SRM
INSTITUTE OF SCIENCE & TECHNOLOGY
(Deemed to be University u/s 3 of UGC Act, 1956)

# 18CSS202J- COMPUTER COMMUNICATION

## UNIT III

- Line Coding Schemes: Unipolar, Polar and Bipolar.
- Amplitude Shift Keying Technique, Frequency Shift Keying Technique and Phase Shift Keying Technique
- Pulse Code Modulation, Delta Modulation
- Guided Media: Twisted pair, coaxial and Fiber optic cables.
- Unguided Media: Radio waves, Microwaves and Infrared.

# DIGITAL-TO-DIGITAL CONVERSION

Data can be either digital or analog. Signals that represent data can also be digital or analog. In this section, we see how we can represent digital data by using digital signals.

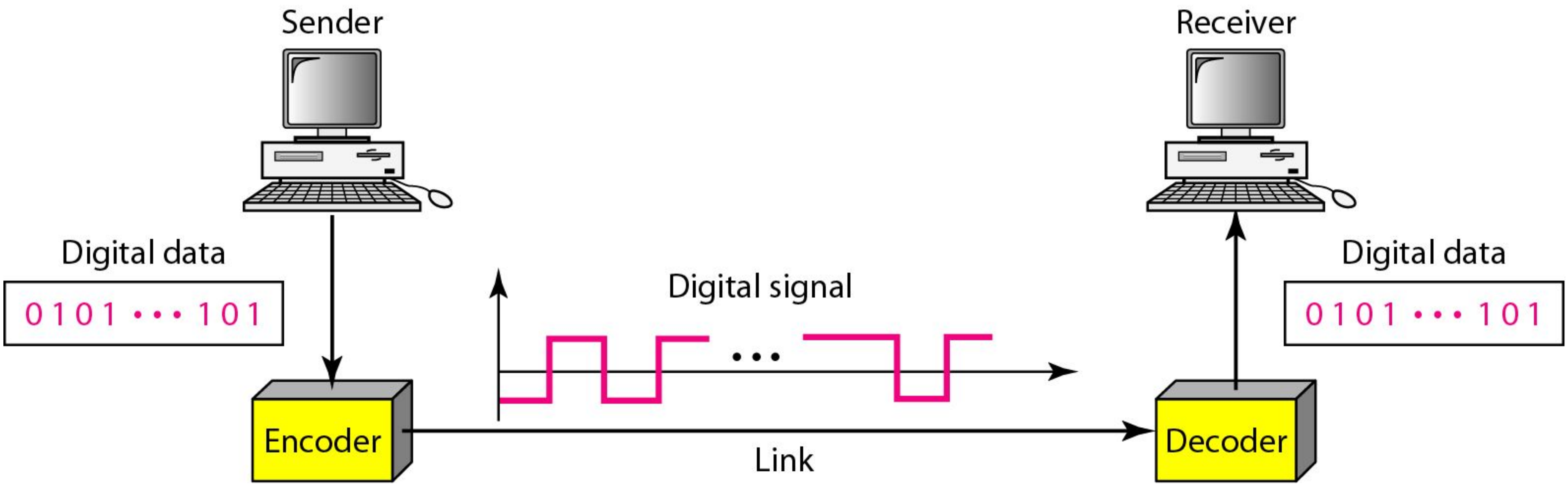
The conversion involves three techniques: line coding, block coding, and scrambling. Line coding is always needed; block coding and scrambling may or may not be needed.

Line coding is used to convert digital data to a digital signal. Line Coding Scheme is discussed here

# Line Coding

- **Line coding** is the process of converting digital data to digital signals
- Converting a string of 1’s and 0’s (digital data) into a sequence of signals that denote the 1’s and 0’s.
- For example a high voltage level (+V) could represent a “1” and a low voltage level (0 or -V) could represent a “0”.

## *Line coding and decoding*



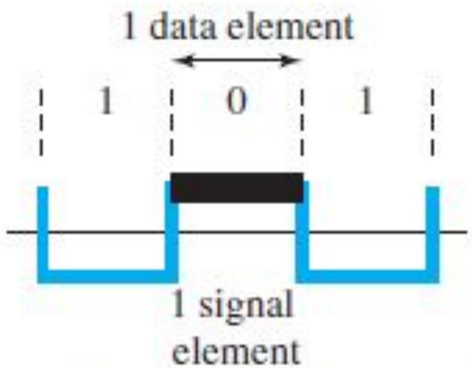
# Characteristics of different line coding schemes

***1. Signal Element Versus Data Element***

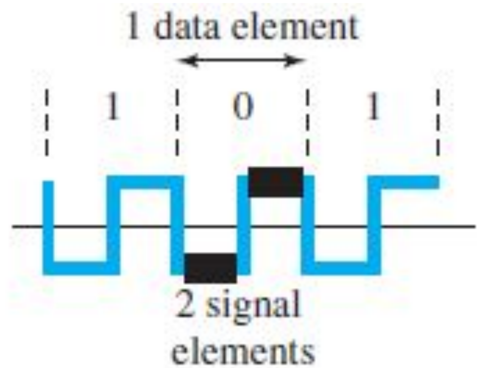
A data element is the smallest entity that can represent a piece of information: this is the bit. In digital data communications, a signal element carries data elements.

In other words, data elements are what we need to send; signal elements are what we can send. Data elements are being carried; signal elements are the carriers.

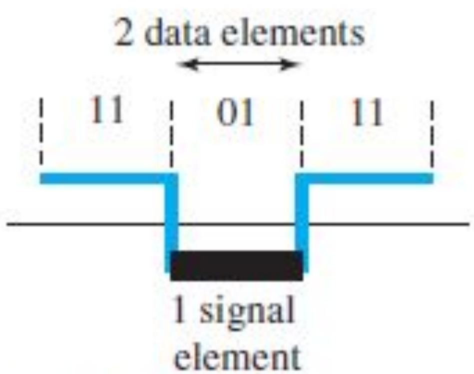
The ratio represented by *r* is defined as the number of data elements carried by each signal element



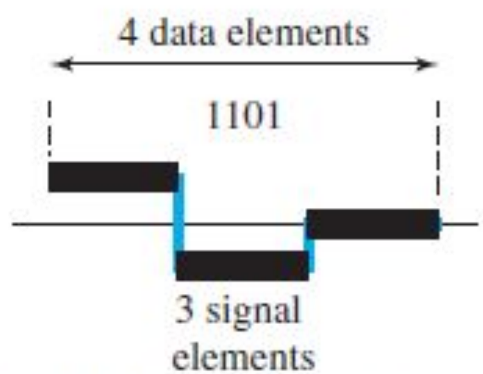

a. One data element per one signal element ($r = 1$)




b. One data element per two signal elements $\left(r = \frac{1}{2}\right)$




c. Two data elements per one signal element ($r = 2$)




d. Four data elements per three signal elements $\left(r = \frac{4}{3}\right)$

# Data Rate Versus Signal Rate

- The **data rate** defines the number of data elements (bits) sent in 1s. The unit is bits per second (bps).

- The **signal rate** is the number of signal elements sent in 1s. 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.**

- One goal in data communications is to increase the data rate while decreasing 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 ($N$) and signal rate ($S$)

$$\boldsymbol{S = N/r}$$

the average signal rate is defined as

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

***C – case factor***

# Mapping Data symbols onto Signal levels

- A data symbol (or element) can consist of a number of data bits:
  - 1 , 0 or
  - 11, 10, 01, ……
- A data symbol can be coded into a single signal element or multiple signal elements
  - 1 -> +V, 0 -> -V
  - 1 -> +V and -V, 0 -> -V and +V
- The ratio ‘r’ is the number of data elements carried by a signal element.

# Exampl
# e

***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***

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

**Not**
**e**

**Although the actual bandwidth of a digital signal is infinite, the effective bandwidth is finite.**

# Exampl e

***The maximum data rate of a channel (see Chapter 3) is Nmax = 2 × B × $\log_2 L$ (defined by the Nyquist formula). Does this agree with the previous formula for $N_{max}$?***

## Solution

*A signal with L levels actually can carry $\log_2 L$ bits per level. If each level corresponds to one signal element and we assume the average case (c = 1/2), then we have*

$$N_{\max} = \frac{1}{c} \times B \times r = 2 \times B \times \log_2 L$$

# Considerations for choosing a good signal element referred to as line encoding

- Baseline wandering - a receiver will evaluate the average power of the received signal (called the baseline) and use that to determine the value of the incoming data elements. If the incoming signal does not vary over a long period of time, the baseline will drift and thus cause errors in detection of incoming data elements.
- A good line encoding scheme will prevent long runs of fixed amplitude.

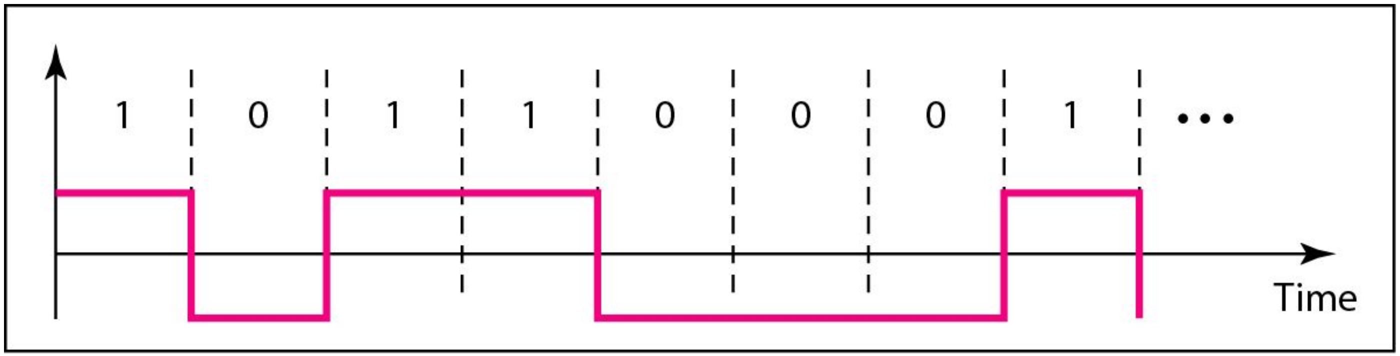
# Line encoding characteristics

- DC components - when the voltage level remains constant for long periods of time, there is an increase in the low frequencies of the signal. Most channels are bandpass and may not support the low frequencies.
- This will require the removal of the dc component of a transmitted signal.

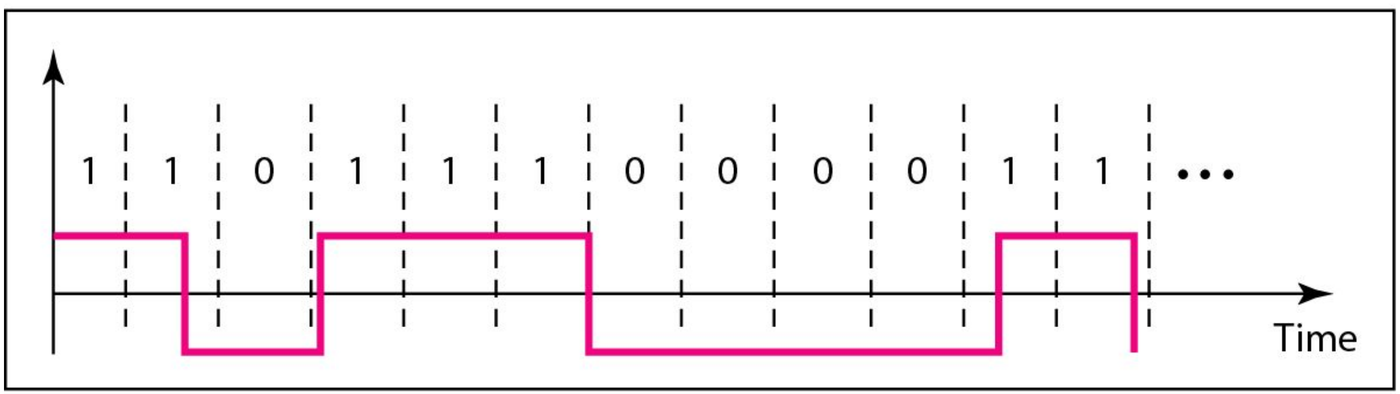
# Line encoding characteristics

- Self synchronization - the clocks at the sender and the receiver must have the same bit interval.
- If the receiver clock is faster or slower it will misinterpret the incoming bit stream.

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




a. Sent




b. Received

# Example 4.3

***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?***

*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 |

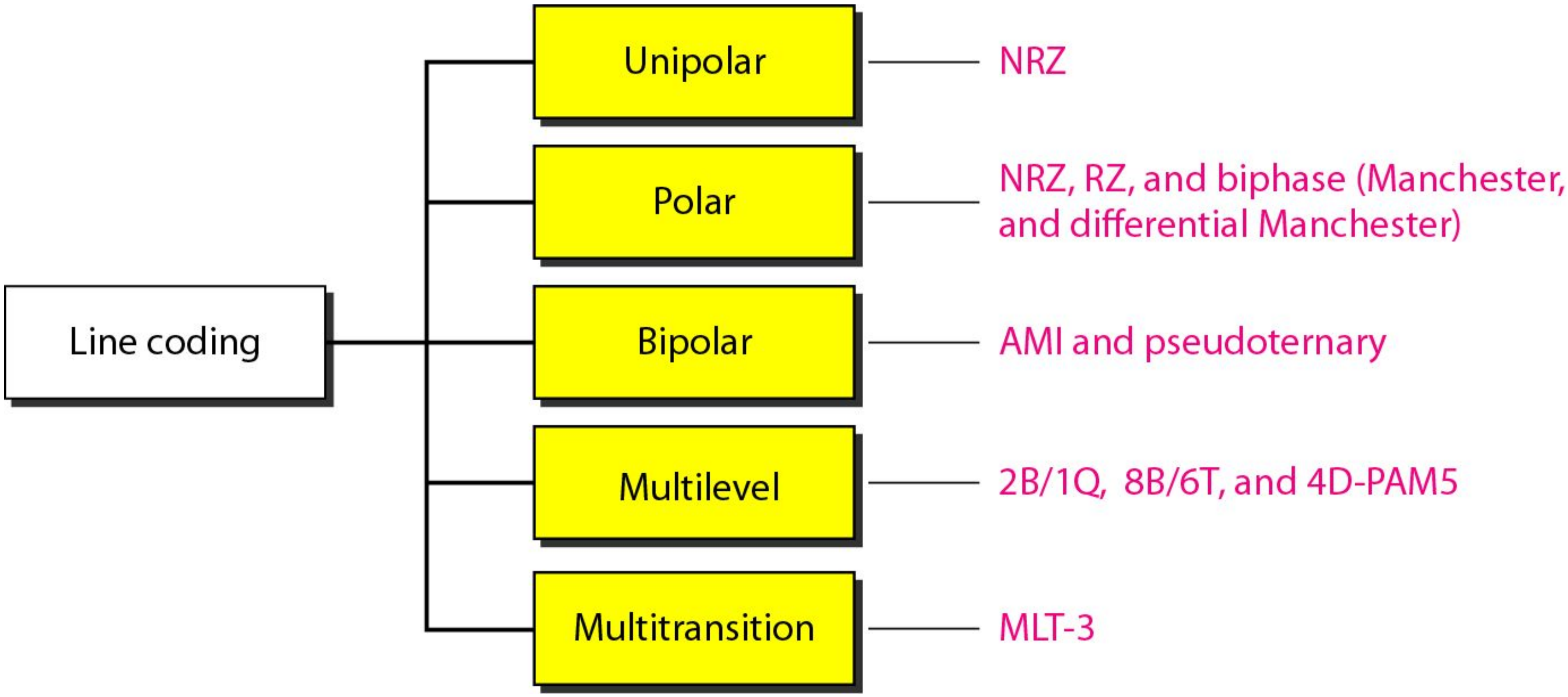
# Line encoding characteristics

- Error detection - errors occur during transmission due to line impairments.
- Some codes are constructed such that when an error occurs it can be detected.

# Line encoding characteristics

- Noise and interference - there are line encoding techniques that make the transmitted signal “immune” to noise and interference.
- This means that the signal cannot be corrupted, it is stronger than error detection.

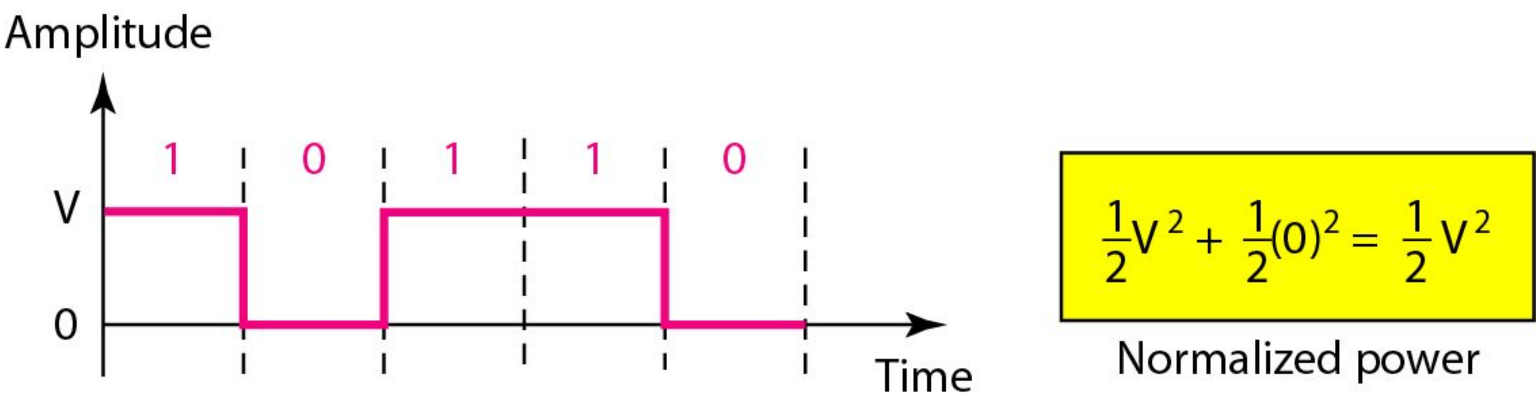
## **Figure** *Line coding schemes*



# Unipolar

- All signal levels are on one side of the time axis , either above or below.
- +v define 1 and –v define 0.
- NRZ - Non Return to Zero scheme is an example of this code. The signal level does not return to zero at a middle of the bit.
- Scheme is prone to baseline wandering and DC components. It has no synchronization or any error detection. It is simple but costly in power consumption.

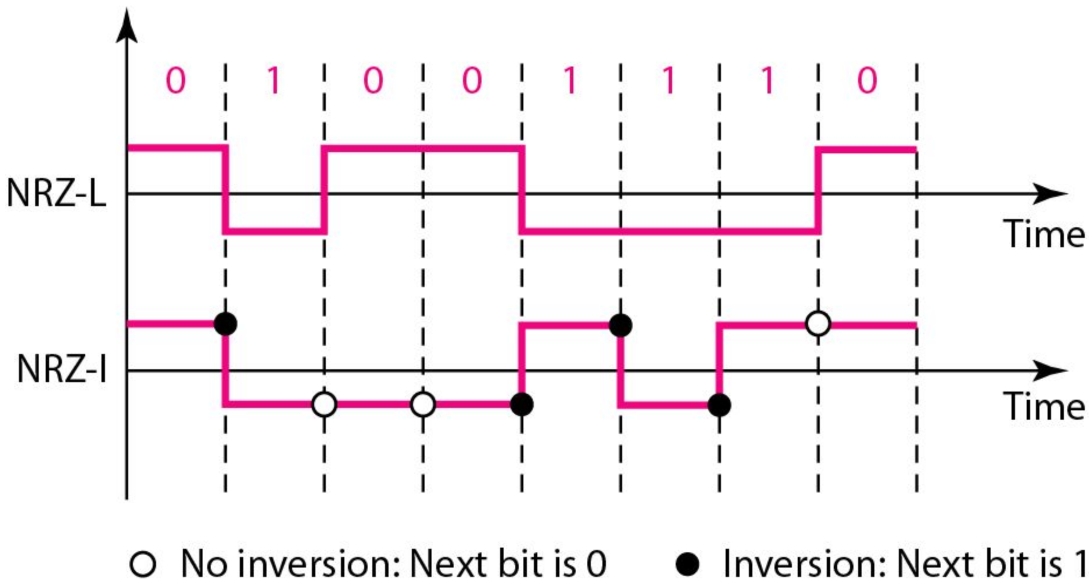
## **Figure** ***Unipolar NRZ scheme***

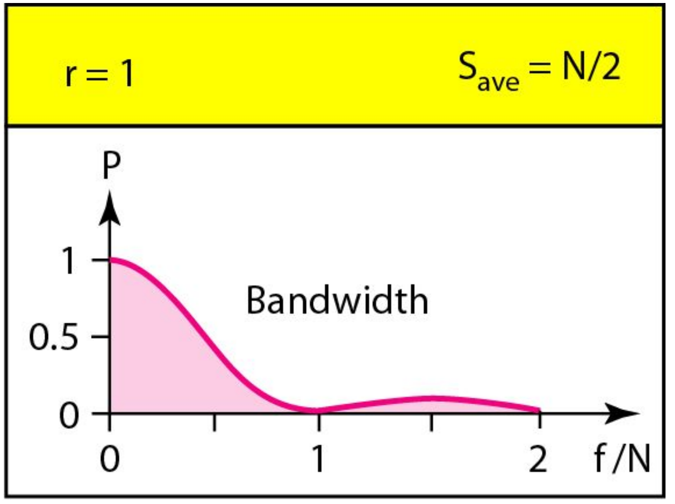


# Polar - NRZ

- The voltages are on both sides of the time axis.
- Polar NRZ scheme can be implemented with two voltages. E.g. +V for 0 and -V for 1.
- There are two versions:
  - NZR - Level (NRZ-L) - positive voltage for one symbol and negative for the other
  - NRZ - Inversion (NRZ-I) - the change or lack of change in polarity determines the value of a symbol. E.g. a “1” symbol inverts the polarity a “0” does not.

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



***Not e***

**In NRZ-L the level of the voltage determines the value of the bit.
In NRZ-I the inversion or the lack of inversion determines the value of the bit.**

**Note**

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

**Not**
**e**

**NRZ-L and NRZ-I both have a DC component problem and baseline wandering, it is worse for NRZ-L. Both have no self synchronization &no error detection. Both are relatively simple to implement.**

## Example

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

## Solution

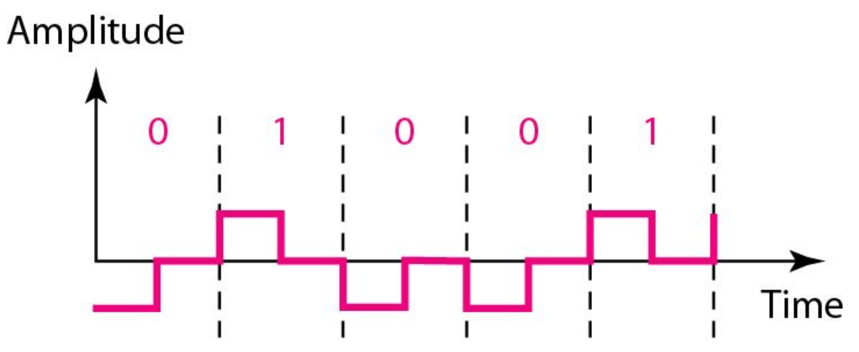
*The average signal rate is S= N / 2 = 500 kbaud. The minimum bandwidth for this average baud rate is Bmin = S = 500 kHz.*

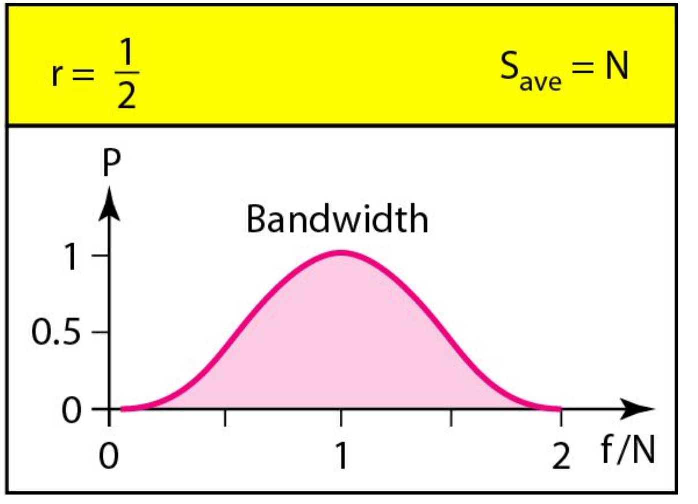
*Note c = 1/2 for the avg. case as worst case is 1 and best case is 0*

# Polar - RZ

- The Return to Zero (RZ) scheme uses three voltage values. +, 0, -.
- Each symbol has a transition in the middle. Either from high to zero or from low to zero.
- This scheme has more signal transitions (two per symbol) and therefore requires a wider bandwidth.
- No DC components or baseline wandering.
- Not synchronized.
- More complex as it uses three voltage level. It has no error detection capability.

## Figure 4.7 *Polar RZ scheme*

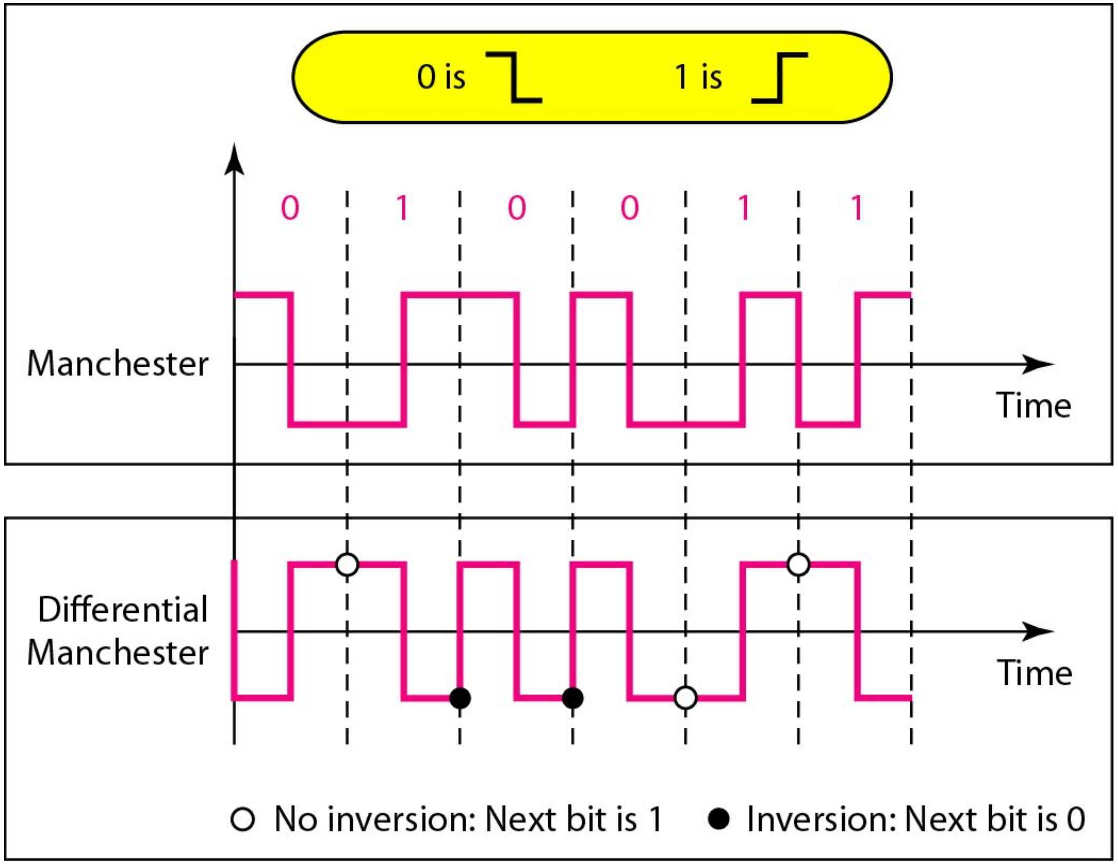


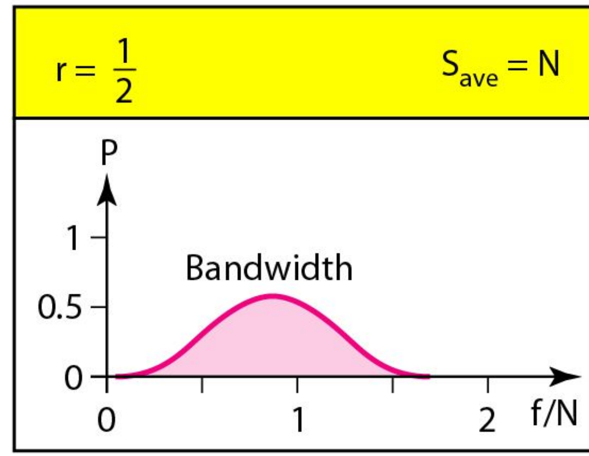



# Polar - Biphase: Manchester and Differential Manchester

- Manchester coding consists of combining the NRZ-L and RZ schemes.
  - Every symbol has a level transition in the middle: from high to low or low to high. Uses only two voltage levels.
- Differential Manchester coding consists of combining the NRZ-I and RZ schemes.
  - Every symbol has a level transition in the middle. But the level at the beginning of the symbol is determined by the symbol value. One symbol causes a level change the other does not.

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



**Not e**

**In Manchester and differential Manchester encoding, the transition at the middle of the bit is used for synchronization.**

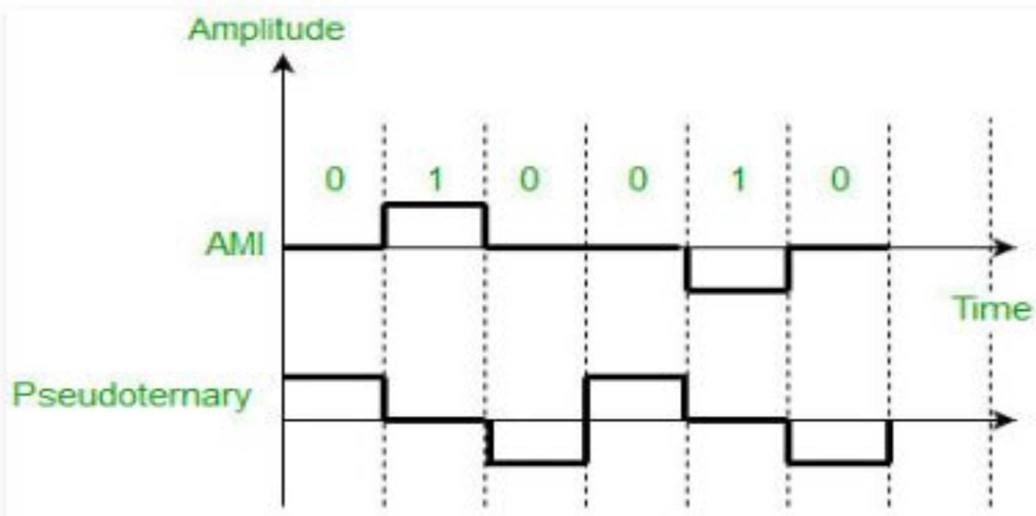
**Note**

**The minimum bandwidth of Manchester and differential Manchester is 2 times that of NRZ. The is no DC component and no baseline wandering. None of these codes has error detection.**

**Bipolar schemes –**

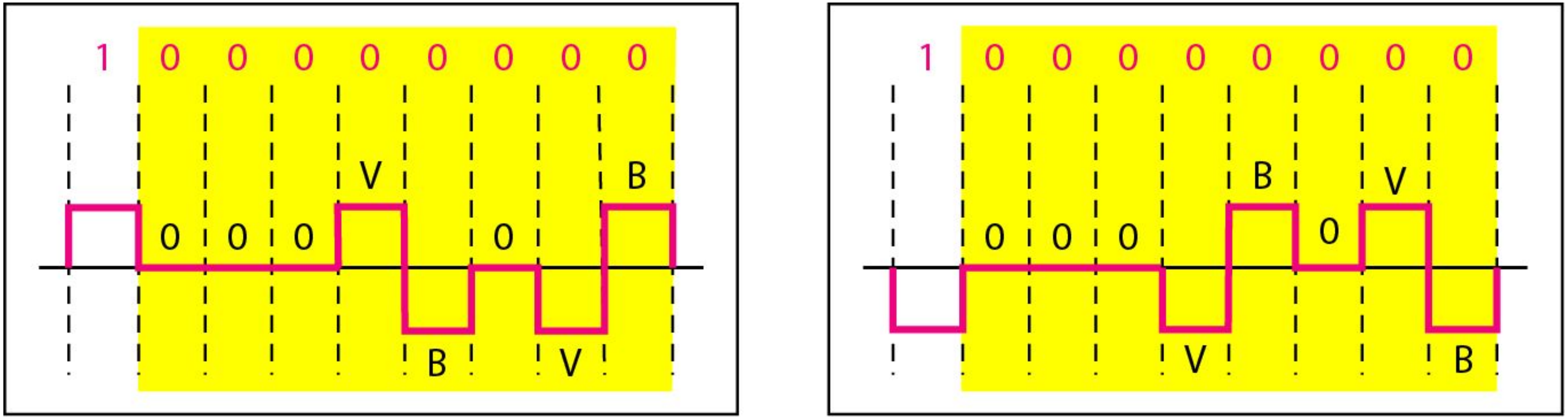
In this scheme there are three voltage levels positive, negative, and zero. The voltage level for one data element is at zero, while the voltage level for the other element alternates between positive and negative.

- **Alternate Mark Inversion (AMI) –** A neutral zero voltage represents binary 0. Binary 1's are represented by alternating positive and negative voltages.
- **Pseudoternary –** Bit 1 is encoded as a zero voltage and the bit 0 is encoded as alternating positive and negative voltages i.e., opposite of AMI scheme. Example: Data = 010010.



**For example: B8ZS substitutes eight consecutive zeros with 000VB0VB.**
**The V stands for violation, it violates the line encoding rule**
**B stands for bipolar, it implements the bipolar line encoding rule**

# Figure 4.19 *Two cases of B8ZS scrambling technique*




a. Previous level is positive.

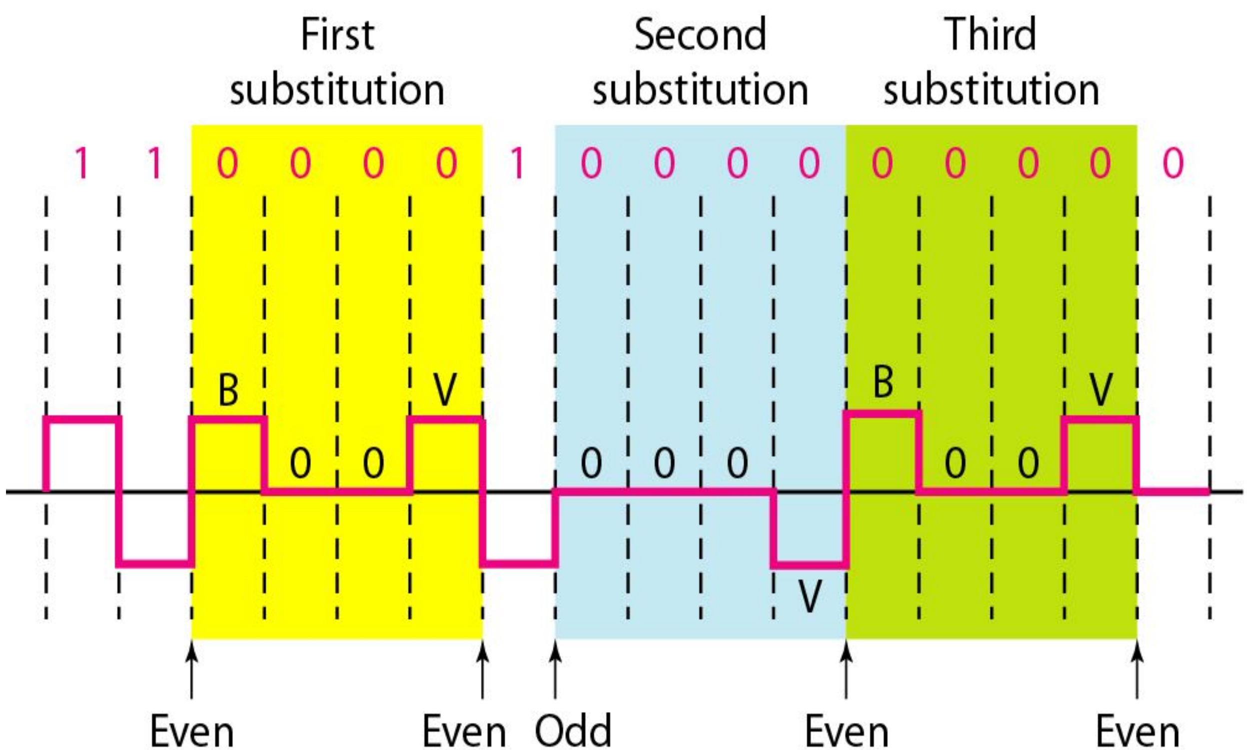
b. Previous level is negative.

**HDB3 substitutes four consecutive zeros with 000V or B00V depending on the number of nonzero pulses after the last substitution.**

**If # of non zero pulses is even the substitution is B00V to make total # of non zero pulse even.**

**If # of non zero pulses is odd the substitution is 000V to make total # of non zero pulses even.**

## **Figure 4.20** ***Different situations in HDB3 scrambling technique***




Even – B00V + □ -00-
- □ +00+

Odd – 000V + □ 000+
- □ 000-

# DIGITAL-TO-ANALOG CONVERSION

# DIGITAL-TO-ANALOG CONVERSION

## ***Topics to be discussed***

- Aspects of Digital-to-Analog Conversion
- Amplitude Shift Keying
- Frequency Shift Keying
- Phase Shift Keying
- Quadrature Amplitude Modulation

# Digital to Analog Conversion

- Converting digital data to a bandpass analog signal is traditionally called 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.*
- Digital data needs to be carried on an analog signal.
- A carrier signal (frequency $f_c$) performs the function of transporting the digital data in an analog waveform.
- The analog carrier signal is manipulated to uniquely identify the digital data being carried.

# Figure 1 *Digital-to-analog conversion*

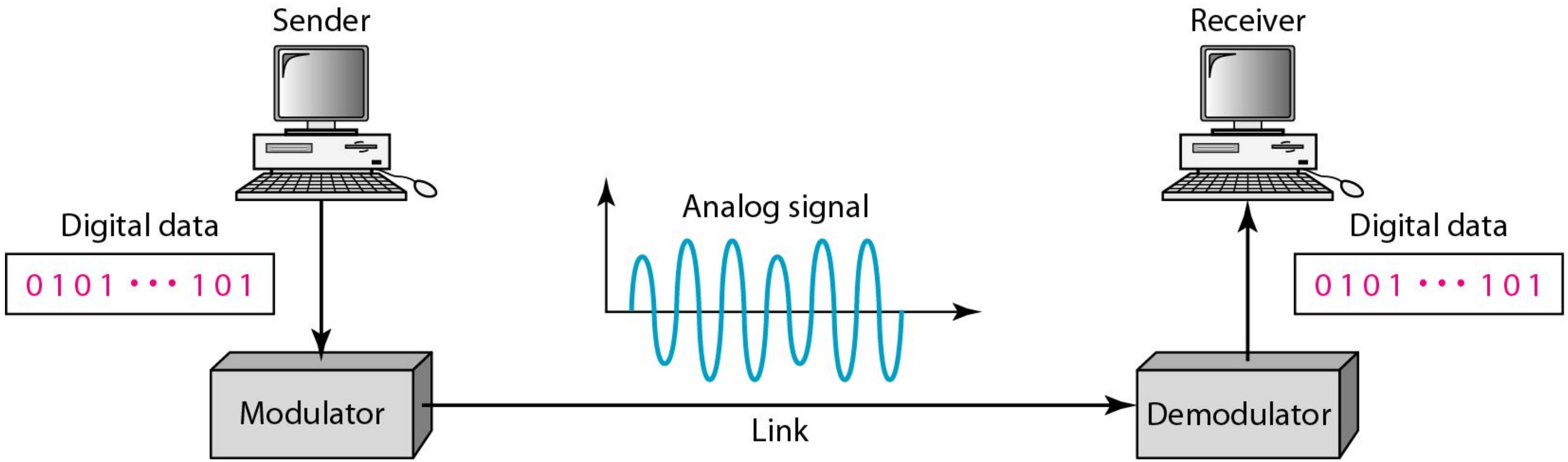



Figure 1 shows the relationship between the digital information, the digital-to-analog modulating process, and the resultant analog signal.

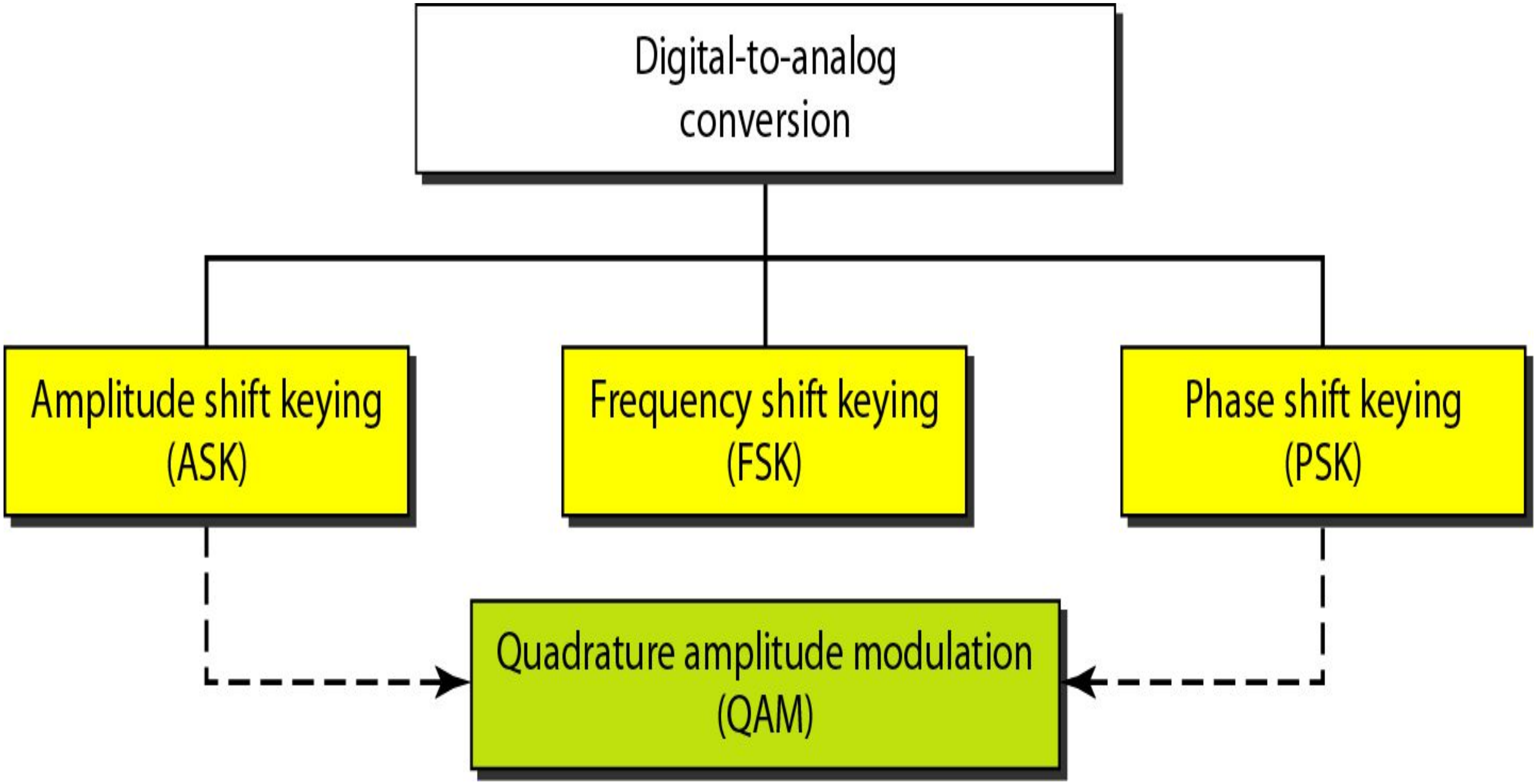
*Types of digital-to-analog conversion*

- Four mechanisms are used for modulating digital data into an analog signal
  amplitude shift keying (ASK)
  frequency shift keying (FSK)
  phase shift keying(PSK)

quadrature amplitude modulation (QAM – combines ASK &PSK)

QAM is the most efficient and commonly used today

# Figure 2 *Types of digital-to-analog conversion*



# Bit and Baud rates and the carrier signal

- **Bit rate**, N, is the number of bits per second (bps). Also called as ***Data Rate***

- Baud rate is the number of signal elements per second (bauds). Also called as ***Signal Rate***

# Cond..

- The relationship between them is

  S=Nx1/r bauds

  Where

  *N - data rate*

  r - number of data bits per signal element.

- In the analog transmission of digital data, the signal or baud rate is less than or equal to the bit rate.

# Example 1

- *An analog signal carries 4 bits per signal element. If 1000 signal elements are sent per second, find the bit rate.*
- *Solution*
- *In this case, r = 4, S = 1000, and N is unknown. We can find the value of N from*

$$S = N \times \frac{1}{r} \qquad \text{or} \qquad N = S \times r = 1000 \times 4 = 4000 \text{ bps}$$

*Example 2*

- An analog signal has a bit rate of 8000 bps and a baud rate of 1000 baud. How many data elements are carried by each signal element? How many signal elements do we need?
- Solution
- In this example, S = 1000, *N =8000, and rand L are unknown. We find first the value of rand* then the value of *L.*

$$S = N \times \frac{1}{r} \longrightarrow r = \frac{N}{S} = \frac{8000}{1000} = 8 \text{ bits/baud}$$

$$r = \log_2 L \longrightarrow L = 2^r = 2^8 = 256$$

# Amplitude Shift Keying (ASK)

- In  ASK, the amplitude of the carrier signal is varied to create signal elements. Both frequency and phase remain constant while the amplitude changes.
- ASK is implemented by changing the amplitude of a carrier signal to reflect amplitude levels in the digital signal.
- For example: a digital “1” could not affect the signal, whereas a digital “0” would, by making it zero.
- The line encoding will determine the values of the analog waveform to reflect the digital data being carried.

# Bandwidth of ASK

- The bandwidth B of ASK is proportional to the signal rate S.

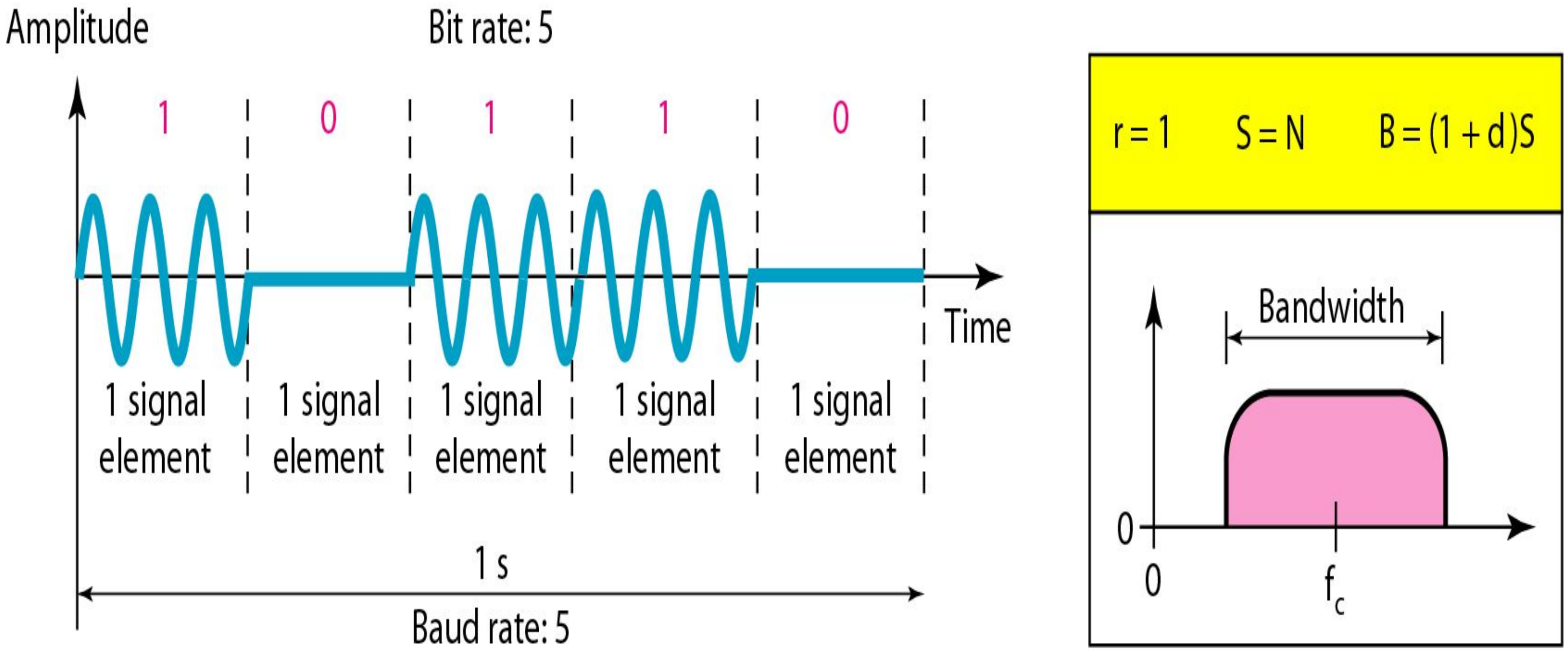
$$B = (1+d)S$$

- “d” is depends on  modulation and filtering process
- “d” value  lies between 0 and 1.

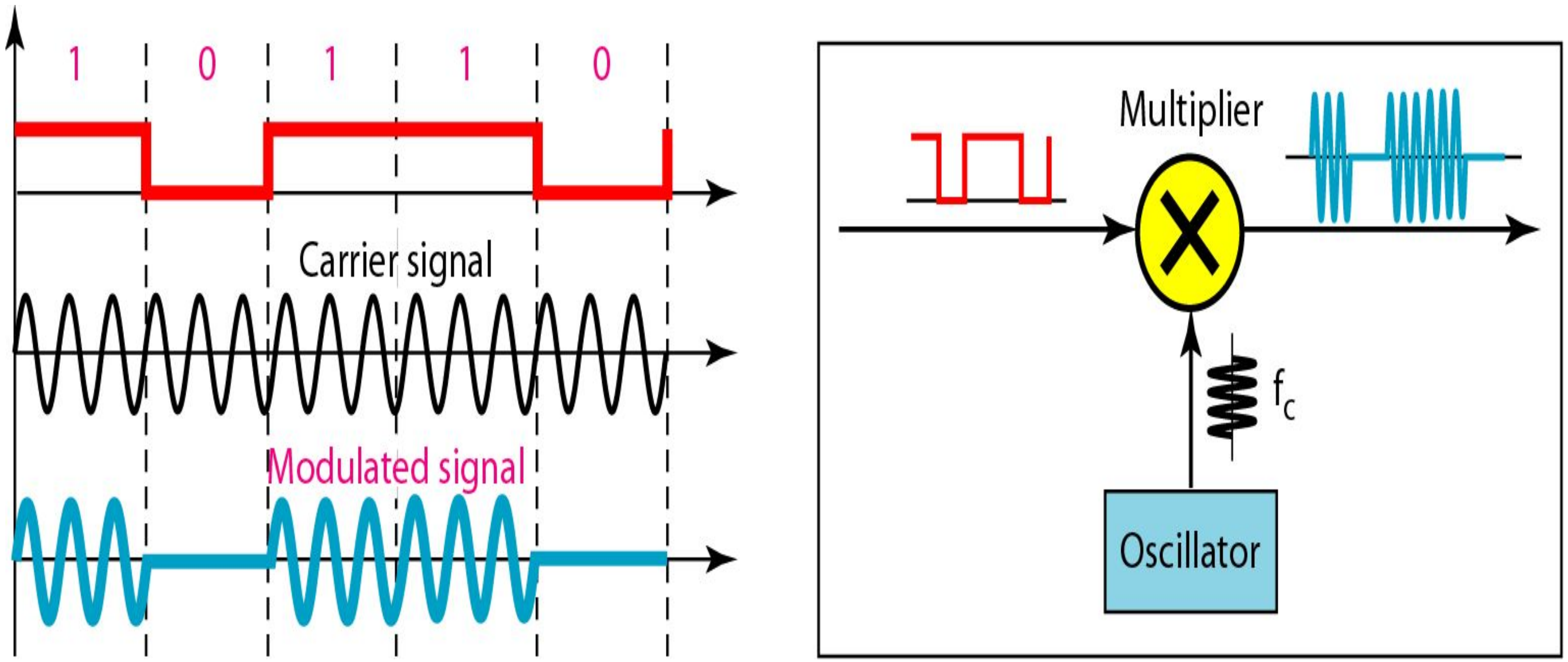
# *Binary ASK (BASK)*

- ASK is normally implemented using only two levels.
- This is referred to as binary amplitude shift keying or *on-off keying (OOK).*
- *The peak amplitude of one signal level* is 0
- the other is the same as the amplitude of the carrier frequency

# Figure 3 *Binary amplitude shift keying*



# Figure 4 *Implementation of binary ASK*



- If digital data are presented as a unipolar NRZ digital signal with a high voltage of I V and a low voltage of 0 V
- Implementation can achieved by multiplying the NRZ digital signal by the carrier signal coming from an oscillator.
- When the amplitude of the NRZ signal is 1, the amplitude of the carrier frequency is held;
- when the amplitude of the NRZ signal is 0, the amplitude of the carrier frequency is zero.

*Example 3*

We have an available bandwidth of 100 kHz which spans from 200 to 300 kHz. What are the carrier frequency and the bit rate if we modulated our data by using ASK with *d =l?*

Solution

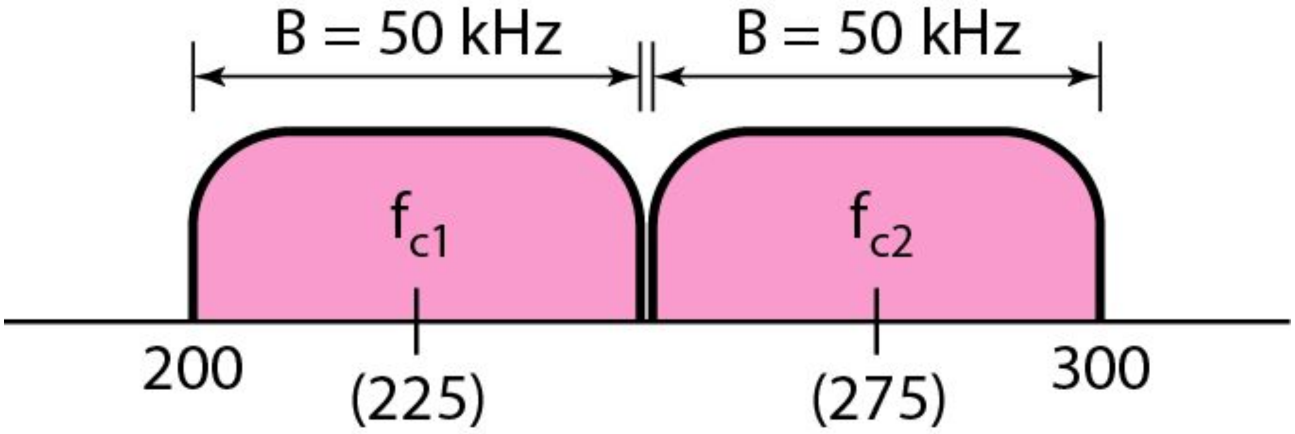
- The middle of the bandwidth is located at 250 kHz. This means that our carrier frequency can be *at fe =250 kHz. We can use the formula for bandwidth to find the bit rate (with d =1 and r =1).*

$$B = (1 + d) \times S = 2 \times N \times \frac{1}{r} = 2 \times N = 100 \text{ kHz} \quad \Rightarrow \quad N = 50 \text{ kbps}$$

*Example 4*

- *In data communications, we normally use full-duplex links with communication in both directions. We need to divide the bandwidth into two with two carrier frequencies, as shown in Figure 5. The figure shows the positions of two carrier frequencies and the bandwidths. The available bandwidth for each direction is now 50 kHz, which leaves us with a data rate of 25 kbps in each direction.*

# Figure 5 *Bandwidth of full-duplex ASK used in Example 4*

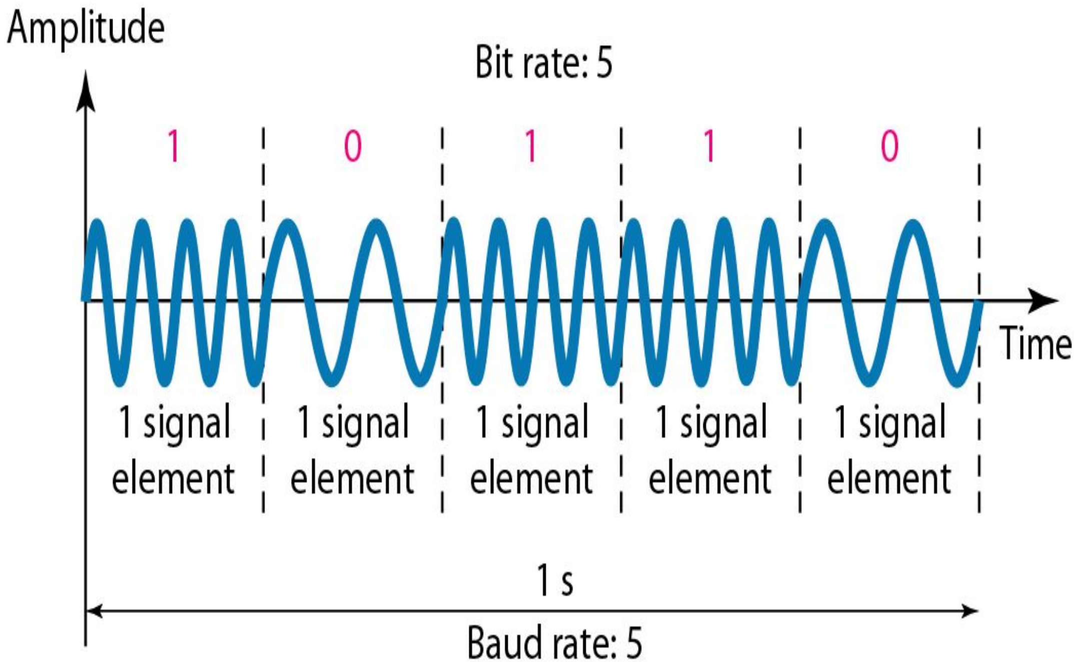


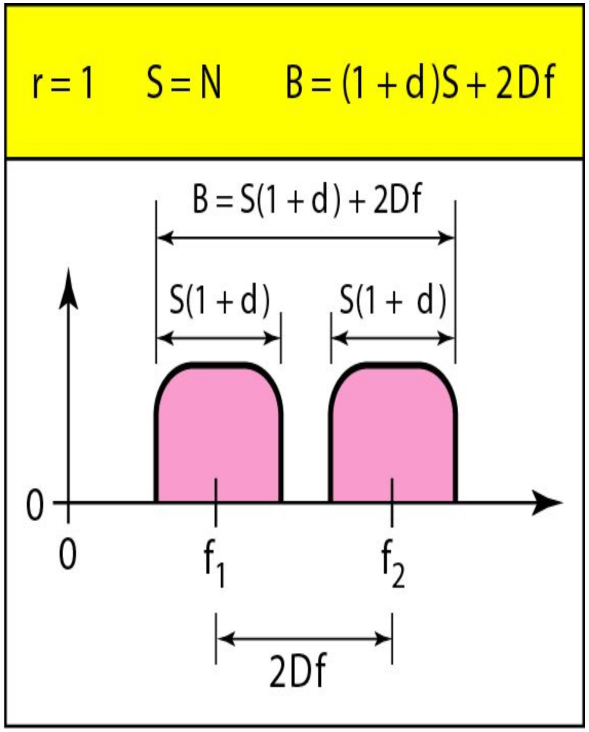
# Frequency Shift Keying

- In frequency shift keying, the frequency of the carrier signal is varied to represent data.
- The frequency of the modulated signal is constant for the duration of one signal element, but changes for the next signal element if the data element changes.
- Both peak amplitude and phase remain constant for all signal elements.
- The digital data stream changes the frequency of the carrier signal, $f_c$.
- For example,

  a "1" could be represented by $f_1=f_c+\Delta f$,

# Figure 6 *Binary frequency shift keying (BFSK)*



# Bandwidth of FSK

- If the difference between the two frequencies ($f_1$ and $f_2$) is $2\Delta f$, then the required BW B will be:

$$B = (1+d)xS + 2\Delta f$$

## *Example 5*

- *We have an available bandwidth of 100 kHz which spans from 200 to 300 kHz. What should be the carrier frequency and the bit rate if we modulated our data by using FSK with d = 1?*

*Solution*

- *This problem is similar to Example 5.3, but we are modulating by using FSK. The midpoint of the band is at 250 kHz. We choose $2\Delta f$ to be 50 kHz; this means*

$$\boxed{B = (1 + d) \times S + 2\Delta f = 100 \;\Rightarrow\; 2S = 50 \text{ kHz} \quad S = 25 \text{ kbaud} \quad N = 25 \text{ kbps}}$$

# Coherent and Non Coherent

- Two implementations of BFSK:

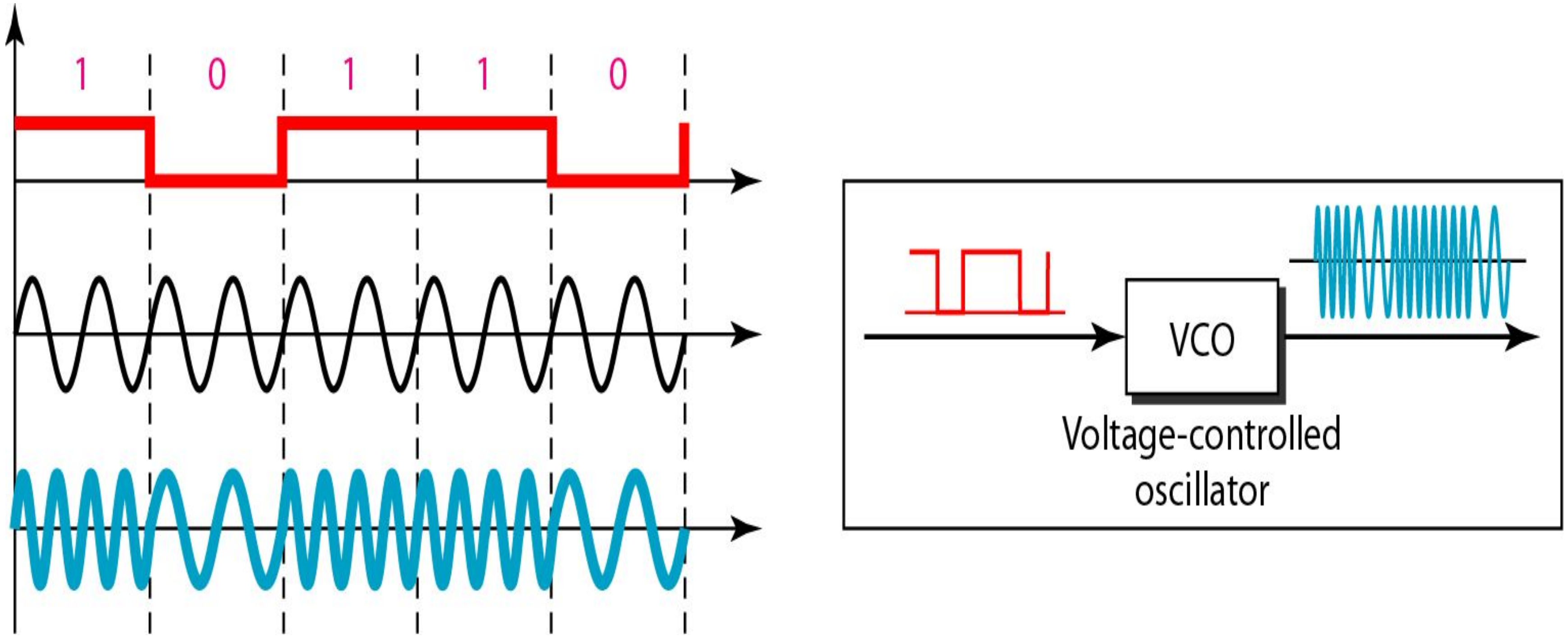
  noncoherent and coherent.
- In a non-coherent FSK scheme, when we change from one frequency to the other, we do not adhere to the current phase of the signal.
- In coherent FSK, the switch from one frequency signal to the other only occurs at the same phase in the signal.

- Noncoherent BFSK can be implemented by treating

  BFSK as two ASK modulations and using two carrier frequencies.
- Coherent BFSK can be implemented by using one *voltage-controlled oscillator (VeO) that changes its frequency* according to the input voltage.
- Figure 7 shows the simplified idea behind the

  second implementation.
- The input to the oscillator is the unipolar NRZ signal.
- When the amplitude of NRZ is zero, the oscillator keeps its regular frequency;
- when the amplitude is positive, the frequency is increased.

# Figure 7 *Implementation of BFSK*



# Multi level FSK

- Similarly to ASK, FSK can use multiple bits per signal element.
- That means we need to provision for multiple frequencies, each one to represent a group of data bits.
- The bandwidth for FSK can be higher

$$B = (1+d)xS + (L-1)/2\Delta f = LxS$$

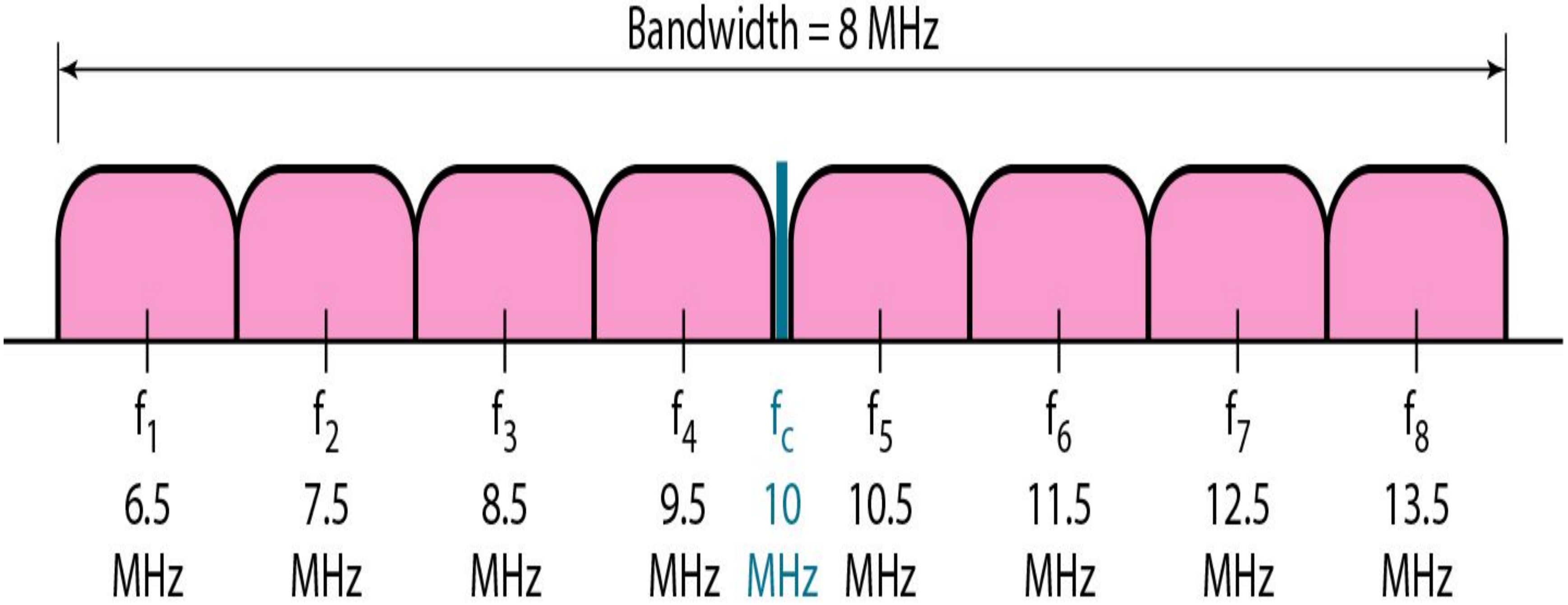
## *Example 6*

- *We need to send data 3 bits at a time at a bit rate of 3 Mbps. The carrier frequency is 10 MHz. Calculate the number of levels (different frequencies), the baud rate, and the bandwidth.*

## *Solution*

- *We can have* $L = 2^3 = 8$*. The baud rate is* $S = 3$ *Mbps/3 = 1 Mbaud. This means that the carrier frequencies must be 1 MHz apart (*$2\Delta f = 1$ *MHz). The bandwidth is* $B = 8 \times 1M = 8M$*. Figure 8 shows the allocation of frequencies and bandwidth.*

# Figure 8 *Bandwidth of MFSK used in Example 6*



# Phase Shift Keying

- In PSK, the phase of the carrier is varied to represent two or more different signal elements. Both peak amplitude and frequency remain constant as the phase changes.
- Today, PSK is more common than ASK or FSK
- We vary the phase shift of the carrier signal to represent digital data.
- The bandwidth requirement, B is:

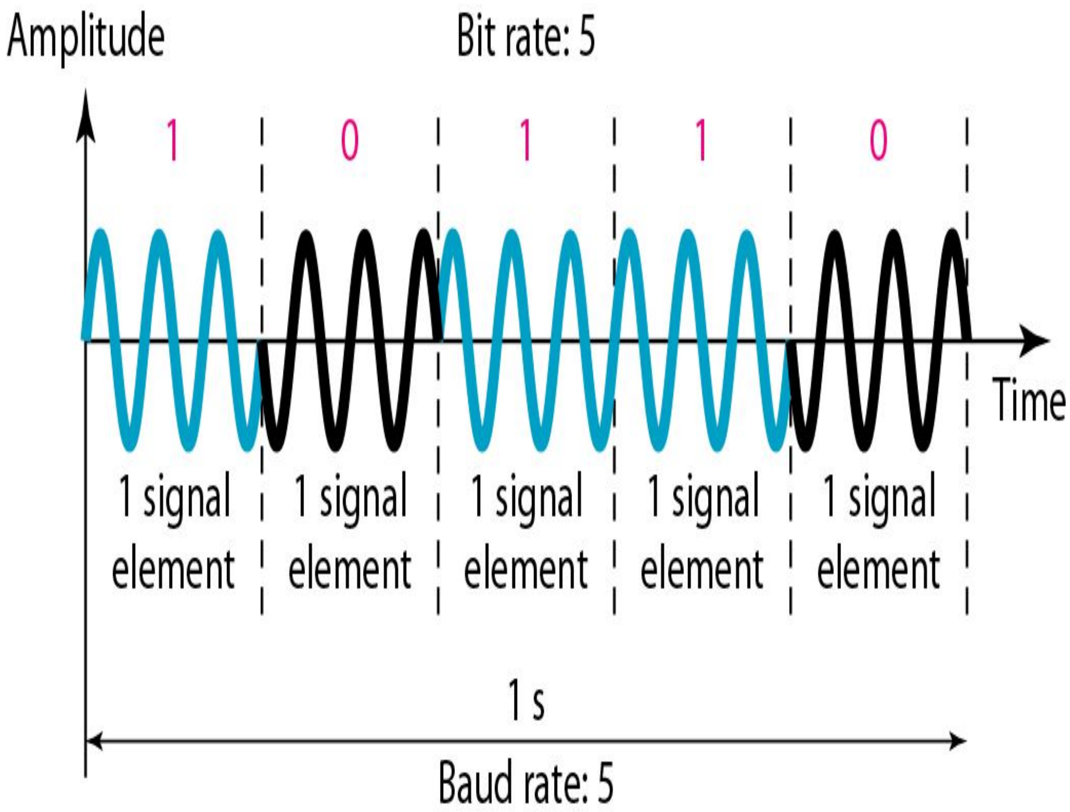
$$B = (1+d)xS$$

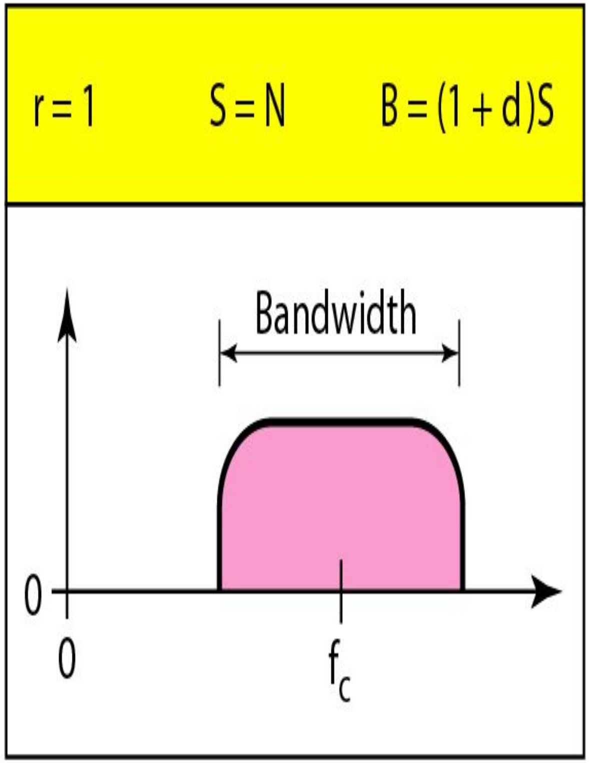
- PSK is much more robust than ASK as it is not that vulnerable to noise, which changes amplitude of the signal.

# *Binary phase shift keying (BPSK)*

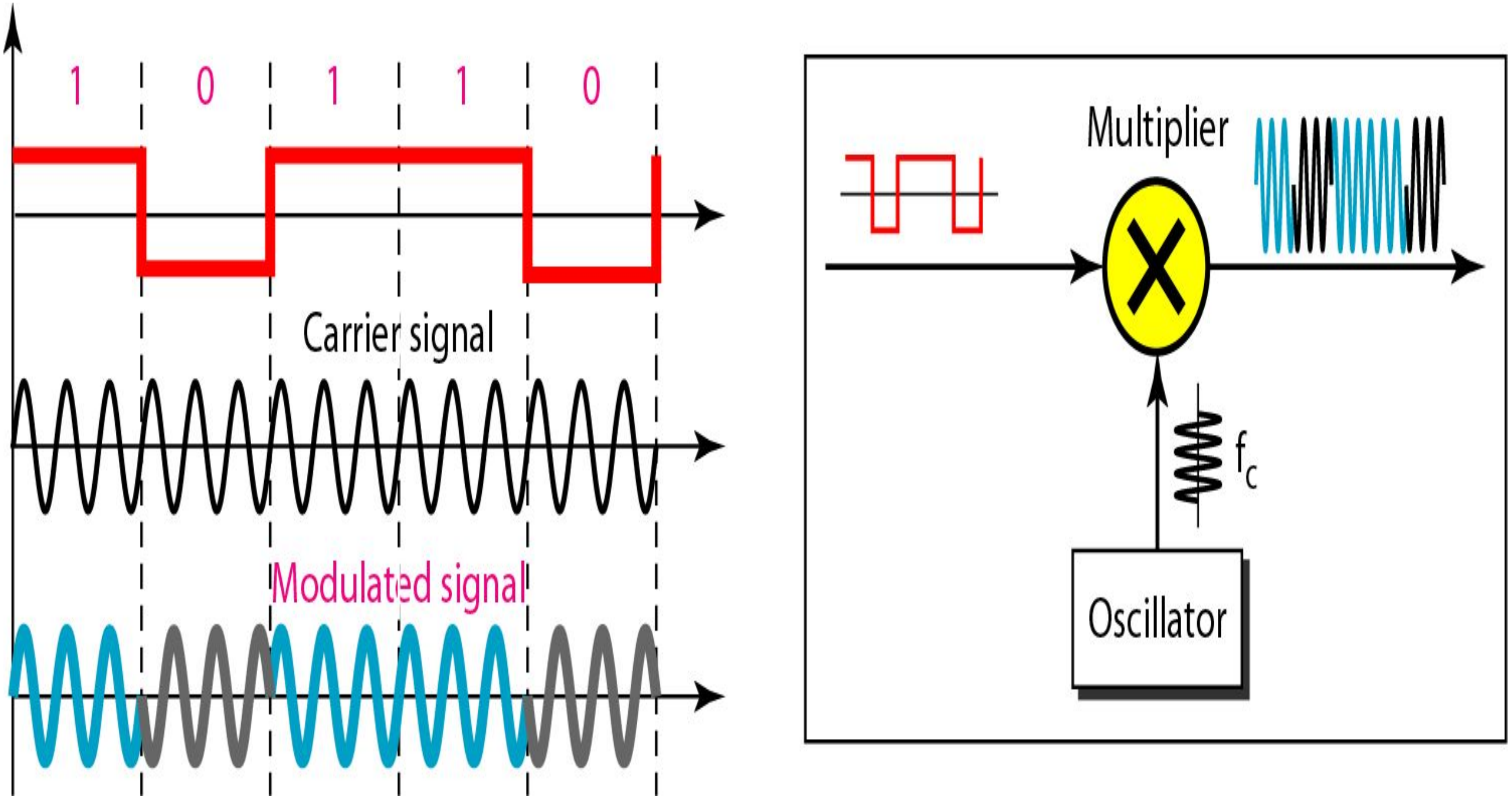
- The simplest PSK is binary PSK, in which we have only two signal elements, one with a phase of 0°, and the other with a phase of 180°
- Binary PSK is as simple as binary ASK
- PSK is less susceptible to noise than ASK.
- PSK is superior to FSK because we do not need two carrier signals.

# Figure 9 *Binary phase shift keying*



# Figure 10  *Implementation of BASK*

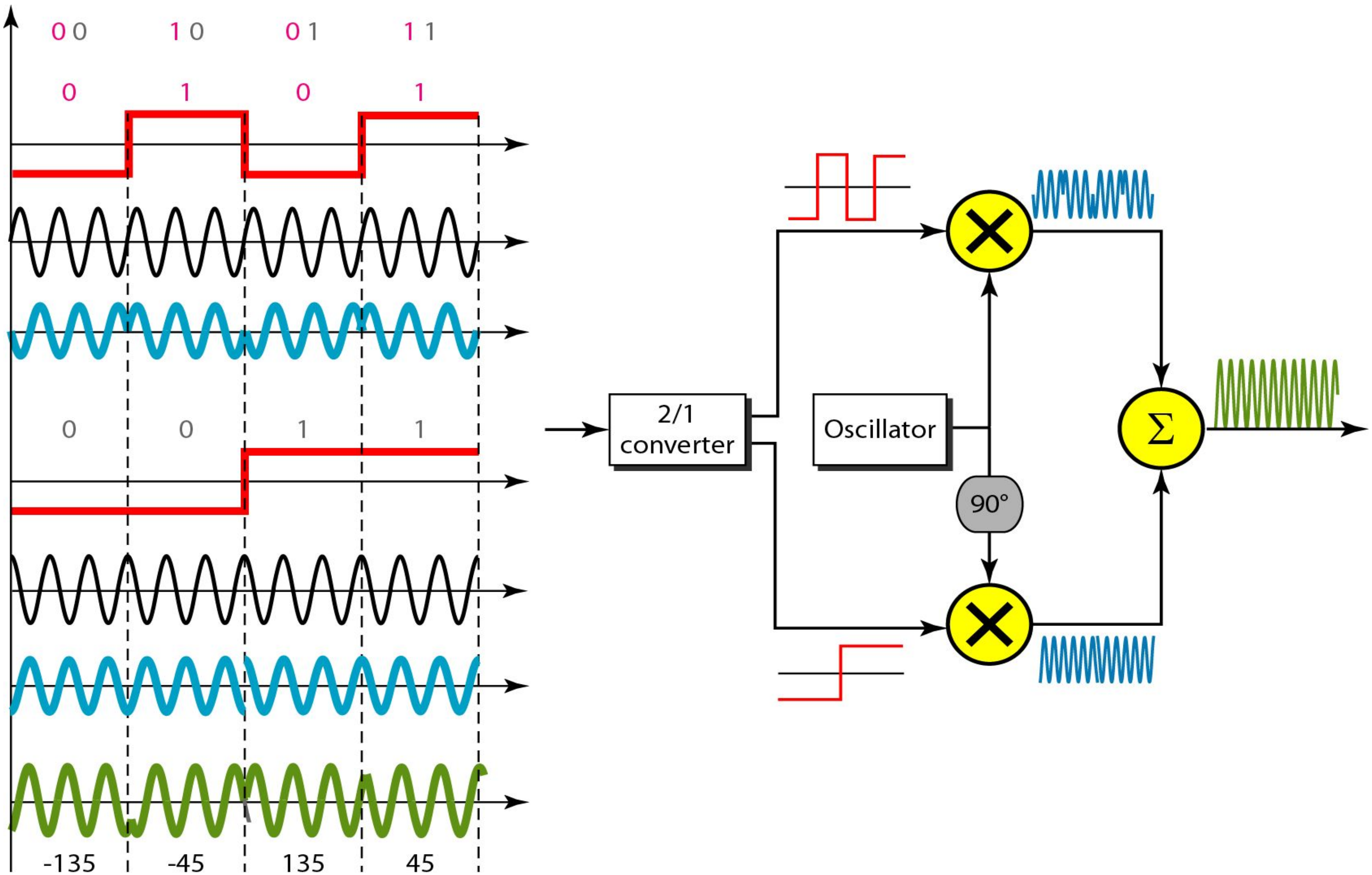


- The implementation of BPSK is as simple as that for ASK
- The reason is that the signal element with phase 180° can be seen as the complement of the signal element with phase 0°
- The polar NRZ signal is multiplied by the carrier frequency;
- the 1 bit (positive voltage) is represented by a phase starting at 0°;
- the 0 bit (negative voltage) is represented by a phase starting at 180°.

# Quadrature PSK

- To increase the bit rate, we can code 2 or more bits onto one signal element.
- In QPSK, we parallelize the bit stream so that every two incoming bits are split up and PSK a carrier frequency. One carrier frequency is phase shifted $90^o$ from the other - in quadrature.
- The two PSKed signals are then added to produce one of 4 signal elements. L = 4 here.

# Figure 11 *QPSK and its implementation*



## *Example 7*

- *Find the bandwidth for a signal transmitting at 12 Mbps for QPSK. The value of $d = 0$.*

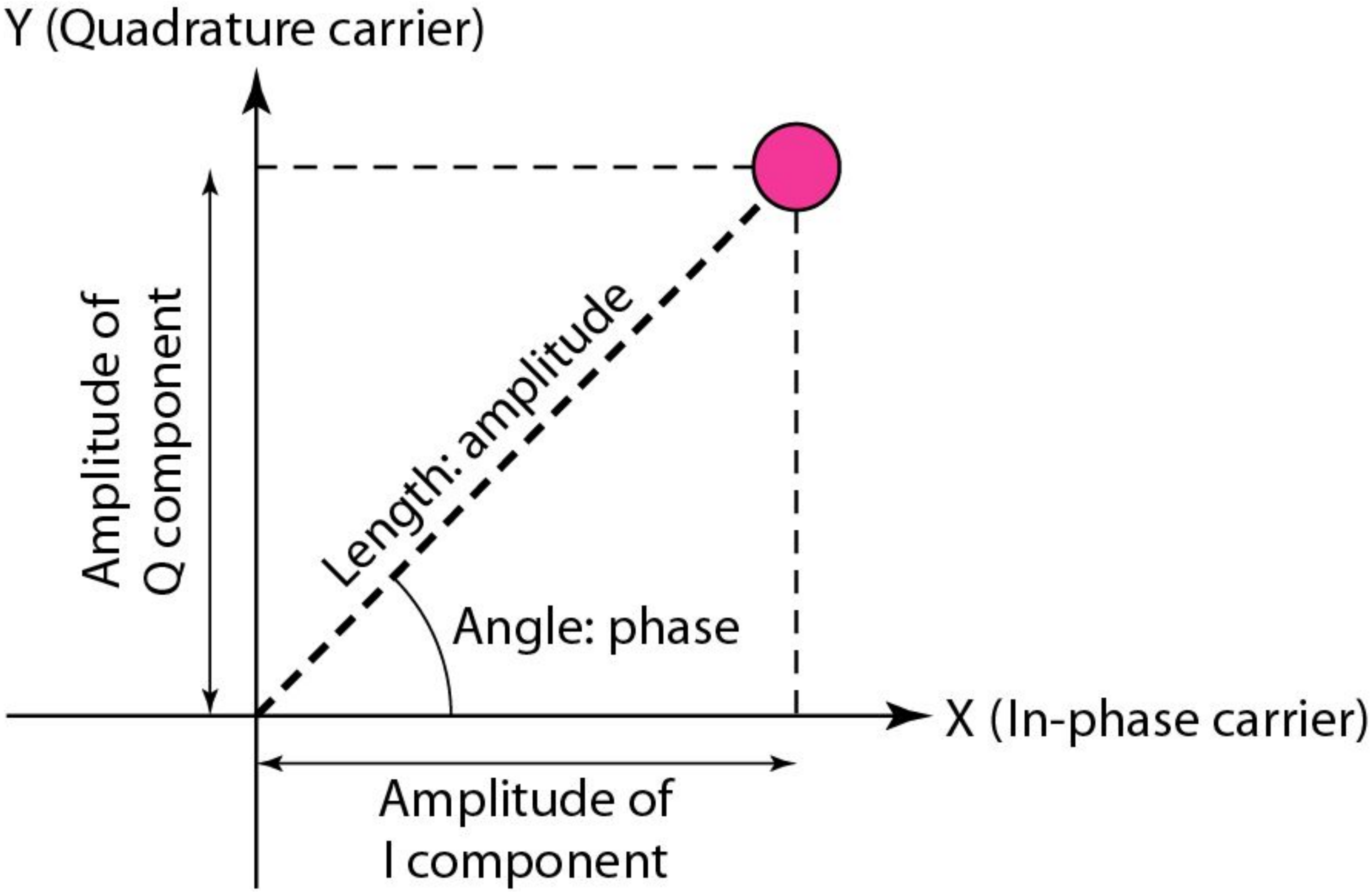
## *Solution*

- *For QPSK, 2 bits is carried by one signal element. This means that $r = 2$. So the signal rate (baud rate) is $S = N \times (1/r) = 6$ Mbaud. With a value of $d = 0$, we have $B = S = 6$ MHz.*

# Constellation Diagrams

- A constellation diagram helps us to define the amplitude and phase of a signal when we are using two carriers, one in quadrature of the other.
- The X-axis represents the in-phase carrier and the Y-axis represents quadrature carrier.

# Figure 12 *Concept of a constellation diagram*



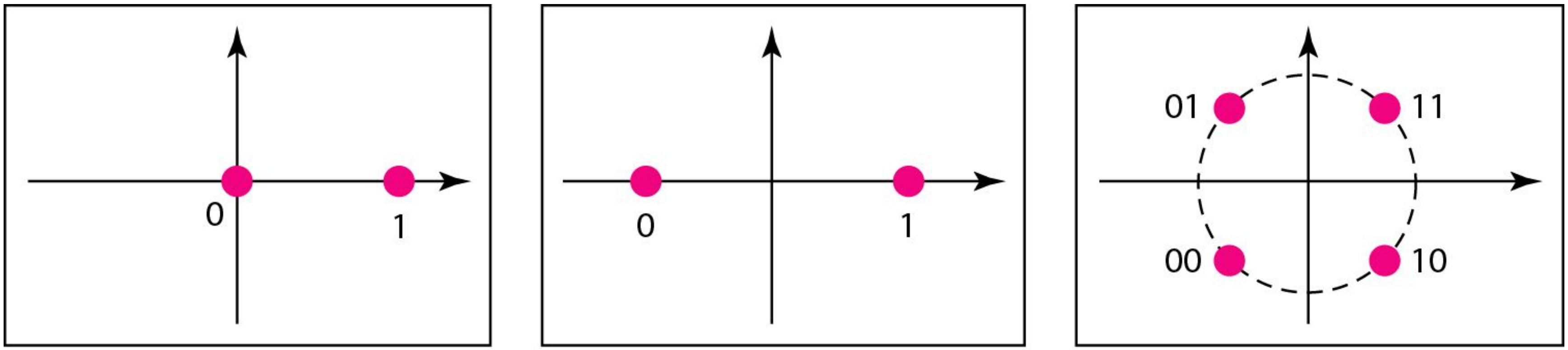
## *Example 8*

- *Show the constellation diagrams for an ASK (OOK), BPSK, and QPSK signals.*

## *Solution*

- *Figure 5.13 shows the three constellation diagrams.*

# Figure 13 *Three constellation diagrams*

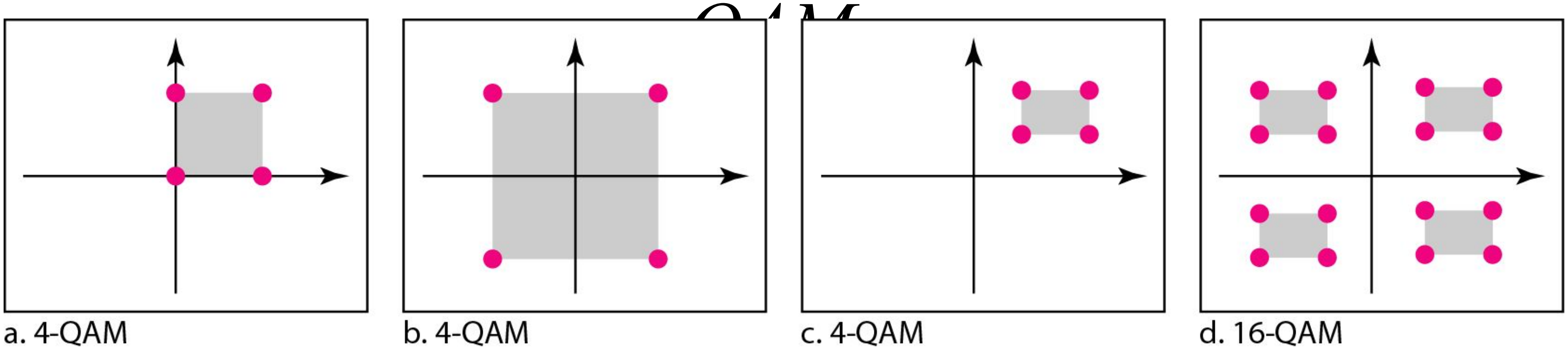



a. ASK (OOK)   b. BPSK   c. QPSK

# Quadrature amplitude modulation (QAM)

QAM is a combination of ASK and PSK.

**Figure 14** *Constellation diagrams for some QAM*



*Bandwidth for QAM*

- The minimum bandwidth required for QAM transmission is the same as that required for ASK and PSK transmission.
- QAM has the same advantages as PSK over ASK.

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# *ANALOG TO DIGITAL CONVERSION*

# ANALOG-TO-DIGITAL CONVERSION

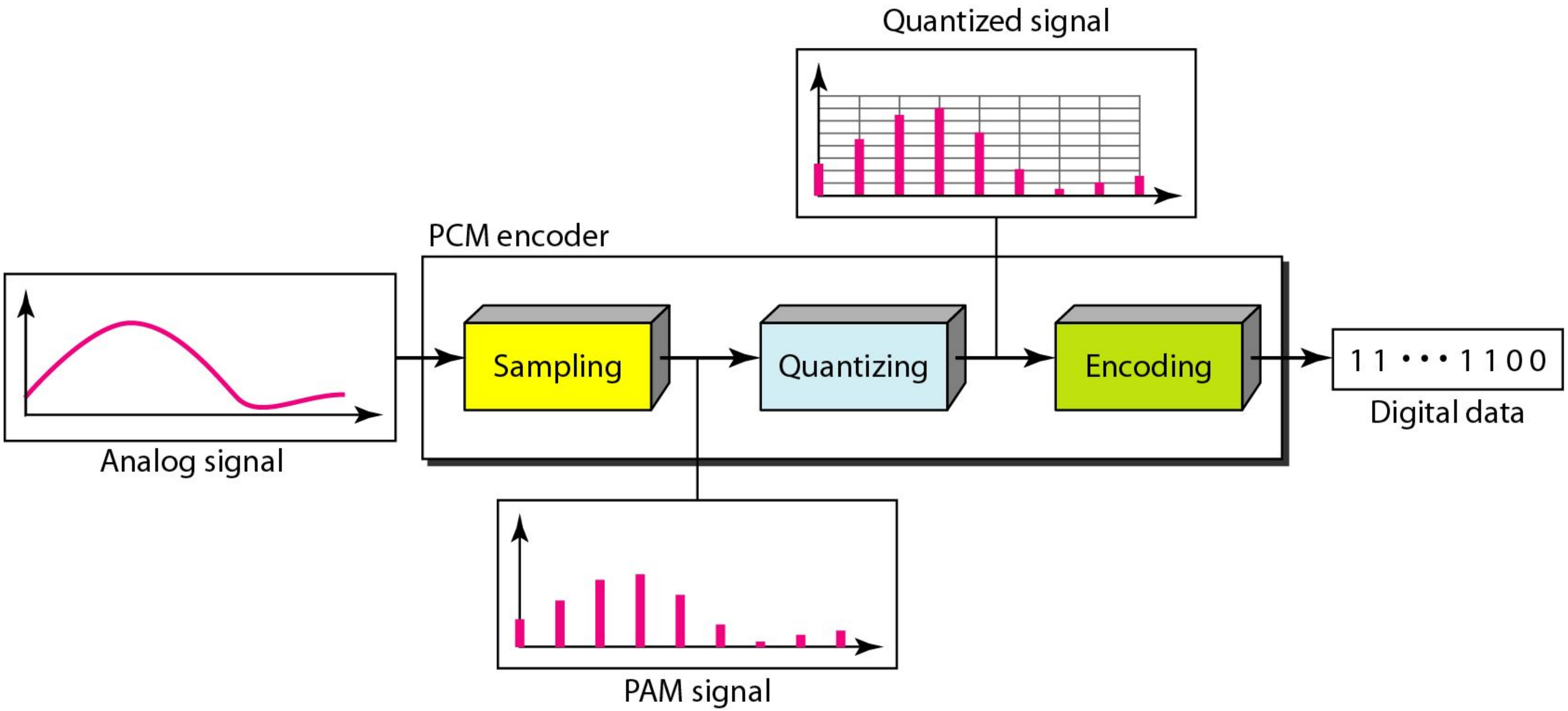
- ***A digital signal is superior to an analog signal because it is more robust to noise and can easily be recovered, corrected and amplified. For this reason, the tendency today is to change an analog signal to digital data. In this section we describe two techniques, pulse code modulation and delta 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, i.e. remove high frequency components that affect the signal shape.

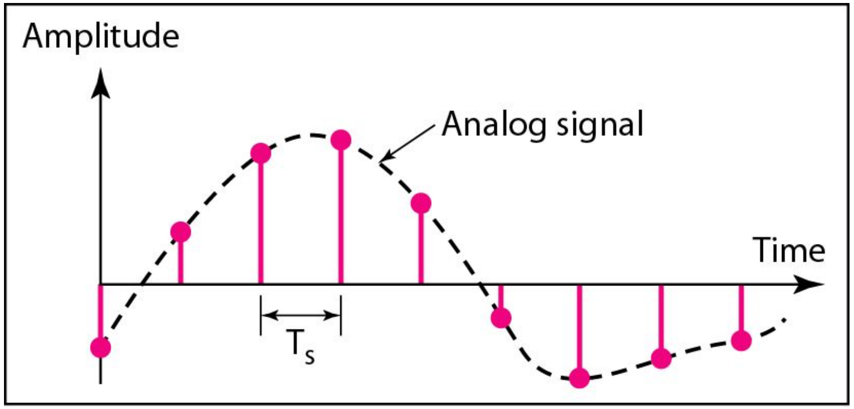
# Components of PCM encoder



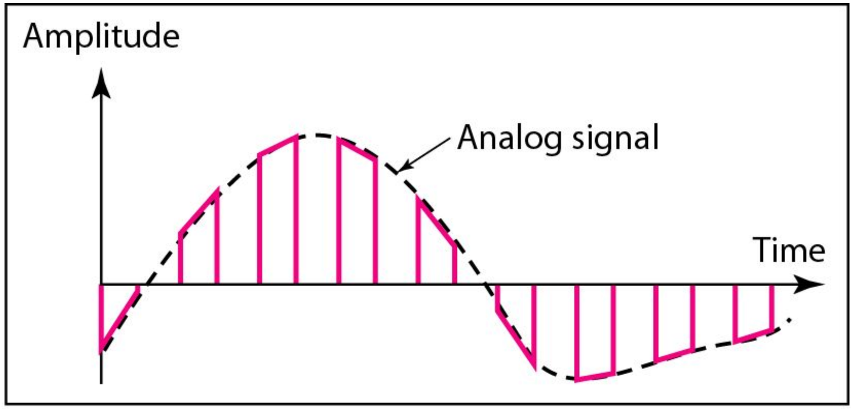
# Sampling

- 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

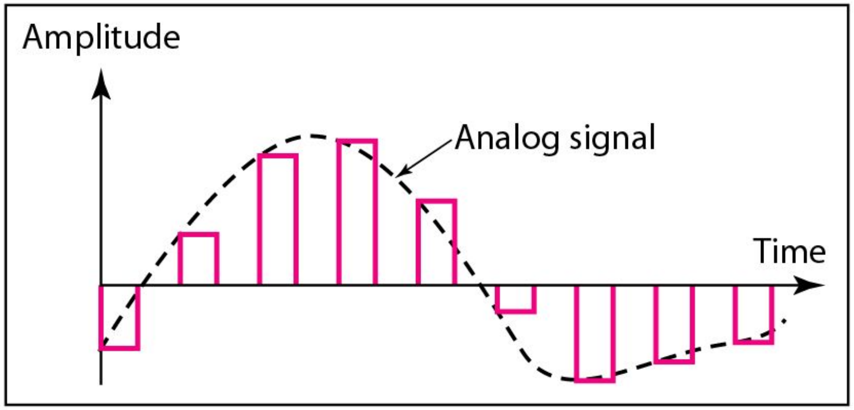
# Three different sampling methods for PCM




a. Ideal sampling




b. Natural sampling

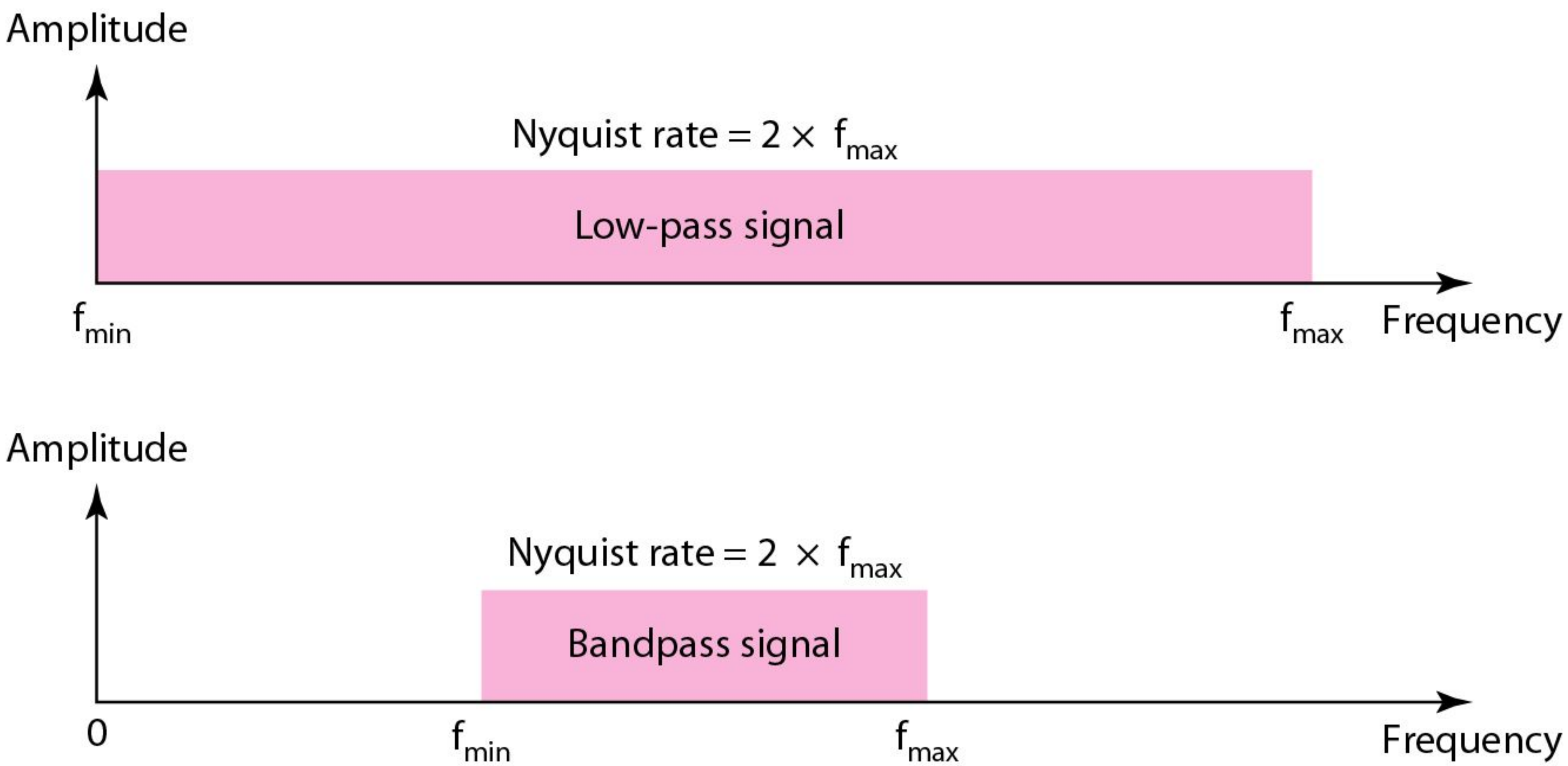



c. Flat-top sampling

# *Note*

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

# *Nyquist sampling rate for low-pass and bandpass signals*

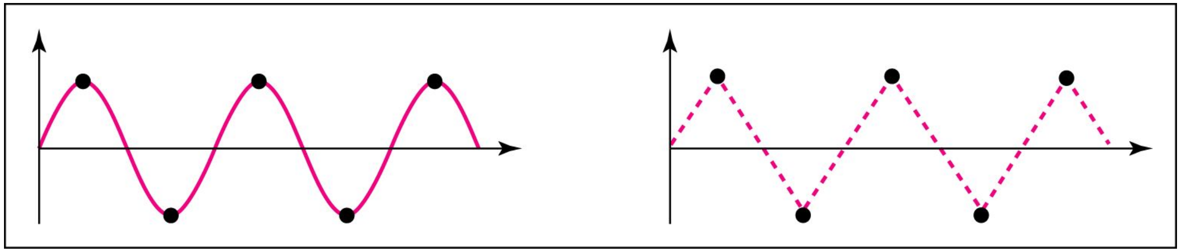


# Example 1

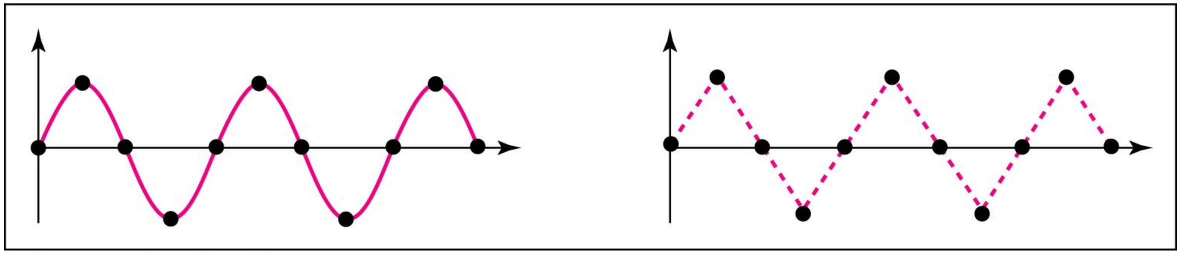
- *For an intuitive example of the Nyquist theorem, let us sample a simple sine wave at three sampling rates: $f_s = 4f$ (2 times the Nyquist rate), $f_s = 2f$ (Nyquist rate), and $f_s = f$ (one-half the Nyquist rate). Figure 4.24 shows the sampling and the subsequent recovery of the signal.*

- *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.*

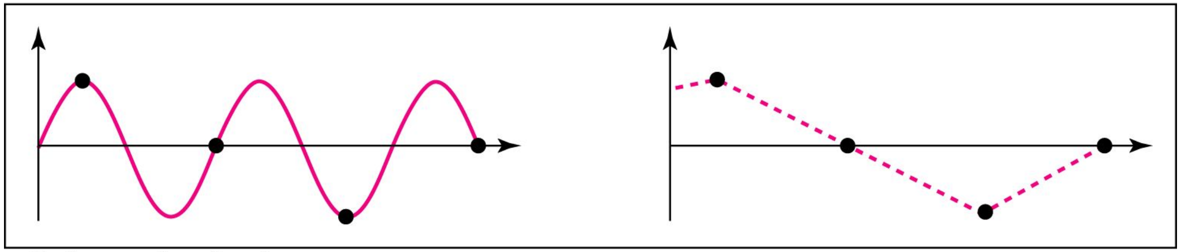
# *Recovery of a sampled sine wave for different sampling rates*



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



b. Oversampling: $f_s = 4f$

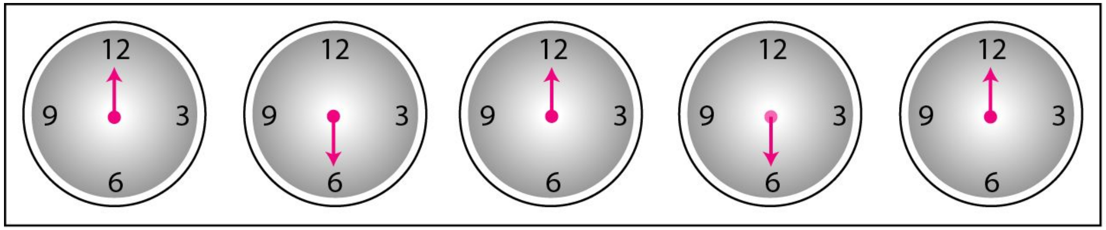


c. Undersampling: $f_s = f$

# Example 2

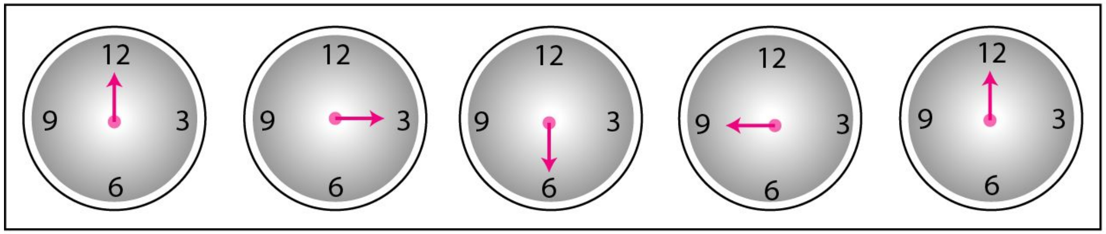
- ***Consider the revolution of a hand of a clock. The second hand of a clock has a period of 60 s. According to the Nyquist theorem, we need to sample the hand every 30 s ($T_s = T$ or $f_s = 2f$ ). In Figure 4.25a, the sample points, in order, are 12, 6, 12, 6, 12, and 6. The receiver of the samples 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.***

# *Sampling of a clock with only one hand*



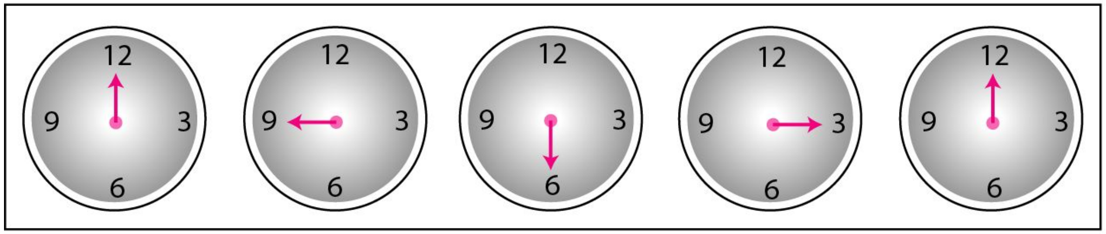

Samples can mean that the clock is moving either forward or backward.
(12-6-12-6-12)

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




Samples show clock is moving forward.
(12-3-6-9-12)

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




Samples show clock is moving backward.
(12-9-6-3-12)

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

# 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 $\Delta$.

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

# *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 amplitudes $V_{min}$=-20V and $V_{max}$=+20V.
- We want to use L=8 quantization levels.
- Zone width $\Delta$ = (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

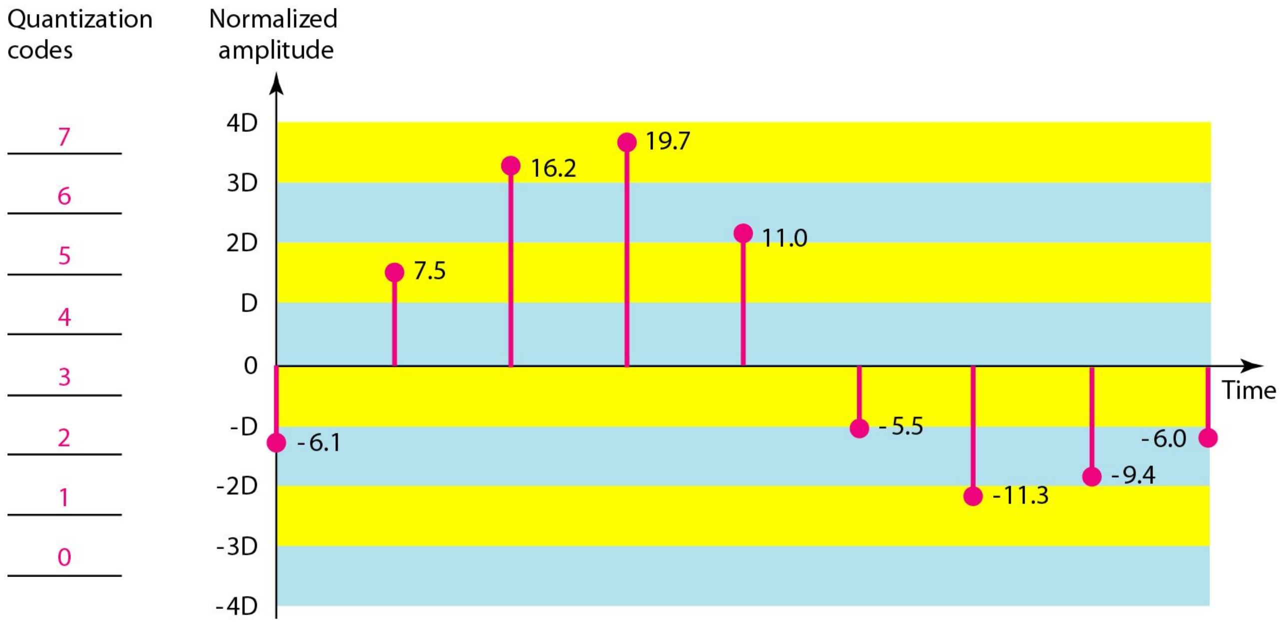
# *Assigning Codes to Zones*

- 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.

# *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 $\Delta$ 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 $\Delta$'s at lower amplitudes and larger $\Delta$'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

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

## Solution

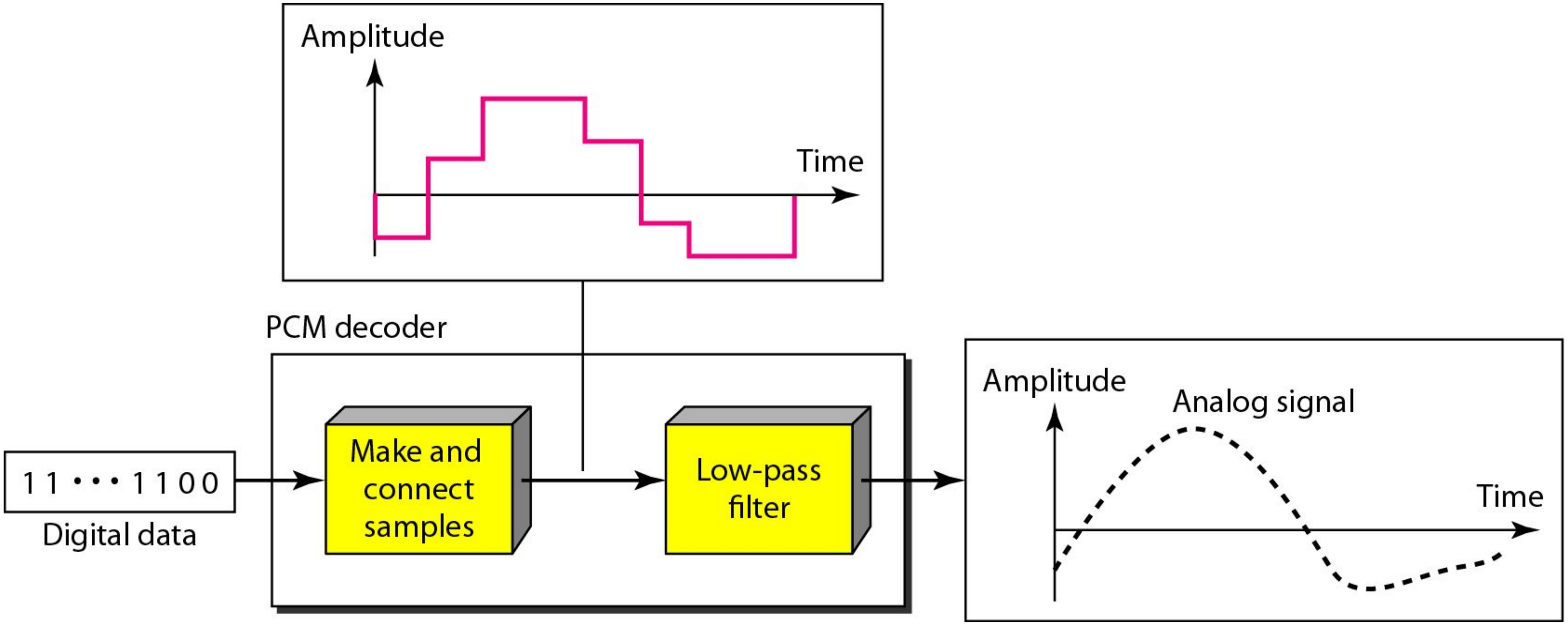
- ***The human voice normally contains frequencies from 0 to 4000 Hz. So the sampling rate and bit rate are calculated as follows:***

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

# 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.

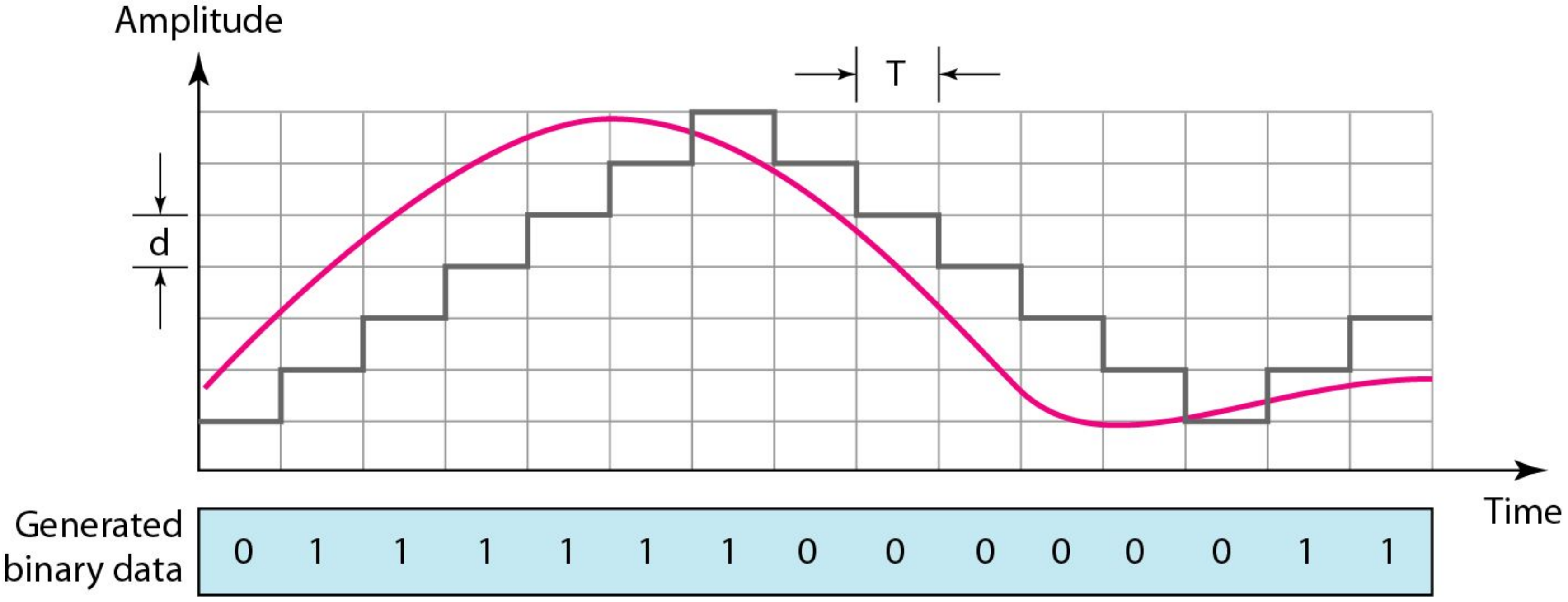
# Components of a PCM decoder



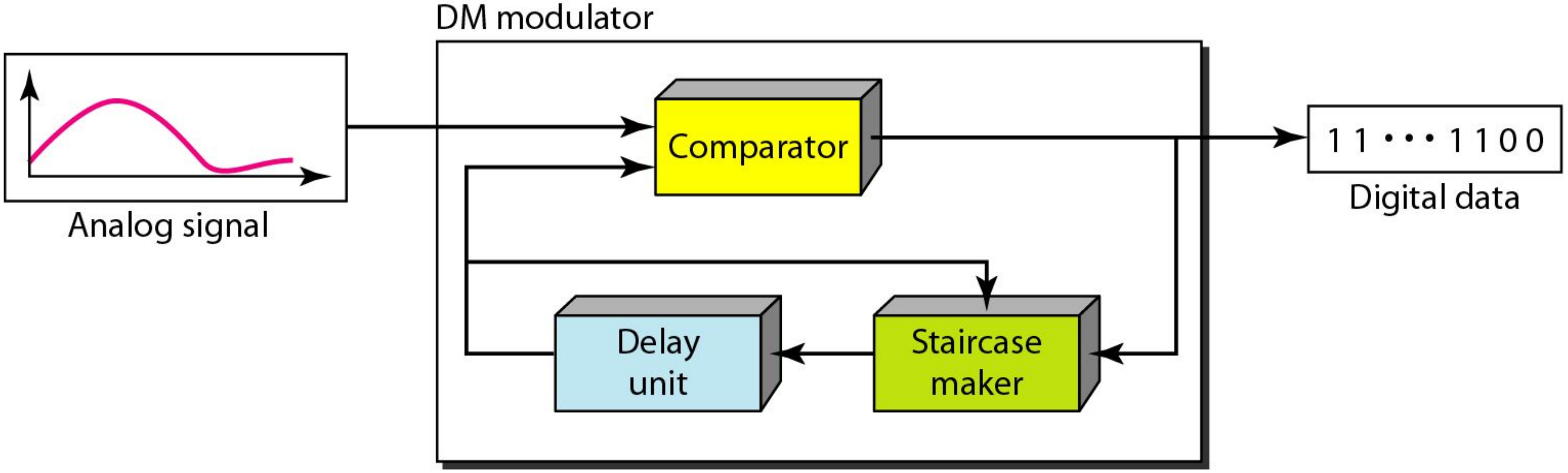
# *Delta Modulation*

- This scheme sends only the difference between pulses, if the pulse at time $t_{n+1}$ is higher in amplitude value than the pulse at time $t_n$, then a single bit, say a "1", is used to indicate the positive value.
- If the pulse is lower in value, resulting in a negative value, a "0" is used.
- This scheme works well for small changes in signal values between samples.
- If changes in amplitude are large, this will result in large errors.

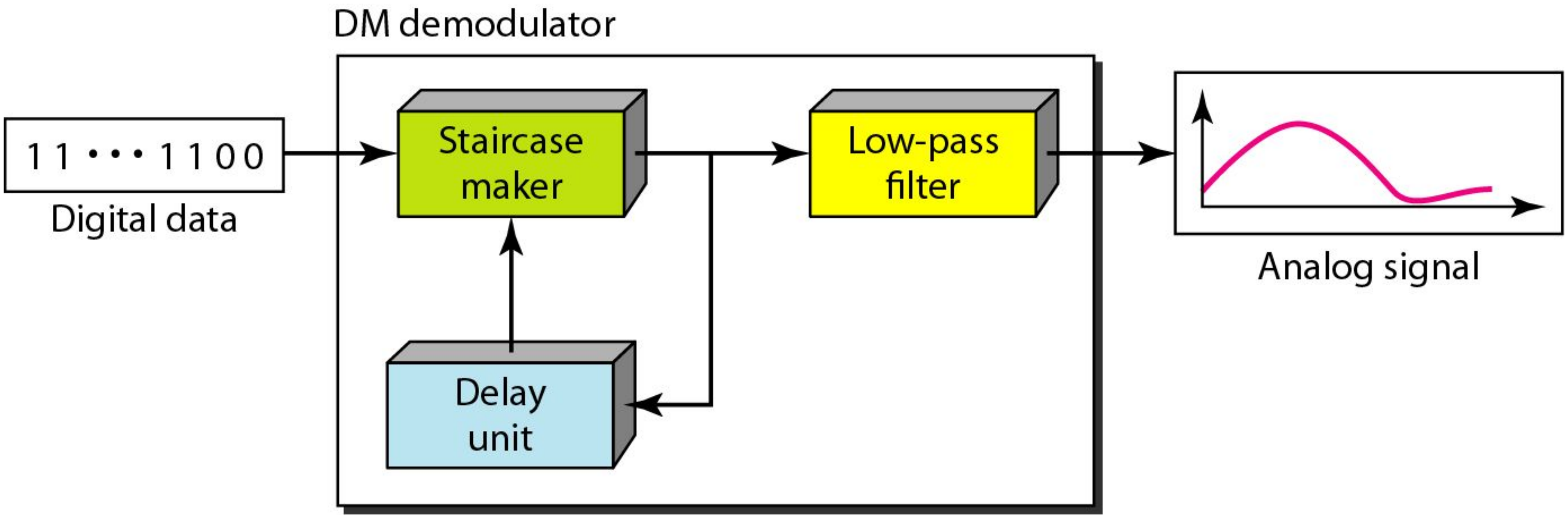
# The process of delta modulation



# *Delta modulation components*



# *Delta demodulation components*



# *Delta PCM (DPCM)*

- Instead of using one bit to indicate positive and negative differences, we can use more bits -> quantization of the difference.
- Each bit code is used to represent the value of the difference.
- The more bits the more levels -> the higher the accuracy.

# Multiplexing : Sharing a Medium

# Introduction

Under the simplest conditions, a medium can carry only one signal at any moment in time.

For multiple signals to share one medium, the medium must somehow be divided, giving each signal a portion of the total bandwidth.

The current techniques that can accomplish this include

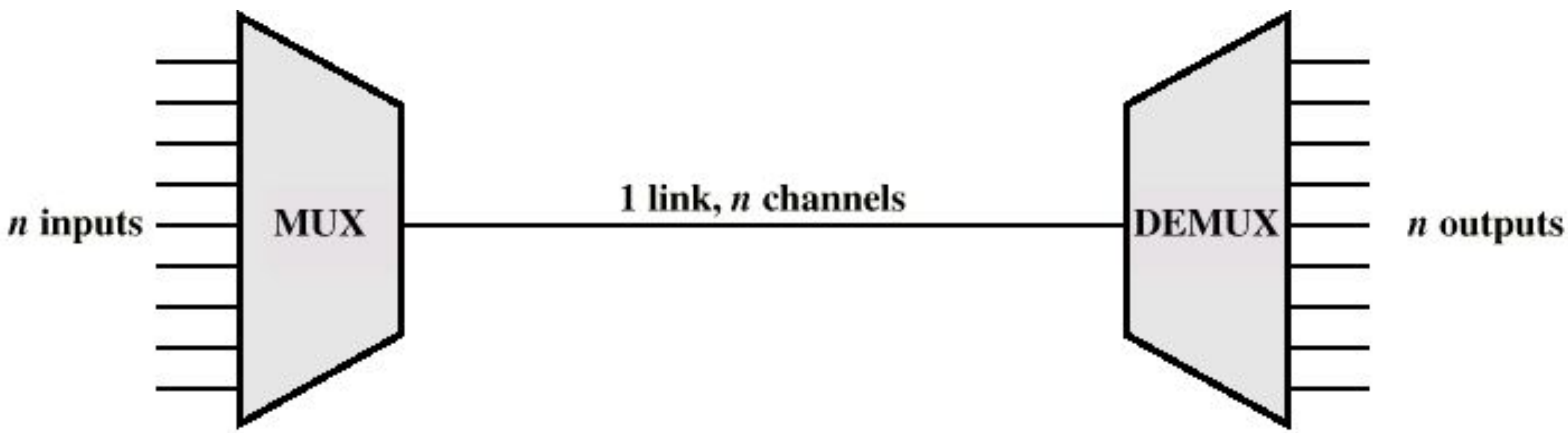
- frequency division multiplexing (FDM)
- time division multiplexing (TDM)
    - Synchronous vs statistical
- wavelength division multiplexing (WDM)
- code division multiplexing (CDM)

# Multiplexing

Multiplexor (MUX)

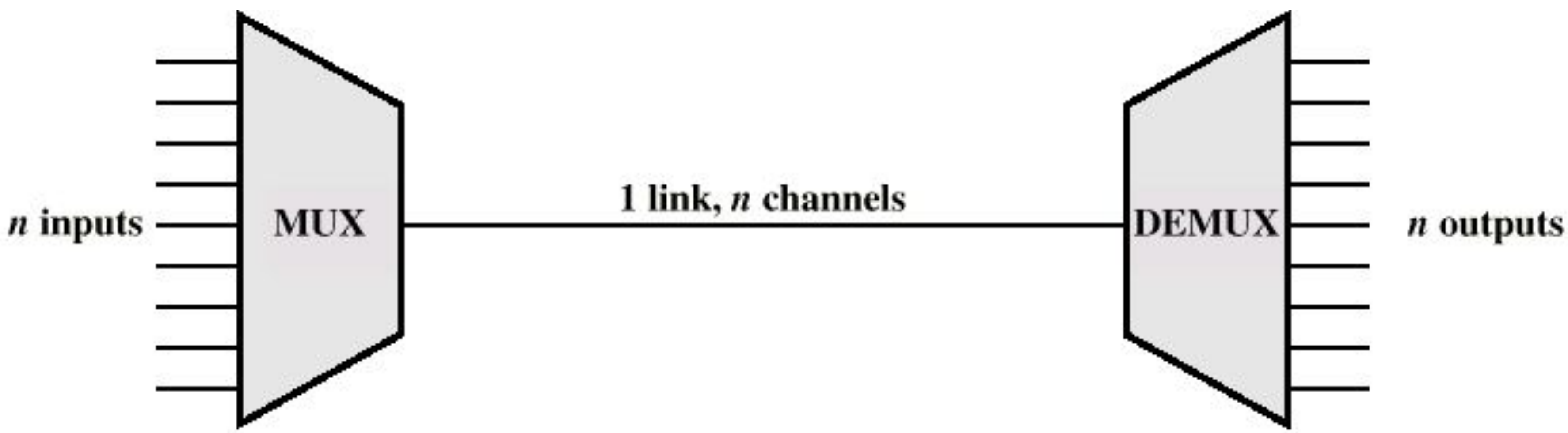
Demultiplexor (DEMUX)

- Sometimes just called a MUX



# Multiplexing

- Two or more simultaneous transmissions on a single circuit.
  - Transparent to end user.
- *Multiplexing costs less*.



# Frequency Division Multiplexing

Assignment of non-overlapping frequency ranges to each "user" or signal on a medium. Thus, all signals are transmitted at the same time, each using different frequencies.

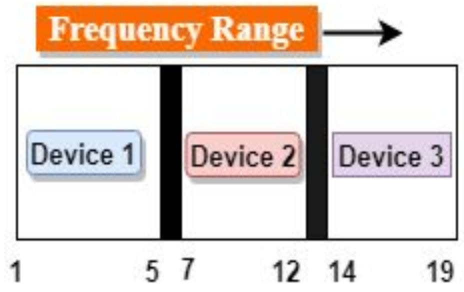
A multiplexor accepts inputs and assigns frequencies to each device.

The multiplexor is attached to a high-speed communications line.

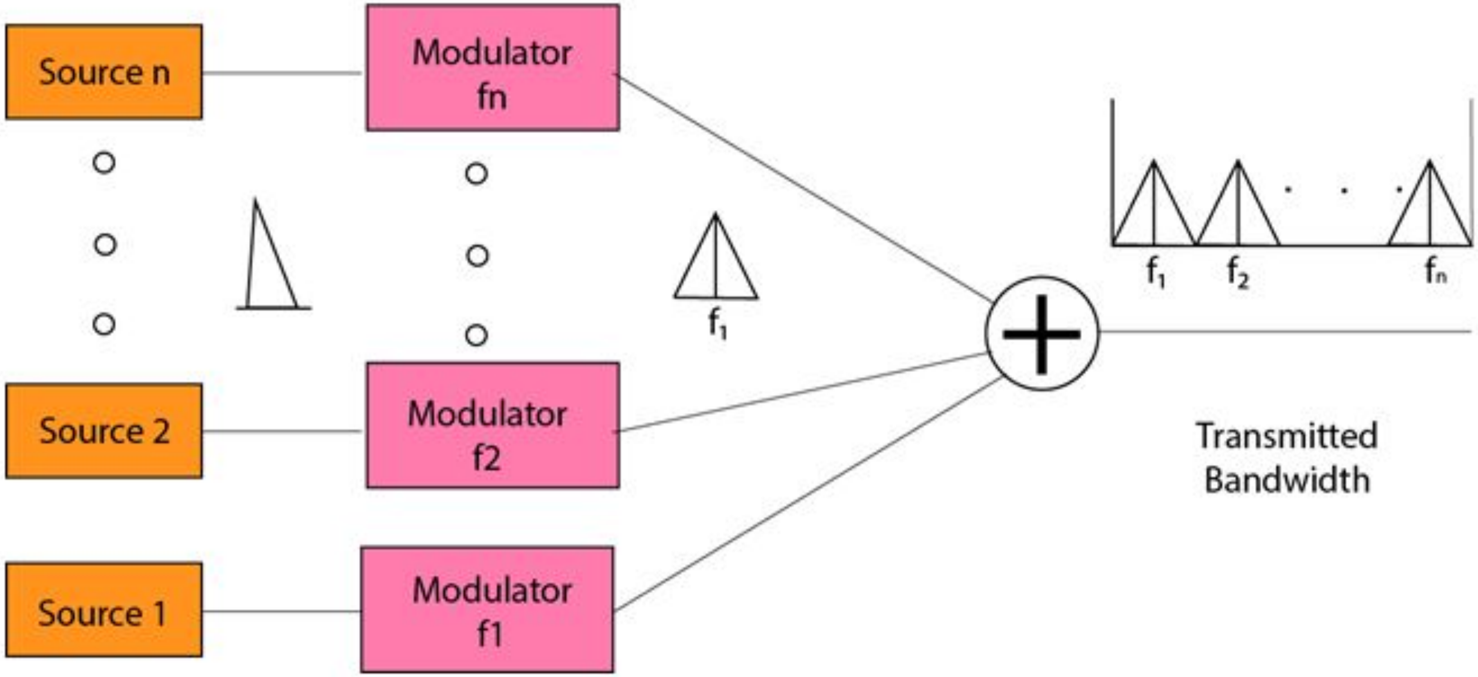
A corresponding multiplexor, or de-multiplexor, is on the end of the high-speed line and separates the multiplexed signals.

# Frequency-division Multiplexing (FDM)

- It is an analog technique.
- **Frequency Division Multiplexing** is a technique in which the available bandwidth of a single transmission medium is subdivided into several channels.




- In the above diagram, a single transmission medium is subdivided into several frequency channels, and each frequency channel is given to different devices. Device 1 has a frequency channel of range from 1 to 5.
- The input signals are translated into frequency bands by using modulation techniques, and they are combined by a multiplexer to form a composite signal.
- The main aim of the FDM is to subdivide the available bandwidth into different frequency channels and allocate them to different devices.
- Using the modulation technique, the input signals are transmitted into frequency bands and then combined to form a composite signal.
- The carriers which are used for modulating the signals are known as **sub-carriers**. They are represented as f1,f2..fn.
- **FDM** is mainly used in radio broadcasts and TV networks.




**Advantages Of FDM:**

- FDM is used for analog signals.
- FDM process is very simple and easy modulation.
- A Large number of signals can be sent through an FDM simultaneously.
- It does not require any synchronization between sender and receiver.

# Frequency Division Multiplexing

Analog signaling is used to transmits the signals.

Broadcast radio and television, cable television, and the AMPS cellular phone systems use frequency division multiplexing.

This technique is the oldest multiplexing technique.

Since it involves analog signaling, it is more susceptible to noise.

*Assignment of frequencies for cable television channels*

| | Channel | Frequency in MHz |
|---|---|---|
| **Low-Band VHF and Cable** | 2 | 54–60 |
| | 3 | 60–66 |
| | 4 | 66–72 |
| | 5 | 76–82 |
| | 6 | 82–88 |
| **Mid-Band Cable** | 95 | 90–96 |
| | 96 | 96–102 |
| | 97 | 102–108 |
| | 98 | 108–114 |
| | 99 | 114–120 |
| | 14 | 120–126 |
| | 15 | 126–132 |
| | 16 | 132–138 |
| | 17 | 138–144 |
| | 18 | 144–150 |
| | 19 | 150–156 |
| | 20 | 156–162 |
| | 21 | 162–168 |
| | 22 | 168–174 |
| **High-Band VHF and Cable** | 7 | 174–180 |
| | 8 | 180–186 |
| | 9 | 186–192 |
| | 10 | 192–198 |
| | 11 | 198–204 |
| | 12 | 204–210 |
| | 13 | 210–216 |

# Time Division Multiplexing

- It is a digital technique.
- In Frequency Division Multiplexing Technique, all signals operate at the same time with different frequency, but in case of Time Division Multiplexing technique, all signals operate at the same frequency with different time.
- In **Time Division Multiplexing technique**, the total time available in the channel is distributed among different users. Therefore, each user is allocated with different time interval known as a Time slot at which data is to be transmitted by the sender.
- A user takes control of the channel for a fixed amount of time.
- In Time Division Multiplexing technique, data is not transmitted simultaneously rather the data is transmitted one-by-one.
- In TDM, the signal is transmitted in the form of frames. Frames contain a cycle of time slots in which each frame contains one or more slots dedicated to each user.
- It can be used to multiplex both digital and analog signals but mainly used to multiplex digital signals.

**There are two types of TDM:**

- Synchronous TDM
- Asynchronous TDM

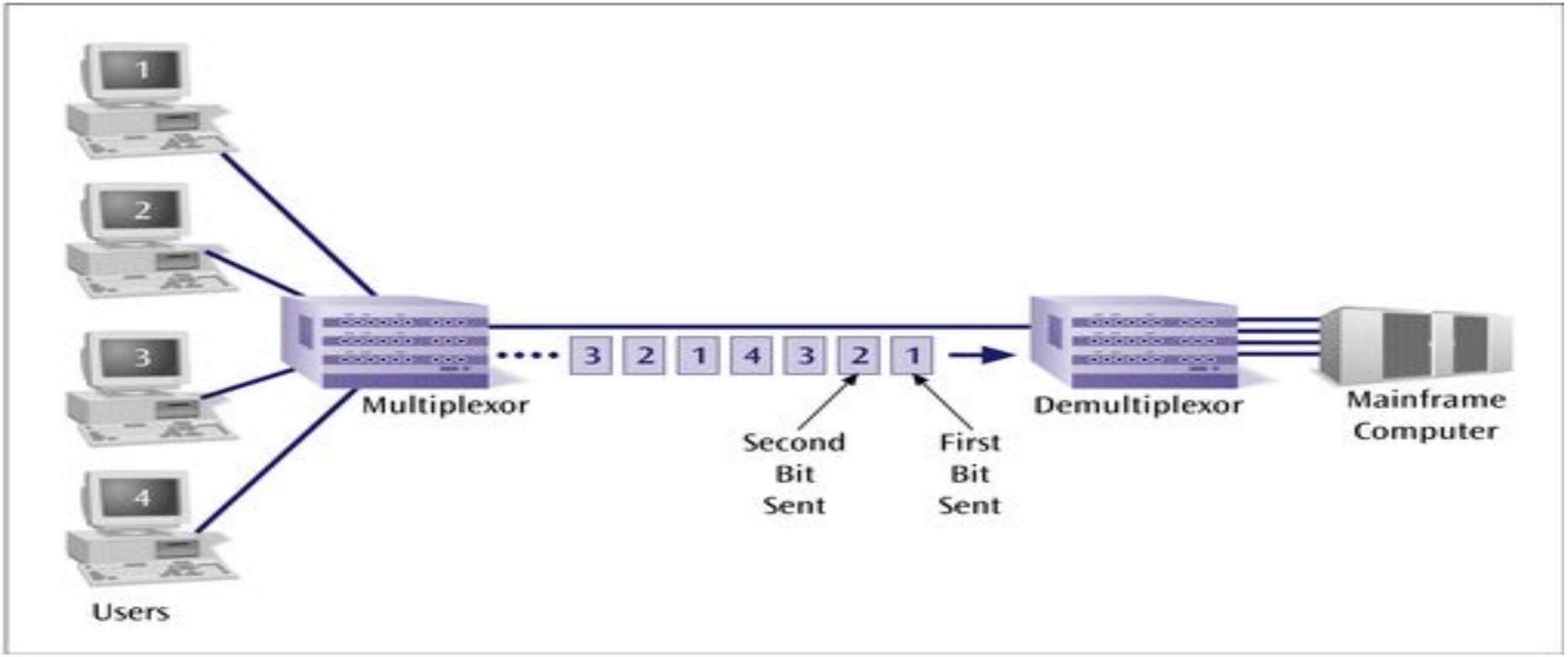
# Synchronous Time Division Multiplexing

The original time division multiplexing.

The multiplexor accepts input from attached devices in a round-robin fashion and transmit the data in a never ending pattern.

T-1 and ISDN telephone lines are common examples of synchronous time division multiplexing.

# Sample Output Stream generated by a Synchronous Time Division Multiplexing



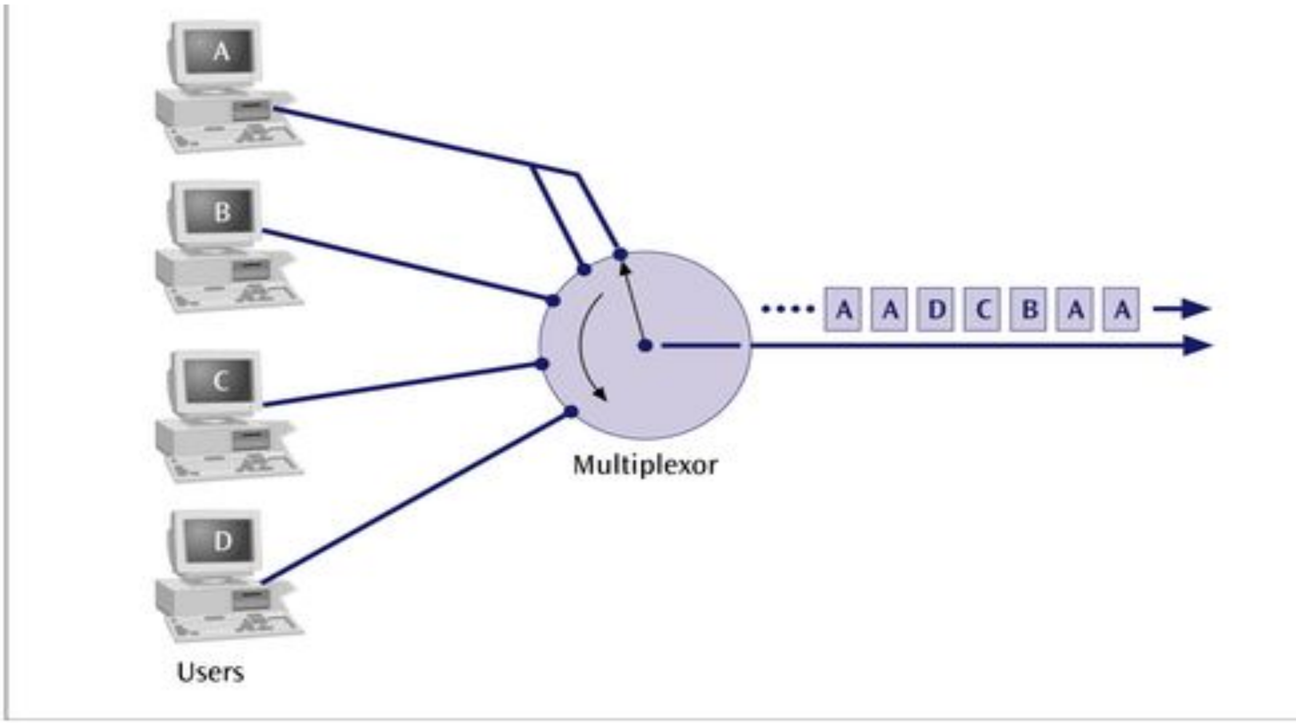
# Synchronous Time Division Multiplexing

If one device generates data at a faster rate than other devices, then the multiplexor must either sample the incoming data stream from that device more often than it samples the other devices, or buffer the faster incoming stream.

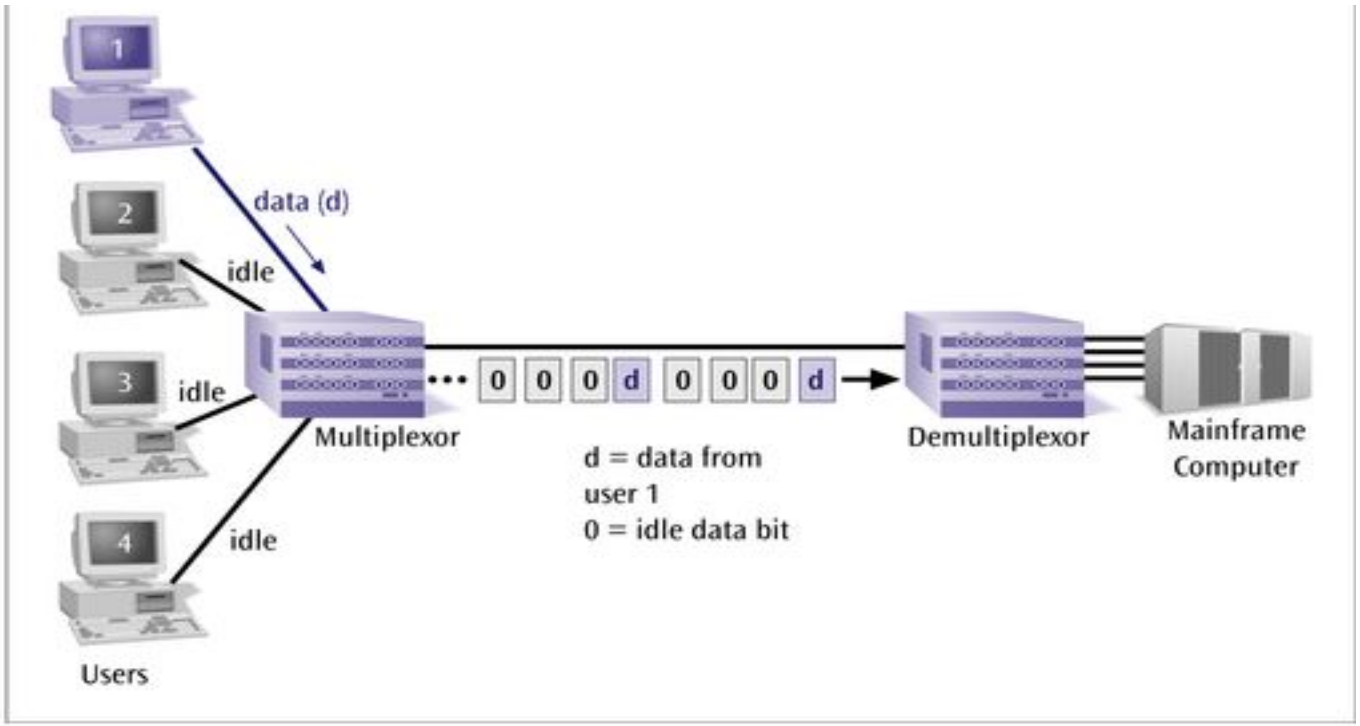
If a device has nothing to transmit, the multiplexor must still insert a piece of data from that device into the multiplexed stream.

# Example

*A synchronous time division multiplexor system which samples device A twice as fast as the other devices*

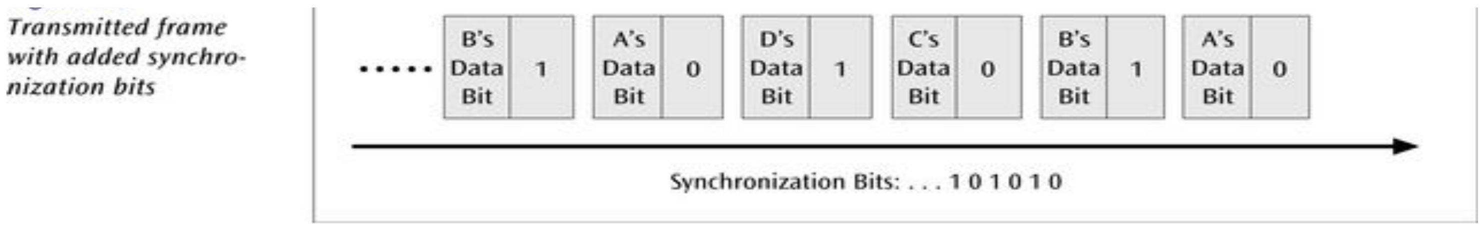


*Multiplexor transmission stream with only one input device transmitting data*



# Synchronous time division multiplexing

So that the receiver may stay synchronized with the incoming data stream, the transmitting multiplexor can insert alternating 1s and 0s into the data stream.

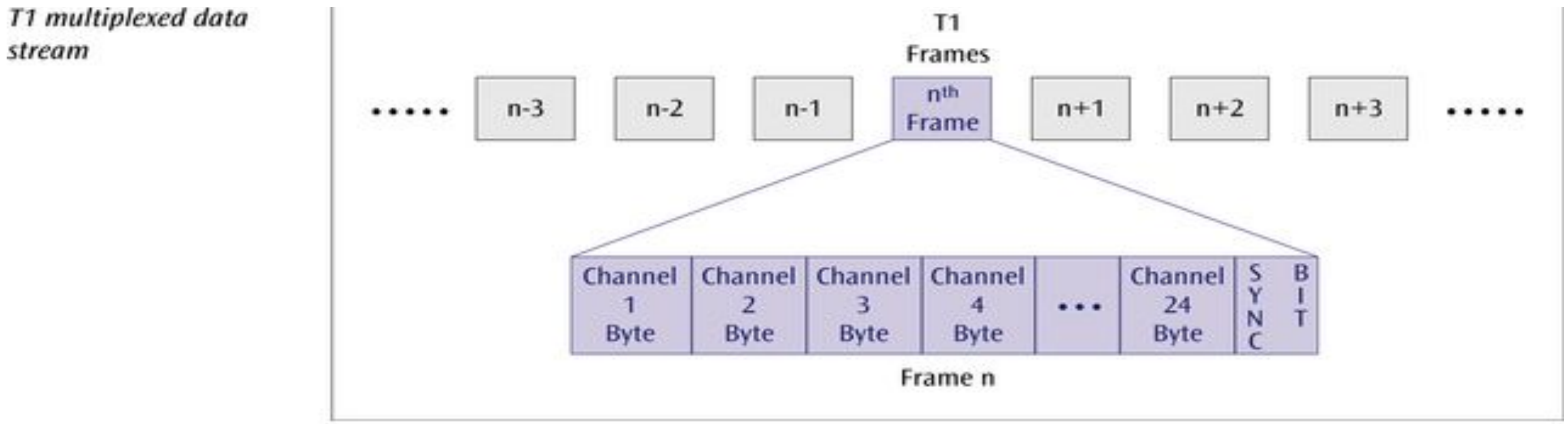


# Synchronous Time Division Multiplexing

Three types popular today:

- T-1 multiplexing (the classic)
- ISDN multiplexing
- SONET (Synchronous Optical NETwork)

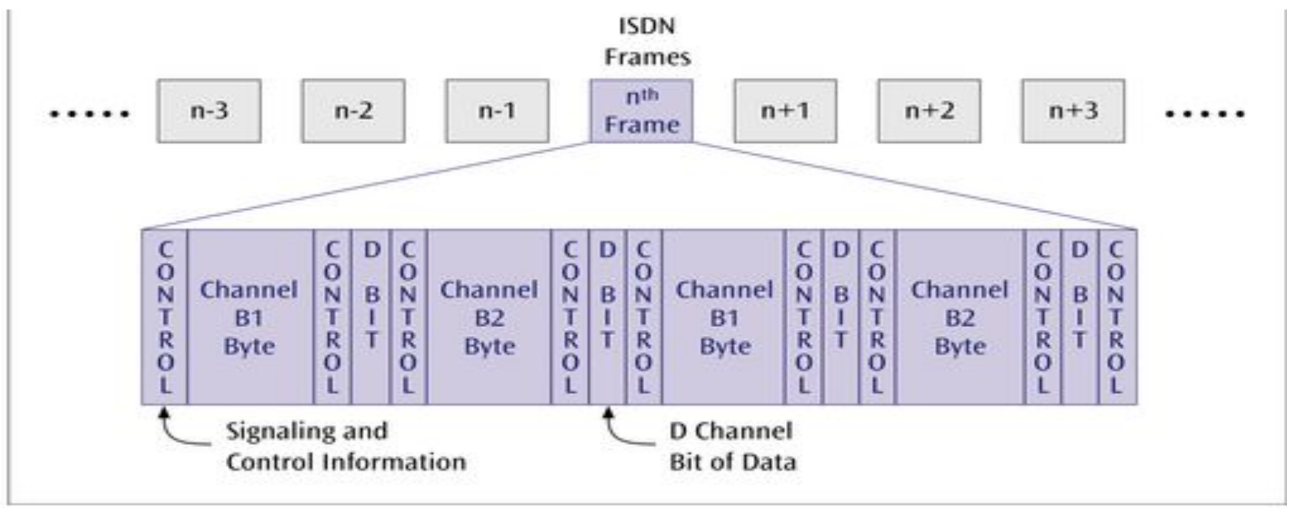
The T1 (1.54 Mbps) multiplexor stream is a *continuous* series of frames of both digitized data and voice channels.



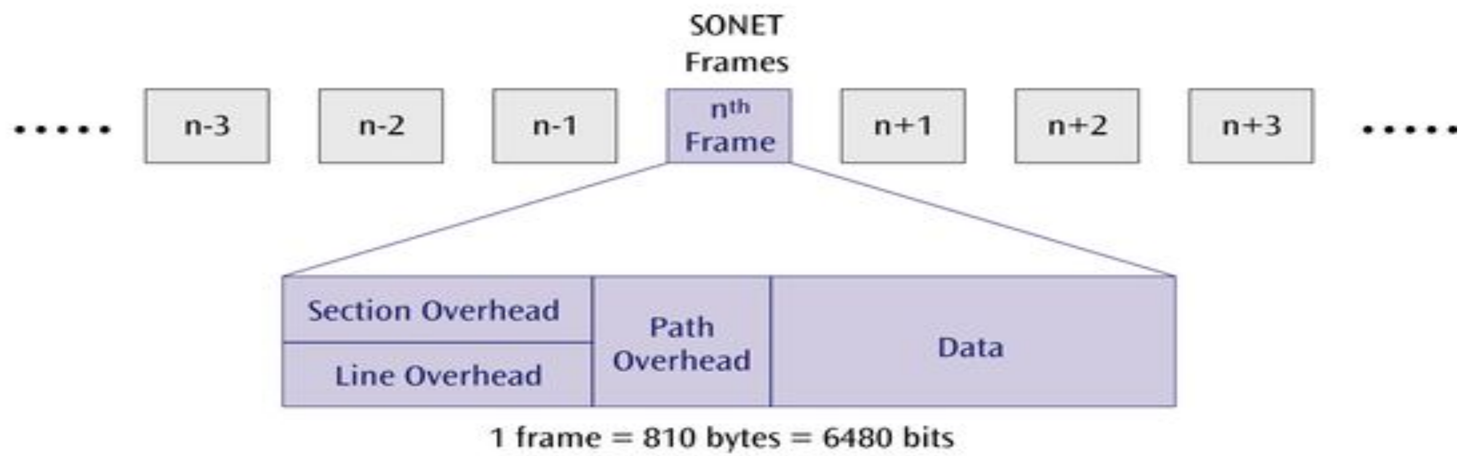

24 separate 64Kbps channels

The ISDN multiplexor stream is also a continuous stream of frames. Each frame contains various control and sync info.

*ISDN frame layout showing B channel bits and signaling control information bits*



# SONET – massive data rates




SONET STS-1 Frame Layout

# Synchronous TDM

- Very popular
- Line will require as much bandwidth as all the bandwidths of the sources

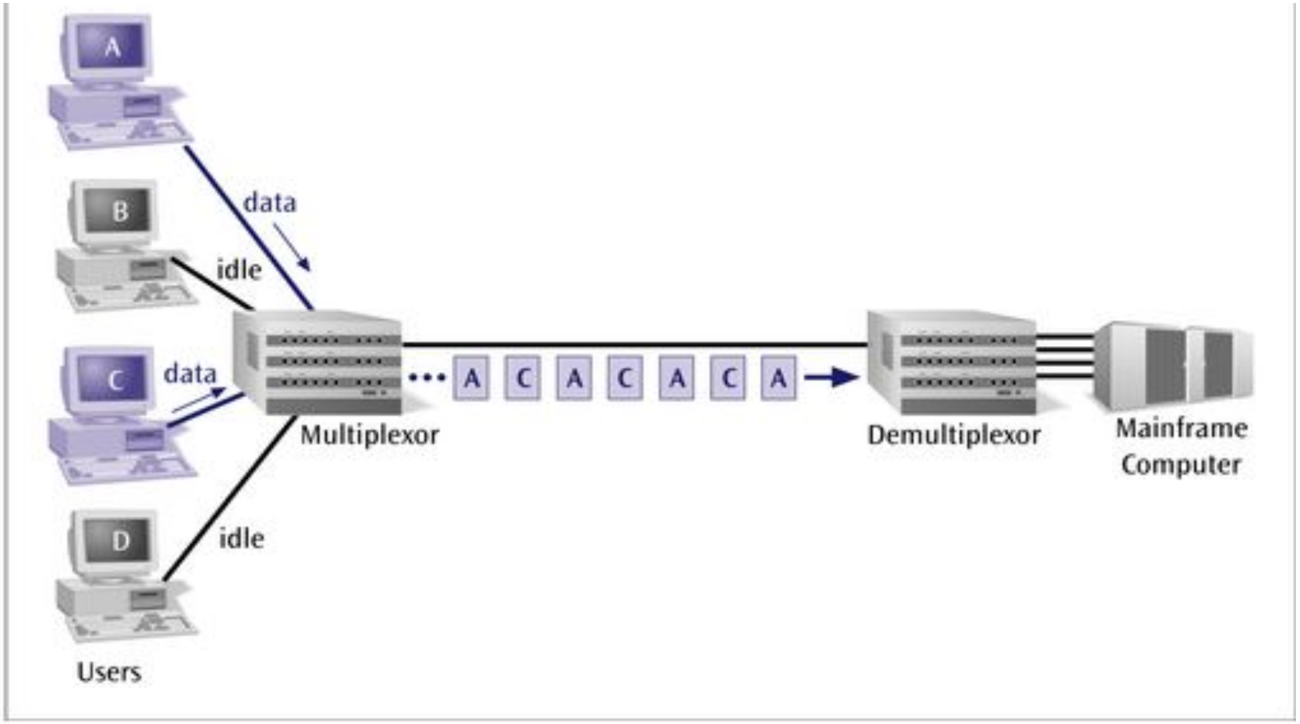
# Statistical Time Division Multiplexing

A statistical multiplexor transmits only the data from active workstations (*or why work when you don't have to*).

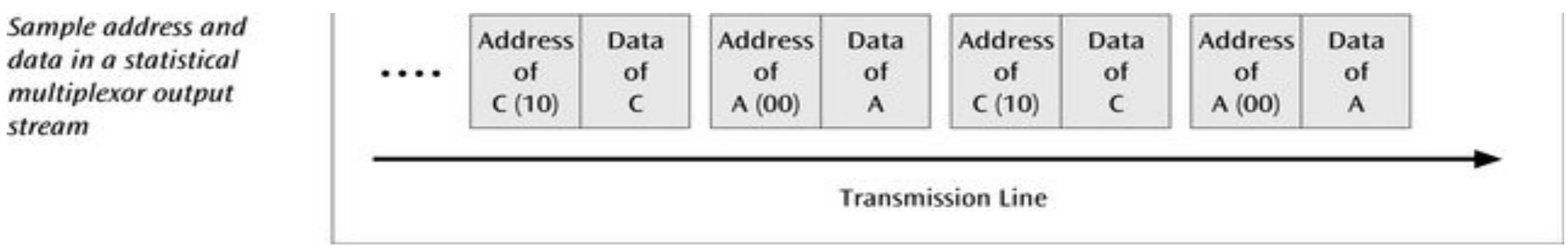
If a workstation is not active, no space is wasted on the multiplexed stream.

A statistical multiplexor accepts the incoming data streams and creates a frame containing only the data to be transmitted.

*Two stations out of four transmitting via a statistical multiplexor*

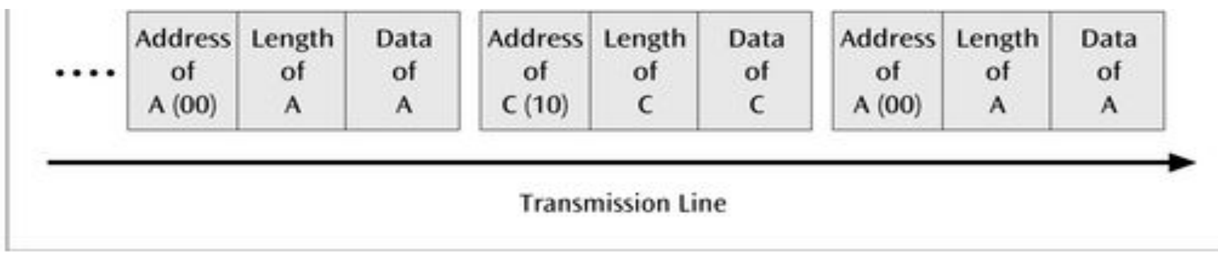


To identify each piece of data, an address is included.



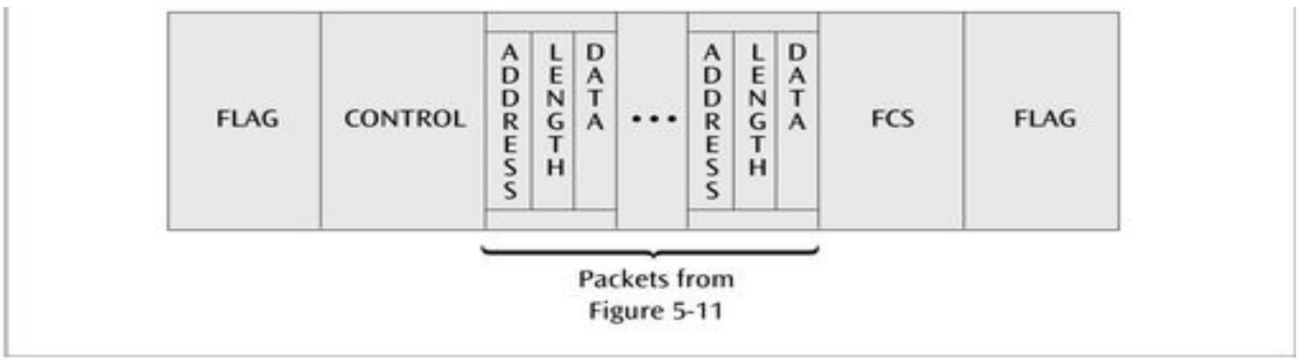
If the data is of variable size, a length is also included.

*Packets of address and data fields in a statistical multiplexor output stream*



More precisely, the transmitted frame contains a collection of data groups.

Frame layout for the information packet transferred between statistical multiplexors



# Statistical Time Division Multiplexing

A statistical multiplexor does not require a line over as high a speed line as synchronous time division multiplexing since STDM does not assume all sources will transmit all of the time!

Good for low bandwidth lines (used for LANs)

*Much more efficient use of bandwidth!*

# Wavelength Division Multiplexing (WDM)

SRM INSTITUTE OF SCIENCE & TECHNOLOGY
(Deemed to be University u/s 3 of UGC Act, 1956)

Give each message a different wavelength (frequency)

Easy to do with fiber optics and optical sources

# Dense Wavelength Division Multiplexing (DWDM)

Dense wavelength division multiplexing is often called just wavelength division multiplexing

Dense wavelength division multiplexing multiplexes multiple data streams onto a single fiber optic line.

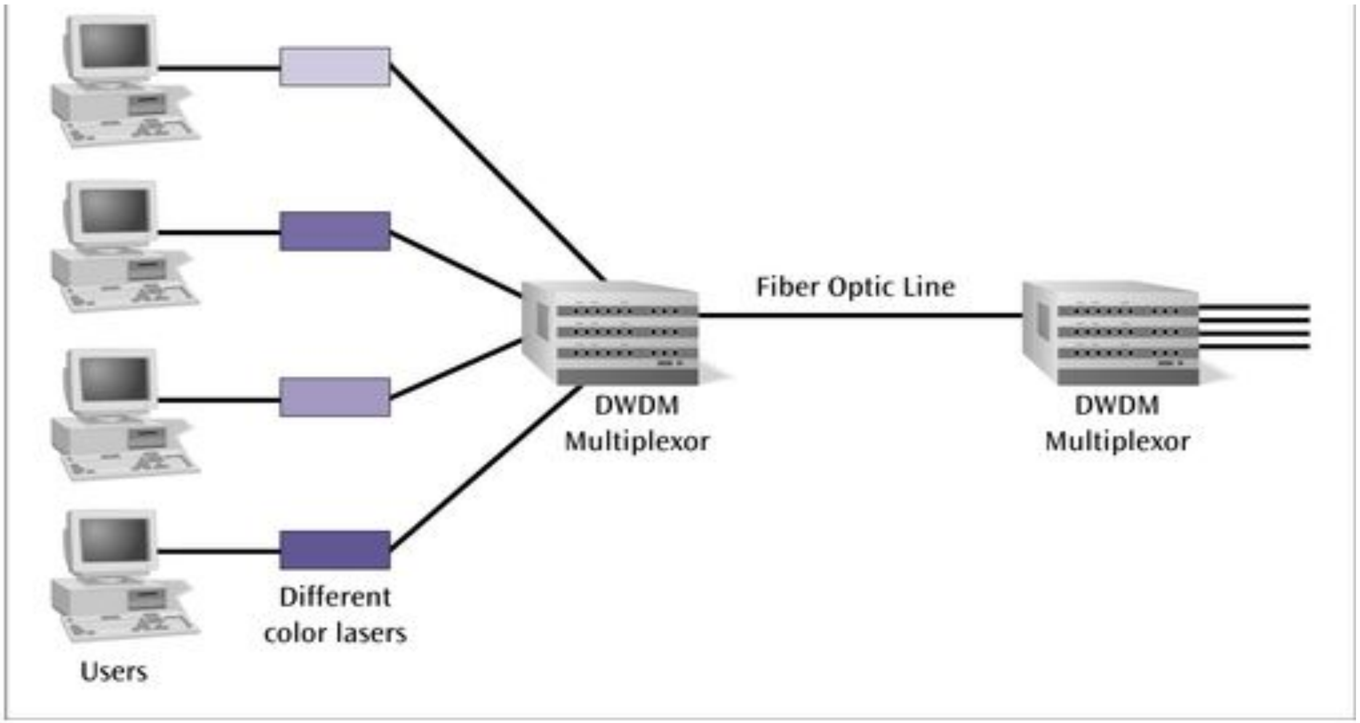
Different wavelength lasers (called lambdas) transmit the multiple signals.

Each signal carried on the fiber can be transmitted at a different rate from the other signals.

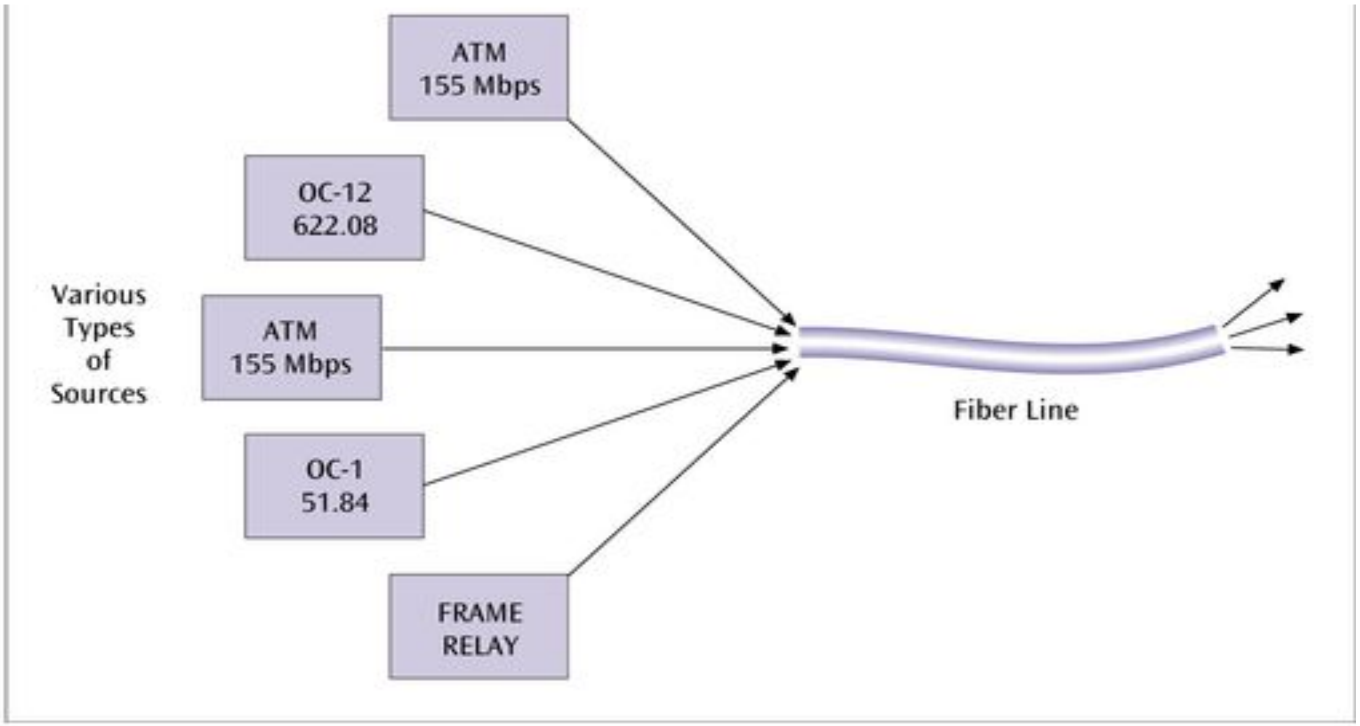
Dense wavelength division multiplexing combines many (30, 40, 50, 60, more?) onto one fiber.

# Data Signals Transmitted

*Multiple lasers transmitting data signals down a single fiber optic line*



*Fiber optic line using dense wavelength division multiplexing and supporting multiple-speed transmissions*



# Code Division Multiplexing (CDM)

Old but now new method

Also known as code division multiple access (CDMA)

An advanced technique that allows multiple devices to transmit on the *same* frequencies at the *same* time using different codes

Used for mobile communications

# Code Division Multiplexing

An advanced technique that allows multiple devices to transmit on the *same* frequencies at the *same* time.

Each mobile device is assigned a unique 64-bit code (chip spreading code)

To send a binary 1, mobile device transmits the unique code

To send a binary 0, mobile device transmits the inverse of code

## Code Division Multiplexing

Receiver gets summed signal, multiplies it by receiver code, adds up the resulting values

Interprets as a binary 1 if sum is near +64

Interprets as a binary 0 if sum is near –64

*Advantages and disadvantages of multiplexing techniques*

| Multiplexing Technique | Advantages | Disadvantages |
|---|---|---|
| **Frequency Division Multiplexing** | Simple | Analog signals only |
| | Popular with radio, TV, cable TV<br>Relatively inexpensive<br>All the receivers, such as cellular telephones, do not need to be at the same location | Limited by frequency ranges |
| **Synchronous Time Division Multiplexing** | Digital signals<br>Relatively simple<br>Commonly used with T-1 and ISDN | Wastes bandwidth |
| **Statistical Time Division Multiplexing** | More efficient use of bandwidth<br>Packets can be various sizes<br>Frame can contain control and error information | More complex than synchronous time division multiplexing |
| **Dense Wavelength Division Multiplexing** | Very high capacities over fiber<br>Scalable<br>Signals can have varying speeds | Cost<br>Complexity |
| **Code Division Multiplexing** | Large capacities<br>Scalable | Complexity |

## Business Multiplexing In Action

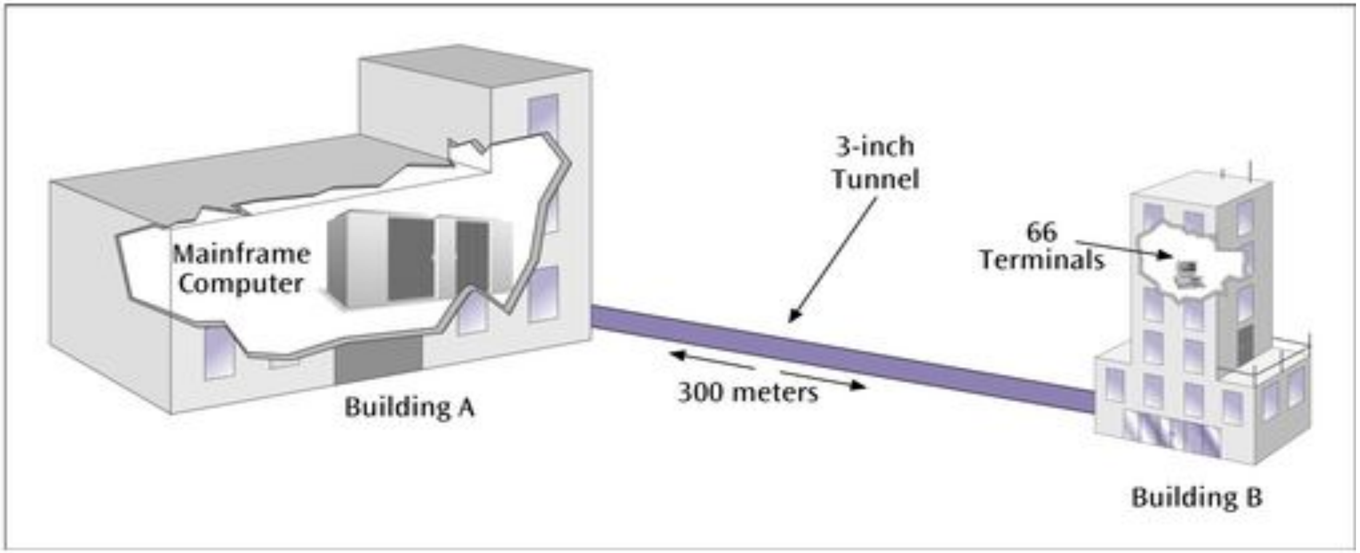
XYZ Corporation has two buildings separated by a distance of 300 meters.

A 3-inch diameter tunnel extends underground between the two buildings.

Building A has a mainframe computer and Building B has 66 terminals.

List some efficient techniques to link the two buildings.

**Figure 5-15**
*Buildings A and B and the 3-inch tunnel connecting the buildings*



# Possible Solutions

Connect each terminal to the mainframe computer using separate point-to-point lines.

Connect all the terminals to the mainframe computer using one multipoint line.

Connect all the terminal outputs and use microwave transmissions to send the data to the mainframe.

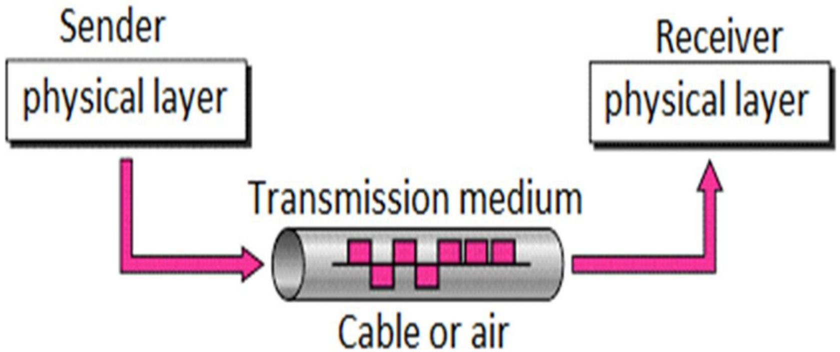
Collect all the terminal outputs using multiplexing and send the data to the mainframe computer using a conducted line.

# Summary

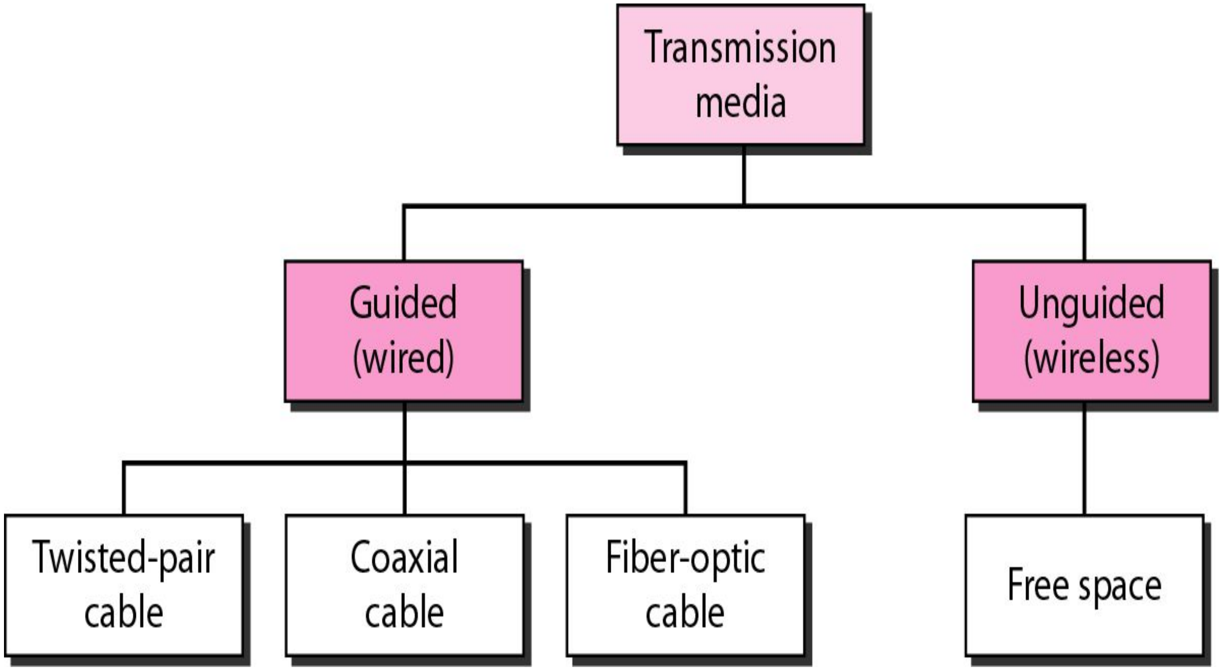
- Multiplexing
- Types of multiplexing
  - TDM
    - Synchronous TDM (T-1, ISDN, optical fiber)
    - Statistical TDM (LANs)
  - FDM (cable, cell phones, broadband)
  - WDM (optical fiber)
  - CDM (cell phones)

# TRANSMISSION MEDIA

- Sending of data from one device to another is called transmission of data.
- Medium used to transmit the data is called media.
- Transmission of data through medium is called transmission media. So, it is a pathway that carries the information from sender to receiver.
- We use different types of cables or waves to transmit data.
- Data is transmitted normally in electrical or electromagnetic signals.
- Transmission media are located below the physical layer.
- Computers use signals to represent data.
- Signals are transmitted in from of electromagnetic energy.

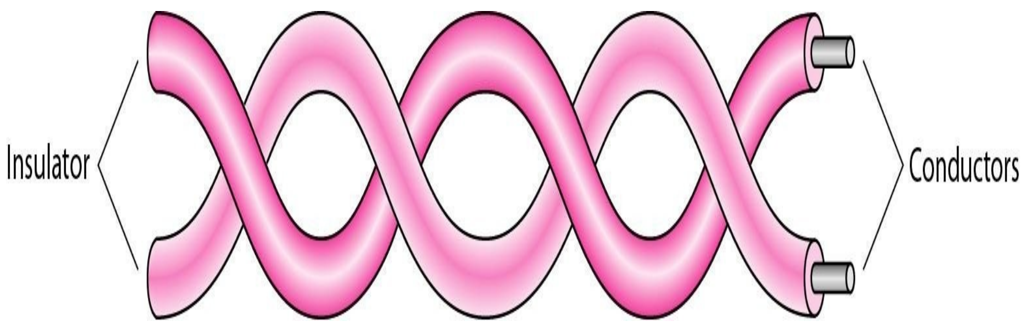


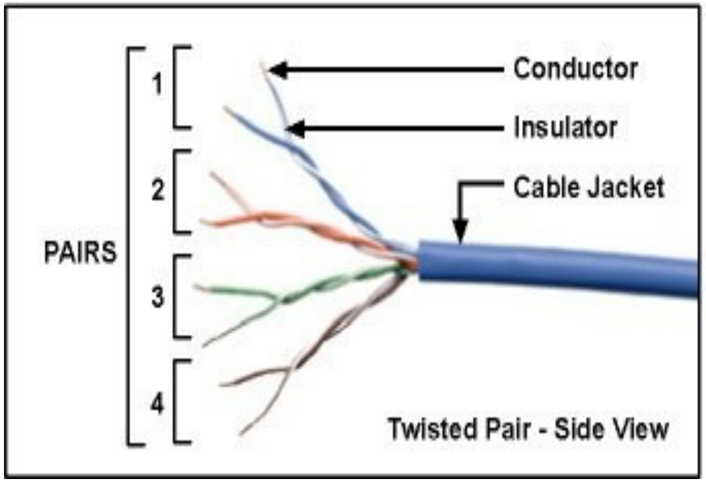
# *Types of* **transmission** *media*



# Twisted-pair Cable

- A twisted pair consists of two conductors (normally copper), each with its own plastic insulation, twisted together
- One of the wires carries signal, the other is used only as a ground reference.
- The receiver uses the difference b/w the two.
- Twisting increases, the probability that both wires are effected by the noise in the same manner, thus the difference at the receiver remains same.
- Therefore, number of twists per unit length determines the quality of the cable.



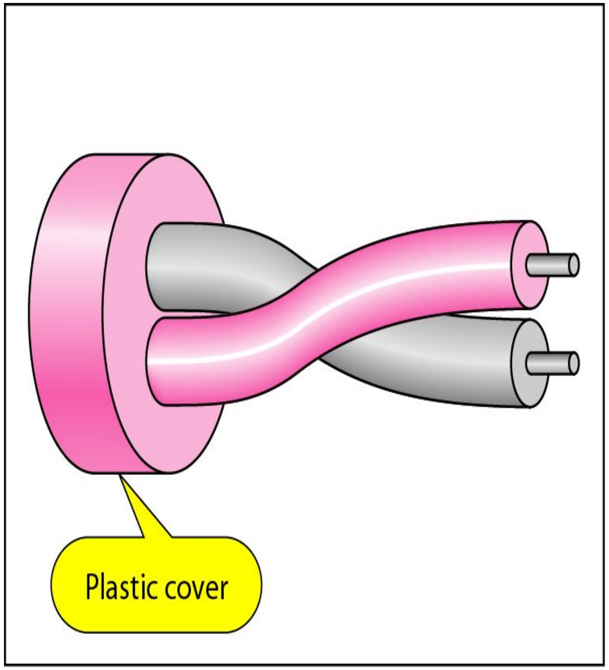



# Twisted Pair - Transmission Characteristics

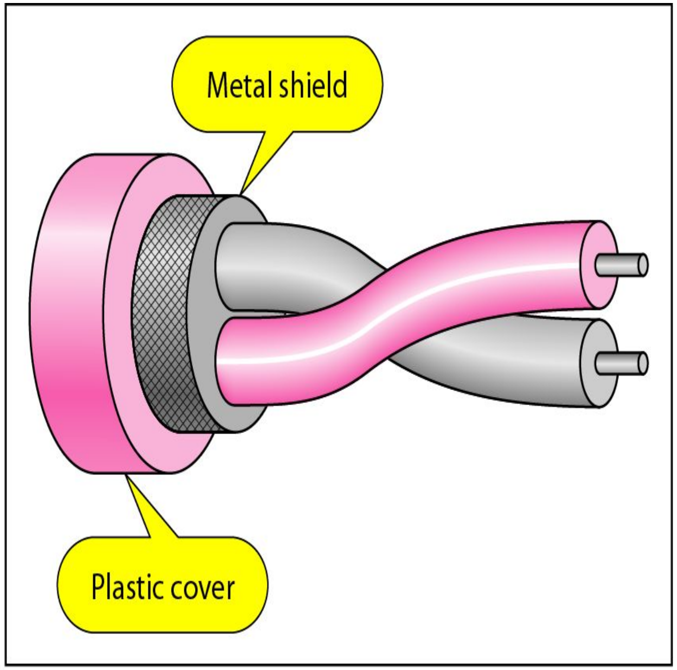
- Analog
  - needs amplifiers every 5km to 6km
- Digital
  - can use either analog or digital signals
  - needs a repeater every 2-3km
- Limited distance
- Limited bandwidth (1MHz)
- Limited data rate (100MHz)
- Susceptible to interference and noise

# Unshielded Versus Shielded Twisted-Pair Cable

UTP and STP cables




a. UTP




b. STP

# Unshielded Twisted Pair (UTP)

- Ordinary telephone wire
- Cheapest
- Easiest to install
- Suffers from external EM interference
- **Advantages of UTP:**
  - Affordable
  - Most compatible cabling
  - Major networking system
- **Disadvantages of UTP:**
- Suffers from external Electromagnetic interference

**Applications**:

- Telephone lines connecting subscribers to the central office
- DSL lines
- LAN – 10Base-T and 100Base-T

# Shielded Twisted Pair (STP)

- Metal braid or sheathing that reduces interference
- More expensive
- Harder to handle (thick, heavy)
- It offers protective sheathing around the copper wire and Provides better performance at lower data rates.
- Not commonly used
- Installation is easy
- Distance is only 100-500 meters
- Special connecters are required.

**STP Application**

- STP is used in IBM token ring networks.
- Higher transmission rates over longer distances.
- **Advantages of STP:**
  - Faster than UTP
- **Disadvantages of STP:**
  - More expensive than UTP
  - High attenuation rate

## *Categories of unshielded twisted-pair cables*

| *Category* | *Specification* | *Data Rate (Mbps)* | *Use* |
|---|---|---|---|
| 1 | Unshielded twisted-pair used in telephone | < 0.1 | Telephone |
| 2 | Unshielded twisted-pair originally used in T-lines | 2 | T-1 lines |
| 3 | Improved CAT 2 used in LANs | 10 | LANs |
| 4 | Improved CAT 3 used in Token Ring networks | 20 | LANs |
| 5 | Cable wire is normally 24 AWG with a jacket and outside sheath | 100 | LANs |
| 5E | An extension to category 5 that includes extra features to minimize the crosstalk and electromagnetic interference | 125 | LANs |
| 6 | A new category with matched components coming from the same manufacturer. The cable must be tested at a 200-Mbps data rate. | 200 | LANs |
| 7 | Sometimes called SSTP (shielded screen twisted-pair). Each pair is individually wrapped in a helical metallic foil followed by a metallic foil shield in addition to the outside sheath. The shield decreases the effect of crosstalk and increases the data rate. | 600 | LANs |

# Twisted Pair - Applications

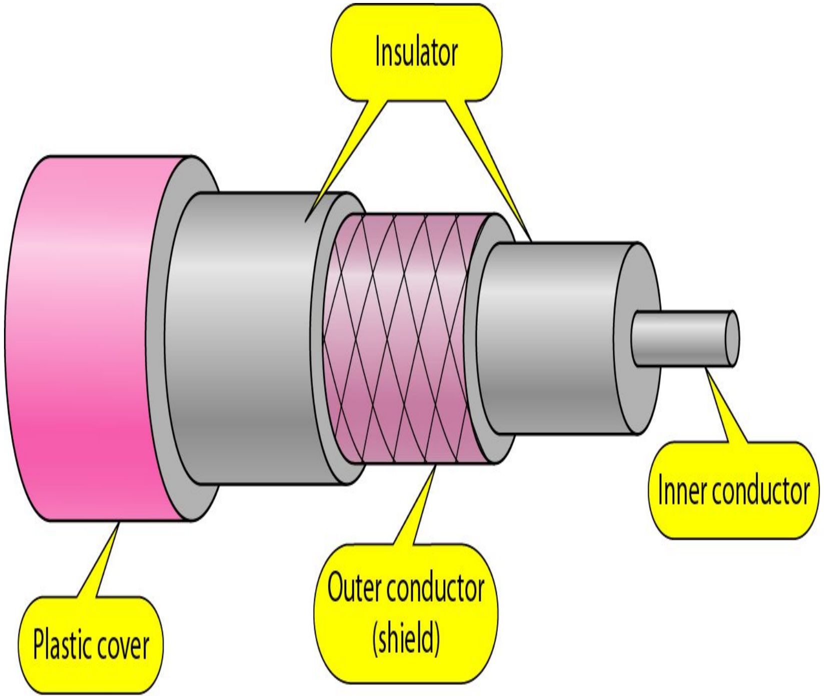
**Applications**

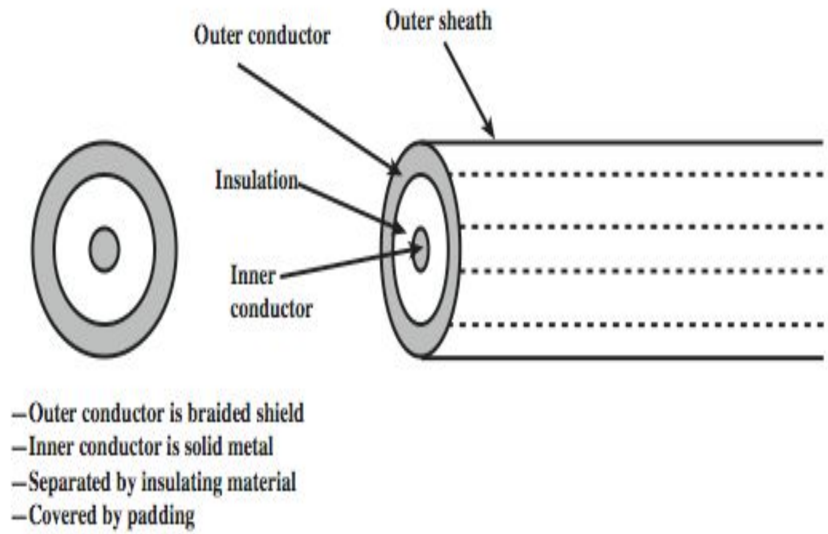
- Most common medium
- Telephone network
- Within buildings
- For local area networks (LAN)

**Pros and Cons**

- Cheap
- Easy to work with
- Low data rate
- Short range

# Guided Media – Coaxial Cable




(b) Coaxial cable

- Inner conductor is a solid wire outer conductor serves both as a shield
- against noise and a second conductor

# Characteristics

- Superior frequency characteristics
- Performance limited by attenuation & noise
- Analog signals
  - amplifiers every few km
  - closer if higher frequency
  - up to 500MHz
- Digital signals
  - repeater every 1km
  - closer for higher data rates

# Applications

- Most versatile medium
- Television distribution
- Long distance telephone transmissi
- Can carry 10,000 voice calls simul
- Short distance computer systems li
- Local area networks
- Analog telephone networks
- Cable TV networks
- Traditional Ethernet LAN – 10Base2, 10Base5

**Categories of coaxial cables**

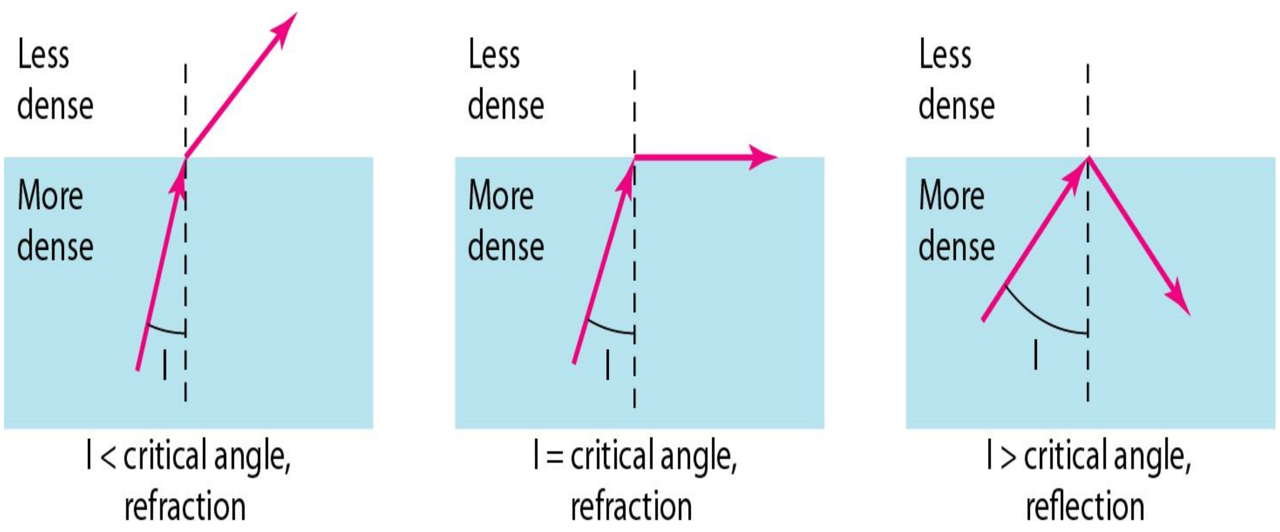
| *Category* | *Impedance* | *Use* |
|---|---|---|
| RG-59 | 75 Ω | Cable TV |
| RG-58 | 50 Ω | Thin Ethernet |
| RG-11 | 50 Ω | Thick Ethernet |

# Guided Media – Fiber-Optic Cable

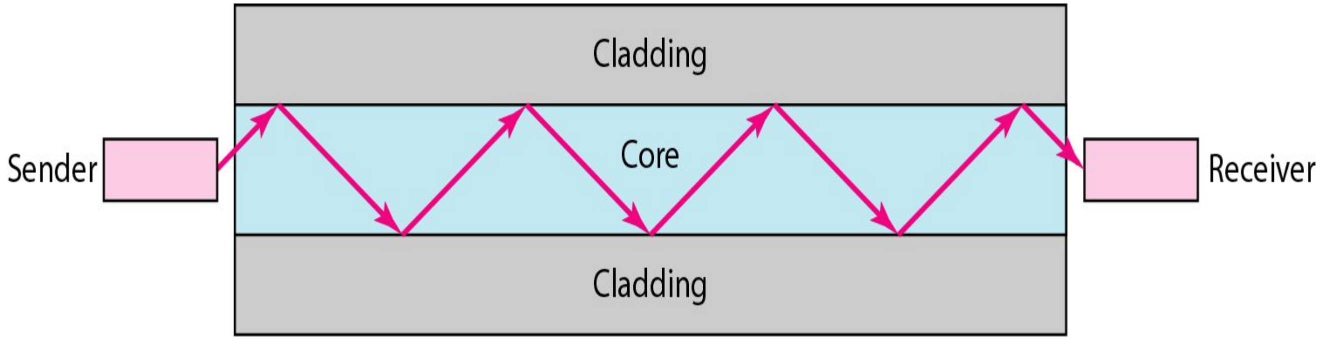
Fiber-optic cable transmit signals in the form of light.

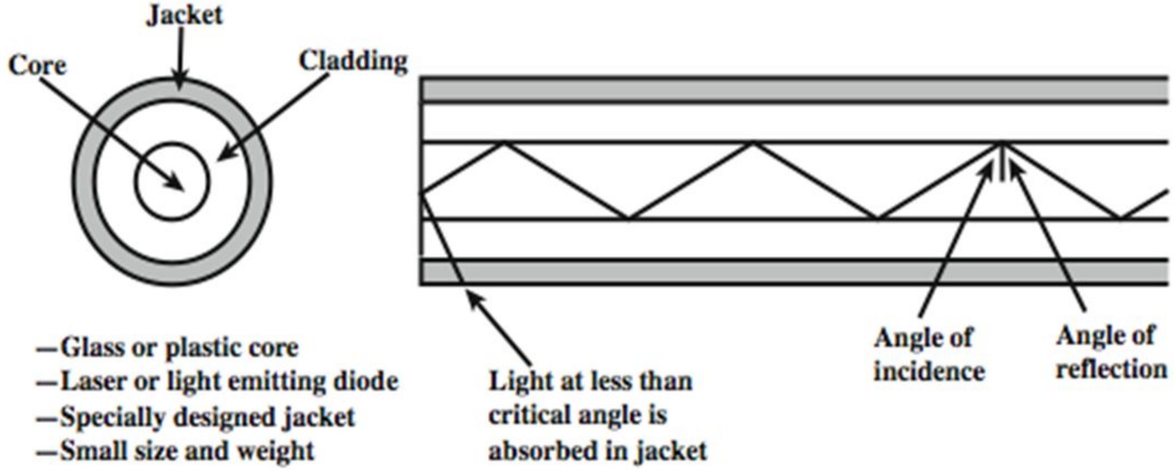
**Bending of light ray**

- Angle of Incidence (I): the angle the ray makes with the line perpendicular to the interface between the two substances
- Critical Angle: the angle of incidence which provides an angle of refraction of 90-degrees.



- Uses reflection to guide light through a channel
- Core is of glass or plastic surrounded by Cladding
- Cladding is of less dense glass or plastic



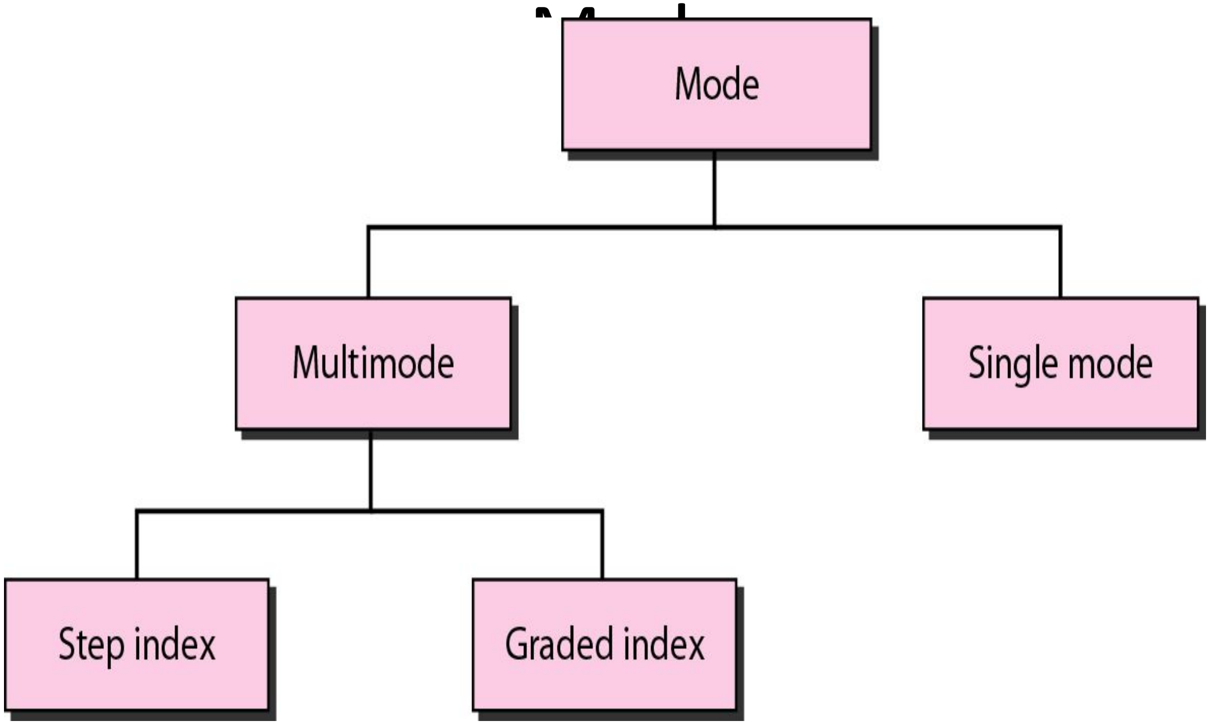

(c) Optical fiber

# Optical Fiber – Transmission Characteristics

- Uses total internal reflection to transmit light
  - effectively acts as wave guide for $10^{14}$ to $10^{15}$ Hz
- Can use several different light sources
  - Light Emitting Diode (LED)
    - cheaper, wider operating temp range, lasts longer
  - Injection Laser Diode (ILD)
    - more efficient, has greater data rate
- Relation of wavelength, type & data rate

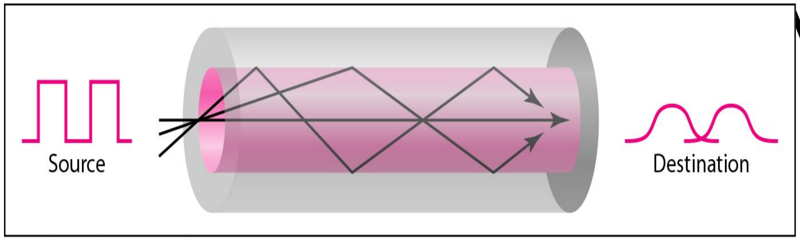
# Guided Media – Fiber-Optic Cable

## Propagation

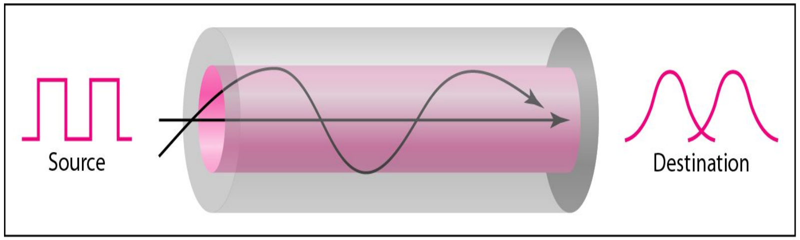


# Guided Media – Fiber-Optic Cable

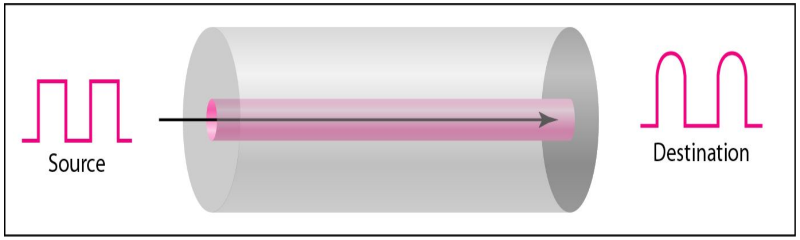
## Propagation Modes



a. Multimode, step index



b. Multimode, graded index



c. Single mode

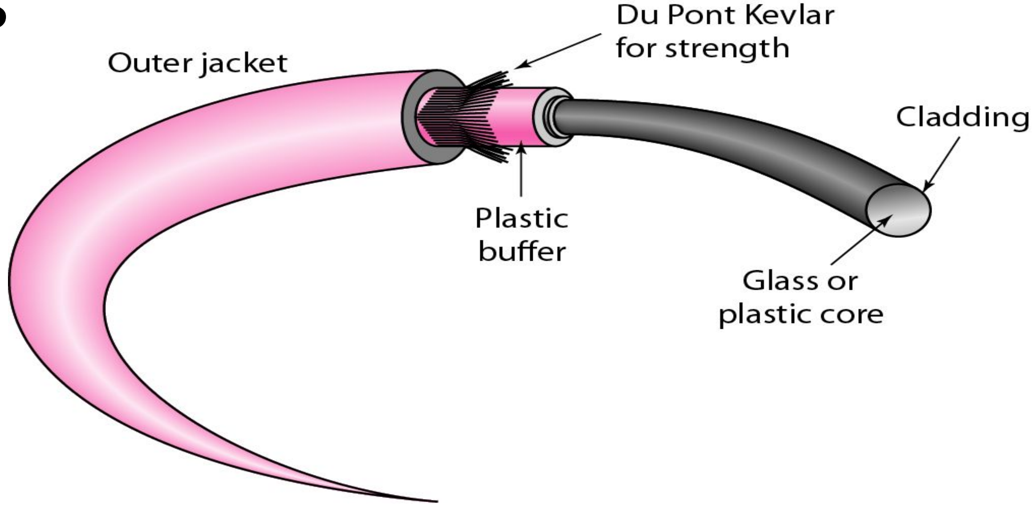
In multimode step-index fiber, the density of the core remains constant from the center to the edges. A beam of light moves through this constant density in a straight line until it reaches the interface of the core and the cladding.

At the interface, there is an abrupt change due to a lower density; this alters the angle of the beam's motion

In multimode graded-index fiber, decreases this distortion of the signal through the cable.

Density is highest at the center of the core and decreases gradually to its lowest at the

# Fiber Constructio




**Applications**:

- The fiber optic cable is often found in backbone networks because its bandwidth is cost effective.
- Telecommunications
- Local Area Networks
  - 100Base-FX network (Fast Ethernet)
  - 100Base-X
- Cable TV– backbone
- CCTV
- Medical Education

## Fiber Optic Advantages

- Greater capacity (bandwidth of up to 2 Gbps) & Smaller size and lighter weight.
- Lower attenuation.
- greater repeater spacing
  - o 10s of km at least
- More resistance to corrosive materials & immunity to environmental interference.
- highly secure due to tap difficulty and lack of signal radiation.

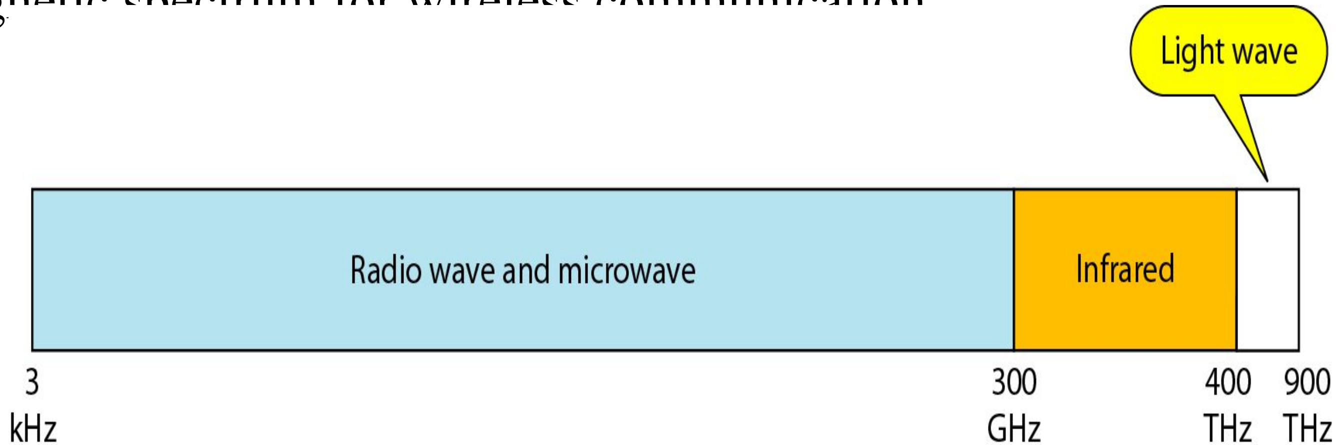
## Fiber Optic Disadvantages

- Installation and maintenance need expertise
- Much more expensive
- requires highly skilled installers
- adding additional nodes is difficult

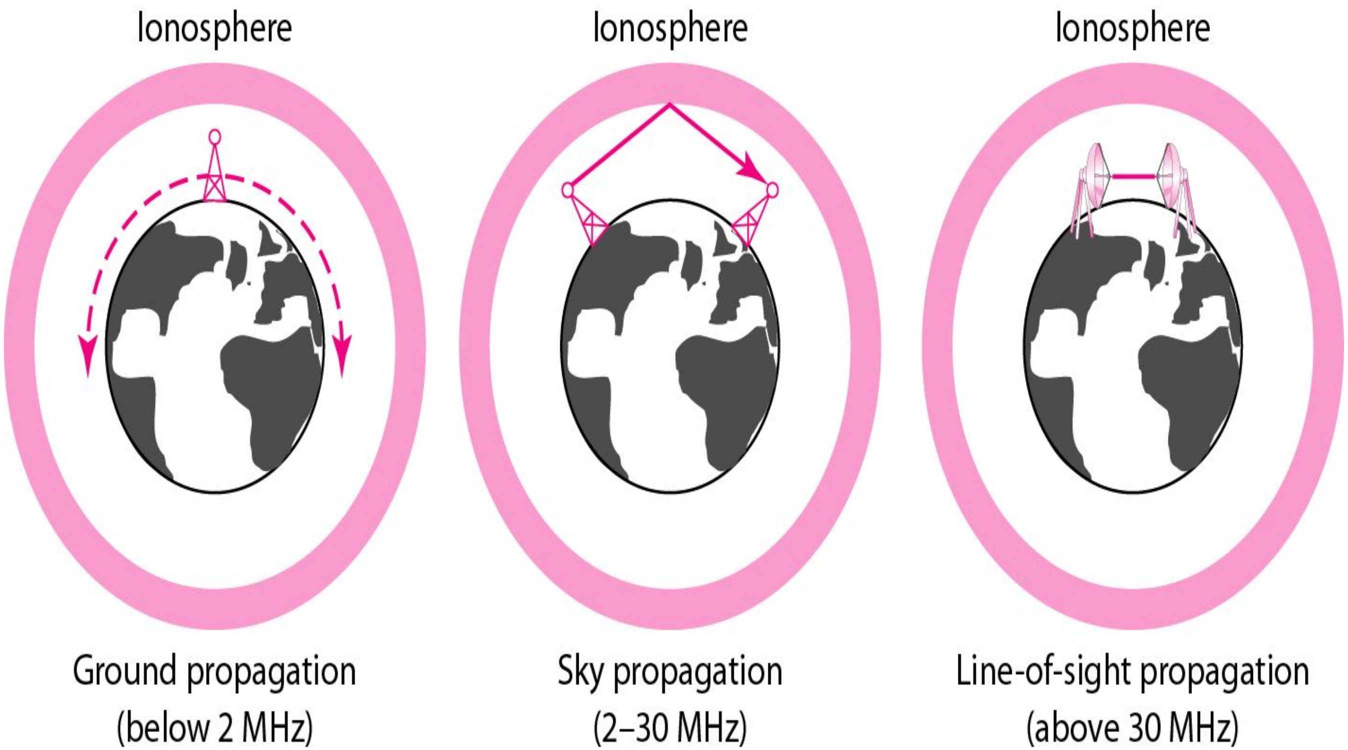
# UNGUIDED MEDIA: WIRELESS

Unguided media transport electromagnetic waves without using a physical conductor. This type of communication is often referred to as wireless communication.

Electromagnetic spectrum for wireless communication



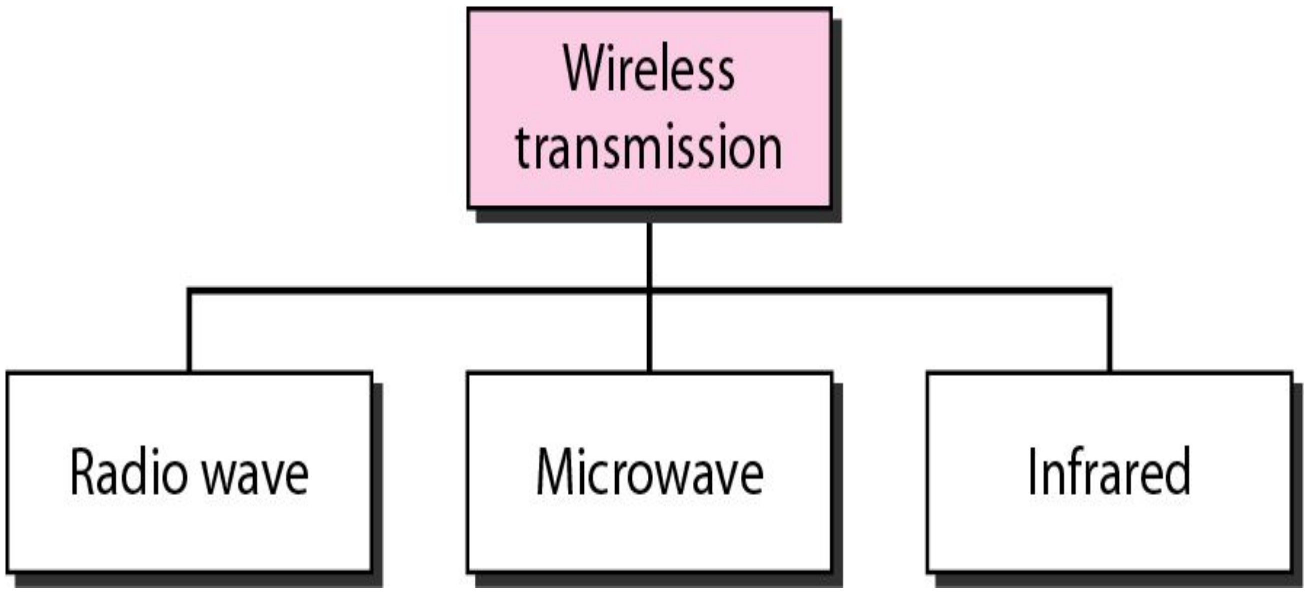
*Propagation methods*



# *Bands*

| *Band* | *Range* | *Propagation* | *Application* |
|---|---|---|---|
| VLF (very low frequency) | 3–30 kHz | Ground | Long-range radio navigation |
| LF (low frequency) | 30–300 kHz | Ground | Radio beacons and navigational locators |
| MF (middle frequency) | 300 kHz–3 MHz | Sky | AM radio |
| HF (high frequency) | 3–30 MHz | Sky | Citizens band (CB), ship/aircraft communication |
| VHF (very high frequency) | 30–300 MHz | Sky and line-of-sight | VHF TV, FM radio |
| UHF (ultrahigh frequency) | 300 MHz–3 GHz | Line-of-sight | UHF TV, cellular phones, paging, satellite |
| SHF (superhigh frequency) | 3–30 GHz | Line-of-sight | Satellite communication |
| EHF (extremely high frequency) | 30–300 GHz | Line-of-sight | Radar, satellite |

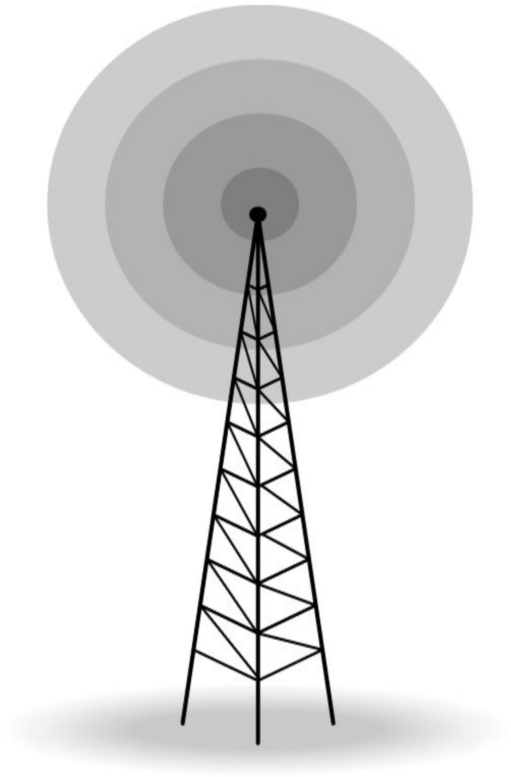
# *Wireless transmission waves*



Radio waves are used for multicast communications, such as radio and television, and paging systems.

They can penetrate through walls.

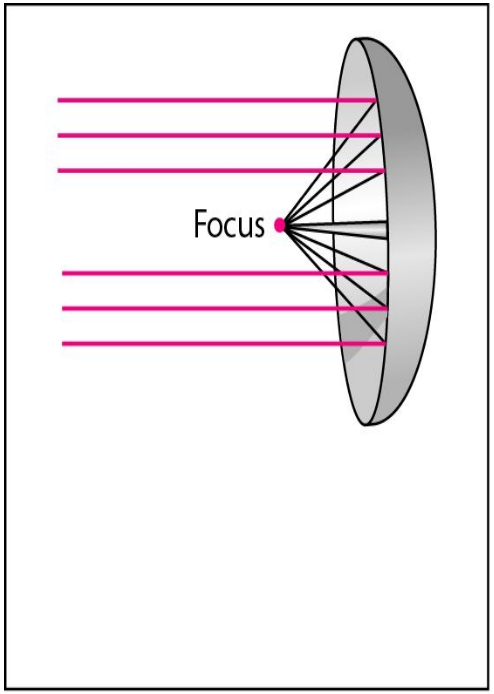
Highly regulated. Use omni directional antennas



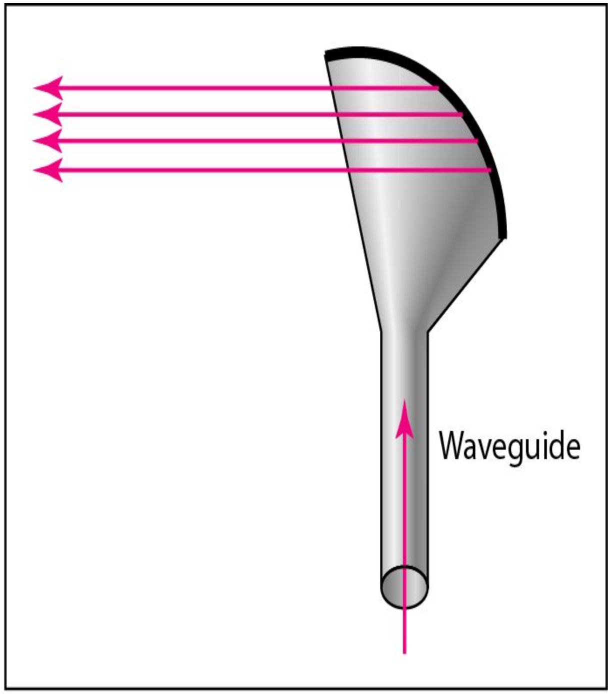
Microwaves are used for unicast communication such as cellular telephones, satellite networks, and wireless LANs.

Higher frequency ranges cannot penetrate walls.

Use directional antennas - point to point line of sight communications




a. Dish antenna




b. Horn antenna

Infrared signals can be used for short-range communication in a closed area using line-of-sight propagation.

## Wireless Channels

Are subject to a lot more errors than guided media channels.

Interference is one cause for errors, can be circumvented with high SNR.

The higher the SNR the less capacity is available for transmission due to the broadcast nature of the channel.

Channel also subject to fading and no coverage holes.