandwidth is defined as the range of frequencies that the channel can carry.
- It is calculated by the difference b/w the maximum frequency and the minimum frequency.

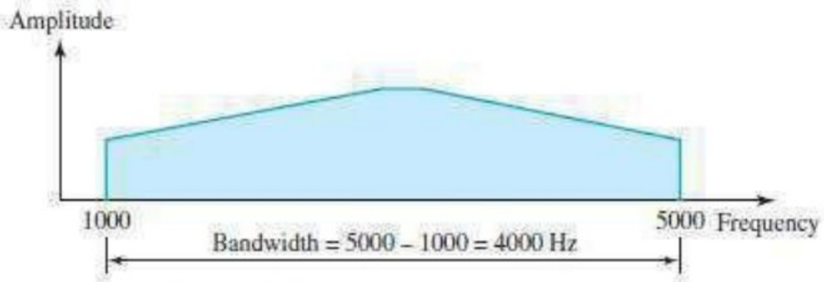



Figure 3.13 *The bandwidth of signals*

In figure 3.13, the signal has a minimum frequency of F1 = 1000Hz and maximum frequency of F2 = 5000Hz.

Hence, the bandwidth is given by F2 - F1= 5000 - 1000 = 4000 Hz

**(2) Bandwidth of a Digital Signal (in bps)**

 Bandwidth refers to the number of bits transmitted in one second in a channel (or link).  For example:

The bandwidth of a Fast Ethernet is a maximum of 100 Mbps. (This means that this network can send 100 Mbps).

**Relationship between (1) and (2)**

• There is an explicit relationship between the bandwidth in hertz and bandwidth in bits per seconds.
• Basically, an increase in bandwidth in hertz means an increase in bandwidth in bits per second.
• The relationship depends on → baseband transmission or
→ transmission with modulation.

**1.12.2 Throughput**

• The throughput is a measure of how fast we can actually send data through a network.
• Although, bandwidth in bits per second and throughput seem the same, they are actually different.
• A link may have a bandwidth of B bps, but we can only send T bps through this link with T always less than B.
• In other words,
1) The bandwidth is a potential measurement of a link.
2) The throughput is an actual measurement of how fast we can send data.

For example:

¤ We may have a link with a bandwidth of 1 Mbps, but the devices connected to the end of the link may handle only 200 kbps.
¤ This means that we cannot send more than 200 kbps through this link.

**Example 1.18**

A network with bandwidth of 10 Mbps can pass only an average of 12,000 frames per minute with each frame carrying an average of 10,000 bits. What is the throughput of this network?

**Solution**

We can calculate the throughput as

$$\text{Throughput} = (12{,}000 \times 10{,}000) / 60 = 2 \text{ Mbps}$$

### 1.12.3 Latency (Delay)

• The latency defines how long it takes for an entire message to completely arrive at the destination from the time the first bit is sent out from the source.

**Latency = propagation time + transmission time + queuing time + processing delay**

#### 1) Propagation Time

- Propagation time is defined as the time required for a bit to travel from source to destination. - Propagation time is given by

**Propagation time = Distance / (Propagation Speed)**

- Propagation speed of electromagnetic signals depends on
  → medium and
  → frequency of the signal.

**Example 1.19**

What is the propagation time if the distance between the two points is 12,000 km? Assume the propagation speed to be $2.4 \times 10^8$ m/s in cable.

**Solution**

We can calculate the propagation time as

$$\text{Propagation time} = (12{,}000 \times 10{,}000) / (2.4 \times 2^8) = 50 \text{ ms}$$

b

#### 2) Transmission Time

- The time required for transmission of a message depends on
  → size of the message and →
  bandwidth of the channel.
- The transmission time is given by

**Transmission time = (Message size) / Bandwidth**

**Example 1.20**

What are the propagation time and the transmission time for a 2.5-KB (kilobyte) message (an e-mail) if the bandwidth of the network is 1 Gbps? Assume that the distance between the sender and the receiver is 12,000 km and that light travels at $2.4 \times 10^8$ m/s.

**Solution**

We can calculate the propagation and transmission time as

$$\text{Propagation time} = (12{,}000 \times 1000) / (2.4 \times 10^8) = 50 \text{ ms}$$

$$\text{Transmission time} = (2500 \times 8) / 10^9 = 0.020 \text{ ms}$$

**Example 1.21**

What are the propagation time and the transmission time for a 5-MB (megabyte) message (an image) if the bandwidth of the network is 1 Mbps? Assume that the distance between the sender and the receiver is 12,000 km and that light travels at $2.4 \times 10^8$ m/s.

**Solution**

We can calculate the propagation and transmission times as

$$\textbf{Propagation time} = (12{,}000 \times 1000) / (2.4 \times 10^8) = 50 \text{ ms}$$

$$\textbf{Transmission time} = (5{,}000{,}000 \times 8) / 10^6 = 40 \text{ s}$$

**3) Queuing Time**

- Queuing-time is the time needed for each intermediate-device to hold the message before it can be processed.
  (Intermediate device may be a router or a switch)
- The queuing-time is not a fixed factor. This is because
  i) Queuing-time changes with the load imposed on the network.
  ii) When there is heavy traffic on the network, the queuing-time increases. An intermediate-device
  → queues the arrived messages and
  → processes the messages one by one.
- If there are many messages, each message will have to wait.

**4) Processing Delay**

- Processing delay is the time taken by the routers to process the packet header.

**1.12.4 Bandwidth Delay Product**

- Two performance-metrics of a link are 1) Bandwidth and 2) Delay
- The bandwidth-delay product is very important in data-communications.
- Let us elaborate on this issue, using 2 hypothetical cases as examples. **Case 1:** The following figure shows case 1 (Figure 3.32).

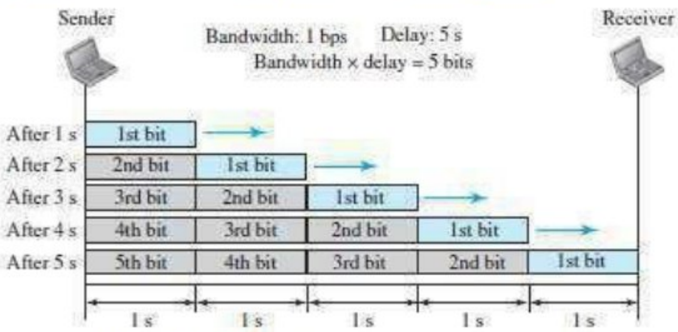



**Figure 3.32** *Filling the link with bits for case I*

- Let us assume,
  Bandwidth of the link = 1 bps Delay of the link = 5s.
- From the figure 3.32, bandwidth-delay product is 1 x 5 = 5. Thus, there can be maximum 5 bits on the line.
- There can be no more than 5 bits at any time on the link. **Case 2:** The following figure shows case 2 (Figure 3.33).

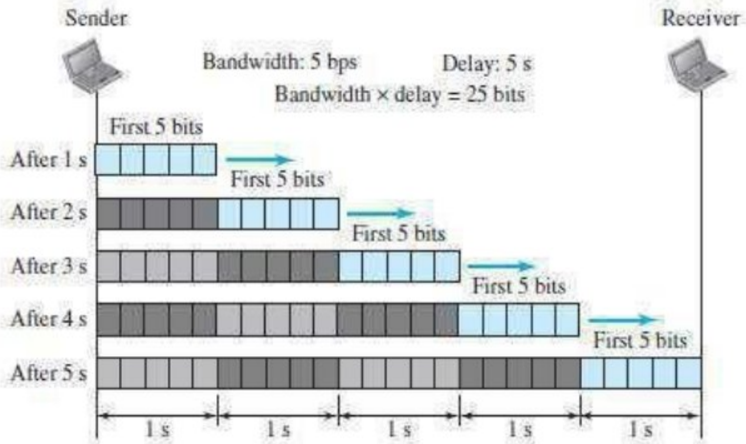



Figure 3.33 *Filling the link with bits in case 2*

- Let us assume,
  Bandwidth of the link = 4 bps
  Delay of the link = 5s.
- From the figure 3.33, bandwidth-delay product is 5 x 5 = 25. Thus, there can be maximum 25 bits on the line.
- At each second, there are 5 bits on the line, thus the duration of each bit is 0.20s.

• The above 2 cases show that the (bandwidth X delay) is the number of bits that can fill the link.
• This measurement is important if we need to
 → send data in bursts and
 → wait for the acknowledgment of each burst.
• To use the maximum capability of the link
 → We need to make the burst-size as (2 x bandwidth x delay). → We need to fill up the full-duplex channel (two directions).
• Amount (2x bandwidth x delay) is the number of bits that can be in transition at any time (Fig 3.34).

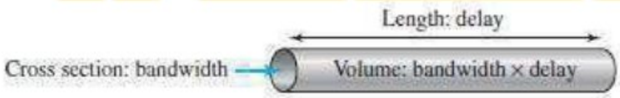



Figure 3.34 *Concept of bandwidth-delay product*

**1.12.5 Jitter**
• Another performance issue that is related to delay is jitter.
• We can say that jitter is a problem
 → if different packets of data encounter different delays and
 → if the application using the data at the receiver site is time-sensitive (for ex: audio/video).
• For example:
 If the delay for the first packet is 20 ms the delay for the second is 45 ms and the delay for the third is 40 ms then the real-time application that uses the packets suffers from jitter.

# MODULE 1(CONT.): DIGITAL TRANSMISSION

## 1.13 DIGITAL TO DIGITAL CONVERSION

- Data can be analog or digital, so can be the signal that represents it.
- Signal encoding is the conversion from analog/digital data to analog/digital signal.
- The possible encodings are:
  1) Digital data to digital signal
  2) Digital data to analog signal
  3) Analog data to digital signal
  4) Analog data to analog signal

### 1.13.1 LINE CODING

- Line-coding is the process of converting digital-data to digital-signals (Figure 4.1). • The data may be in the form of text, numbers, graphical images, audio, or video
- The data are stored in computer memory as sequences of bits (0s or 1s).
- Line-coding converts a sequence of bits to a digital-signal.
- At the sender, digital-data is encoded into a digital-signal.
  At the receiver, digital-signal is decoded into a digital-data.

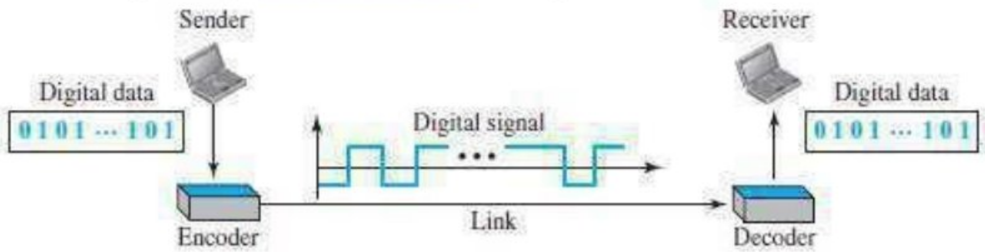



Figure 4.1 *Line coding and decoding*

#### 1.13.1.1 Characteristics

- Different characteristics of digital signal are
  1) Signal Element Vs Data Element
  2) Data Rate Vs Signal Rate
  3) Bandwidth
  4) Baseline Wandering
  5) DC Components
  6) Built-in Error Detection
  7) Self-synchronization
  8) Immunity to Noise and Interference
  9) Complexity

**1) Data Element vs. Signal Element**

| Data Element | Signal Element |
|---|---|
| A data-element is the smallest entity that can represent a piece of information (Figure 4.2). | A signal-element is shortest unit (timewise) of a digital-signal. |
| A data-element is the bit. | A signal-element carries data-elements. |
| Data-elements are being carried. | Signal-elements are the carriers. |

 Ratio r is defined as number of data-elements carried by each signal-element.

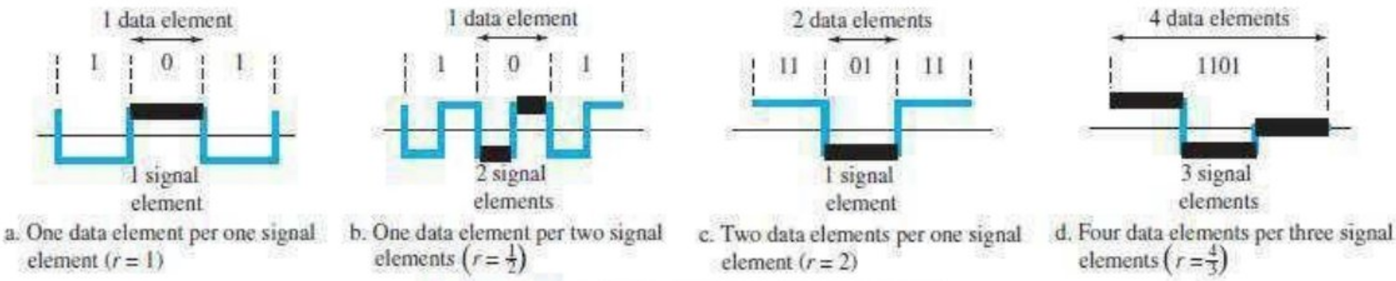



Figure 4.2 *Signal element versus data element*

**2) Data Rate vs. Signal Rate**

| Data Rate | Signal Rate |
|---|---|
| The data-rate defines the number of data-elements (bits) sent in 1 sec. | The signal-rate is the number of signal-elements sent in 1 sec. |
| The unit is bits per second (bps). | The unit is the baud. |
| The data-rate is sometimes called the bit-rate. | The signal-rate is sometimes called the pulse rate, the modulation rate, or the baud rate |
| Goal in data-communications: increase the data-rate. | Goal in data-communications: decrease the signal-rate. |
| Increasing the data-rate increases the speed of transmission. | Decreasing the signal-rate decreases the bandwidth requirement. |

- The relationship between data-rate and signal-rate is given by

$$S_{ave} = c \times N \times (1/r) \quad \text{baud}$$

where N = data-rate (in bps) c = case factor, which varies for each case S = number of signal-elements and r = previously defined factor.

- This relationship depends on → value of r.
  → data pattern.

(If we have a data pattern of all 1s or all 0s, the signal-rate may be different from a data pattern of alternating 0s and 1s).

DC MODULE 1 NOTE BY stufa

**3) Bandwidth**

- Digital signal that carries information is non-periodic.
- The bandwidth of a non-periodic signal is continuous with an infinite range.
- However, most digital-signals we encounter in real life have a bandwidth with finite values. - The effective bandwidth is finite.
- The baud rate, not the bit-rate, determines the required bandwidth for a digital-signal. - More changes in the signal mean injecting more frequencies into the signal. (Frequency means change and change means frequency.)
- The bandwidth refers to range of frequencies used for transmitting a signal.
- Relationship b/w baud rate (signal-rate) and the bandwidth (range of frequencies) is given as

$$B_{min} = c \times N \times (1 / r)$$

where N = data-rate (in bps) c = case factor, which varies for each case
r = previously defined factor
$B_{min}$ = minimum bandwidth **4)**

**4) Baseline Wandering**

- While decoding, the receiver calculates a running-average of the received signal-power. This average is called the baseline.
- The incoming signal-power is estimated against this baseline to determine the value of the data-element.
- A long string of 0s or 1s can cause a drift in the baseline (baseline wandering). Thus, make it difficult for the receiver to decode correctly.
- A good line-coding scheme needs to prevent baseline wandering.

**5) DC Components**

- When the voltage-level in a digital-signal is constant for a while, the spectrum creates very low frequencies.
- These frequencies around zero are called DC (direct-current) components.
- DC components present problems for a system that cannot pass low frequencies. For example: Telephone line cannot pass frequencies below 200 Hz. For Telephone systems, we need a scheme with no DC component.

**6) Built-in Error Detection**

- Built-in error-detecting capability has to be provided to detect the errors that occurred during transmission.

**7) Self Synchronization**

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

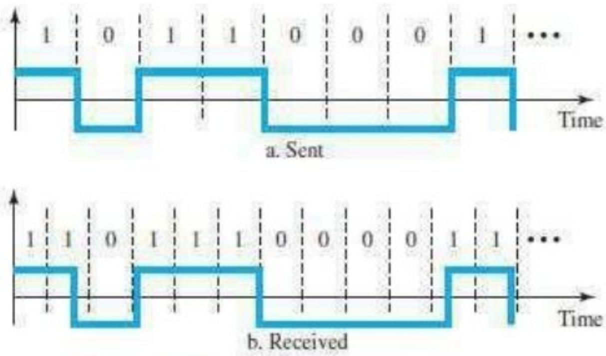



**Figure 4.3** *Effect of lack of synchronization*

- As shown in figure 4.3, we have a situation where the receiver has shorter bit duration.
- The sender sends 10110001, while the receiver receives 110111000011.
- 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.

**8) Immunity to Noise & Interference 9) Complexity**

- The code should be immune to noise and other interferences.
- A complex scheme is more costly to implement than a simple one.
- For ex: A scheme that uses 4 signal-levels is more difficult to interpret than one that uses only 2 levels.

**Example 1.22**

A signal is carrying data in which one data element is encoded as one signal element ($r = 1$). If the bit rate is 100 kbps, what is the average value of the baud rate if $c$ is between 0 and 1?

**Solution**

We assume that the average value of $c$ is 1/2. The baud rate is then

$$S = c \times N \times (1 / r) = 1/2 \times 100{,}000 \times (1/1) = 50{,}000 = 50 \text{ kbaud}$$

**Example 1.23**

In a digital transmission, the receiver clock is 0.1 percent faster than the sender clock. How many extra bits per second does the receiver receive if the data rate is 1 kbps? How many if the data rate is 1 Mbps?

**Solution**

At 1 kbps, the receiver receives 1001 bps instead of 1000 bps.

**1000 bits sent → 1001 bits received → 1 extra bps**

At 1 Mbps, the receiver receives 1,001,000 bps instead of 1,000,000 bps.

**1,000,000 bits sent → 1,001,000 bits received → 1000 extra bps**

### 1.13.2 LINE CODING SCHEMES

• The Line Coding schemes are classified into 3 broad categories (Figure 4.4):

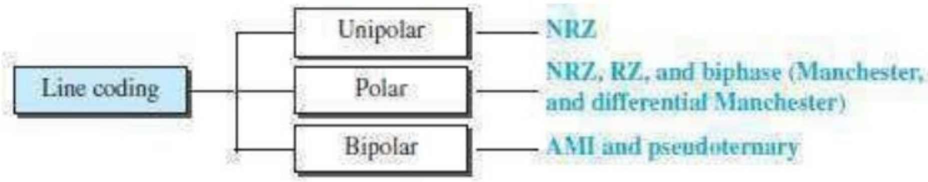



Figure 4.4 *Line coding schemes*

### 1.13.2.1 Unipolar Scheme

• All signal levels are either above or below the time axis. **NRZ (Non-Return-to-Zero)**

- The positive voltage defines bit 1 and the zero voltage defines bit 0 (Figure 4.5).
- It is called NRZ because the signal does not return to 0 at the middle of the bit.

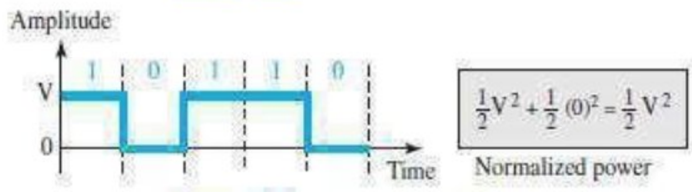



Figure 4.5 *Unipolar NRZ scheme*

- Disadvantages:
  1) Compared to polar scheme, this scheme is very costly.
  2) Also, the normalized power is double that for polar NRZ.
  3) Not suitable for transmission over channels with poor performance around zero frequency.

(Normalized power ⇾ power needed to send 1 bit per unit line resistance)

### 1.13.2.2 Polar Schemes

• The voltages are on the both sides of the time axis.
• Polar NRZ scheme can be implemented with two voltages (V).

For example: -V for bit 1
+V for bit 0.

**a) Non-Return-to-Zero (NRZ)**

- We use 2 levels of voltage amplitude.
- Two versions of polar NRZ (Figure 4.6):

**i) NRZ-L (NRZ-Level)**

¤ The level of the voltage determines the value of the bit.
¤ For example: i) Voltage-level for 0 can be positive and
ii) Voltage-level for 1 can be negative.

**ii) NRZ-I (NRZ-Invert)**

¤ The change or lack of change in the level of the voltage determines the value of the bit.
¤ If there is no change, the bit is 0;
If there is a change, the bit is 1.

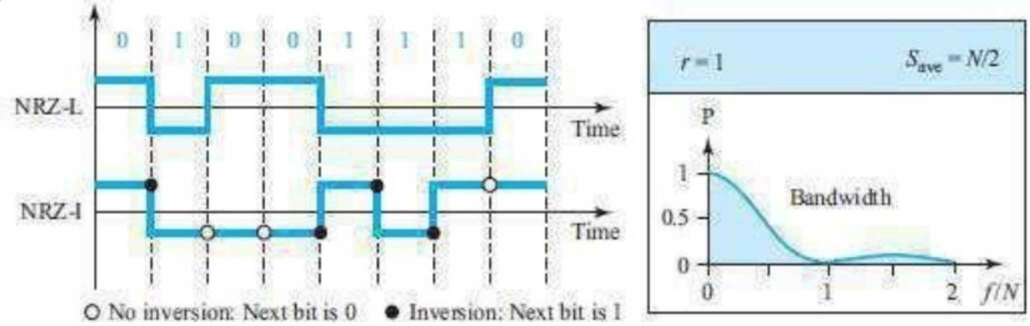



**Figure 4.6** *Polar NRZ-L and NRZ-I schemes*

 Disadvantages:

1) **Baseline wandering** is a problem for both variations (NRZ-L NRZ-I).

 i) In NRZ-L, if there is a long sequence of 0s or 1s, the average signal-power becomes skewed.

 The receiver might have difficulty discerning the bit value. ii)

 In NRZ-I, this problem occurs only for a long sequence of 0s.

 If we eliminate the long sequence of 0s, we can avoid baseline wandering.

2) The **synchronization problem** also exists in both schemes.

 → A long sequence of 0s can cause a problem in both schemes.

 → A long sequence of 1s can cause a problem in only NRZ-L.

3) In NRZ-L, problem occurs when there is a sudden **change of polarity** in the system.

 ¤ For example:

 In twisted-pair cable, a change in the polarity of the wire results in

 → all 0s interpreted as 1s and → all 1s interpreted as 0s.

 ¤ NRZ-I does not have this problem.

 ¤ Both schemes have an average signal-rate of N/2 Bd.

4) NRZ-L and NRZ-I both have a **DC component problem**.

**Example 1.24**

A system is using NRZ-I to transfer 10-Mbps data. What are the average signal rate and minimum bandwidth?

**Solution**

The average signal rate is $S = N/2 = 500$ kbaud. The minimum bandwidth for this average baud rate is $B_{min} = S = 500$ kHz.

**b) Return-to-Zero (RZ)**

- In NRZ encoding, problem occurs when the sender-clock and receiver-clock are not synchronized.
- Solution: Use return-to-zero (RZ) scheme (Figure 4.7).
- RZ scheme uses 3 voltages: positive, negative, and zero.
- There is always a transition at the middle of the bit. Either
  i) from high to zero (for 1) or ii)
  from low to zero (for 0)

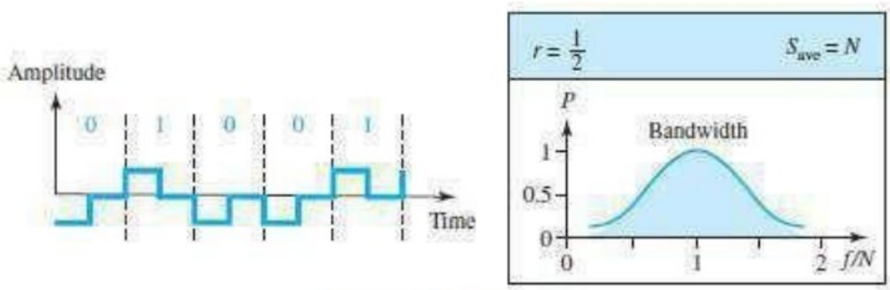



Figure 4.7 *Polar RZ scheme*

- Disadvantages:
  1) RZ encoding requires 2 signal-changes to encode a bit & .’. occupies greater bandwidth.
  2) Complexity: RZ uses 3 levels of voltage, which is more complex to create and detect.
  3) Problem occurs when there is a sudden change of polarity in the system. This result in
     → all 0s interpreted as 1s &
     → all 1s interpreted as 0s.

**c) Biphase: Manchester & Differential Manchester i) Manchester Encoding**

- This is a combination of NRZ-L & RZ schemes (RZ transition at the middle of the bit). There is always a transition at the middle of the bit. Either
  i) from high to low (for 0) or ii) from low to high (for 1).
- It uses only two voltage levels (Figure 4.8).
- The duration of the bit is divided into 2 halves. The voltage
  → remains at one level during the first half & → moves to the other level in the second half.
- The transition at the middle of the bit provides synchronization. **ii) Differential Manchester**
- This is a combination of NRZ-I and RZ schemes.
- There is always a transition at the middle of the bit, but the bit-values are determined at the beginning of the bit.
- If the next bit is 0, there is a transition. If the next bit is 1, there is none.

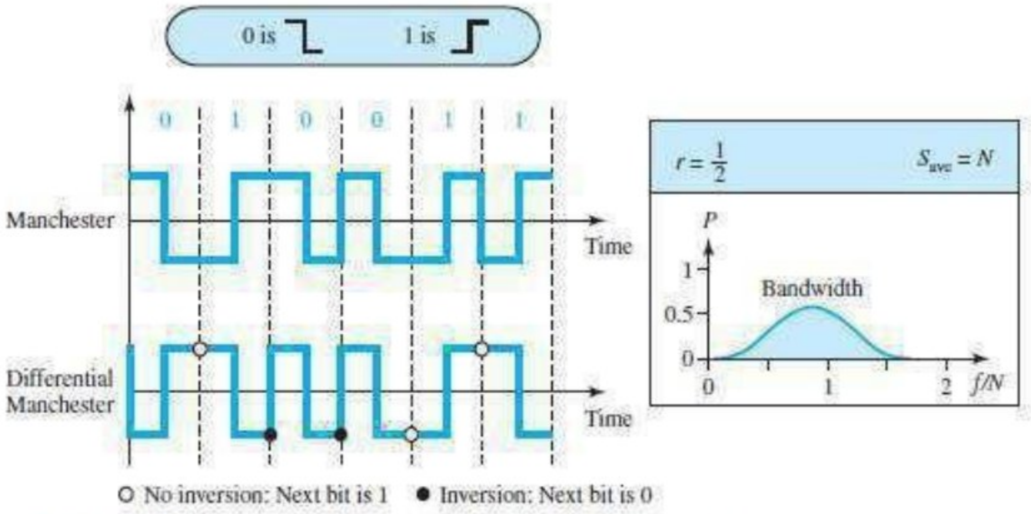



Figure 4.8 *Polar biphase: Manchester and differential Manchester schemes*

- Advantages:
  1) The Manchester scheme overcomes problems associated with NRZ-L. Differential Manchester overcomes problems associated with NRZ-I.
  2) There is no baseline wandering.
  3) There is no DC component '.' each bit has a positive & negative voltage contribution. Disadvantage:
  1) Signal-rate: Signal-rate for Manchester & diff. Manchester is double that for NRZ.



**1.13.2.3 Bipolar Schemes (or Multilevel Binary)**

- This coding scheme uses 3 voltage levels (Figure 4.9):
  i) positive ii) negative & iii) zero.
- Two variations of bipolar encoding:
  i) AMI (Alternate Mark Inversion) ii) Pseudoternary

**i) AMI**

- Binary 0 is represented by a neutral 0 voltage (AMI - Alternate 1 Inversion). - Binary 1s are represented by alternating positive and negative voltages.

**ii) Pseudoternary**

- Binary 1 is represented by a neutral 0 voltage.
- Binary 0s are represented by alternating positive and negative voltages.

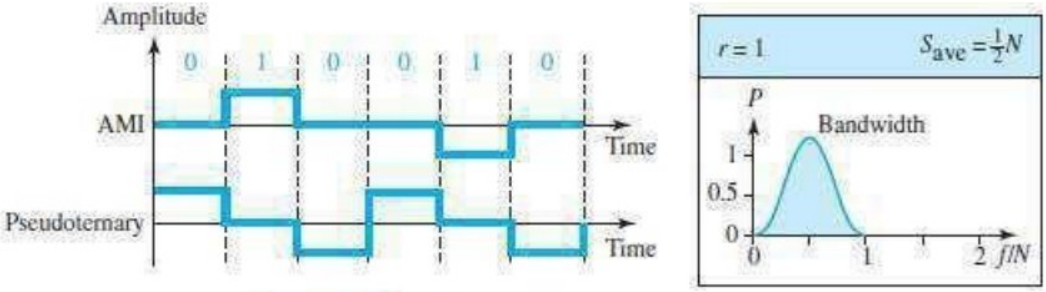



**Figure 4.9** *Bipolar schemes: AMI and pseudoternary*

- Advantages:
  1) The bipolar scheme has the same signal-rate as NRZ.
  2) There is no DC component '.' each bit has a positive & negative voltage contribution. 3) The concentration of the energy is around frequency N/2.
- Disadvantage:
  1) AMI has a synchronization problem when a long sequence of 0s is present in the data.

**Table 4.1** *Summary of line coding schemes*

| *Category* | *Scheme* | *Bandwidth (average)* | *Characteristics* |
|---|---|---|---|
| Unipolar | NRZ | $B = N/2$ | Costly, no self-synchronization if long 0s or 1s, DC |
| Polar | NRZ-L | $B = N/2$ | No self-synchronization if long 0s or 1s, DC |
| | NRZ-I | $B = N/2$ | No self-synchronization for long 0s, DC |
| | Biphase | $B = N$ | Self-synchronization, no DC, high bandwidth |
| Bipolar | AMI | $B = N/2$ | No self-synchronization for long 0s, DC |

# MODULE 2: DIGITAL TRANSMISSION

## 2.1 ANALOG TO DIGITAL CONVERSION

- An analog-signal may created by a microphone or camera.
- To change an analog-signal to digital-data, we use PCM (pulse code modulation).
- After the digital-data are created (digitization), then we convert the digital-data to a digital-signal.

### 2.1.1 PCM

- PCM is a technique used to change an analog signal to digital data (digitization).
- PCM has encoder at the sender and decoder at the receiver.
- The encoder has 3 processes (Figure 4.21):

  1) Sampling
  2) Quantization &
  3) Encoding.

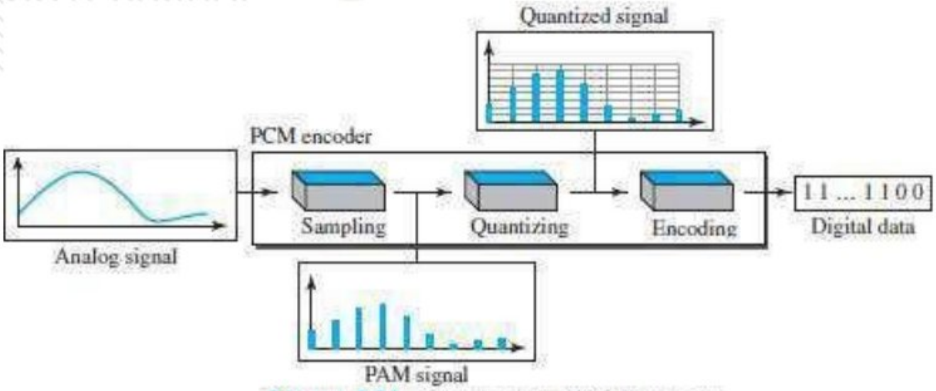



**Figure 4.21** *Components of PCM encoder*

#### 2.1.1.1 Sampling

- We convert the continuous time signal (analog) into the discrete time signal (digital). • Pulses from the analog-signal are sampled every $T_s$ sec where $T_s$ is the sample-interval or period.
- The inverse of the sampling-interval is called the sampling-frequency (or sampling-rate).
- Sampling-frequency is given by

$$f_s = 1/T_s$$

- Three sampling methods (Figure 4.22):

  **1) Ideal Sampling**

   This method is difficult to implement.

  **2) Natural Sampling**

  - A high-speed switch is turned ON for only the small period of time when the sampling occurs.  The result is a sequence of samples that retains the shape of the analog-signal. **3) Flat Top Sampling**
  - The most common sampling method is sample and hold.  Sample and hold method creates flat-top samples.
  - This method is sometimes referred to as *PAM* (pulse amplitude modulation).

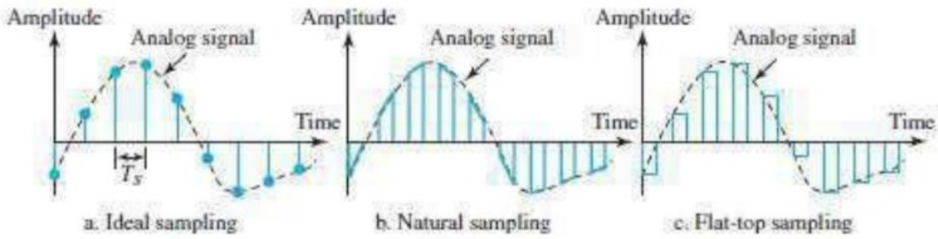



**Figure 4.22** *Three different sampling methods for PCM*

### 2.1.1.1.1 Sampling Rate

• According to Nyquist theorem,

“The sampling-rate must be at least 2 times the highest frequency, not the bandwidth“.

i) If the analog-signal is **low-pass**, the bandwidth and the highest frequency are the same value (Figure 4.23a). ii) If the analog-signal is **bandpass**, the bandwidth value is lower than the value of the maximum frequency (Figure 4.23b).

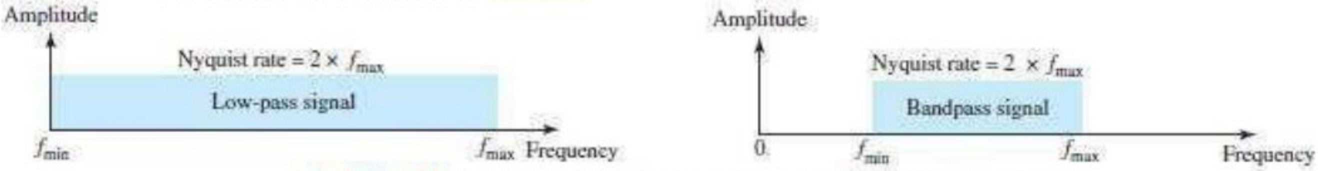



**Figure 4.23** *Nyquist sampling rate for low-pass and bandpass signals*

### 2.1.2 Quantization

- The sampled-signal is quantized.
- Result of sampling is a set of pulses with amplitude-values b/w max & min amplitudes of the signal.
- Four steps in quantization:

1) We assume that the original analog-signal has amplitudes between $V_{min}$ & $V_{max}$. 2) We divide the range into L zones, each of height Δ(delta).

$$\Delta = \frac{V_{max} - V_{min}}{L}$$

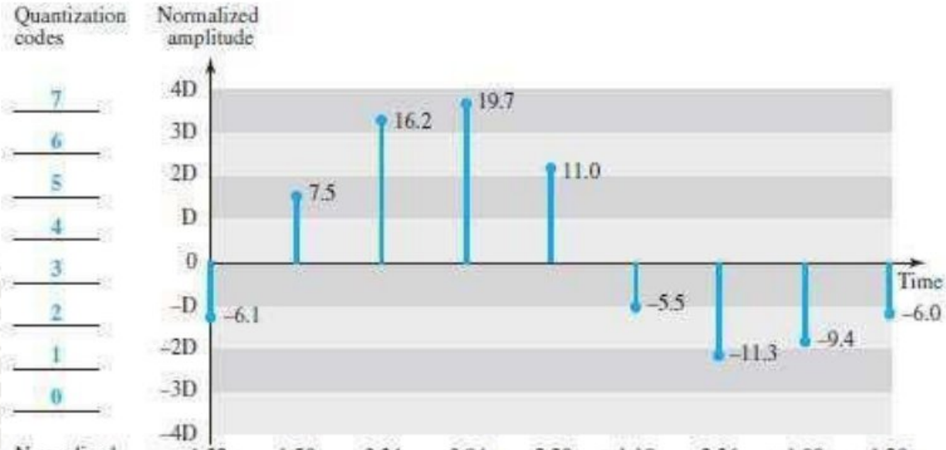
where L = number of levels.

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.

- For example: Let $V_{min}$=-20 $V_{max}$ =+20 V L = 8 Therefore, Δ = [+20-(-20)]/8= 5 V
- In the chart (Figure 4.26),

1) First row is normalized-PAM-value for each sample.
2) Second row is normalized-quantized-value for each sample.
3) Third row is normalized error (which is diff. b/w normalized PAM value & quantized values).
4) Fourth row is quantization code for each sample.
5) Fifth row is the encoded words (which are the final products of the conversion).




| | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| 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 |

**Figure 4.26** *Quantization and encoding of a sampled signal*

**2.1.2.1 Quantization Level**

- Let L = number of levels.
- The choice of L depends on

  → range of the amplitudes of the analog-signal and →

  how accurately we need to recover the signal.
- If the signal has only 2 amplitude values, we need only 2 quantization-levels.

  If the signal (like voice) has many amplitude values, we need more quantization-levels.
- In audio digitizing, L is normally chosen to be 256. In video digitizing, L is normally thousands.
- Choosing lower values of L increases the quantization-error.

**2.1.2.2 Quantization Error**

- Quantization-error is the difference b/w normalized PAM value & quantized values
- Quantization is an approximation process.
- The input values to the quantizer are the real values.

  The output values from the quantizer are the approximated values.
- The output values are chosen to be the middle value in the zone.
- If the input value is also at the middle of the zone, Then, there is no error.

  Otherwise, there is an error.
- In the previous example,

  The normalized amplitude of the third sample is 3.24, but the normalized quantized value is 3.50. This means that there is an error of +0.26.

**2.1.2.3 Uniform vs. Non Uniform Quantization**

- Non-uniform quantization can be done by using a process called companding and expanding.

  1) The signal is companded at the sender before conversion. 2) The signal is expanded at the receiver after conversion.

- Companding means reducing the instantaneous voltage amplitude for large values.
  Expanding means increasing the instantaneous voltage amplitude for small values.
- It has been proved that non-uniform quantization effectively reduces the $SNR_{dB}$ of quantization.

### 2.1.3 Encoding

- The quantized values are encoded as n-bit code word.
- In the previous example,
  A quantized value 2 is encoded as 010.
  A quantized value 5 is encoded as 101.
- Relationship between number of quantization-levels (L) & number of bits (n) is given by $n=\log_2 L$ or $2^n=L$
- The bit-rate is given by:

$$\textbf{Bit rate} = \textbf{sampling rate} \times \textbf{number of bits per sample} = f_s \times n$$

**Example 2.1**

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.

**Example 2.2**

What is the $SNR_{dB}$ 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 $SNR_{dB} = 6.02(3) + 1.76 = 19.82$ dB. Increasing the number of levels increases the SNR.

**Example 2.3**

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

**Solution**

We can calculate the number of bits as

$$SNR_{dB} = 6.02n_b + 1.76 = 40 \quad \rightarrow \quad n = 6.35$$

Telephone companies usually assign 7 or 8 bits per sample.

**Example 2.4**

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:

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

#### 2.1.3.1 Original Signal Recovery

- PCM decoder is used for recovery of the original signal.
- Here is how it works (Figure 4.27):
  1) The decoder first uses circuitry to convert the code words into a pulse that holds theamplitude until the next pulse.

2) Next, the staircase-signal is passed through a low-pass filter to smooth the staircase signal into an analog-signal.

- The filter has the same cut-off frequency as the original signal at the sender.
- If the signal is sampled at the Nyquist sampling-rate, then the original signal will be re-created.
- The maximum and minimum values of the original signal can be achieved by using amplification.

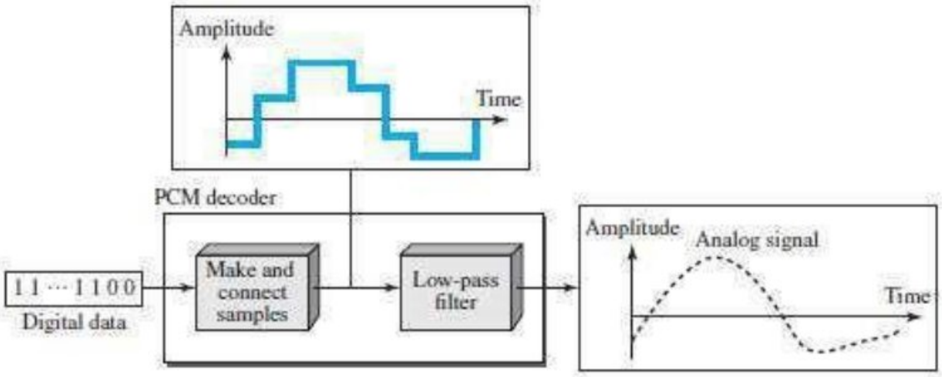



Figure 4.27 *Components of a PCM decoder*

**2.1.3.2 PCM Bandwidth**

- The minimum bandwidth of a line-encoded signal is

$$B_{min} = c \times N \times \frac{1}{r}$$

- We substitute the value of N in above formula:

$$B_{min} = c \times N \times \frac{1}{r} = c \times n_b \times f_s \times \frac{1}{r} = c \times n_b \times 2 \times B_{analog} \times \frac{1}{r}$$

- When 1/r = 1 (for a NRZ or bipolar signal) and c = (1/2) (the average situation), the minimumbandwidth is

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

- This means the minimum bandwidth of the digital-signal is $n_b$ times greater than the bandwidth of the analog-signal.

**2.1.3.3 Maximum Data Rate of a Channel**

- The Nyquist theorem gives the data-rate of a channel as

$$N_{max} = 2 \times B \times \log_2 L$$

- We can deduce above data-rate from the Nyquist sampling theorem by using the following arguments.
  1) We assume that the available channel is low-pass with bandwidth B.
  2) We assume that the digital-signal we want to send has L levels, where each level is a signalelement. This means r = 1/log$_2$L.
  3) We first pass digital-signal through a low-pass filter to cut off the frequencies above B Hz. 4) We treat the resulting signal as an analog-signal and sample it at 2 x B samples per second and quantize it using L levels. 5) The resulting bit-rate is

$$N = f_s \times n_b = 2 \times B \times \log_2 L$$

This is the maximum bandwidth; if the case factor c increases, the data-rate is reduced.

$$N_{max} = 2 \times B \times \log_2 L \quad \text{bps}$$

**2.1.3.4 Minimum Required Bandwidth**

• The previous argument can give us the minimum bandwidth if the data-rate and the number of signal-levels are fixed. We can say

$$B_{\min} = \frac{N}{(2 \times \log_2) L} \quad \text{Hz}$$

## 2.2 TRANSMISSION MODES

• Two ways of transmitting data over a link (Figure 4.31): 1) Parallel mode & 2) Serial mode.

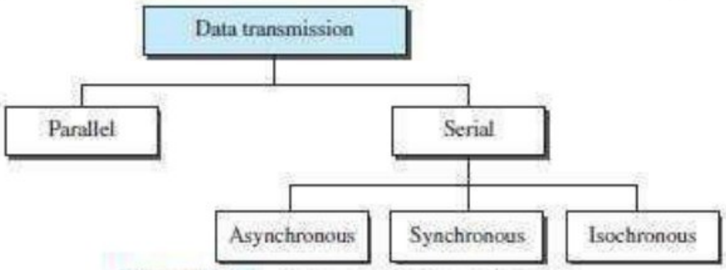



Figure 4.31 *Data transmission and modes*

### 2.1.1 PARALLEL TRANSMISSION

- Multiple bits are sent with each clock-tick (Figure 4.32).
- „n" bits in a group are sent simultaneously.
- „n" wires are used to send „n" bits at one time.
- Each bit has its own wire.
- Typically, the 8 wires are bundled in a cable with a connector at each end.

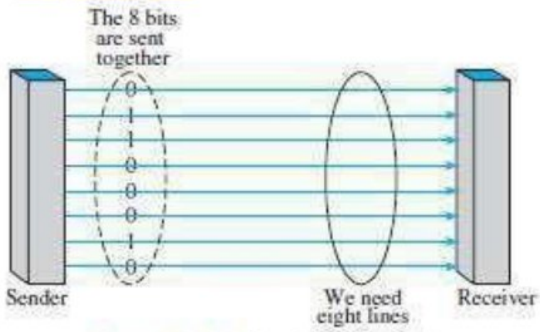



Figure 4.32 *Parallel transmission*

- Advantage:

  1) Speed: Parallel transmission can increase the transfer speed by a factor of n over serialtransmission.

- Disadvantage:

  1) Cost: Parallel transmission requires n communication lines just to transmit the data-stream. Because this is expensive, parallel transmission is usually limited to short distances.

### 2.2.2 SERIAL TRANSMISSION

- One bit is sent with each clock-tick using only a single link (Figure 4.33).

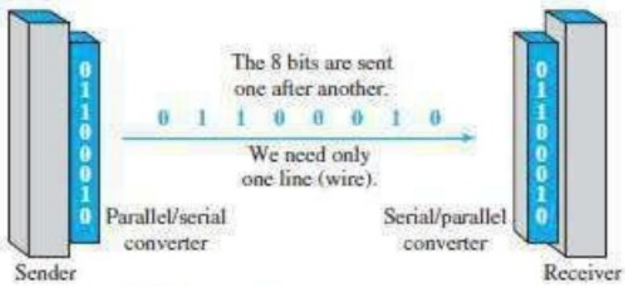



Figure 4.33 *Serial transmission*

- Advantage:

1) Cost: Serial transmission reduces cost of transmission over parallel by a factor of n.

- Disadvantage:

    1) Since communication within devices is parallel, following 2 converters are required at interface: i) Parallel-to-serial converter ii) Serial-to-parallel converter

- Three types of serial transmission: asynchronous, synchronous, and isochronous.

**2.2.2.1 Asynchronous Transmission**

- Asynchronous transmission is so named because the timing of a signal is not important (Figure 4.34).
- Prior to data transfer, both sender & receiver agree on pattern of information to be exchanged.
- Normally, patterns are based on grouping the bit-stream into bytes.
- The sender transmits each group to the link without regard to a timer.
- As long as those patterns are followed, the receiver can retrieve the info. without regard to a timer.
- There may be a gap between bytes.
- We send

    → 1 start bit (0) at the beginning of each byte → 1
    stop bit (1) at the end of each byte.

- Start bit alerts the receiver to the arrival of a new group.

    Stop bit lets the receiver know that the byte is finished.

- Here, the term asynchronous means "asynchronous at the byte level".

    However, the bits are still synchronized & bit-durations are the same.

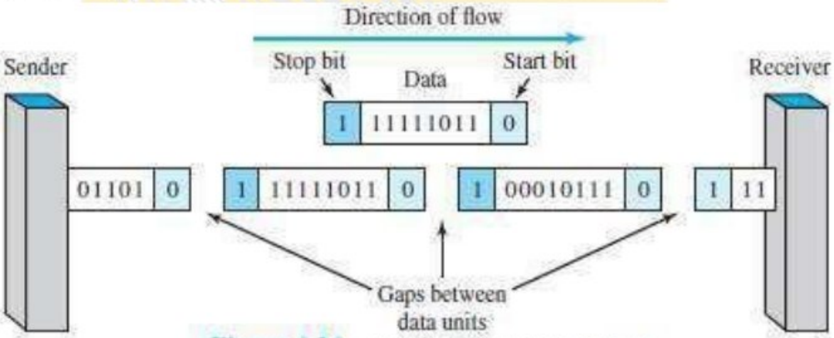



Figure 4.34 *Asynchronous transmission*

- Disadvantage:

    1) Slower than synchronous transmission. (Because of stop bit, start bit and gaps)

- Advantages:

    1) Cheap & effective.
    2) Useful for low-speed communication.

**2.2.2.2 Synchronous Transmission**

- We send bits one after another without start or stop bits or gaps (Figure 4.35).
- The receiver is responsible for grouping the bits.
- The bit-stream is combined into longer "frames," which may contain multiple bytes.
- If the sender wants to send data in separate bursts, the gaps b/w bursts must be filled with a special sequence of 0s & 1s (that means idle).

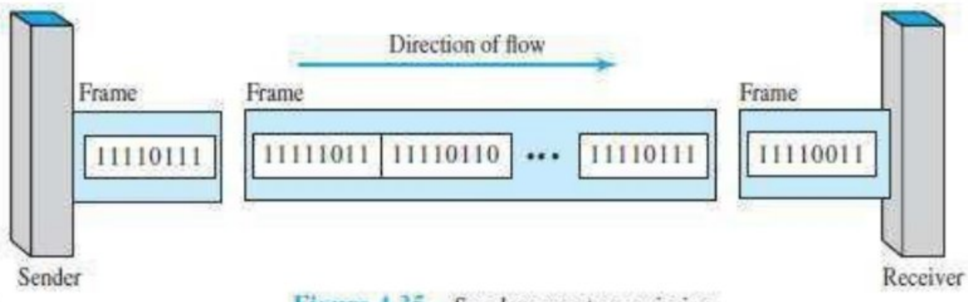



Figure 4.35 *Synchronous transmission*

- Advantages:
  1) Speed: Faster than asynchronous transmission. („." of no stop bit, start bit and gaps).
  2) Useful for high-speed applications such as transmission of data from one computer to another.

**2.2.2.3 Isochronous**

- Synchronization between characters is not enough; the entire stream of bits must be synchronized.
- The isochronous transmission guarantees that the data arrive at a fixed rate.
- In real-time audio/video, jitter is not acceptable. Therefore, synchronous transmission fails.
- For example: TV images are broadcast at the rate of 30 images per second. The images must beviewed at the same rate.

**ANALOG TRANSMISSION**

**2.3 DIGITAL TO ANALOG CONVERSION**

- Digital-to-analog conversion is the process of changing one of the characteristics of an analog-signal based on the information in digital-data (Figure 5.1).

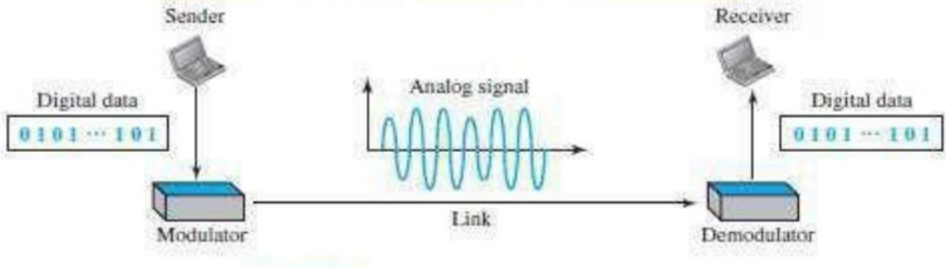



Figure 5.1 *Digital-to-analog conversion*

- A sine wave can be defined by 3 attributes:
  1) Amplitude 2)
  Frequency & 3) Phase.
- When anyone of the 3 attributes of a wave is varied, a different version of the wave will be created.
- So, by changing one attribute of an analog signal, we can use it to represent digital-data.
- Four methods of digital to analog conversion (Figure 5.2):
  1) Amplitude shift
  keying (ASK) 2)
  Frequency shift keying
  (FSK)
  3) Phase shift keying (PSK)
  4) Quadrature amplitude modulation (QAM).
- QAM is a combination of ASK and PSK i.e. QAM combines changing both the amplitude and phase. QAM is the most efficient of these 4 methods.