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Internal Assessment Test III – May2023

Sub:	**Computer Networks**								**Sub Code:**	**22MCA14**
Date:	**24/05/2023**		**Duration:**	**90 min's**	**Max Marks:**	**50**	**Sem:**	**I**	**Branch:**	**MCA**

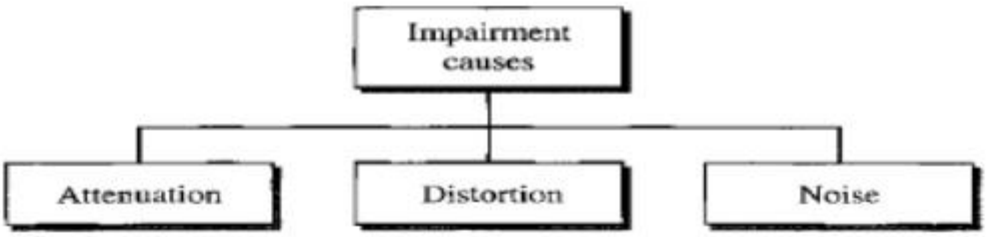
Note : Answer FIVE FULL Questions, choosing ONE full question from each Module

	PART I	MARKS	OBE	
			CO	RBT
1	a) Define Transmission impairments. Explain different causes of transmission impairment during signal transmission. **OR**	[10]	CO2	L2
2	a) Explain briefly about Shannon capacity and Nyquist bit rate for communication channel.	[10]	CO2	L2
3	**PART II** a) With a neat diagram, explain the most common technique to change analog signal to digital signal. **OR**	[10]	CO2	L3
4	a) With a neat diagram explain ASK,FSK and PSK.	[10]	CO2	L3

5	**PART III** a) Explain NRZ line coding scheme with example. Give the NRZ, NRZ-I and Manchester encodings for bit stream 0011111000. **OR**	[10]	CO2	L4
6	a) Explain parallel and serial transmission modes.	[10]	CO2	L2
7	PART IV a) What is Multiplexing? b) What is De Multiplexing? c) Explain Wavelength division Multiplexing technique. **OR**	[2] [2] [6]	CO3	L3
8	d) Describe about Direct Sequence spread spectrum.	[10]	CO3	L3
9	PART V a) Explain Virtual Circuit Network Set-up request and set-up acknowledgement with example. **OR**	[10]	C03	L3
10	a) Explain Circuit switching with example.	[10]	CO3	L2

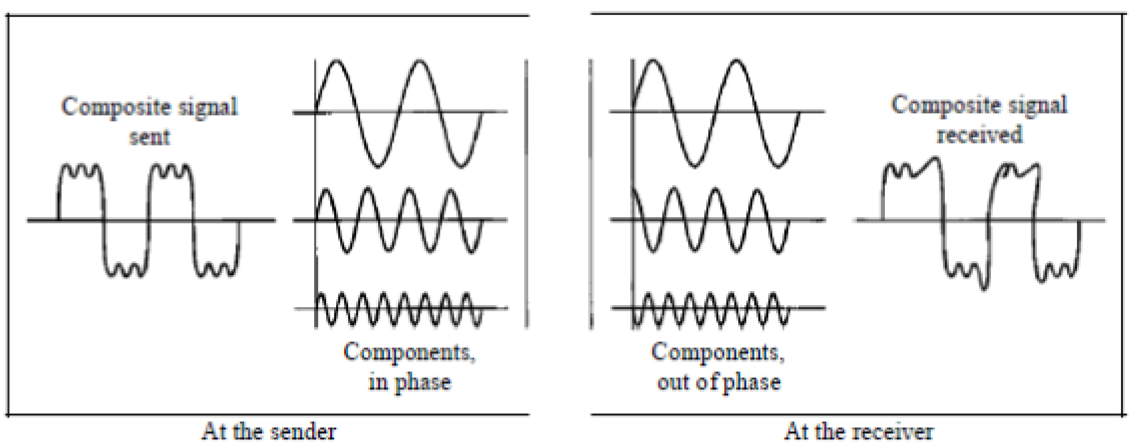
1) TRANSMISSION IMPAIRMENT Signals travel through transmission media, which are not perfect. The imperfection causes signal impairment. This means that the signal at the beginning of the medium is not the same as the signal at the end of the medium. What is sent is not what is received. Three causes of impairment are attenuation, distortion, and noise.

Causes of impairment



1. Attenuation :
   Attenuation means a loss of energy. When a signal, simple or composite, travels through a medium, it loses some of its energy in overcoming the resistance of the medium. That is why a wire carrying electric signals gets warm, if not hot, after a while. Some of the electrical energy in the signal is converted to heat. To compensate for this loss, amplifiers are used to amplify the signal. Attenuation is measured in terms of Decibels. The decibel (dB) measures the relative strengths of two signals or one signal at two different points. Note that the decibel is negative if a signal is attenuated and positive if a signal is amplified.
   dB=10log10 P2/P1
   Variables PI and P2 are the powers of a signal at points 1 and 2, respectively.
2. Distortion:
   Distortion means that the signal changes its form or shape. Distortion can occur in a composite signal made of different frequencies. Each signal component has its own propagation speed through a medium and, therefore, its own delay in arriving at the final destination. Differences in delay may create a difference in phase if the delay is not exactly the same as the period duration. In other words, 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 3.28 shows the effect of distortion on a composite signal.

   Figure 3.28 *Distortion*



3. Noise Noise is another cause of impairment. Several types of noise, such as thermal noise, induced noise, crosstalk, and impulse noise, may corrupt the signal. Thermal noise is the random motion of electrons in a wire which creates an extra signal not originally sent by the transmitter. Induced noise comes from sources such as motors and appliances. These devices act as a sending antenna, and the transmission medium acts as the receiving antenna. Crosstalk is the effect of one

wire on the other. One wire acts as a sending antenna and the other as the receiving antenna. Impulse noise is a spike (a signal with high energy in a very short time) that comes from power lines, lightning, and so on.
Signal-to-Noise Ratio (SNR) The signal-to-noise ratio is defined as
SNR= Average Signal power / 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 two powers, it is often described in decibel units, SNR dB, defined as
SNRdB = l0logl0 SNR

2) **DATA RATE LIMITS** A very important consideration in data communications is how fast we can send data, in bits per second. over a channel. Data rate depends on three factors:
1. The bandwidth available
2. The level of the signals we use
3. The quality of the channel (the level of noise)
Two theoretical formulas were developed to calculate the data rate: one by Nyquist for a noiseless channel. another by Shannon for a noisy channel.

Noiseless Channel: **Nyquist Bit Rate**
For a noiseless channel, the Nyquist bit rate formula defines the theoretical maximum bit rate

BitRate = 2 x bandwidth x 10g2 L

In this formula, bandwidth is the bandwidth of the channel, L is the number of signal levels used to represent data, and BitRate is the bit rate in bits per second. According to the formula, we might think that, given a specific bandwidth, we can have any bit rate we want by increasing the number of signa11eve1s. Although the idea is theoretically correct, practically there is a limit. When we increase the number of signal1eve1s, we impose a burden on the receiver. If the number of levels in a signal is just 2, the receiver can easily distinguish between a 0 and a 1. If the level of a signal is 64, the receiver must be very sophisticated to distinguish between 64 different levels. In other words, increasing the levels of a signal reduces the reliability of the system.

Noisy Channel: Shannon Capacity

In reality, we cannot have a noiseless channel; the channel is always noisy. In 1944, Claude Shannon introduced a formula, called the Shannon capacity, to determine the theoretical highest data rate for a noisy channel:

Capacity =bandwidth X log2 (1 +SNR)

In this formula, bandwidth is the bandwidth of the channel, SNR is the signal-to-noise ratio, and capacity is the capacity of the channel in bits per second. Note that in the Shannon formula there is no indication of the signal level, which means that no matter how many levels we have, we cannot achieve a data rate higher than the capacity of the channel. In other words, the formula defines a characteristic of the channel, not the method of transmission.

Bandwidth in Bits per Seconds The term bandwidth can also refer to the number of bits per second that a channel, a link, or even a network can transmit. For example, one can say the bandwidth of a Fast Ethernet network is a maximum of 100 Mbps. This means that this network can send 100 Mbps.

3) **ANALOG TO DIGITAL CONVERSION**
- An analog-signal may created by a microphone or camera.
- To change an analog-signal to digital-data, we use PCM (pulse code modulation).

• After the digital-data are created (digitization), then we convert the digital-data to a digital-signal.

2.1.1 PCM

• PCM is a technique used to change an analog signal to digital data (digitization). • PCM has encoder at the sender and decoder at the receiver.

• The encoder has 3 processes (Figure 4.21):

1) Sampling
2) Quantization &
3) Encoding.

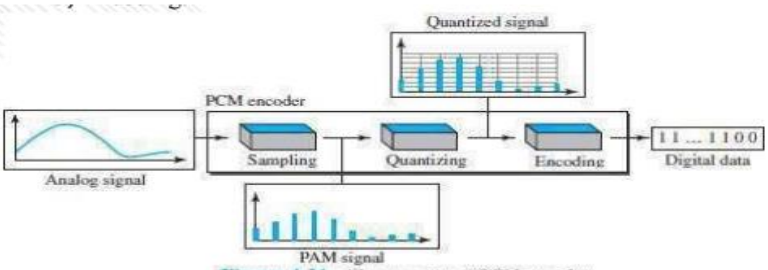


Figure 4.21 Components of PCM encoder

2.1.1.1 Sampling

• We convert the continuous time signal (analog) into the discrete time signal (digital). • Pulses from the analog-signal are sampled every T_s sec where T_s is the sample-interval or period.

• The inverse of the sampling-interval is called the sampling-frequency (or sampling-rate).

• Sampling-frequency is given by $f_s = 1/T_s$

• Three sampling methods (Figure 4.22):

1) Ideal Sampling

This method is difficult to implement.

2) Natural Sampling

A high-speed switch is turned ON for only the small period of time when the sampling occurs. The result is a sequence of samples that retains the shape of the analog-signal.

3) Flat Top Sampling The most common sampling method is sample and hold. Sample and hold method creates flat-top samples.

This method is sometimes referred to as *PAM* (pulse amplitude modulation).

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).

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).

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:

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

2.1.3.1 Original Signal Recovery

• PCM decoder is used for recovery of the original signal.

• Here is how it works (Figure 4.27):

1) The decoder first uses circuitry to convert the code words into a pulse that holds theamplitude until the next pulse.
2) Next, the staircase-signal is passed through a low-pass filter to smooth the staircase signal into an analog-signal.

• The filter has the same cut-off frequency as the original signal at the sender.

- If the signal is sampled at the Nyquist sampling-rate, then the original signal will be re-created.
- The maximum and minimum values of the original signal can be achieved by using amplification.

2.1.3.2 PCM Bandwidth

- The minimum bandwidth of a line-encoded signal is
- We substitute the value of N in above formula:
- When 1/r = 1 (for a NRZ or bipolar signal) and c = (1/2) (the average situation), the minimumbandwidth is
- This means the minimum bandwidth of the digital-signal is n_b times greater than the bandwidth of the analog signal.

2.1.3.3 Maximum Data Rate of a Channel

- The Nyquist theorem gives the data-rate of a channel as
- We can deduce above data-rate from the Nyquist sampling theorem by using the following arguments. 1) We assume that the available channel is low-pass with bandwidth B.

2) We assume that the digital-signal we want to send has L levels, where each level is a signalelement. This means $r = 1/\log_2 L$.

3) We first pass digital-signal through a low-pass filter to cut off the frequencies above B Hz. 4) We treat the resulting signal as an analog-signal and sample it at 2 x B samples per second and quantize it using L levels. 5) The resulting bit-rate is

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

2.1.3.4 Minimum Required Bandwidth

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

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

4)

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).

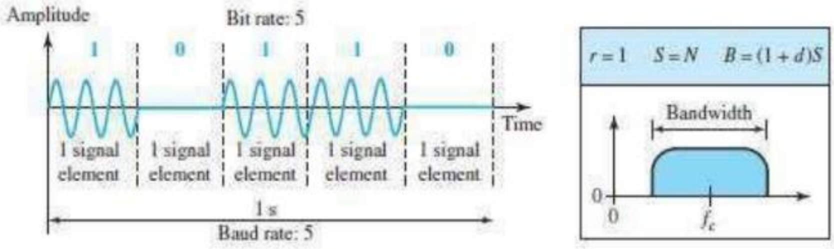


Figure 5.3 *Binary amplitude shift keying*

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.

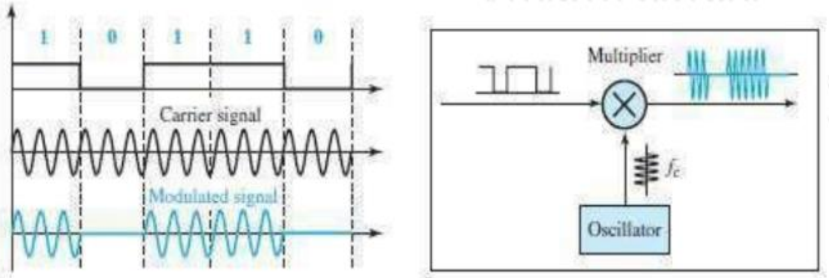


Figure 5.4 *Implementation of binary ASK*

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.

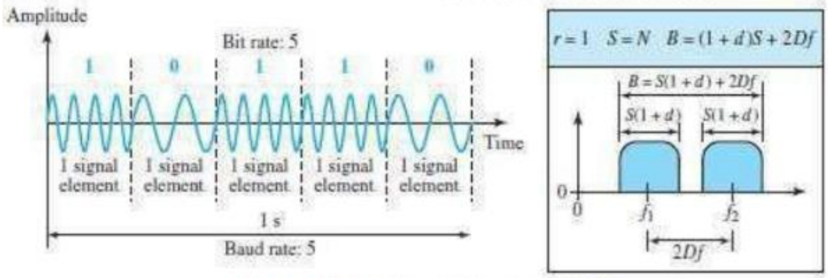


Figure 5.6 *Binary frequency shift keying*

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.	

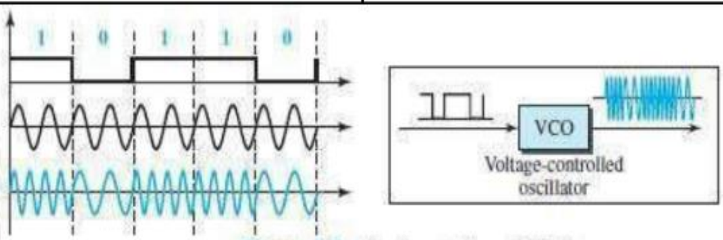


Figure 5.7 *Implementation of BFSK*

2.3.4 Phase Shift Keying (PSK)

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

2.3.4.1 Binary PSK (BPSK)

- We have only two signal-elements:
  1) First signal-element with a phase of 0°.
  2) Second signal-element with a phase of 180° (Figure 5.9).
- ASK vs. PSK
   In ASK, the criterion for bit detection is the amplitude of the signal. In PSK, the criterion for bit detection is the phase.
- Advantages:
  1) PSK is less susceptible to noise than ASK.
  2) PSK is superior to FSK because we do not need 2 carrier-frequencies.
- Disadvantage:
  1) PSK is limited by the ability of the equipment to distinguish small differences in phase.

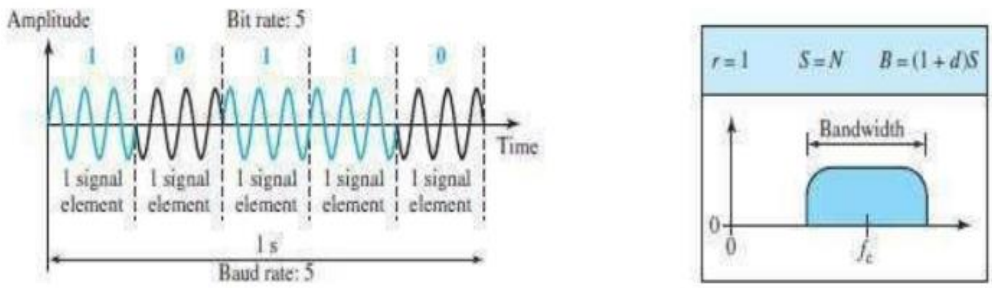


Figure 5.9 *Binary phase shift keying*

2.3.4.1.1 Implementation

- The implementation of BPSK is as simple as that for ASK. (Figure 5.10).
- The signal-element with phase 180° can be seen as the complement of the signal-element with phase 0°.
- Here, line coding method used: polar NRZ.
- The polar NRZ signal is multiplied by the carrier-frequency coming from an oscillator.
  1) When data-element = 1, the phase starts at 0°.
  2) When data-element = 0, the phase starts at 180°.

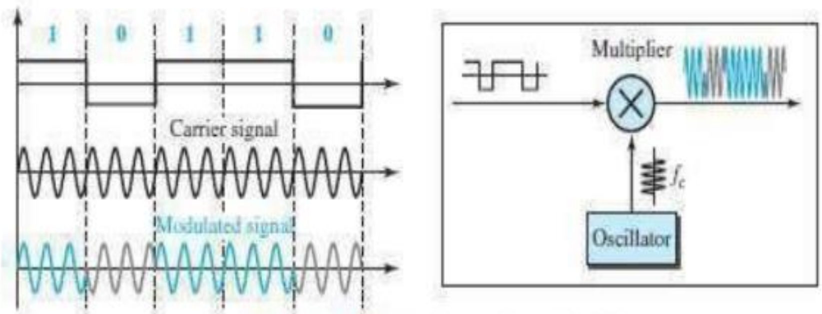


Figure 5.10 *Implementation of BASK*

2.3.4.1.2 Bandwidth for BPSK

- The bandwidth is the same as that for BASK, but less than that for BFSK. (Figure 5.9b)
- No bandwidth is wasted for separating 2 carrier-signals.

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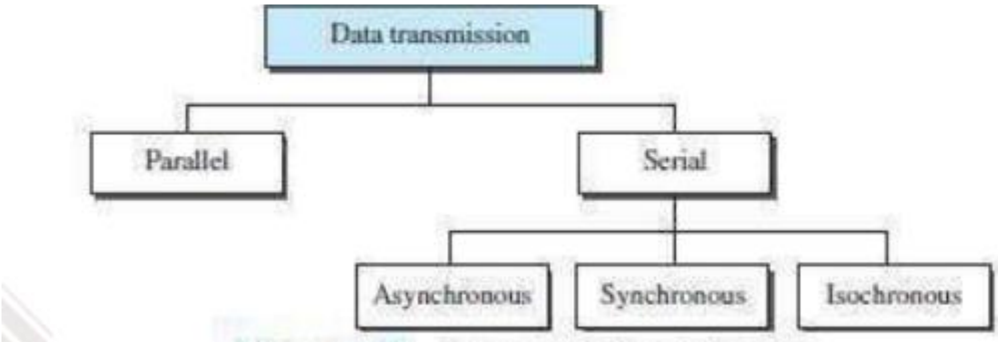
6) TRANSMISSION MODES

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



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

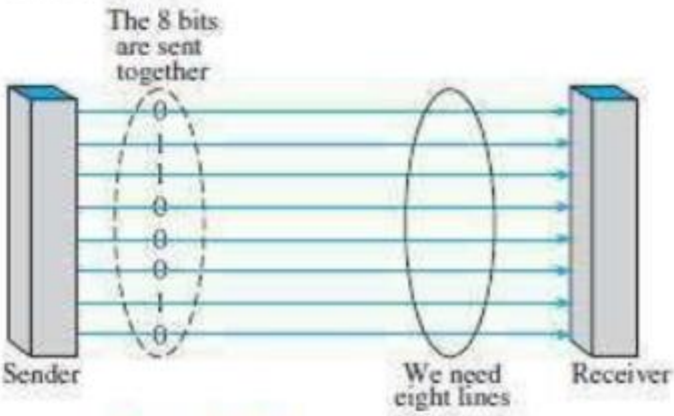


Figure 4.32 *Parallel transmission*

• Advantage:
1) Speed: Parallel transmission can increase the transfer speed by a factor of n over serialtransmission.
• Disadvantage:
1) Cost: Parallel transmission requires n communication lines just to transmit the data-stream. Because this is expensive, parallel transmission is usually limited to short distances.
2.2.2 SERIAL TRANSMISSION
• One bit is sent with each clock-tick using only a single link (Figure 4.33).

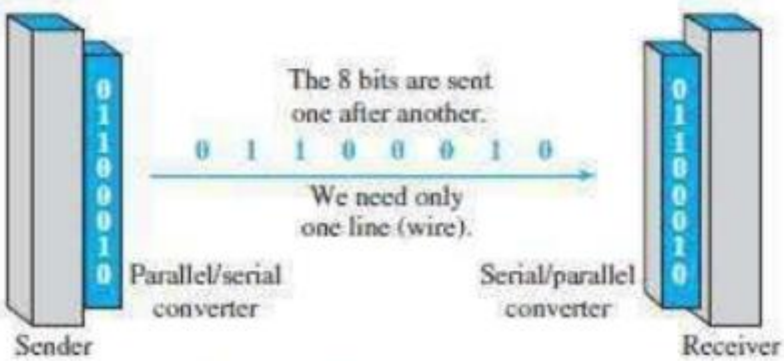


Figure 4.33 *Serial transmission*

• Advantage: 1) Cost: Serial transmission reduces cost of transmission over parallel by a factor of n.
• Disadvantage:
1) Since communication within devices is parallel, following 2 converters are required at interface: i) Parallel-to-serial converter ii) Serial-to-parallel converter
• Three types of serial transmission: asynchronous, synchronous, and isochronous.
2.2.2.1 Asynchronous Transmission
• Asynchronous transmission is so named because the timing of a signal is not important (Figure 4.34).

• Prior to data transfer, both sender & receiver agree on pattern of information to be exchanged.
• Normally, patterns are based on grouping the bit-stream into bytes.
• The sender transmits each group to the link without regard to a timer.
• As long as those patterns are followed, the receiver can retrieve the info. without regard to a timer.
• There may be a gap between bytes.
• We send
→ 1 start bit (0) at the beginning of each byte → 1
stop bit (1) at the end of each byte.
• Start bit alerts the receiver to the arrival of a new group.
Stop bit lets the receiver know that the byte is finished.
• Here, the term asynchronous means "asynchronous at the byte level".
However, the bits are still synchronized & bit-durations are the same.

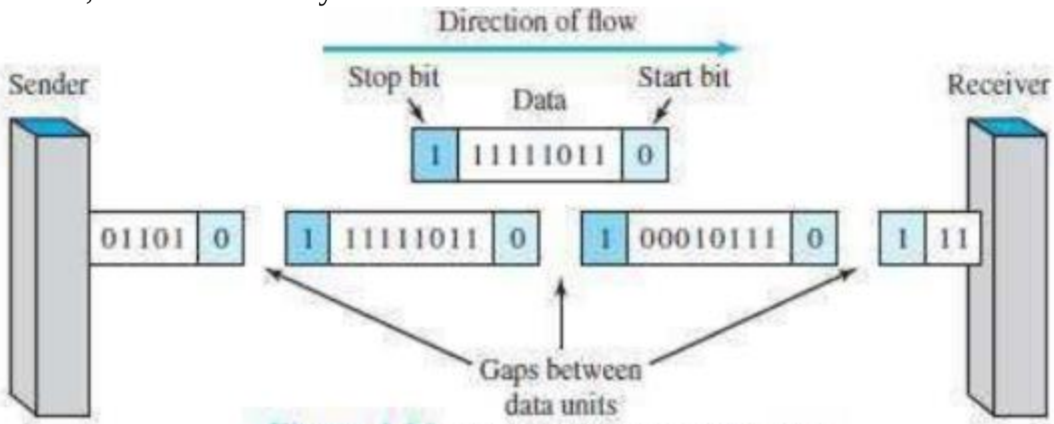


Figure 4.34 *Asynchronous transmission*

• Disadvantage:
1) Slower than synchronous transmission. (Because of stop bit, start bit and gaps)
• Advantages:
1) Cheap & effective.
2) Useful for low-speed communication.
2.2.2.2 Synchronous Transmission
• We send bits one after another without start or stop bits or gaps (Figure 4.35).
• The receiver is responsible for grouping the bits.
• The bit-stream is combined into longer "frames," which may contain multiple bytes.
• If the sender wants to send data in separate bursts, the gaps b/w bursts must be filled with a special sequence of 0s & 1s (that means idle).

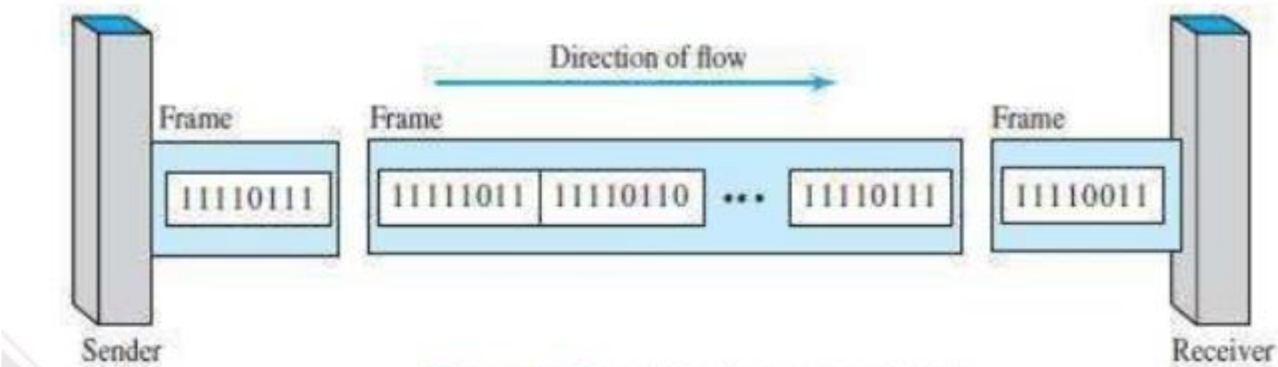
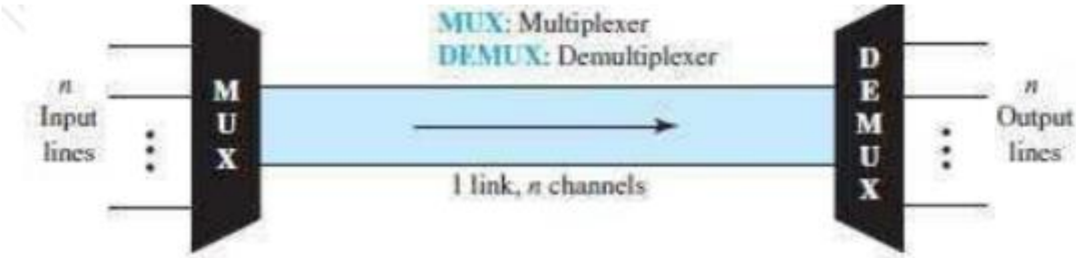


Figure 4.35 *Synchronous transmission*

• Advantages:
1) Speed: Faster than asynchronous transmission. („.‟ of no stop bit, start bit and gaps).
2) Useful for high-speed applications such as transmission of data from one computer to another.
2.2.2.3 Isochronous
• Synchronization between characters is not enough; the entire stream of bits must be synchronized.
• The isochronous transmission guarantees that the data arrive at a fixed rate.
• In real-time audio/video, jitter is not acceptable. Therefore, synchronous transmission fails.
• For example: TV images are broadcast at the rate of 30 images per second. The images must beviewed at the

same rate.

7) MULTIPLEXING
• When bandwidth of a medium is greater than bandwidth needs of the devices, the link can be shared.
• Multiplexing allows simultaneous transmission of multiple signals across a single data-link (Fig 4.21).
• The traffic increases, as data/telecommunications use increases.
• We can accommodate this increase by
→ adding individual links, each time a new channel is needed or →
installing higher-bandwidth links to carry multiple signals.
• Today's technology includes high-bandwidth media such as optical-fiber and satellite microwaves.
• Each has a bandwidth far in excess of that needed for the average transmission-signal.
• If the bandwidth of a link is greater than the bandwidth needs of the devices connected to it, the bandwidth is wasted.
• An efficient system maximizes the utilization of all resources; bandwidth is one of the most precious resources we have in data communications.



• In a multiplexed-system, „n" lines share the bandwidth of one link.
• MUX combines transmission-streams from different input-lines into a single stream (many-to-one).
• At the receiving-end, that stream is fed into a demultiplexer (DEMUX).
• DEMUX
→ separates the stream back into its component-transmissions (one-to-many) and → directs the transmission-streams to different output-lines.
• Link vs. Channel:
1) The link refers to the physical path.
2) The channel refers to the portion of a link that carries a transmission between a given pair oflines. One link can have many channels.
• Three multiplexing techniques (Figure 6.2):
1) Frequency-division multiplexing (FDM)
2) Wavelength-division multiplexing (WDM) and
3)Time-division multiplexing (TDM).

Wavelength Division Multiplexing (WDM)
• WDM is an analog multiplexing technique that combines analog signals .
• WDM is designed to use the high-data-rate capability of fiber optical-cable.
• The data-rate of optical-cable is higher than the data-rate of metallic-cable.
• Using an optical-cable for one single line wastes the available bandwidth.
• Multiplexing allows combining several lines into one line.
• WDM is same as FDM with 2 exceptions:
1) Multiplexing & demultiplexing involve optical-signals transmitted through optical-cable. 2) The frequencies are very high.

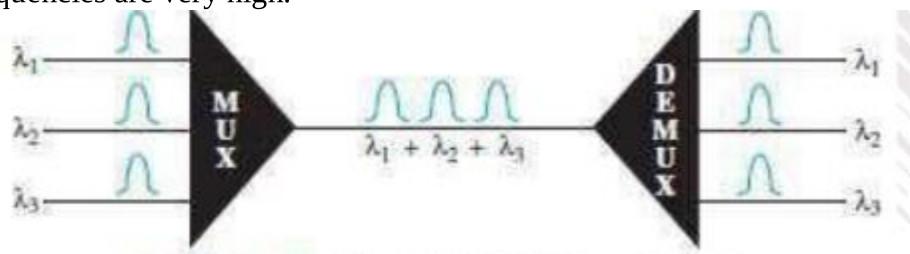


Figure 6.10 *Wavelength-division multiplexing*

• Here is how it works (Figure 6.11):
A multiplexer combines several narrow-bands of light into a

wider-band of light. A demultiplexer divides a wider-band
of light into several narrow-bands of light. A prism is used
for combining and splitting of light sources
A prism bends a beam of light based on → angle of incidence
and → frequency.

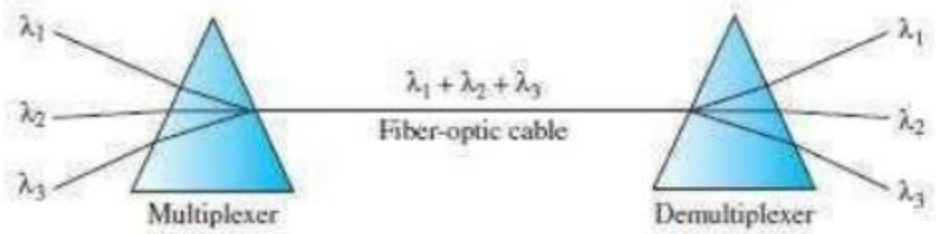


Figure 6.11 *Prisms in wavelength-division multiplexing and demultiplexing*

• Applications of WDM:
1) SONET network: Multiple optical-fiber lines can be multiplexed and demultiplexed.
2) Dense WDM (DWDM) can multiplex a very large number of channels by spacing channels very close to one another. DWDM achieves even greater efficiency.

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8)
Direct Sequence Spread Spectrum (DSSS)
• This technique expands the bandwidth of the original signal.
• Each data-bit is replaced with „n" bits using a spreading-code.
• Each bit is assigned a code of „n" bits called chips.
• The chip-rate is „n" times that of the data-bit .

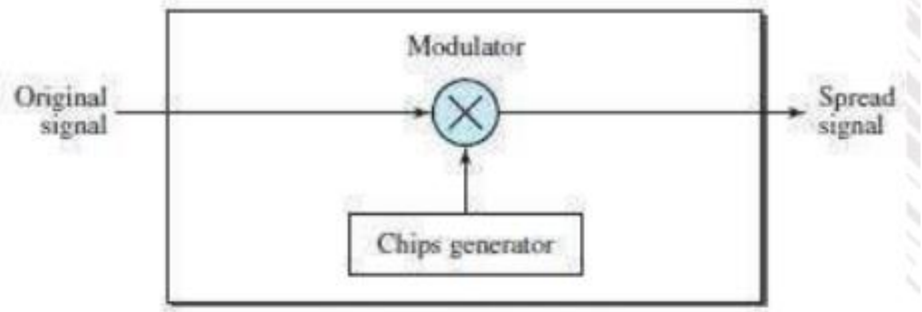

Figure 6.32 *DSSS*

For example (Figure 6.33):

Consider the Barker sequence used in a wireless LAN. Here n =11.
Assume that the original signal and the chips in the chip-generator use polar NRZ encoding. The spreading-code is 11 chips having the pattern 10110111000.
If the original signal-rate is N, the rate of the spread signal is 1/N.
This means that the required bandwidth for the spread signal is 11 times larger than the bandwidth of the original signal.

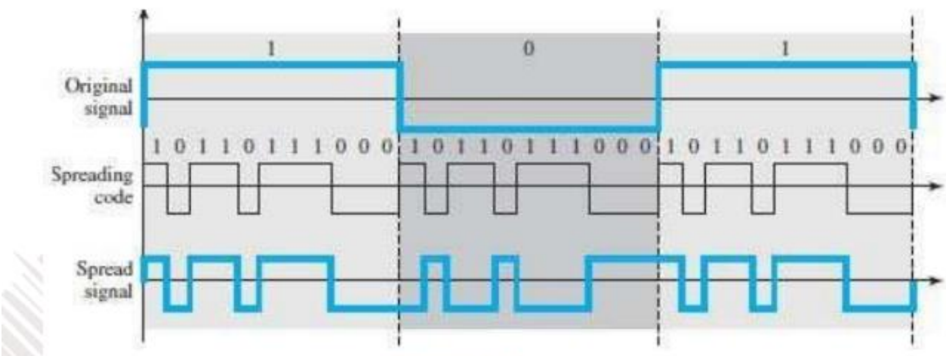

Figure 6.33 *DSSS example*

The spread signal can provide privacy if the attacker does not know the code.

• It can also provide immunity against interference if each station uses a different code.
2.5.2.1 Bandwidth Sharing
• Can we share a bandwidth in DSSS?
• The answer is no and yes.
1) If we use a spreading-code that spreads signals that cannot be combined and separated, we cannot share a bandwidth.
For example:
Some wireless LANs use DSSS and the spread bandwidth cannot be shared.
2) If we use a special spreading-code that spreads signals that can be combined and separated, we can share a bandwidth.
For example:
Cellular telephony uses DSSS and the spread bandwidth is shared b/w several users.

---

9)
2.8.2 Virtual Circuit Network (VCN) •
This is similar to telephone system.
• A virtual-circuit network is a combination of circuit-switched-network and datagram-network.
• Five characteristics of VCN:
1) As in a circuit-switched-network, there are setup & teardown phases in addition to the data transfer phase.
2) As in a circuit-switched-network, resources can be allocated during the setup phase.
As in a datagram-network, resources can also be allocated on-demand.
3) As in a datagram-network, data is divided into packets.
Each packet carries an address in the header.
However, the address in the header has local jurisdiction, not end-to-end jurisdiction. 4) As in a circuit-switched-network, all packets follow the same path established during the connection.
5) A virtual-circuit network is implemented in the data link layer.
A circuit-switched-network is implemented in the physical layer.
A datagram-network is implemented in the network layer.

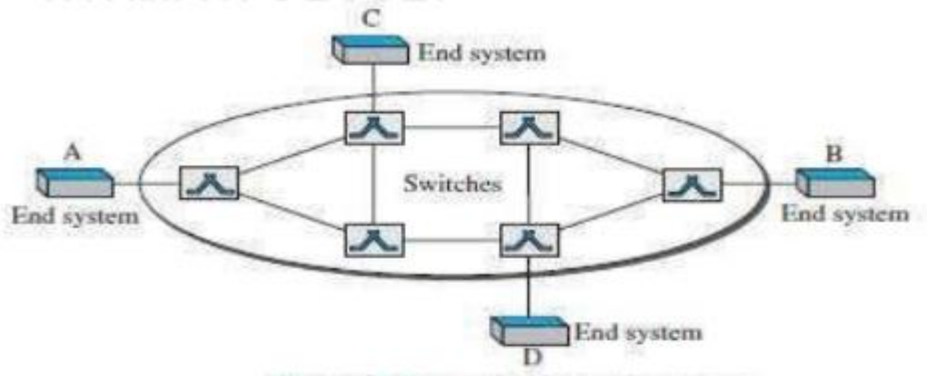

Figure 8.10 Virtual-circuit network

The Figure 8.10 is an example of a virtual-circuit network.
The network has switches that allow traffic from sources to destinations.
A source or destination can be a computer, packet switch, bridge, or any other device that connects other networks.
2.8.2.1 Addressing
• Two types of addressing: 1) Global and 2) Local (virtual-circuit identifier).
1) Global Addressing
A source or a destination needs to have a global address.
Global address is an address that can be unique in the scope of the network or internationally if the network is part of an international network.
2) Virtual Circuit Identifier
The identifier used for data-transfer is called the virtual-circuit identifier (VCI). A VCI, unlike a global address, is a small number that has only switch scope. VCI is used by a frame between two switches.
When a frame arrives at a switch, it has a VCI.
When the frame leaves, it has a different VCI.

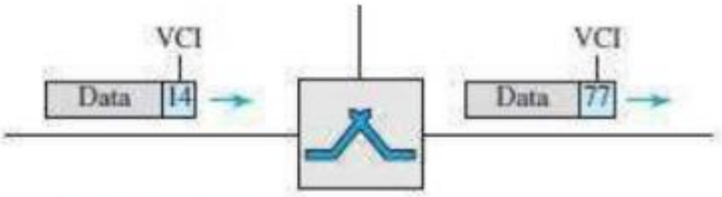

Figure 8.11 *Virtual-circuit identifier*

Three Phases

• A source and destination need to go through 3 phases: setup, data-transfer, and teardown.

1) In setup phase, the source and destination use their global addresses to help switchesmake table entries for the connection.
2) In the teardown phase, the source and destination inform the switches to delete thecorresponding entry.
3) Data-transfer occurs between these 2 phases.

2.8.2.2.2 Setup Phase

- A switch creates an entry for a virtual-circuit.
- For example, suppose source A needs to create a virtual-circuit to B. • Two steps are required: 1) Setup-request and 2) Acknowledgment.

1) Setup Request

⍰ A setup-request frame is sent from the source to the destination (Figure 8.14).

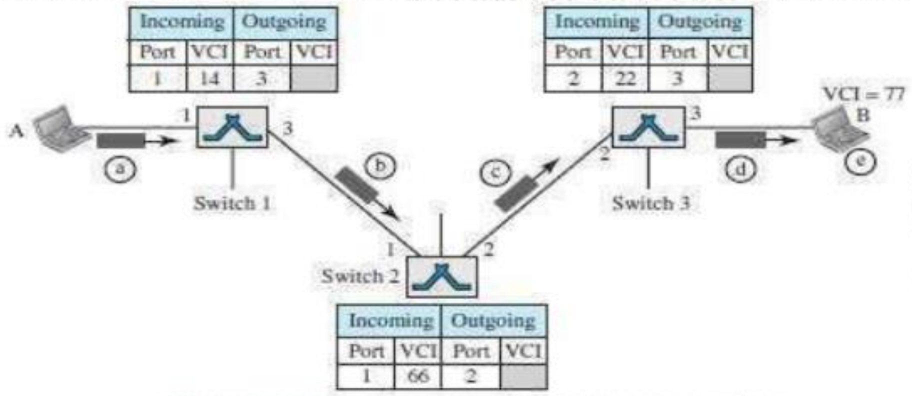

Figure 8.14 *Setup request in a virtual-circuit network*

⍰ Following events occurs:

a) Source-A sends a setup-frame to switch-1.
b) Switch-1 receives the setup-frame.

¤ Switch-1 knows that a frame going from A to B goes out through port 3. ¤ The switch-1 has a routing table. ¤ The switch
→ creates an entry in its table for this virtual-circuit
→ is only able to fill 3 of the 4 columns.

The switch
→ assigns the incoming port (1) and
→ chooses an available incoming-VCI (14) and the outgoing-port (3).
→does not yet know the outgoing VCI, which will be found during the acknowledgment step.
¤ The switch then forwards the frame through port-3 to switch-2.
c) Switch-2 receives the setup-request frame.
¤ The same events happen here as at switch-1.

¤ Three columns of the table are completed: In this case, incoming port (1), incoming-VCI (66), and outgoing port (2).

d) Switch-3 receives the setup-request frame.

¤ Again, three columns are completed: incoming port (2), incoming-VCI (22), and outgoing-port (3).
e) Destination-B

→ receives the setup-frame
→ assigns a VCI to the incoming frames that come from A, in this case 77. ¤
This VCI lets the destination know that the frames come from A, and no other sources.

2) **Acknowledgment**
A special frame, called the acknowledgment-frame, completes the entries in the switchingtables (Figure 8.15).

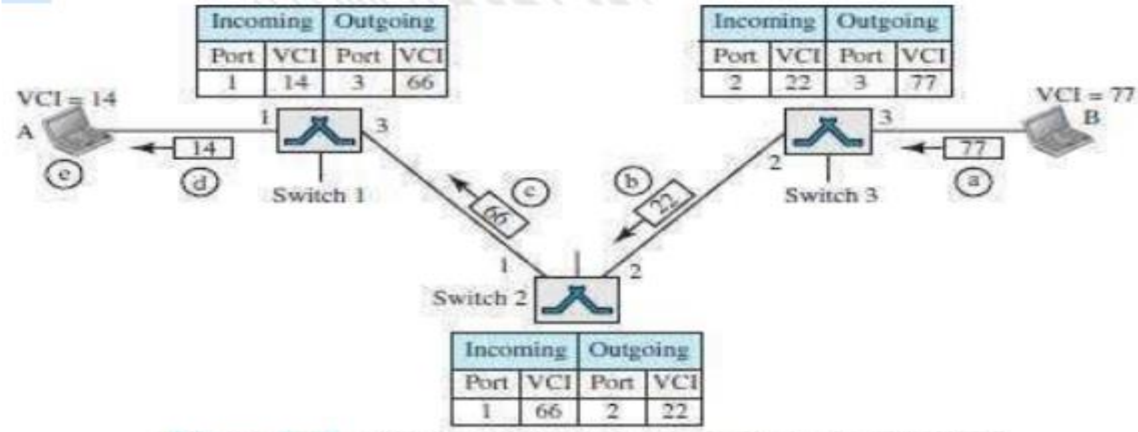

Figure 8.15 *Setup acknowledgment in a virtual-circuit network*

a) The destination sends an acknowledgment to switch-3.
¤ The acknowledgment carries the global source and destination-addresses so the switch knows which entry in the table is to be completed.
¤ The frame also carries VCI 77, chosen by the destination as the incoming-VCI for frames from A.
¤ Switch 3 uses this VCI to complete the outgoing VCI column for this entry.
b) Switch 3 sends an acknowledgment to switch-2 that contains its incoming-VCI in the table, chosen in the previous step.
¤ Switch-2 uses this as the outgoing VCI in the table.
c) Switch-2 sends an acknowledgment to switch-1 that contains its incoming-VCI in the table, chosen in the previous step.
¤ Switch-1 uses this as the outgoing VCI in the table.
d) Finally switch-1 sends an acknowledgment to source-A that contains its incoming-VCI in the table, chosen in the previous step.

e) The source uses this as the outgoing VCI for the data-frames to be sent to destination-B.

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10)
CIRCUIT SWITCHED NETWORK
• This is similar to telephone system.
• Fixed path (connection) is established between a source and a destination prior to the transfer ofpackets.
• A circuit-switched-network consists of a set of switches connected by physical-links (Figure 8.3).
• A connection between 2 stations is a dedicated-path made of one or more links.
• However, each connection uses only one dedicated-channel on each link.
• Normally, each link is divided into „n" channels by using FDM or TDM.
• The resources need to be reserved during the setup phase.

The resources remain dedicated for the entire duration of data transfer until the teardown phase.

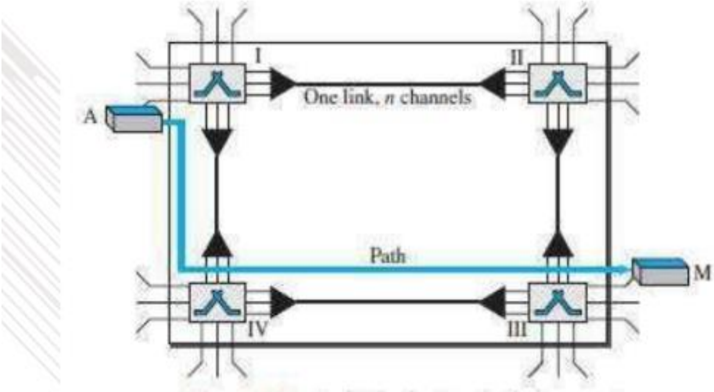

Figure 8.3 A trivial circuit-switched network

• The virtual-circuit setup procedure
→ first determines a path through the network &
→ sets parameters in the switches by exchanging connect-request & connect-confirm messages

• If a switch does not have enough resources to set up a virtual circuit, the switch responds with a connect-reject message and the setup procedure fails (Figure 7.15).

• A connection-release procedure may also be required to terminate the connection.
2.7.1 Three Phases
• The communication requires 3 phases: 1) Connection-setup
2) Data-transfer 3)
Connection teardown.

1) Setup Phase
Before the 2 parties can communicate, a dedicated-circuit needs to be established.
Normally, the end-systems are connected through dedicated-lines to the switches.
So, connection-setup means creating dedicated-channels between the switches.
For ex: Assume system-A needs to connect to system-M. For this, following events occur:
i) System-A sends a setup-request to switch-I.
ii) Switch-I finds a channel between itself and switch-IV that can be dedicated for this purpose.
iii) Switch-I then sends the request to switch-IV, which finds a dedicated-channel between itself and switch-III.
iv) Switch-III informs system-M of system-A's intention at this time.
v) Finally, an acknowledgment from system-M needs to be sent in the opposite direction to system-A.
Only after system A receives this acknowledgment is the connection established.
2) Data Transfer Phase
After the establishment of the dedicated-circuit (channels), the two parties can transfer data. 3)
Teardown Phase
When one of the parties needs to disconnect, a signal is sent to each switch to release the resources.
2.7.2 Efficiency
• Circuit-switched-networks are inefficient when compared to other two types of networks because 1) Resources are allocated during the entire duration of the connection. 2) These resources are unavailable to other connections.
2.7.3 Delay
• Circuit-switched-networks have minimum delay when compared to other two types of networks • During data-transfer,

1) The data are not delayed at each switch.
2) The resources are allocated for the duration of the connection.

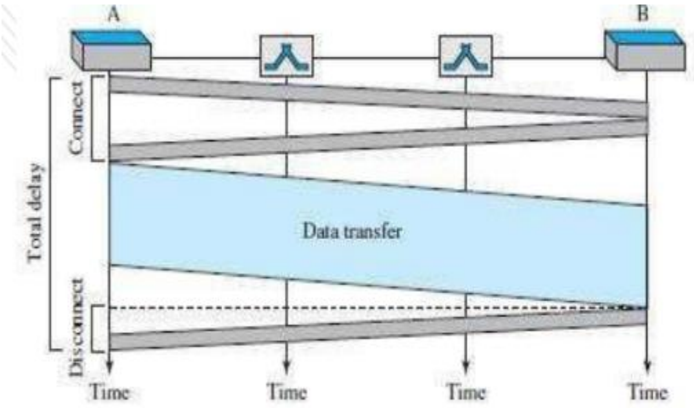

Figure 8.6 *Delay in a circuit-switched network*

As in the above figure, there is no waiting time at each switch.
The total delay is the time needed to
1) Create the connection 2) Transfer-data and
3) Disconnect the circuit.