Solutions for Data Communication(15CS46) IAT-1

1 a) List and explain different components used in data communication system (4)
b) With neat diagram, explain any three network topologies (6)

Solution:
1a)
- Five components of a communication-system (Figure 1.1):
  1) Message
  2) Sender
  3) Receiver
  4) Transmission-Medium
  5) Protocol

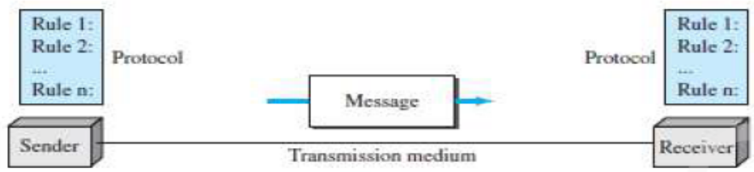


Figure 1.1 *Five components of data communication*

1) Message
- ➤ Message is the information (or data) to be communicated.
- ➤ Message may consist of
  - → number/text
  - → picture or
  - → audio/video

2) Sender
- ➤ Sender is the device that sends the data-message.
- ➤ Sender can be
  - → computer and
  - → mobile phone

3) Receiver
- ➤ Receiver is the device that receives the message.
- ➤ Receiver can be
  - → computer and
  - → mobile phone

4) Transmission Medium
- ➤ Transmission-medium is physical-path by which a message travels from sender to receiver.
- ➤ Transmission-medium can be wired or wireless.
- ➤ Examples of wired medium:
  - → twisted-pair wire (used in landline telephone)
  - → coaxial cable (used in cable TV network)
  - → fiber-optic cable
- ➤ Examples of wireless medium:
  - → radio waves
  - → microwaves
  - → infrared waves (ex: operating TV using remote control)

5) Protocol
- ➤ A protocol is a set of rules that govern data-communications.
- ➤ In other words, a protocol represents an agreement between the communicating-devices.
- ➤ Without a protocol, 2 devices may be connected but not communicating.

b) Three different topologies described below:

1.2.2.2.1 Bus Topology

- All the devices are connected to the single cable called bus (Figure 1.4).
- Every device communicates with the other device through this bus.
- A data from the source is broadcasted to all devices connected to the bus.
- Only the intended-receiver, whose physical-address matches, accepts the data.

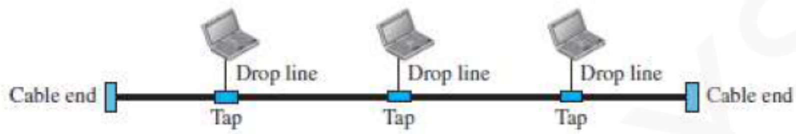


Figure 1.4 *A bus topology connecting three stations*

- Devices are connected to the bus by drop-lines and taps.
- A drop-line is a connection running between the device and the bus.
- A tap is a connector that links to the bus or
- Advantages:
  1) Easy installation.
  2) Cable required is the least compared to mesh/star topologies.
  3) Redundancy is eliminated.
  4) Costs less (Compared to mesh/star topologies).
  5) Mostly used in small networks. Good for LAN.
- Disadvantages:
  1) Difficult to detect and troubleshoot fault.
  2) Signal reflection at the taps can cause degradation in quality.
  3) A fault/break in the cable stops all transmission.
  4) There is a limit on
     i) Cable length
     ii) Number of nodes that can be connected.
  5) Security is very low because all the devices receive the data sent from the source.

ii)

1.2.2.2.2 Star Topology

- All the devices are connected to a central controller called a hub (Figure 1.5).
- There exists a dedicated point-to-point link between a device & a hub.
- The devices are not directly linked to one another. Thus, there is no direct traffic between devices.
- The hub acts as a junction:

  If device-1 wants to send data to device-2,
  the device-1 sends the data to the hub,
  then the hub relays the data to the device-2.

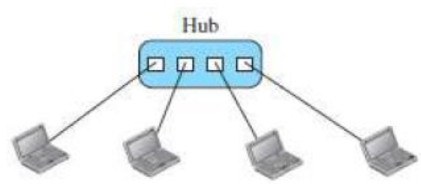


Figure 1.5 *A star topology connecting four stations*

- Advantages:
  1) Less expensive: Each device needs only one link & one I/O port to connect it to any devices.
  2) Easy installation & reconfiguration: Nodes can be added/removed w/o affecting the network.
  3) Robustness: If one link fails, it does not affect the entire system.
  4) Easy to detect and troubleshoot fault.
  5) Centralized management: The hub manages and controls the whole network.
- Disadvantages:
  1) Single point of failure: If the hub goes down, the whole network is dead.
  2) Cable length required is the more compared to bus/ring topologies.
  3) Number of nodes in network depends on capacity of hub.

1.2.2.2.3 Ring Topology

- Each device is connected to the next, forming a ring (Figure 1.6).
- There are only two neighbors for each device.
- Data travels around the network in one direction till the destination is reached.
- Sending and receiving of data takes place by the help of token.
- Each device has a repeater.
- A repeater
  - → receives a signal on transmission-medium &
  - → regenerates & passes the signal to next device.

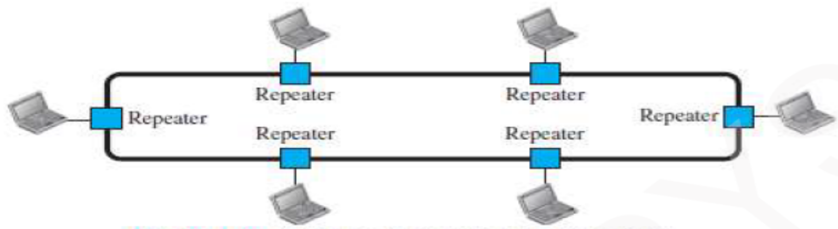


Figure 1.6 *A ring topology connecting six stations*

- Advantages:
  1) Easy installation and reconfiguration.
     To add/delete a device, requires changing only 2 connections.
  3) Fault isolation is simplified.
     If one device does not receive a signal within a specified period, it can issue an alarm.
     The alarm alerts the network-operator to the problem and its location.
  3) Congestion reduced: Because all the traffic flows in only one direction.
- Disadvantages:
  1) Unidirectional traffic.
  2) A fault in the ring/device stops all transmission.
     The above 2 drawbacks can be overcome by using dual ring.
  3) There is a limit on
     i) Cable length &
     ii) Number of nodes that can be connected.
  4) Slower: Each data must pass through all the devices between source and destination.

2) With a neat schematic, explain the working of different layers in TCP/IP network model (10)

1.6.2 Layers in the TCP/IP Protocol Suite

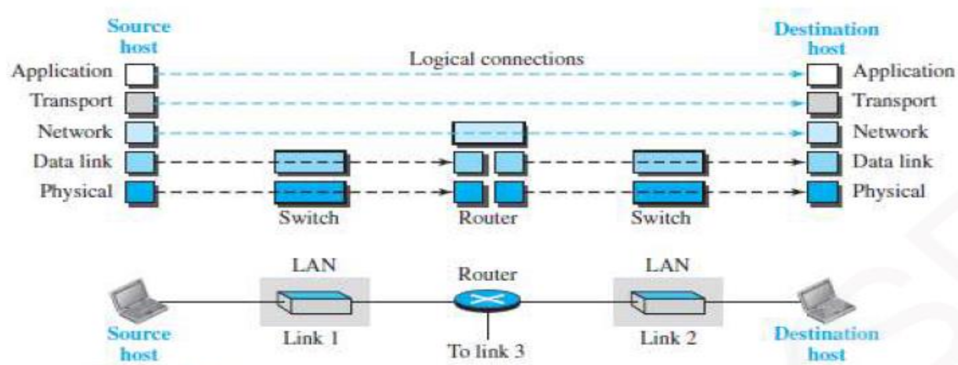


Figure 2.6 *Logical connections between layers of the TCP/IP protocol suite*

- As shown in the figure 2.6, the duty of the application, transport, and network layers is end-to-end.
- However, the duty of the data-link and physical layers is hop-to-hop. A hop is a host or router.
- The domain of duty of the top three layers is the internet.
  The domain of duty of the two lower layers is the link.
- In top 3 layers, the data unit should not be changed by any router or link-layer switch.
  In bottom 2 layers, the data unit is changed only by the routers, not by the link-layer switches.

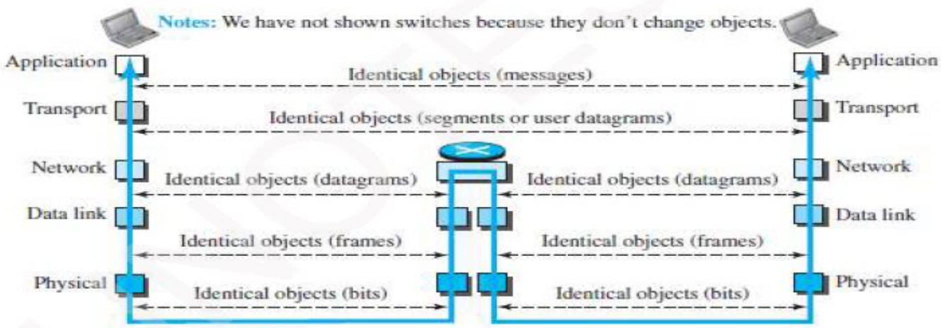
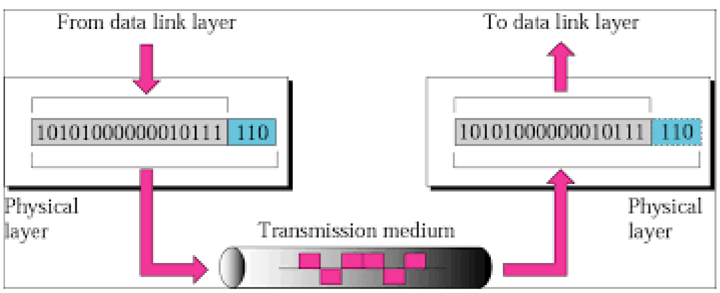


Figure 2.7 *Identical objects in the TCP/IP protocol suite*

- Identical objects exist between two hops. Because router may fragment the packet at the network layer and send more packets than received (Figure 2.7).
- The link between two hops does not change the object.

Physical Layer

The physical layer coordinates the functions required to carry a bit stream over a physical medium. It deals with the mechanical and electrical specifications of the interface and transmission medium. It also defines the procedures and functions that physical devices and interfaces have to perform for transmission to occur. The following figure shows the position of the physical layer with respect to the transmission medium and the data link layer.



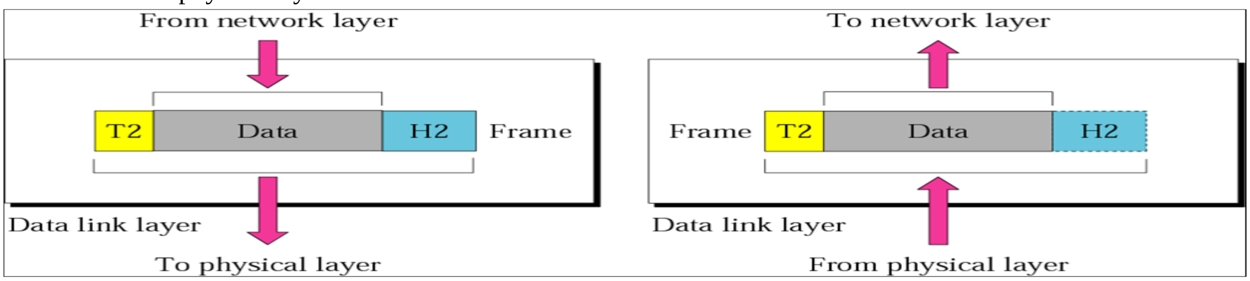
The physical layer is also concerned with the following:

- **Physical characteristics of interfaces and medium.** The physical layer defines the characteristics of the interface between the devices and the transmission medium. It also defines the type of transmission medium.
- **Representation of bits.** The physical layer data consists of a stream of bits (sequence of 0s or ls) with no interpretation. To be transmitted, bits must be encoded into signals-electrical or optical. The physical layer defines the type of encoding (how 0s and 1s are changed to signals).
- **Data rate.** The transmission rate--the number of bits sent each second--is also defined by the physical layer. In other words, the physical layer defines the duration of a bit, which is how long it lasts.
- **Synchronization of bits.**The sender and receiver not only must use the same bit rate but also must be synchronized at the bit level. In other words, the sender and the receiver clocks must be synchronized.
- **Line configuration.** The physical layer is concerned with the connection of devices to the media. In a point-to-point configuration, two devices are connected through a dedicated link. In a multipoint configuration, a link is shared among several devices.
- **Physical topology.** The physical topology defines how devices are connected to make a network. Devices can be connected by using a mesh topology (every device is connected to every other device), a star topology (devices are connected through a central device), a ring topology (each device is connected to the next, forming a ring), a bus topology(every device is on a common link), or a hybrid topology (this is a combination of two or more topologies).

- **Transmission mode.** The physical layer also defines the direction of transmission between two devices: simplex, half-duplex, or full-duplex. In simplex mode, only one device can send; the other can only receive.The simplex mode is a one-way communication. In the half-duplex mode, two devices can send and receive, but not at the same time. In a full-duplex (or simply duplex) mode, two devices can send and receive at the same time.

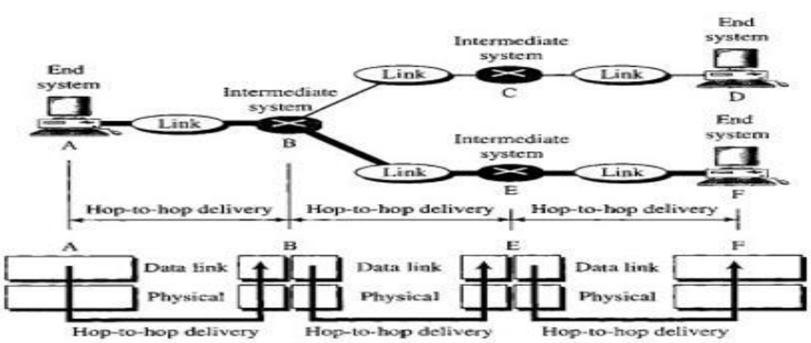
Data Link Layer

The data link layer transforms the physical layer, a raw transmission facility, to a reliable link. It makes the physical layer appear error-free to the upper layer (network layer). The figure shows the relationship of the data link layer to the network and physical layers.



Other responsibilities of the data link layer include the following:

- **Framing.** The data link layer divides the stream of bits received from the network layer into manageable data units called frames.
- **Physical addressing.** If frames are to be distributed to different systems on the network, the data link layer adds a header to the frame to define the sender and/or receiver of the frame.If the frame is intended for a system outside the sender's network, the receiver address is the address of the device that connects the network to the next one.
- **Flow control.** If the rate at which the data are absorbed by the receiver is less than the rate at which data are produced in the sender, the data link layer imposes a flow control mechanism to avoid overwhelming the receiver.
- **Error control.** The data link layer adds reliability to the physical layer by adding mechanisms to detect and retransmit damaged or lost frames. It also uses a mechanism to recognize duplicate frames. Error control is normally achieved through a trailer added to the end of the frame.
- **Access control.** When two or more devices are connected to the same link, data link layer protocols are necessary to determine which device has control over the link at any given time.

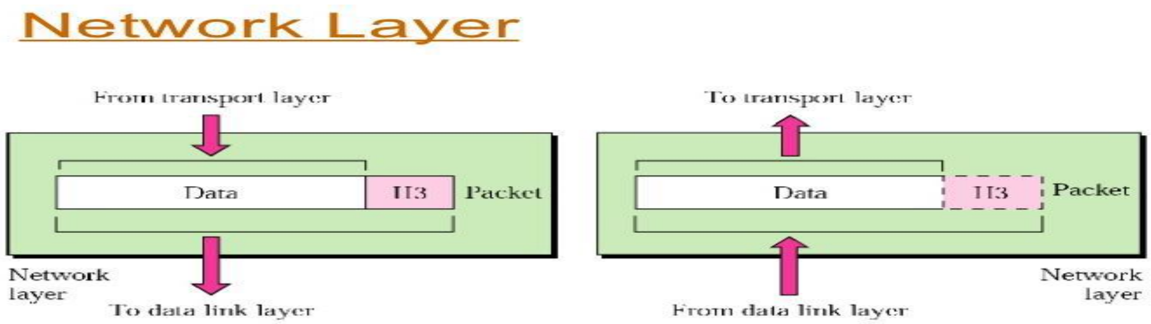


Communication at the data link layer occurs between two adjacent nodes. To send data from A to F, three partial deliveries are made. First, the data link layer at A sends a frame to the data link layer at B (a router). Second, the data link layer at B sends a new frame to the data link layer at E. Finally, the data link layer at E sends a new frame to the data link layer at F.

Network Layer

The network layer is responsible for the source-to-destination delivery of a packet, possibly across multiple networks (links). Whereas the data link layer oversees the delivery of the packet between two systems on the same network (links), the network layer ensures that each packet gets from its point of origin to its final destination.

If two systems are connected to the same link, there is usually no need for a network layer. However, if the two systems are attached to different networks (links) with connecting devices between the networks (links), there is often a need for the network layer to accomplish source-to-destination delivery. The figure shows the relationship of the network layer to the data link and transport layers

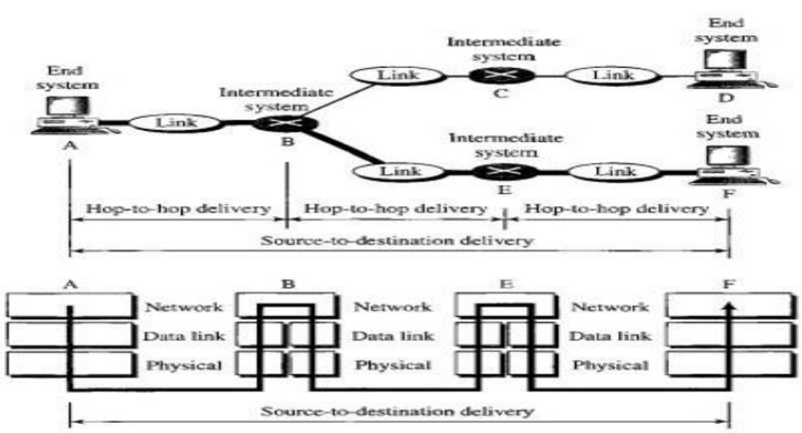


Other responsibilities of the network layer include the following:

□□**Logical□addressing**.The physical addressing implemented by the data link layer handles the addressing problem locally.If a packet passes the network boundary, we need another addressing system to help distinguish the source and destination systems. The network layer adds a header to the packet coming from the upper layer that, among other things includes the logical addresses of the sender and receiver.

□□**Routing.** When independent networks or links are connected to create internetworks (network of networks) or a large network, the connecting devices (called routers or switches) route or switch the packets to their final destination. One of the functions of the network layer is to provide this mechanism.

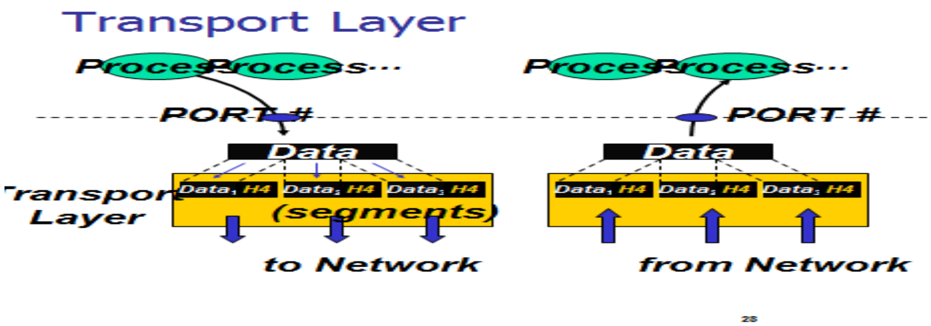
The figure illustrates end-to-end delivery by the network layer.



The network layer at A sends the packet to the network layer at B. When the packet arrives at router B, the router makes a decision based on the final destination (F) of the packet. As we will see in later chapters, router B uses its routing table to find that the next hop is router E. The network layer at B, therefore, sends the packet to the network layer at E. The network layer at E, in turn, sends the packet to the network layer at F.

Transport Layer

The transport layer is responsible for process-to-process delivery of the entire message.A process is an application program running on a host. Whereas the network layer oversees source-to-destination delivery of individual packets, it does not recognize any relationship between those packets. It treats each one independently, as though each piece belonged to a separate message, whether or not it does. The transport layer, on the other hand, ensures that the whole message arrives intact and in order, overseeing both error control and flow control at the source-to-destination level. The figure shows the relationship of the transport layer to the network layer.



Other responsibilities of the transport layer include the following:

 Service-point addressing. Computers often run several programs at the same time. For this reason, source-to-destination delivery means delivery not only from one computer to the next but also from a specific process (running program) on one computer to a specific process (running program) on the other. The transport layer header must therefore include a type of address called a service-point address (or port address). The network layer gets each packet to the correct computer; the transport layer gets the entire message to the correct process on that computer.

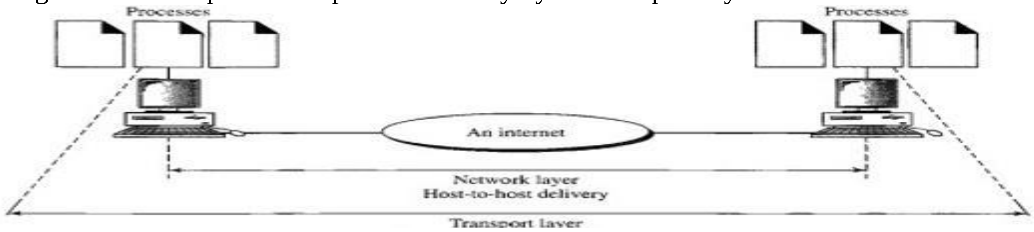
 Segmentation and reassembly. A message is divided into transmittable segments, with each segment containing a sequence number. These numbers enable the transport layer to reassemble the message correctly upon arriving at the destination and to identify and replace packets that were lost in transmission.

 Connection control. The transport layer can be either connectionless or connection-oriented. A connectionless transport layer treats each segment as an independent packet and delivers it to the transport layer at the destination machine. A connection-oriented transport layer makes a connection with the transport layer at the destination machine first before delivering the packets. After all the data are transferred, the connection is terminated.

Flow control. Like the data link layer, the transport layer is responsible for flow control.However, flow control at this layer is performed end to end rather than across a single link.

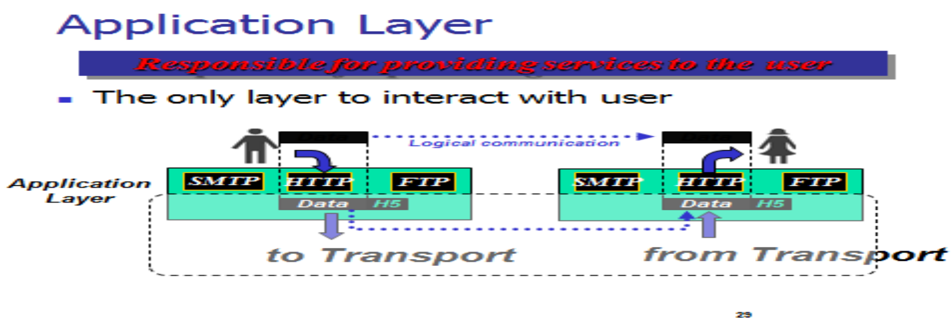
 Error control. Like the data link layer, the transport layer is responsible for error control. However, error control at this layer is performed process-to-process rather than across a single link. The sending transport layer makes sure that the entire message arrives at the receiving transport layer without error (damage, loss, or duplication). Error correction is usually achieved through retransmission.

The figure illustrates process-to-process delivery by the transport layer.



Application Layer

The application layer enables the user, whether human or software, to access the network. The two application layers exchange messages between each other as though there were a bridge between the two layers. However, we should know that the communication is done through all the layers. Communication at the application layer is between two processes (two programs running at this layer). To communicate, a process sends a request to the other process and receives a response. Process-to-process communication is the duty of the application layer. The application layer in the Internet includes many predefined protocols, but a user can also create a pair of processes to be run at the two hosts.It provides user interfaces and support for services such as electronic mail, remote file access and transfer, shared database management, and other types of distributed information services.



- The Simple Mail Transfer Protocol (SMTP) is the main protocol used in electronic mail (e-mail) service.
- The File Transfer Protocol (FTP) is used for transferring files from one host to another.

The Terminal Network (TELNET) and Secure Shell (SSH) are used for accessing a site remotely.

- The Simple Network Management Protocol (SNMP) is used by an administrator to manage the Internet at global and local levels.
- The Domain Name System (DNS) is used by other protocols to find the network-layer address of a Computer

3a) Write a short note on transmission impairments (06)

b) i) If a telephone line has a bandwidth of 4KHz. When the signal is 10 V, the noise is 5 mV. What is the maximum data rate supported by this telephone line? (02)

ii) A network with bandwidth of 10Mbps can pass only an average of 18000 frames per minute with each frame carrying an average of 10,000bits. Find the throughput of the network (02)

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.

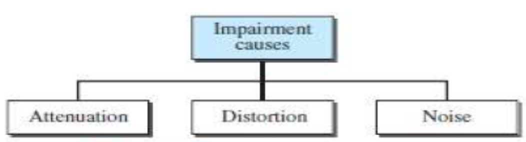


Figure 3.26 *Causes of impairment*

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.

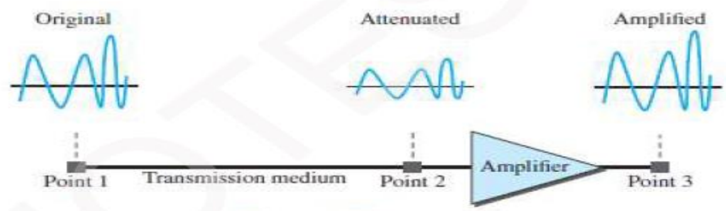


Figure 3.27 *Attenuation*

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.

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.

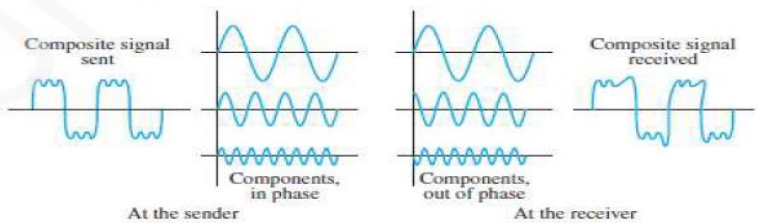


Figure 3.29 *Distortion*

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)

1.10.3.1 Signal-to-Noise Ratio (SNR)

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

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

- 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, SNR_{dB}, defined as

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

b) i) Capacity= Bandwidth* $\log_2(1+SNR)$

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

SNR= (10 V)/ (5 mV)=10/.005=2,000

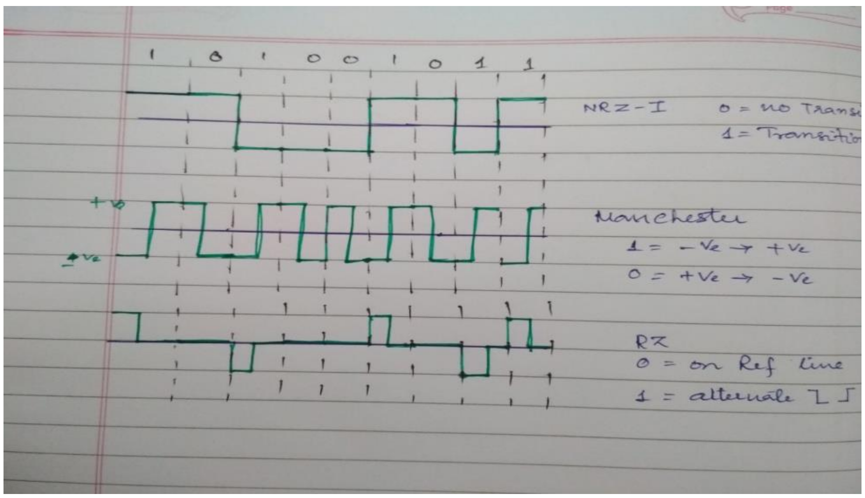
Capacity=4*1000*2000=8,000,000Hz= 8MHz

ii) Throughput= (18,000*10,000)/60= 3Mbps

4a) Represent the give sequence 101001011 in NRZ-I, Manchester, RZ, AMI (04)

b) Write a short note on Baseband transmission (06)

a)



b)

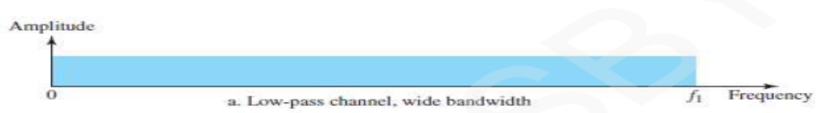
• Two methods for transmitting a digital signal:
1) Baseband transmission 2) Broadband transmission (using modulation).

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.



b. Low-pass channel, narrow bandwidth

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.

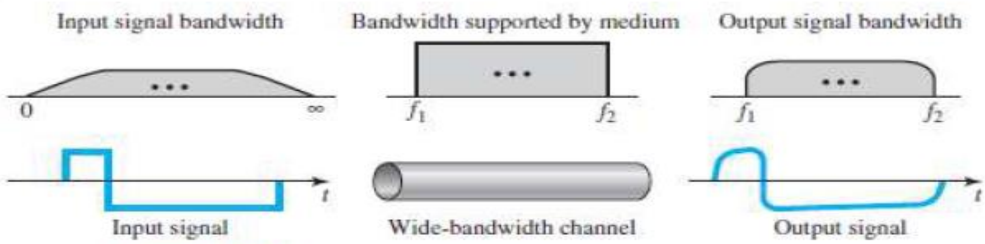


Figure 3.21 *Baseband transmission using a dedicated medium*

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. . . .

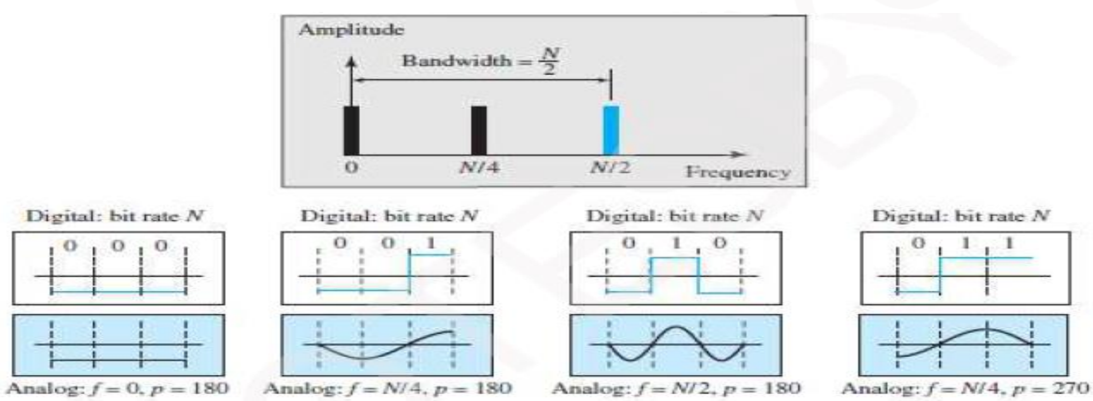
⌘ To simulate these two cases, we need an analog signal of frequency f = N/2.

⌘ 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

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



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.

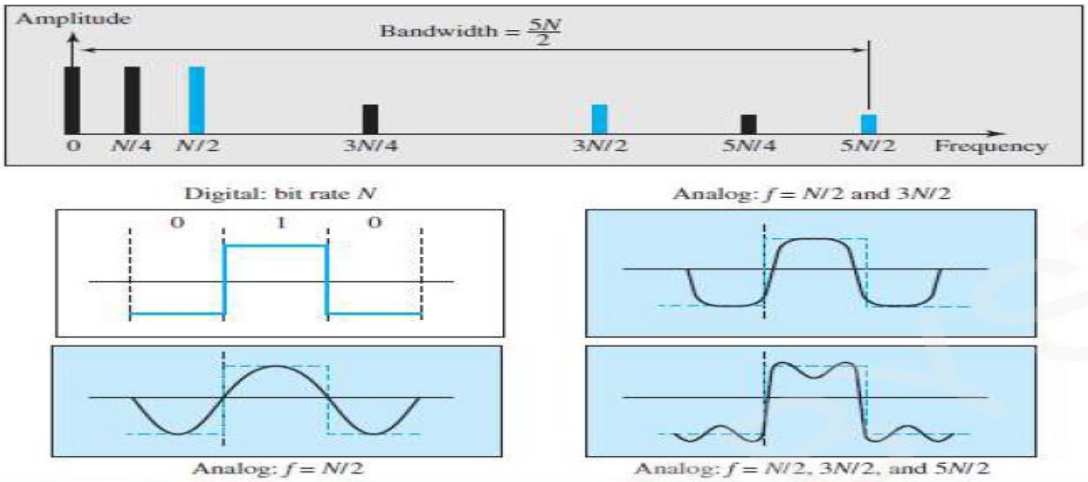


Figure 3.23 *Simulating a digital signal with first three harmonics*

5a) Explain serial and parallel transmission modes (08)
b)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? (02)

2.2 TRANSMISSION MODES

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

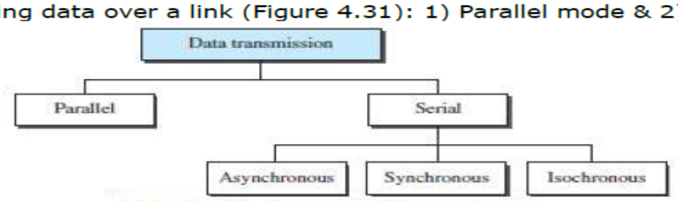


Figure 4.31 *Data transmission and modes*

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.

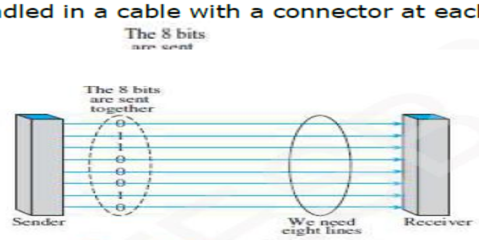


Figure 4.32 *Parallel transmission*

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

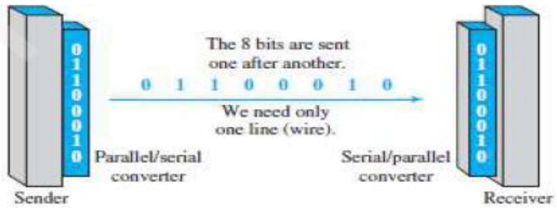


Figure 4.33 *Serial transmission*

- 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.

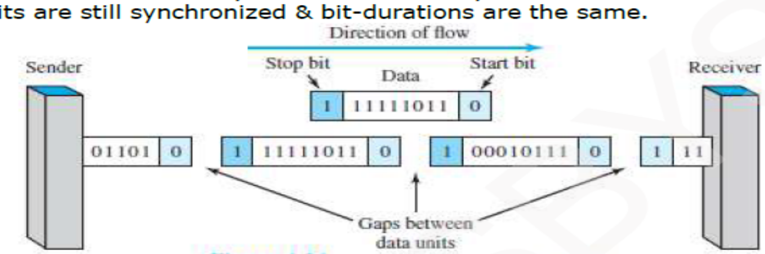


Figure 4.34 *Asynchronous transmission*

- 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).

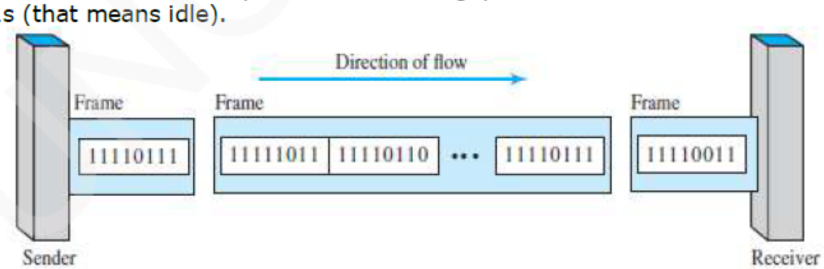


Figure 4.35 *Synchronous transmission*

- Advantages:

Sender Receiver

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 be viewed at the same rate.

b) baud rate= (bit rate)/r r=no. of data elements carried by each signal element
so, r= (bit rate)/ baud rate= 8000/1000= 8bits/SE
$r=\log_2 L$, $L=2^r=2^8=256$

6) Explain the working of PCM encoder with an example. (10)

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.

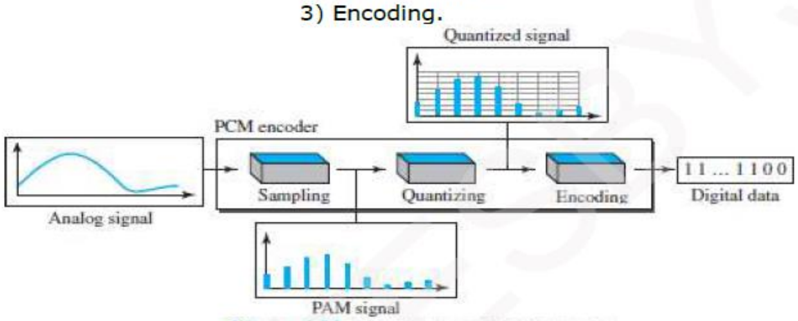


Figure 4.21 *Components of PCM encoder*

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).

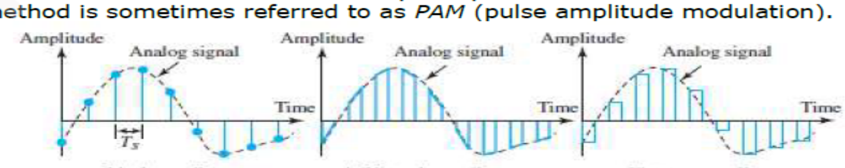


Figure 4.22 *Three different sampling methods for PCM*

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).

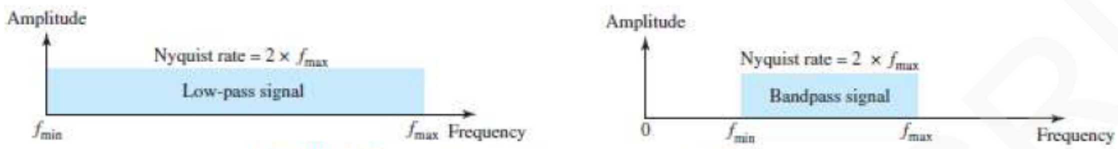


Figure 4.23 *Nyquist sampling rate for low-pass and bandpass signals*

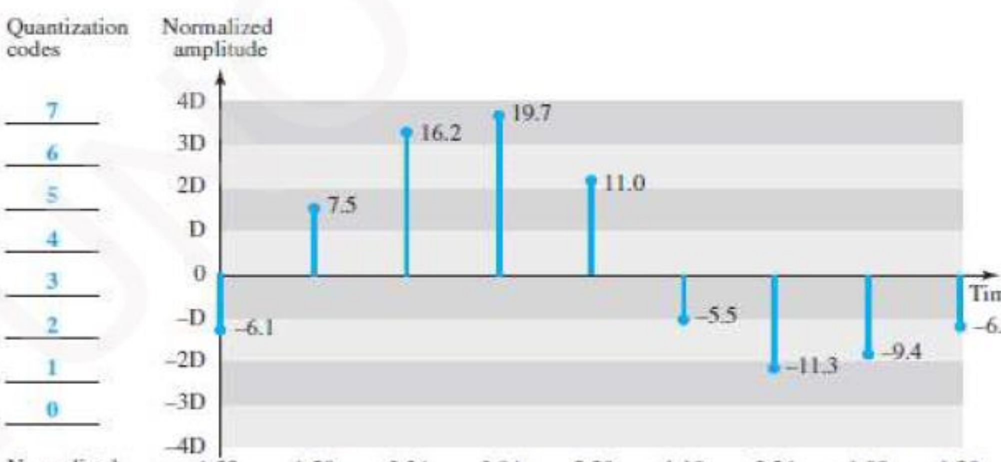
2.1.2 Quantization

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

7) Explain the working the of BASK and BFSK with example (10)

2.3.2 Amplitude Shift Keying (ASK)
•The amplitude of the carrier-signal is varied to represent different signal-elements.
• Both frequency and phase remain constant for all signal-elements.

2.3.2.1 Binary ASK (BASK)
• BASK is implemented using only 2 levels. (Figure 5.3)
• This is also referred to as OOK (On-Off Keying).

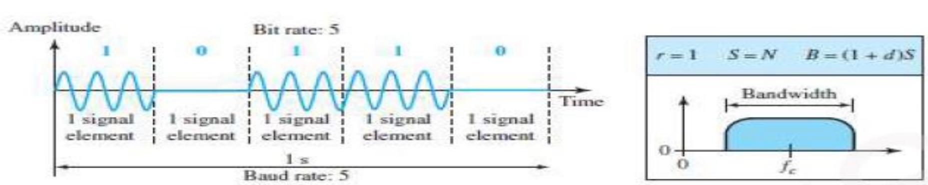


Figure 5.3 *Binary amplitude shift keying*

2.3.2.1.1 Implementation of BASK
• Here, line coding method used = unipolar NRZ (Figure 5.4).
• The unipolar NRZ signal is multiplied by the carrier-frequency coming from an oscillator.
1) When amplitude of the NRZ signal = 0, amplitude of the carrier-signal = 0.
2) When amplitude of the NRZ signal = 1, the amplitude of the carrier-signal is held.

3) Oscillator generates the carrier signal with fixed frequency and dependa on the binary stream multiplexer generates the modulated signal according to 1 & 2

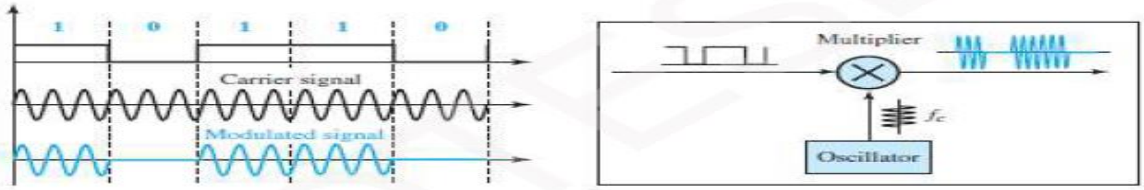


Figure 5.4 *Implementation of binary ASK*

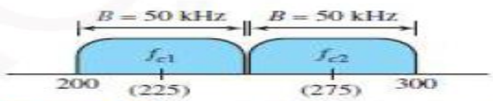


Figure 5.5 *Bandwidth of full-duplex ASK*

2.3.2.1.2 Bandwidth for ASK

- Here, the bandwidth (B) is proportional to the signal-rate (S) (Figure 5.5)
- The bandwidth is given by

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

where d(0<d<1)= this factor depends on modulation and filtering-process.

2.3.3 Frequency Shift Keying (FSK)

- The frequency of the carrier-signal is varied to represent different signal-elements.
- 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 amplitude and phase remain constant for all signal-elements.

2.3.3.1 Binary FSK (BFSK)

- This uses 2 carrier-frequencies: f1 and f2. (Figure 5.6)
  1) When data-element = 1, first carrier frequency(f1) is used.
  2) When data-element = 0, second carrier frequency(f2) is used.

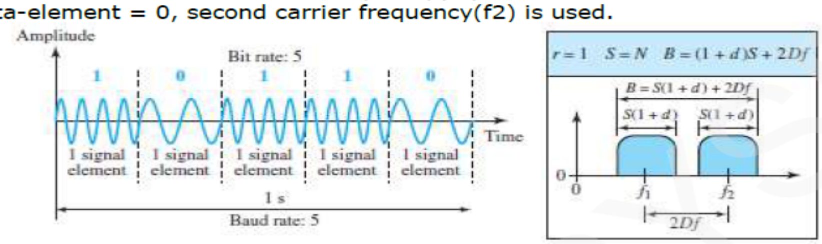


Figure 5.6 *Binary frequency shift keying*

2.3.3.1.1 Implementation

- Here, line coding method used = unipolar NRZ.
- Two implementations of BFSK: i) Coherent and ii) Non-Coherent.

Coherent BFSK	Non Coherent BFSK
The phase continues through the boundary of two signal-elements (Figure 5.7).	There may be discontinuity in the phase when one signal-element ends and the next begins.
This is implemented by using one voltage-controlled oscillator (VCO). VCO changes frequency according to the input voltage.	This is implemented by → treating BFSK as 2 ASK modulations and → using 2 carrier-frequencies
When the amplitude of NRZ signal = 0, the VCO keeps its regular frequency. When the amplitude of NRZ signal = 0, the VCO increases its frequency.	

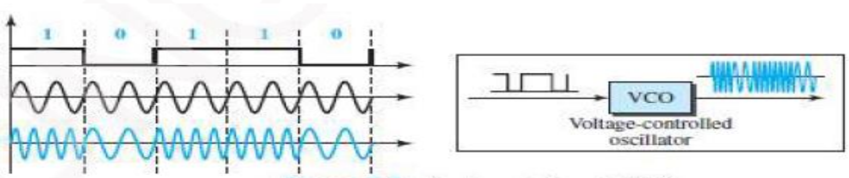


Figure 5.7 *Implementation of BFSK*

.3.3.1.2 Bandwidth for BFSK

FSK has two ASK signals, each with its own carrier-frequency f1 or f2. (Figure 5.6)

The bandwidth is given by

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

where 2Δf is the difference between f1 and f2,

8) Explain the working of QPSK. Generate the analog signal for the bit pattern 10110100 which transmits 2 bits/time period (10)

2.3.4.2 Quadrature PSK (QPSK)

- The scheme is called QPSK because it uses 2 separate BPSK modulations (Figure 5.11):
  1) First modulation is in-phase,
  2) Second modulation is quadrature (out-of-phase).
- A serial-to-parallel converter
  - → accepts the incoming bits
  - → sends first bit to first modulator and
  - → sends second bit to second modulator.
- The bit to each BPSK signal has one-half the frequency of the original signal.
- Advantages:
  1) Decreases the baud rate.
  2) Decreases the required bandwidth.

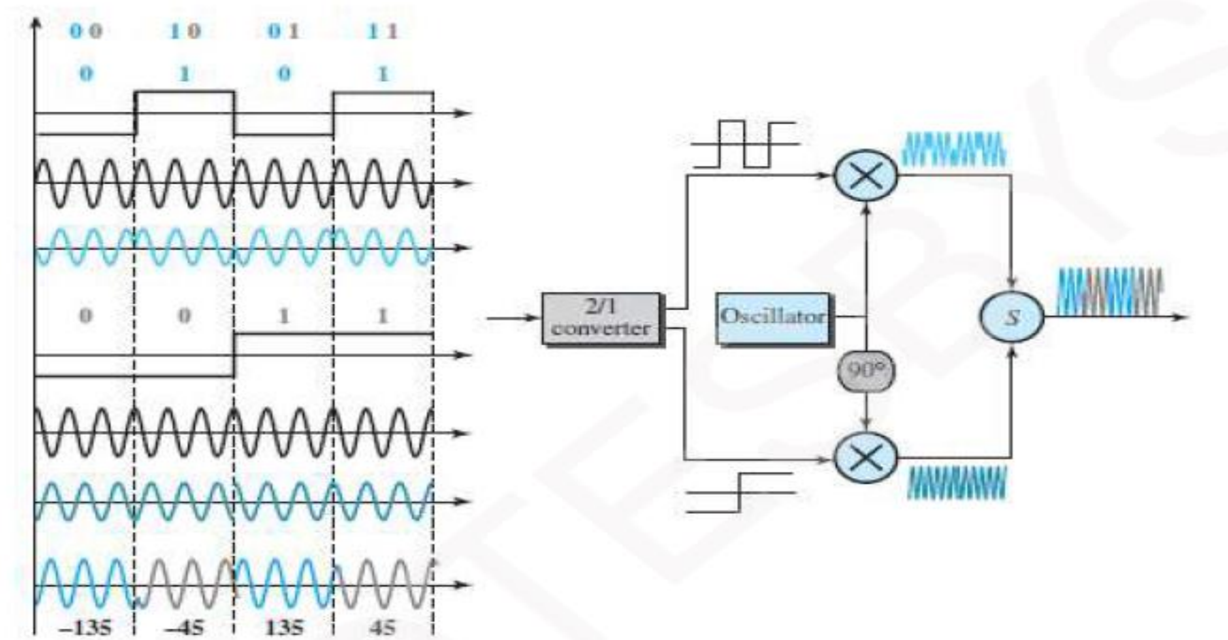


Figure 5.11 *QPSK and its implementation*

- As shown in Figure 5.11, the 2 composite-signals created by each multiplier are 2 sine waves with the same frequency, but different phases.
- When the 2 sine waves are added, the result is another sine wave, with 4 possible phases: 45°, -45°, 135°, and -135°.
- There are 4 kinds of signal-elements in the output signal (L=4), so we can send 2 bits per signal-element (r=2).

Now with the given bit pattern 10110100, taking 2 bits at a time, we will get

Bit Pattern	Modulated signal equation	Phases for modulated signal
00	-Acosw(t) - Asinw(t)	-135
01	-Acosw(t) + Asinw(t)	135
11	Acosw(t) + Asinw(t)	45
10	Acosw(t) - Asinw(t)	-45

So the resultant signal will follow the phase shift according to the above example (sequence is different)