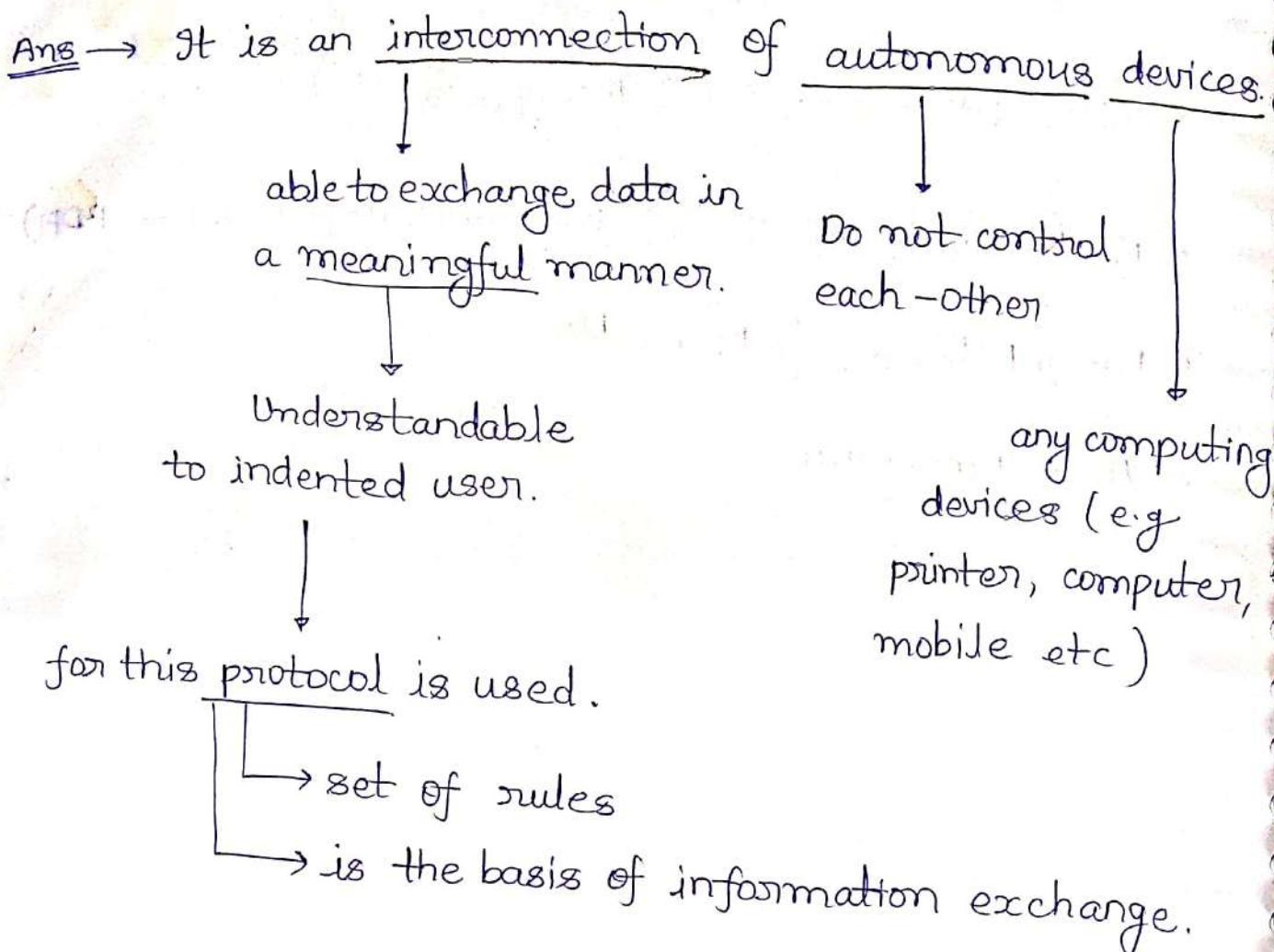


Computer Network

Date - 05/08/19

- * 5 Credit course (3 Theory, 2 Lab)
- * Beij's Guide of Socket Programming (Online PDF)
- * Larry Peterson → Bruce Davie
- * A. S. Tanabuam

* What is Computer Network?



* Network Facilitates →

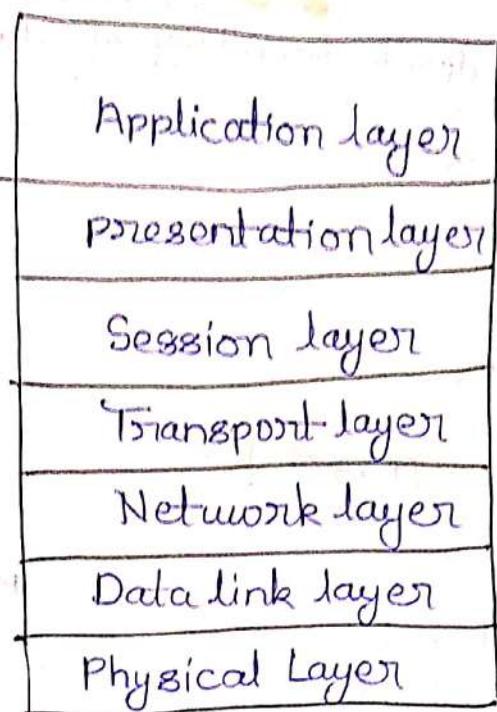
- (1) Data Sharing
- (2) Hardware sharing
- (3) Resource Sharing (Software or Hardware resources)

* Network reduces the cost of deployment also.

* How to visualize a network?

for this, OSI (Open System Interconnection) model is used which contains seven layers namely Application, presentation, session, transport, Network, Datalink, physical, given by ISO in 1984.

OSI model



* Responsibilities/Services of physical layer? (802x mod)

(1) To convert bits into signals. (Digital encoder/Analog Modulator)

(2) Define the medium of communication.

telephone → twisted pair

Computer → Cat 6 cable

WLAN → 2.4 GHz / 5 GHz band

(3) Interface for connection.

* To fulfill these responsibilities, protocols is used.

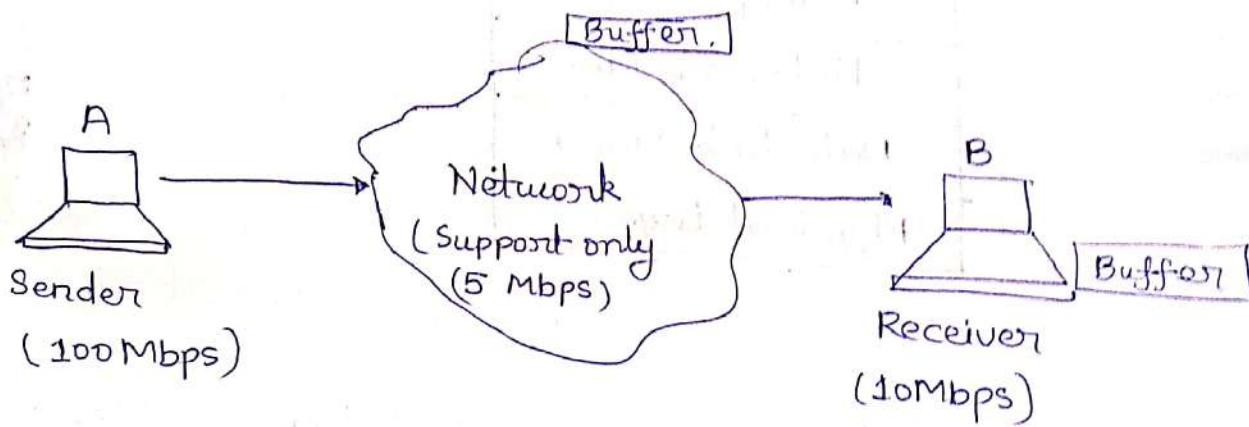
Cat 6 → 100 Gbps

Cat 5 → 500 Mbps

802.3 Lan → Manchester encoding is used.

* Services / Responsibilities of Data Link Layer (DLL)

- (1) Framing → To decide boundary of data packets.
- (2) Flow control
- (3) Error control



- * Network has only 5 Mbps data speed, so it will accumulate 95 Mbps data in its buffer and after some point of time, the network buffer become full and no further packets will be accepted.
- * And same thing also happens at receiver end as it also has the capacity of only 10 Mbps.
- * That's why, flow control protocols are necessary to limit the data rate.

$$\min [\max (\text{Network}, \text{Receiver})]$$

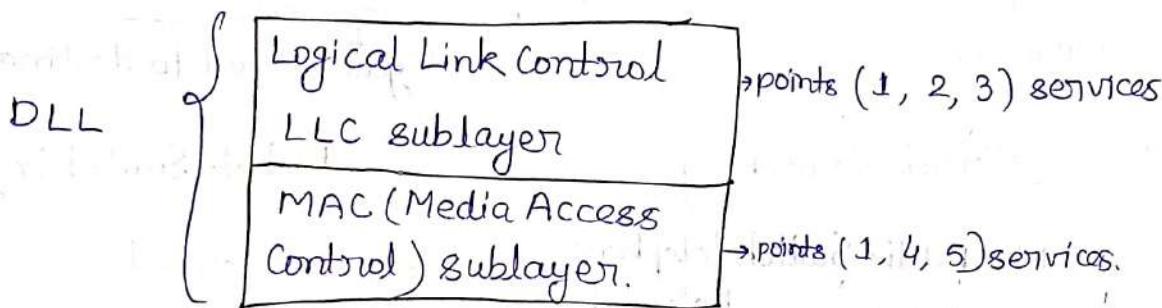
$$= \min (5 \text{ Mbps}, 10 \text{ Mbps})$$

$$= 5 \text{ Mbps}$$

* Error control
↓
Error detection and congestion protocols are used at Data Link Layer.

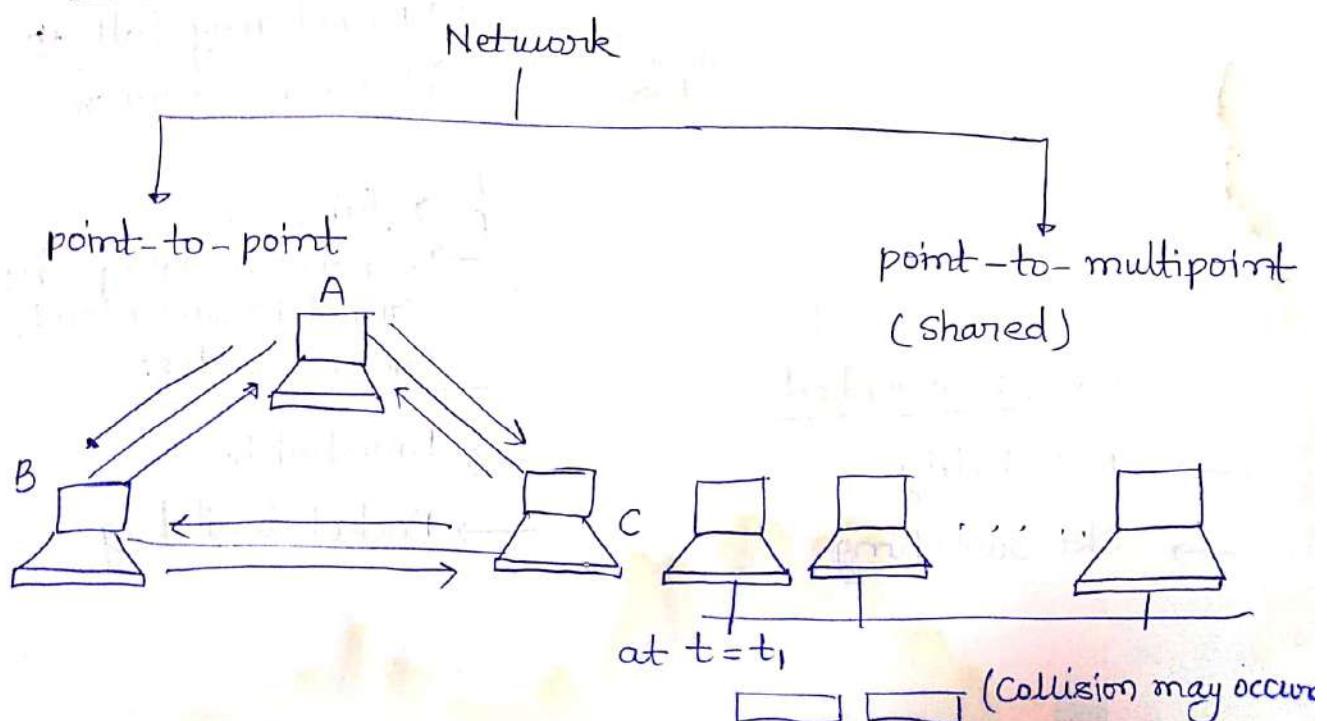
(4) Addressing (48 bits Ethernet address)

↓
also known as MAC address / Global address



(5) Medium Access Protocols.

* Types of Network



- * Bluetooth, WiMax etc are wireless sharing medium.
- * MAC layer is required only in case of local area network (Sharing Medium)

802.2 → for LLC.

- * For error control, CRC (cyclic Redundancy Check) and checksum is used.

Responsibility/Services of Network layer

- (1) To provide connectionless based effort data delivery services.
not guaranteed to destination

Circuit Switching

e.g. PSTN → Public Switch Telephone Network
(Land line telephone)

Packet Switching

- e.g. Internet
(Postal System)
- delay
 - loss of packets.
 - Packet may follow different routes.
 - fairness.
 - Sequence of data packets can not be guaranteed.

connection less.

Connection oriented

→ Reliability

→ Ckt Switching

Connection less

→ Unreliable

→ Packet Switching

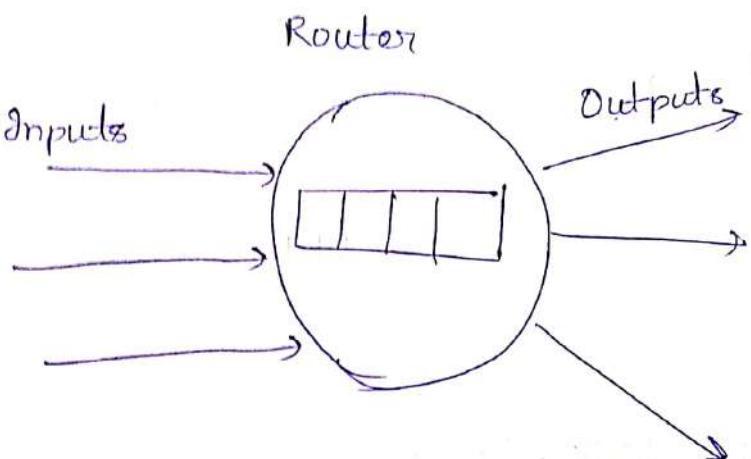
* Ckt switching

* Connection orientation can be achieved over packet switched networks as well as circuit switched networks.

* Connectionless means always Packet switching.

(2) Routing

→ RIP
→ OSPF



(3) Logical address →

IPv4 and IPv6

↓
32-bits
address

↓
128-bits
address.

(4) It is also responsible for congestion feedback.

* Transport Layer →

- It provides connection oriented as well as connectionless services.
(Reliable)
(Unreliable)
- Logical Addressing (Port Number - 16 bits)
0 to 2^{16}
- Congestion control.

TCP → Connection Oriented

UDP → Connectionless

* Responsibilities / Services of Session Layer →

- (1) Dialog Control
- (2) Session Management
 - Initialization
 - Maintenance
 - termination.
- (3) Synchronization of the communication.

* Responsibilities of presentation layer →

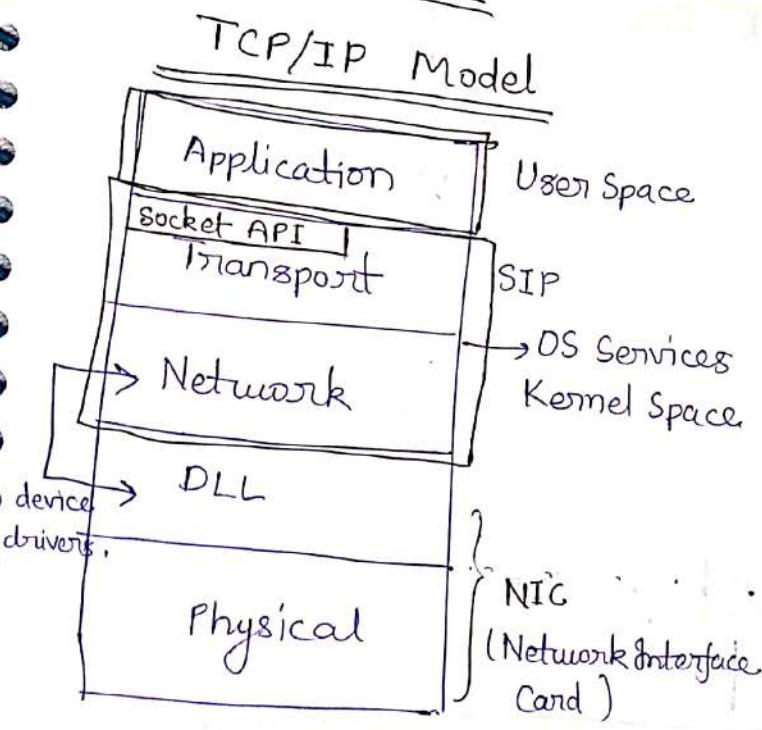
- (1) Syntax of data
- (2) Semantics of data
- (3) Encoding

Services of application layer

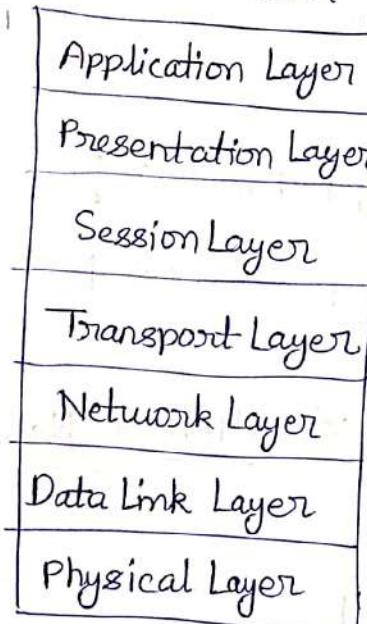
19A 10308

→ DNS, HTTP, Telnet, FTP all protocols runs on application layer.

Working Model



OSI model



htons → Host to Network short (16 byte)

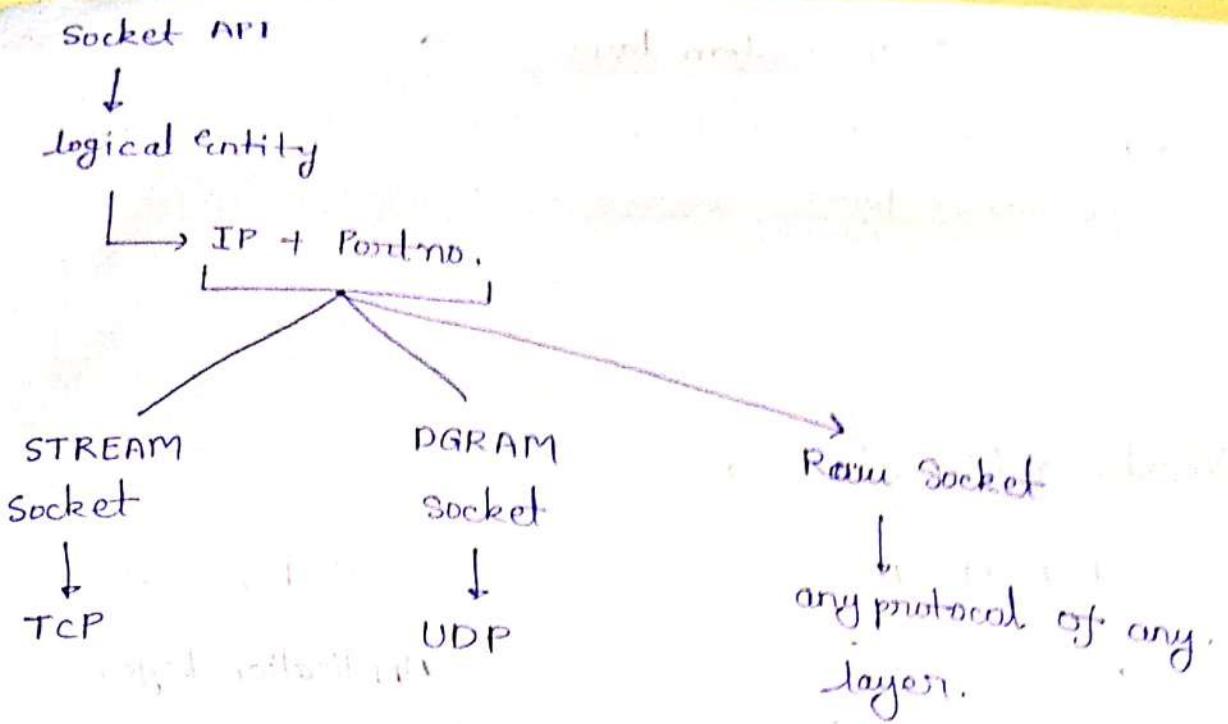
htonl → Host to Network long (32 byte)

ntohs → Network to Host short

ntohl → Network to Host long

System calls.

API → Application Program Interface



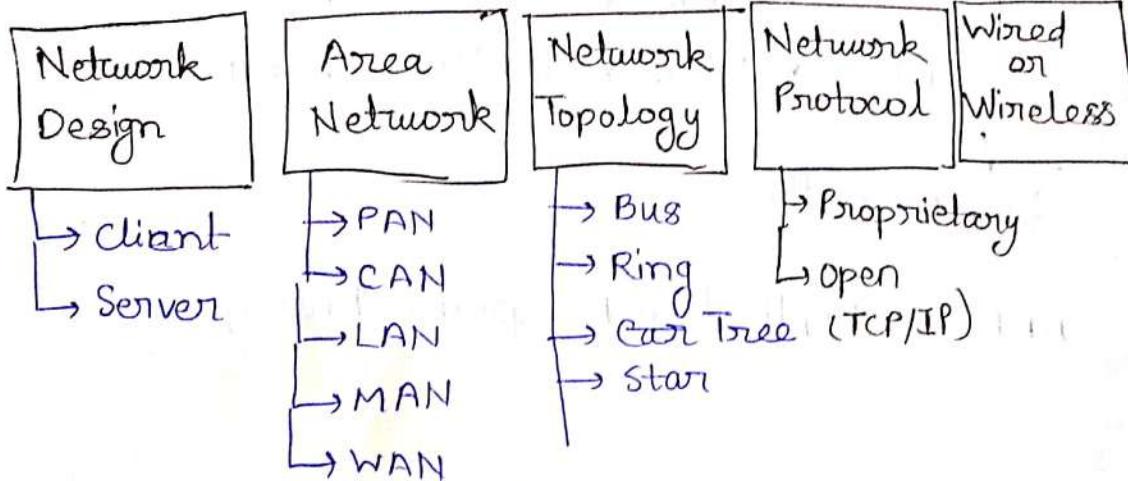
`char* tcp1(char *s)`

`char* tcp2(char *s)`

* Classification of Networks

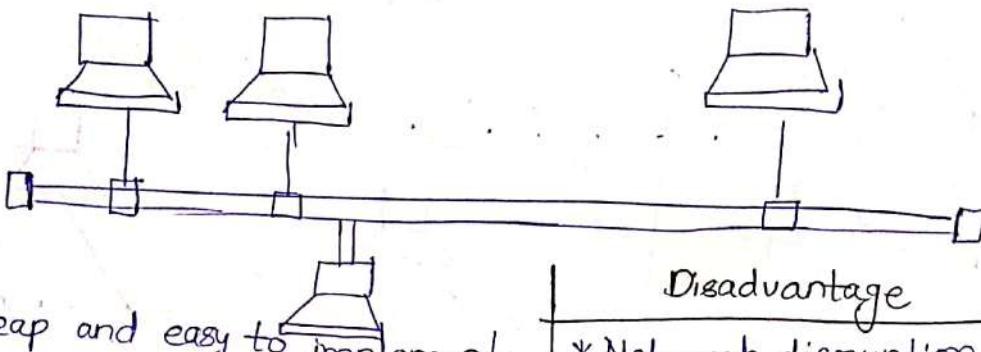


- PAN
- LAN
- MAN
- WAN



Topologies for Network deployment

(1) BUS topologies



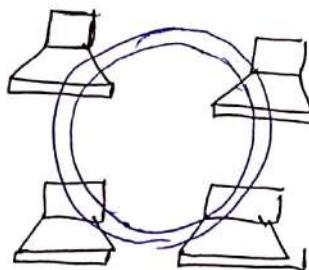
- * Cheap and easy to implement.
- * Requires less cable.
- * Does not use any specialized network equipment.

(2) RING topologies

- * Cable faults are easily located, making troubleshooting easier.

Disadvantage

- * Network disruption when computers are added or removed
- * A break in the cable will prevent all systems from accessing the network difficult to troubleshoot



Disadvantage

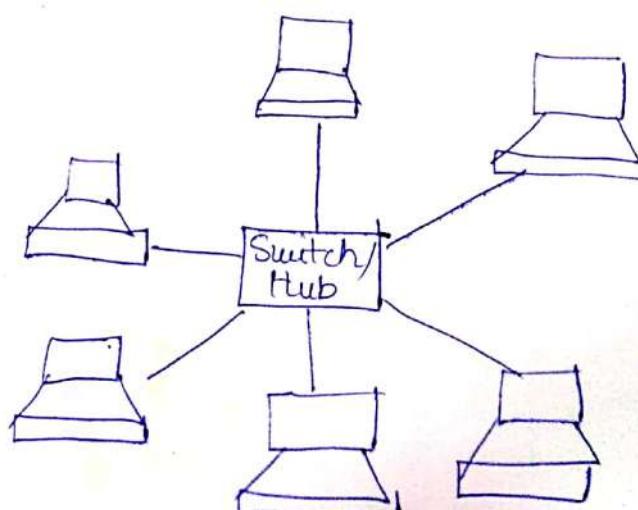
- * A single break in the cable can disrupt the entire network.
- * Expansion to the network can cause network disruption.

(3) STAR Topologies

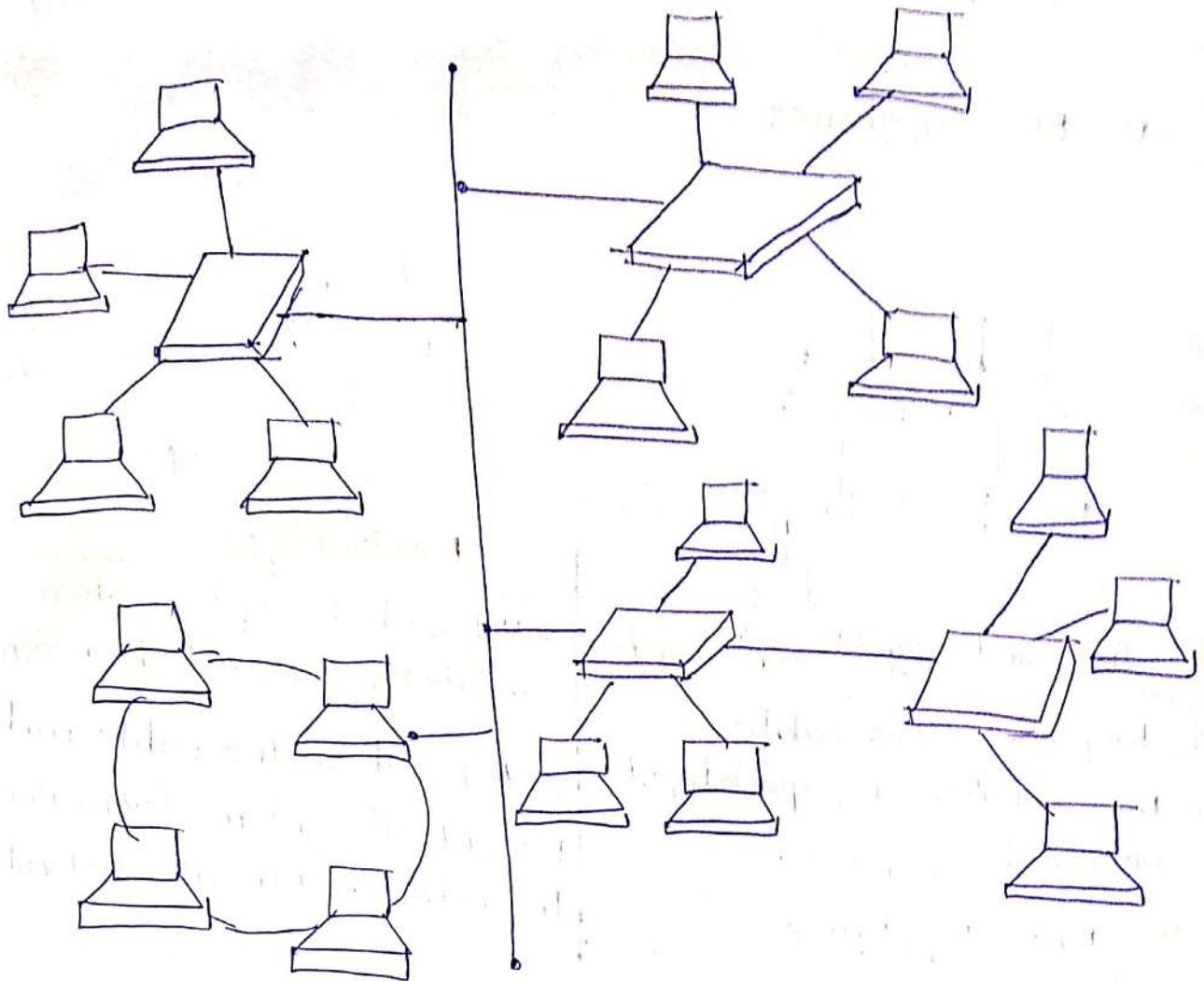
- * Most widely implemented.

Disadvantages →

- * Requires more cable.
- * More difficult to implement.

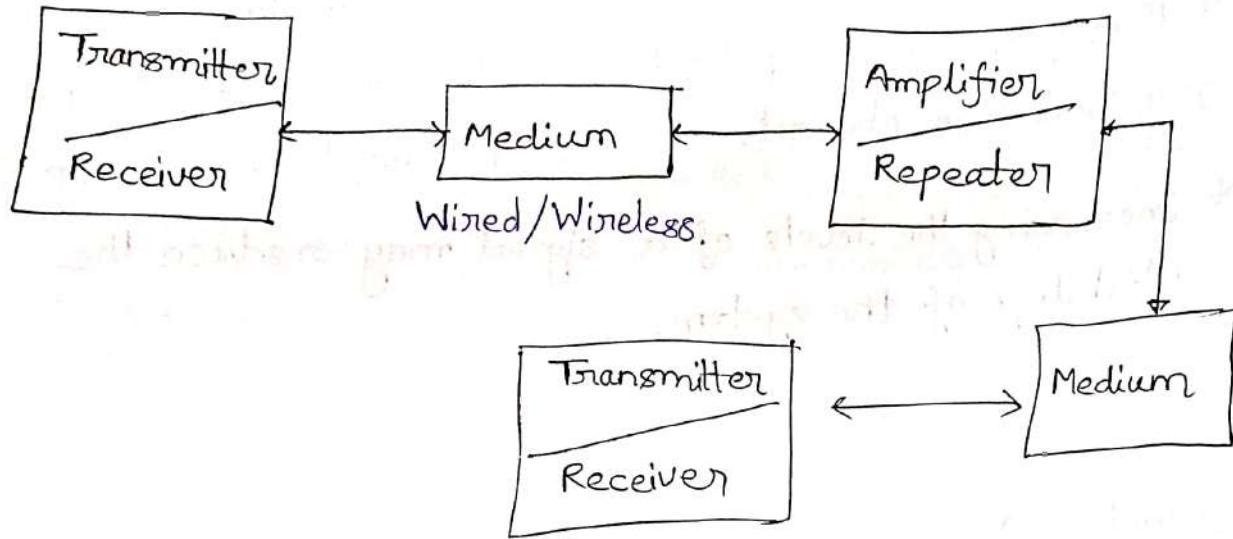


(4) Hybrid Topology



Physical Layer

Abstract Communication Model



Amplifier → Used for Analog signal

Repeater → Used for Repeat. Digital signal.

Channel →

It is a logical point to point connection between the sender and receiver.

Capacity of the channel →

It is defined in terms of bandwidth/spectrum.

Bandwidth →

$$BW = f_{\max} - f_{\min} \text{ Hz}$$

f_{\max}

f_{\min}

Nyquist Theorem →

$$C = 2H \log_2 V \quad \text{bits/sec}$$

H = Bandwidth of medium.

* This formula is applicable for noiseless channel. V = no. of discrete level used in the signal

* Increasing the levels of a signal may reduce the reliability of the system.

Signal →

It is electrical or electromagnetic encoding of data.

Ques Channel BW = 4000 Hz

Signal level used = 4

How much data rate is achievable through this channel.

Soln

$$H = 4000 \text{ Hz}$$

$$V = 4$$

$$C = 2 \times 4000 \log_2 4$$

$$= 8000 \times 2$$

$$C = 16000 \text{ bits/s}$$

Shannon Capacity theorem →

$$C = H \log_2 \left(1 + \frac{S}{N} \right) \text{ bit/sec}$$

$\frac{S}{N}$ = signal to Noise Ratio (SNR)

$$H = BW (\text{Hz})$$

* Unit of SNR is dB.

$$\left(\frac{S}{N} \right)_{\text{dB}} = 10 \log_{10} \left(\frac{S}{N} \right)$$

* Shannon capacity is used to determine the theoretical highest data rate for a noisy channel.

Ques Signal is sent over a channel of 3kHz, post signal to noise ratio is 20dB. What is max achievable data rate over the channel.

Sol^n

$$\frac{S}{N} = 20 \text{ dB}$$

$$\Rightarrow 20 = 10 \log_{10} \left(\frac{S}{N} \right)$$

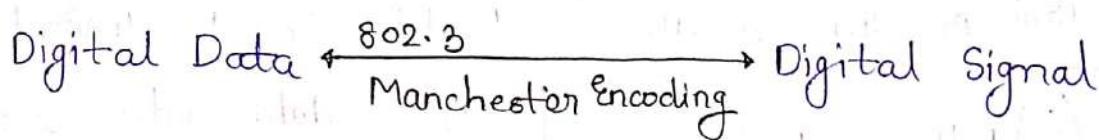
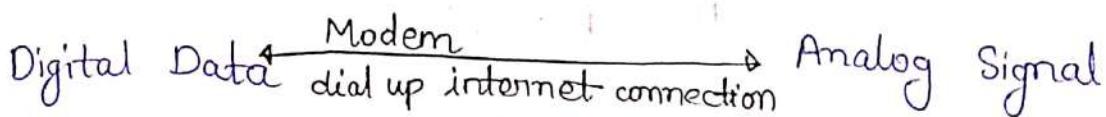
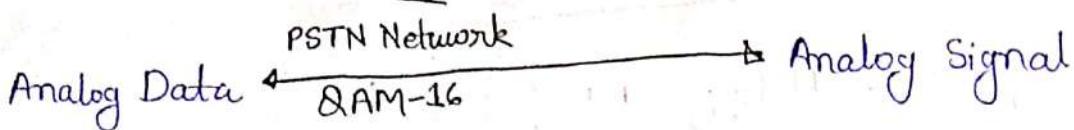
$$\Rightarrow \boxed{\frac{S}{N} = 100}$$

$$C = 3 \times 10^3 \log_{10} \left(1 + 100 \right)$$

$$= 3 \times 10^3 \log_{10} 101$$

$$\boxed{C = 3000 \log_{10} 101 \text{ bit/sec}}$$

Analog Vs Digital



Advantages of Digital Signal

- * Repeater, dsp (Digital Signal Processing)
- * The effect of distortion, noise and interference is much less in digital signals as they are less affected.
- * The signal is un-altered as the pulse needs a high disturbance to alter its properties, which is very difficult.
- * Combining digital signal using TDM is easier than combining analog signal using FDM.
- * The configuring process of digital signals is easier than analog signals.
- * Digital signals can be saved and retrieved more conveniently than analog signals.
- * The capacity of the channel is effectively utilized by digital signals.

Disadvantage of digital Signal

- * Generally, more bandwidth is required than that for analog systems.
- * Synchronization is required.
- * High power consumption (Due to various stages of conversion)
- * Introduce sampling error.
- * As square wave is more affected by noise, That's why while communicating through channel we send sine waves but while operating on device we use square pulse.

Advantage and disadvantage of Analog Signal

Advantages →

- * Major advantages of the analog signal is infinite amount of data.
- * Density is much higher.
- * Easy processing.

Disadvantages →

- * Unwanted noise in recording.
- * If we transmit data at long distance then unwanted disturbance is there.
- * Generation loss is also a big cons of analog signal.

Medium

* Properties of Medium →

Guided Media →

Features →

- * High speed
- * Secure
- * Used for comparatively shorter distances.

(a) Twisted Pair Cable →

(i) UTP →

- * Least expensive
- * Easy to install
- * High speed capacity.

Disadvantages →

- * Susceptible to external interference
- * Lower capacity and performance in comparison to STP.
- * Short distance transmission due to attenuation.

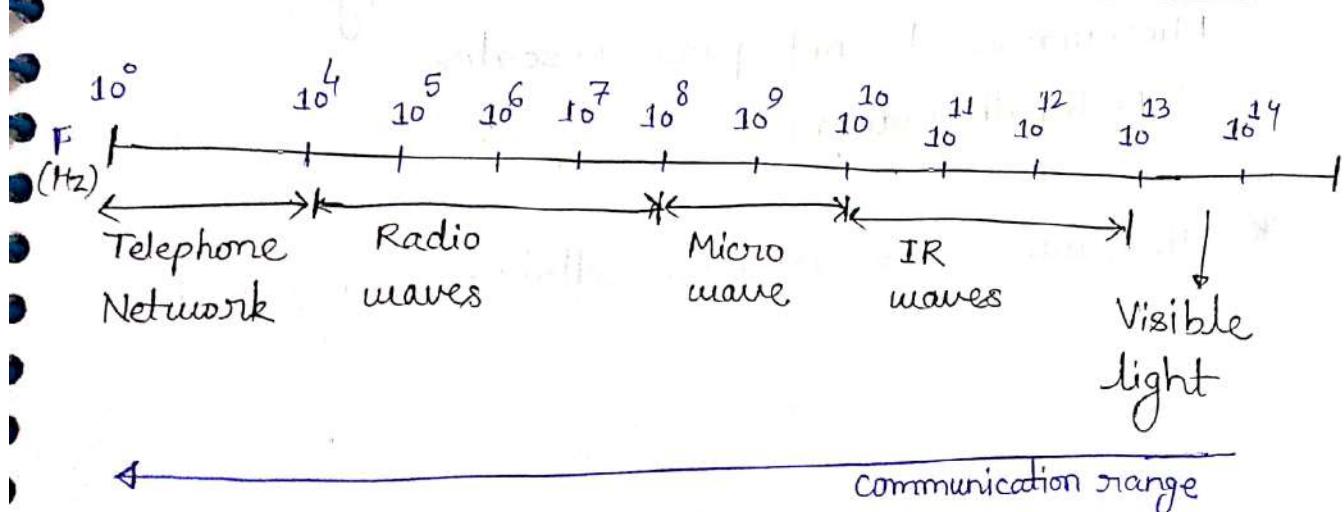
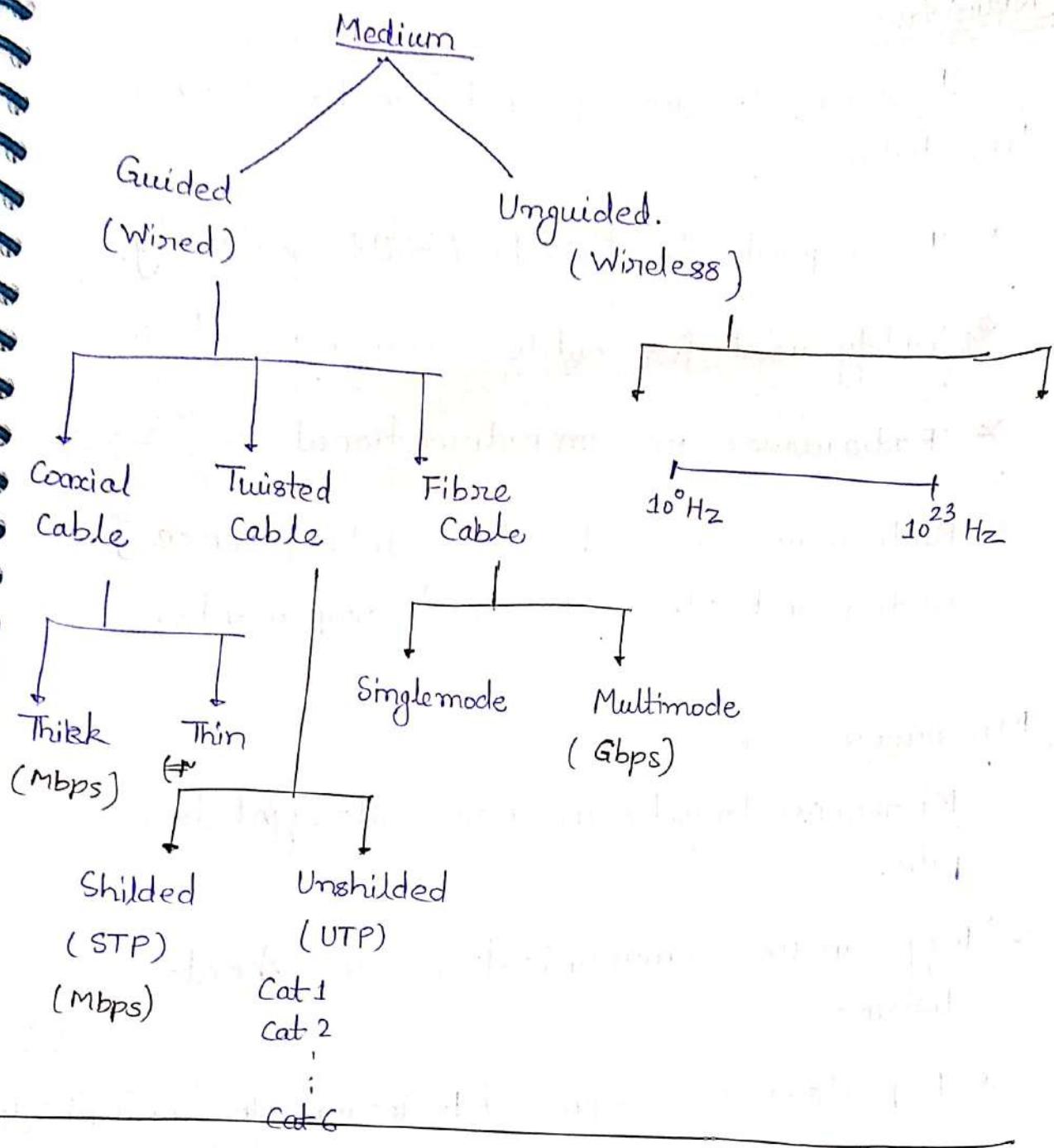
(ii) STP

Advantage →

- * Better performance at a higher data rate in comparison to UTP.
- * Eliminates crosstalk.
- * Comparitively faster.

Disadvantages →

- * Comparitively difficult to install and manufacture
- * More expensive
- * Bulky.



Omnidirectional $\xrightarrow{\text{penetration power}}$ directional

Radio Wave

It is easy to generate and can travel over long distances.

- * It can penetrate obstacles (buildings) easily.
- * Widely used for outdoor communication.
- * Radio waves are omnidirectional.
- * Radio waves are subject to interference from motor and other electrical equipments.

Microwave →

Microwave travels in near straight line path.

- * They can use communication over short distances.
- * Repeaters are required to increase the communication range.
- * Microwaves do not pass obstacles.
(Multipath fading)
- * Microwaves are used for cellular communication.

Infrared (IR) waves →

- * It ranges from 700nm to 1mm.
- * IR waves are longer than those of visible light but shorter than those of radio waves.
- * IR light is invisible to the human eye, although longer IR waves can be sensed as heat.

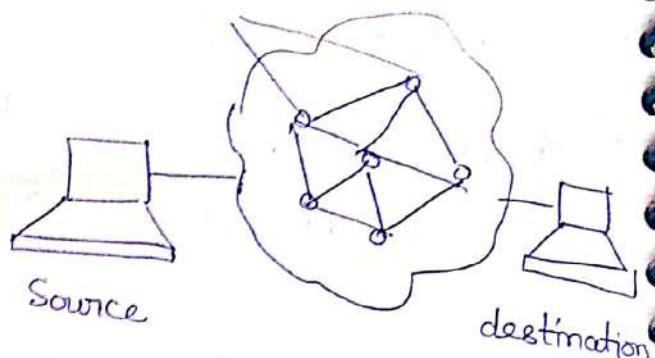
Light waves →

- * LEDs are used to generate light waves.
- * Visible light is just one particular type of electromagnetic radiation.

Network Performance

(1) End to End delay

Routers.

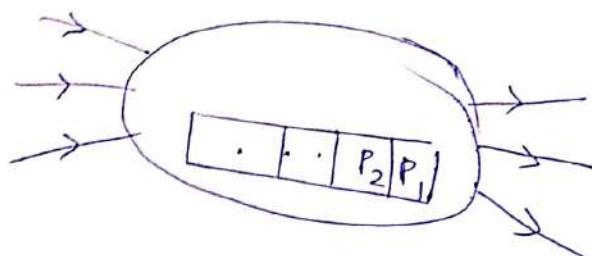


Channel Utilization

It is defined as fraction of time channel occupied.

Throughput

It is defined as bits per second (successfully received data by receiver)



(1) Transmission delay

Time elapse between first and last bit of a packet.

(2) Propagation delay

Time travel to length of link/wire.

(3) Processing delay at each end

(4) Queuing delay

① end to end delay (minimum)

$$= \sum_{\text{no. of intermediate nodes}} (\text{Transmission delay} + \text{propagation delay} + \text{Processing delay} + \text{Queuing delay})$$

② Throughput (Maximum)

Physical layer continues.....

* Converting bits into signal →

(1) digital data

digital signal

[Encoding Techniques]

There are four parameters regarding encoding techniques

- Net DC component present in the signal
 - Clock information
 - Error detection
 - cost [bits / signal]
 - Complexity
-] -ve
] +ve

- (1) NRZ (Non return to zero) series of encoding
- (2) Multilevel Binary
- (3) Biphasic Encoding
- (4) Encoding for long distance transmission.
- (5) Block Encoding

NRZ

|

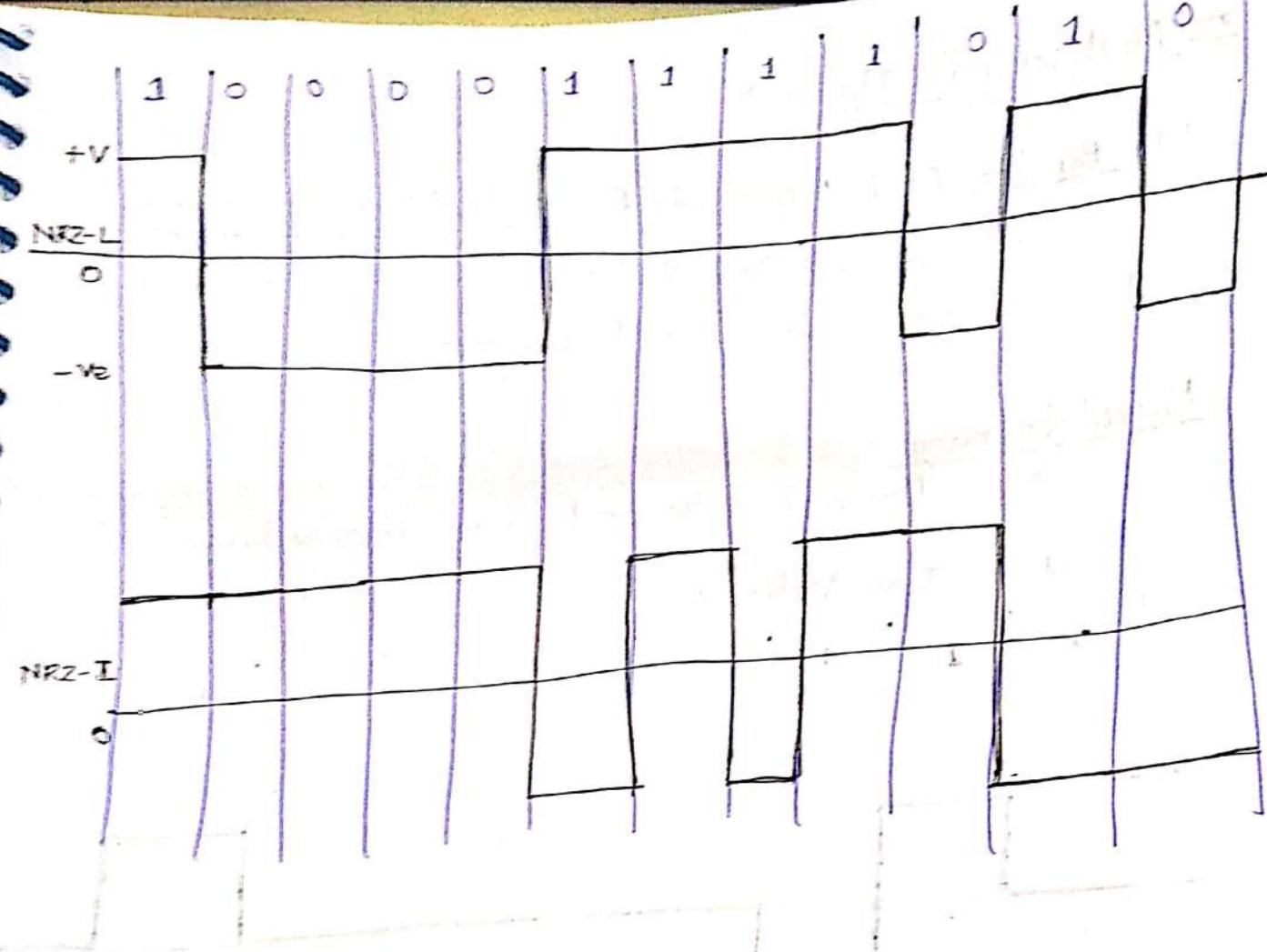


NRZ-I (differential Encoding)

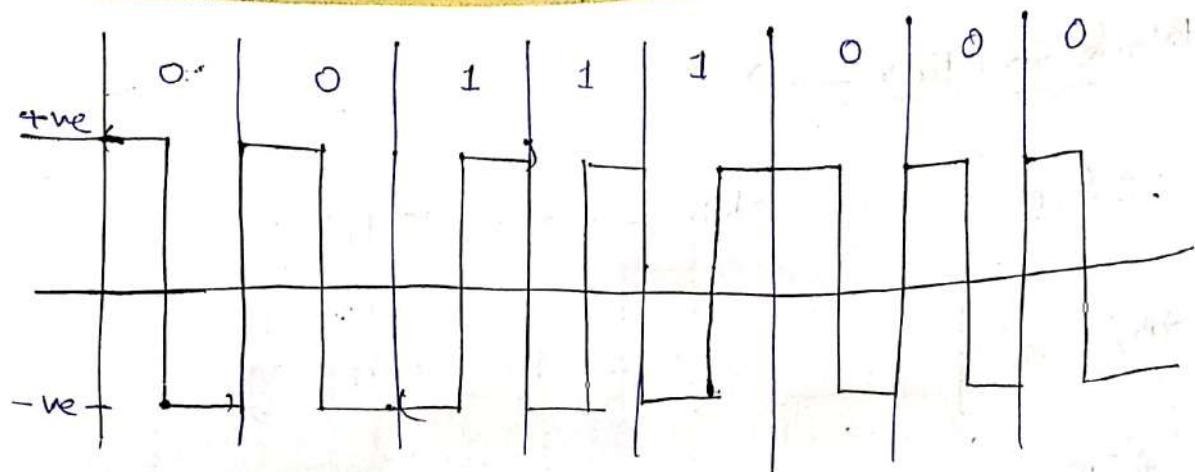
- 0 : No transition at the beginning of bit interval
- 1 : transition

NRZ-BEL

- 0 : high voltage level
- 1 : low voltage level



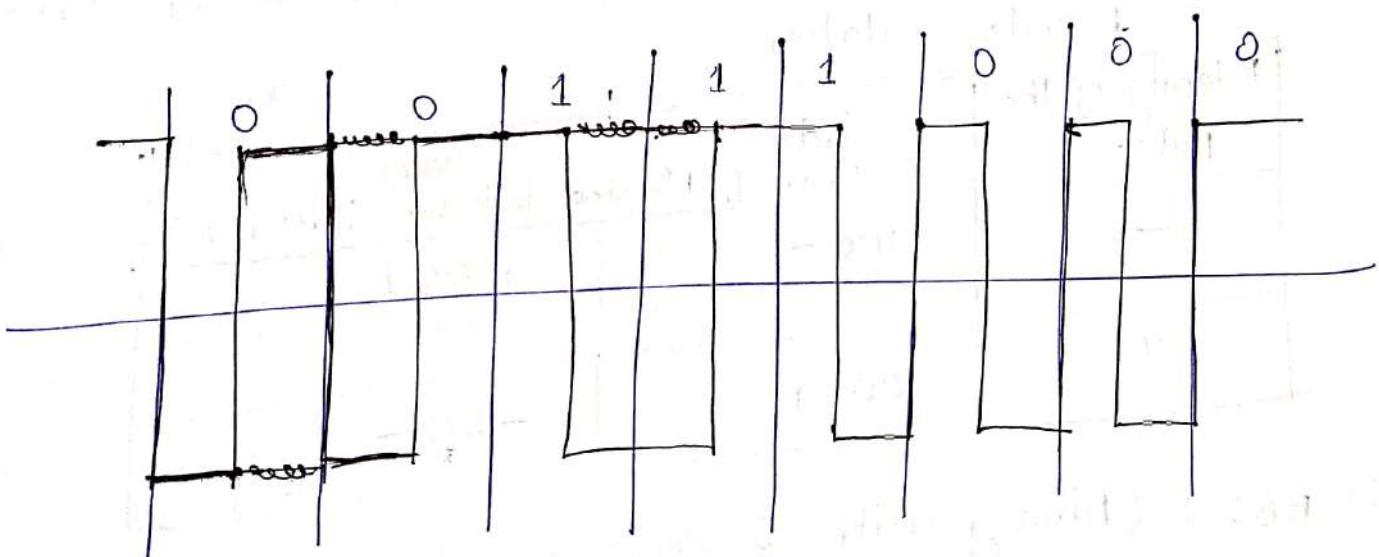
- * Net DC component undesirable in the signal
- * DC component is not desirable in signal because it cannot pass through some communication system like transformer and also lead to distortion.
- * DC component also results loss of power in unwanted signal.



(b) Differential Manchester Encoding

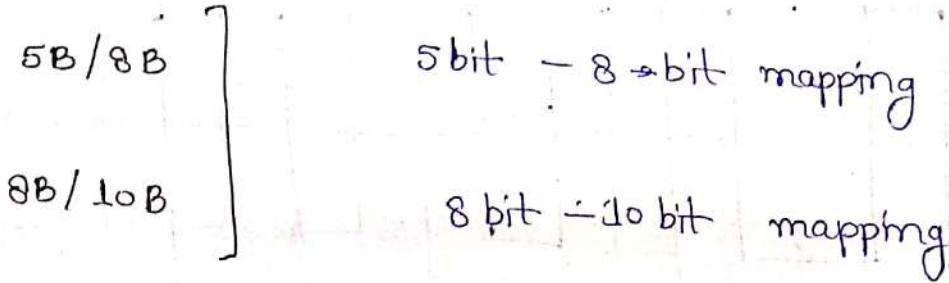
0 : Transition at the beginning of bit interval

1 : Absense of transition at the beginning of bit interval.

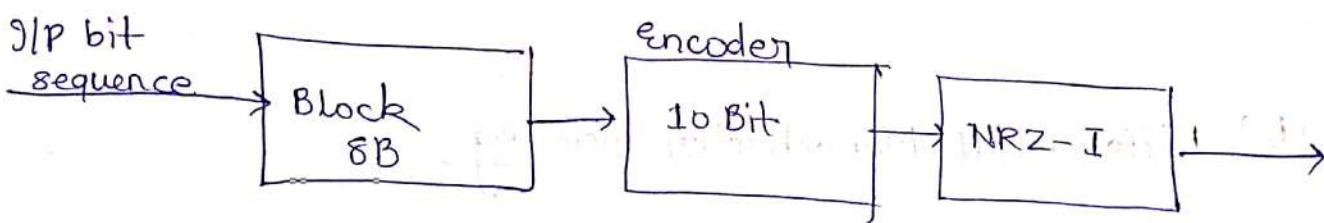


- * Manchester technique is used in Ethernet IEEE 802.3
- * Differential Manchester technique is used in token ring.

* Block Encoding



* They use NRZ-I to transmit the data.



* Encoding for long distance → (Based on Bipolar AMI)

(1) HDB3 (High Density Binary with 4 zero Substitution)

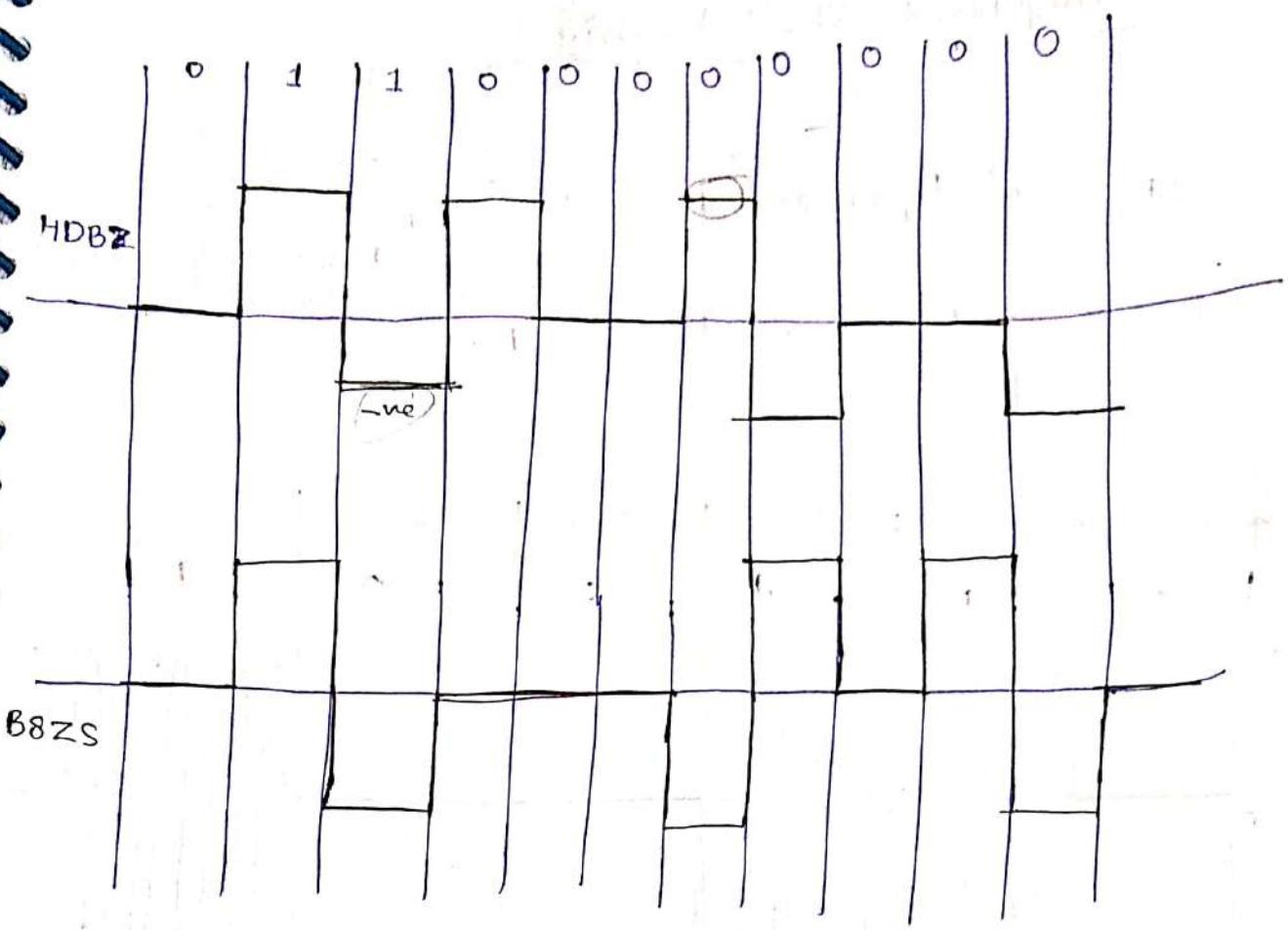
- 1 code violation

Polarity of the pulse	odd (no. of 1's since last substitution)	even
-	000-	+ 00+
+	000+	- 00-

(2) B8ZS (Binary with 8 zero Substitution)

by substituting 2 code violation.

Polarity of the pulse	odd	even
+	0 0 0 + - 0 + -	
-	0 0 0 - + 0 + -	



(2) Digital Data — Analog Encoding Signal

$$s(t) = A \sin(\omega t + \phi)$$

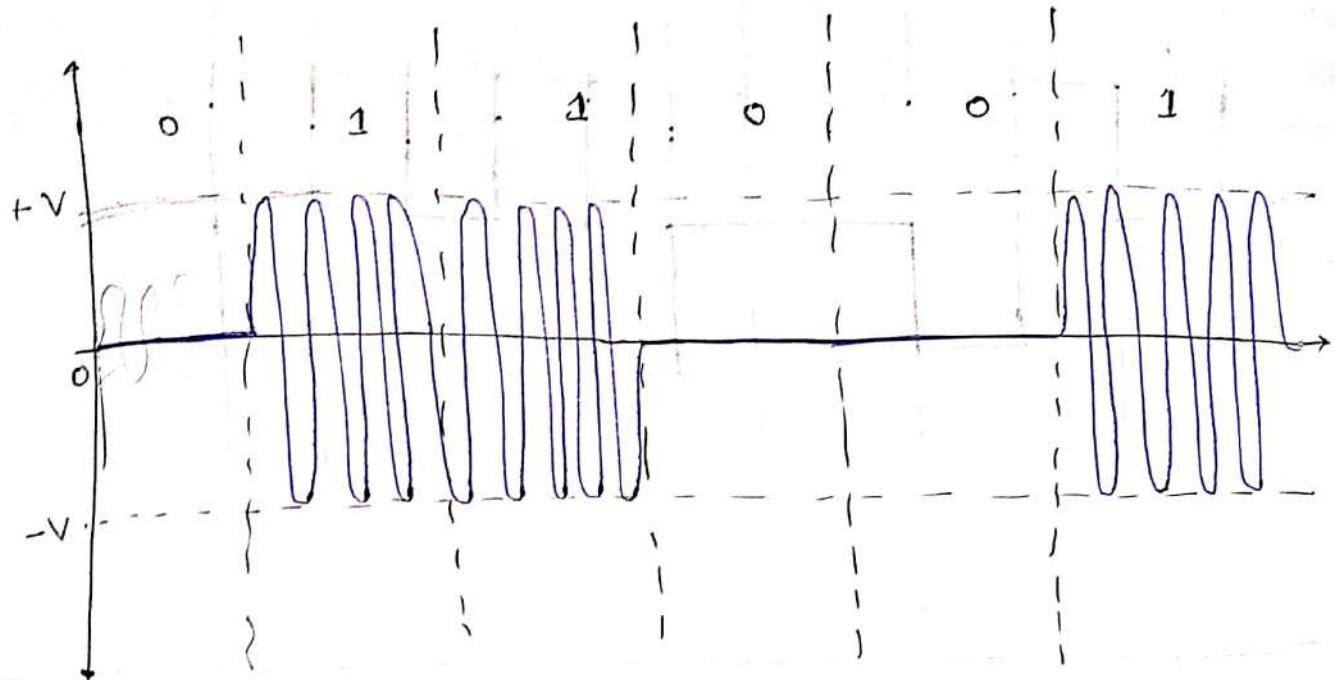
↓ ↗
 Amplitude frequency Phase

* Analog signal generation is known as Modulation techniques.

(1) ASK (Amplitude Shift Keying)

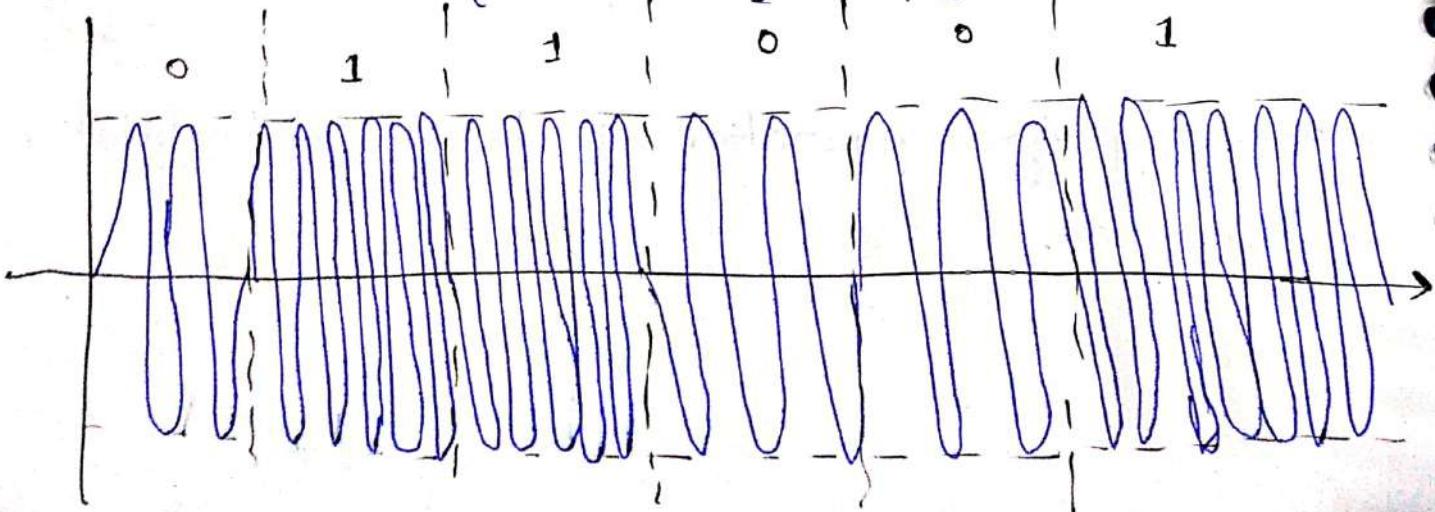
$$s(t) = \begin{cases} A_1 \sin 2\pi f_c t & : 0 \\ A_2 \sin 2\pi f_c t & : 1 \end{cases}$$

Phase frequency } constant



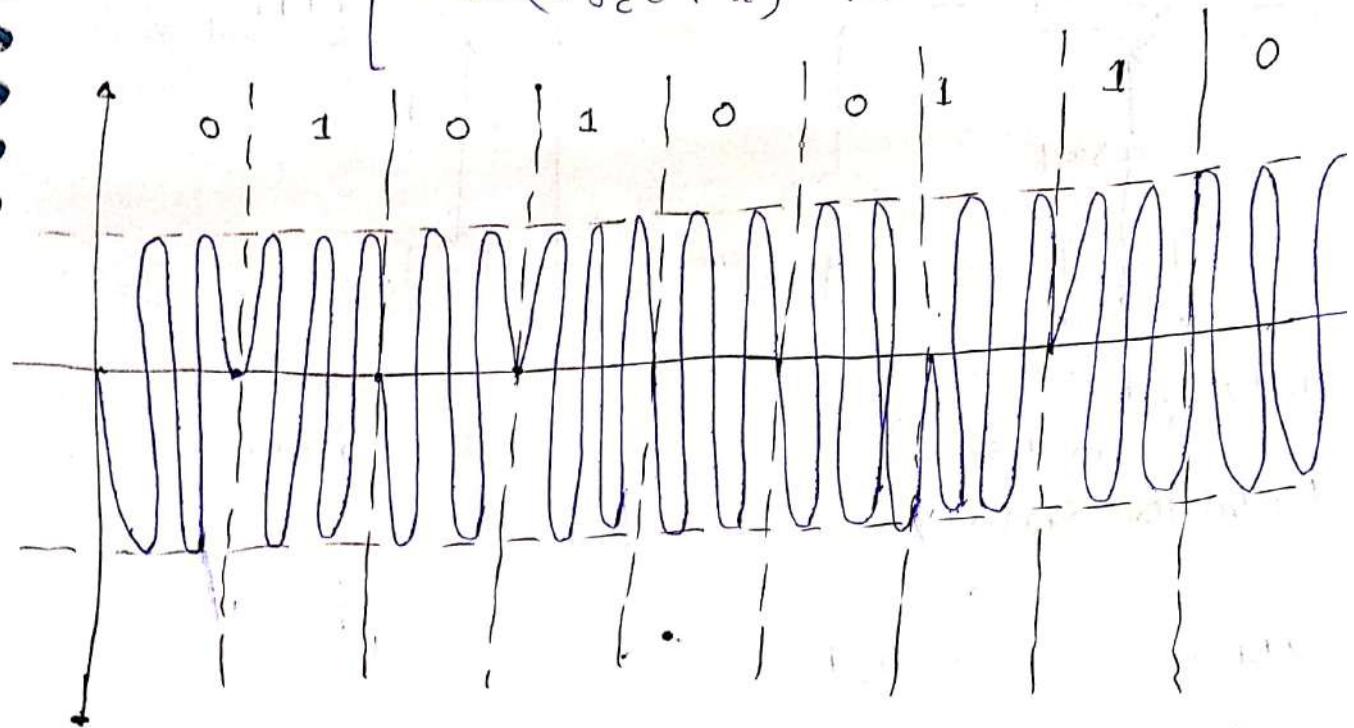
(2) FSK (Frequency Shift Keying)

$$s(t) = \begin{cases} A \sin 2\pi f_{c1} t & : 0 \\ A \sin 2\pi f_{c2} t & : 1 \end{cases}$$

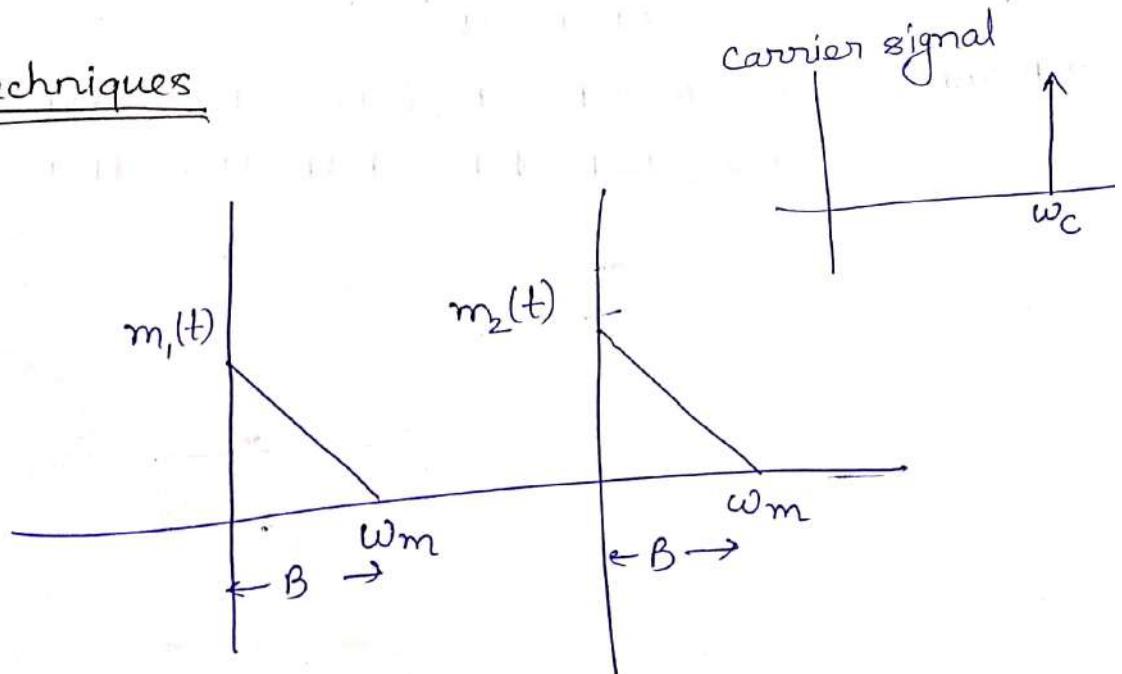


(3) PSK (Phase Shift Keying)

$$s(t) = \begin{cases} A \sin 2\pi f_c t & : 0 \\ A \sin(2\pi f_c t + \pi) & : 1 \end{cases}$$



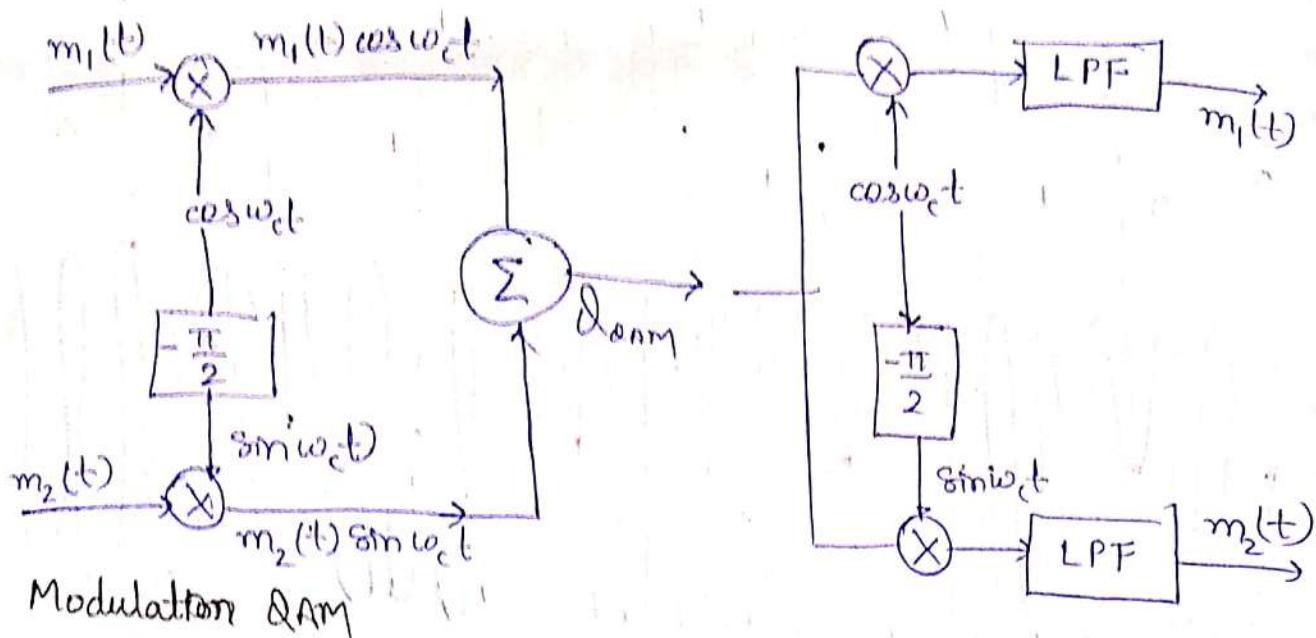
QAM techniques



$$Q_{QAM} = m_1 \cos \omega_c t + m_2(t) \sin \omega_c t$$

Block diagram of QAM

$$Q_{\text{QAM}} = m_1(t) \cos \omega_c t + m_2(t) \sin \omega_c t,$$



→ BPSK

0, 1

→ QPSK

(00, 01, 10, 11)

→ 16 QAM

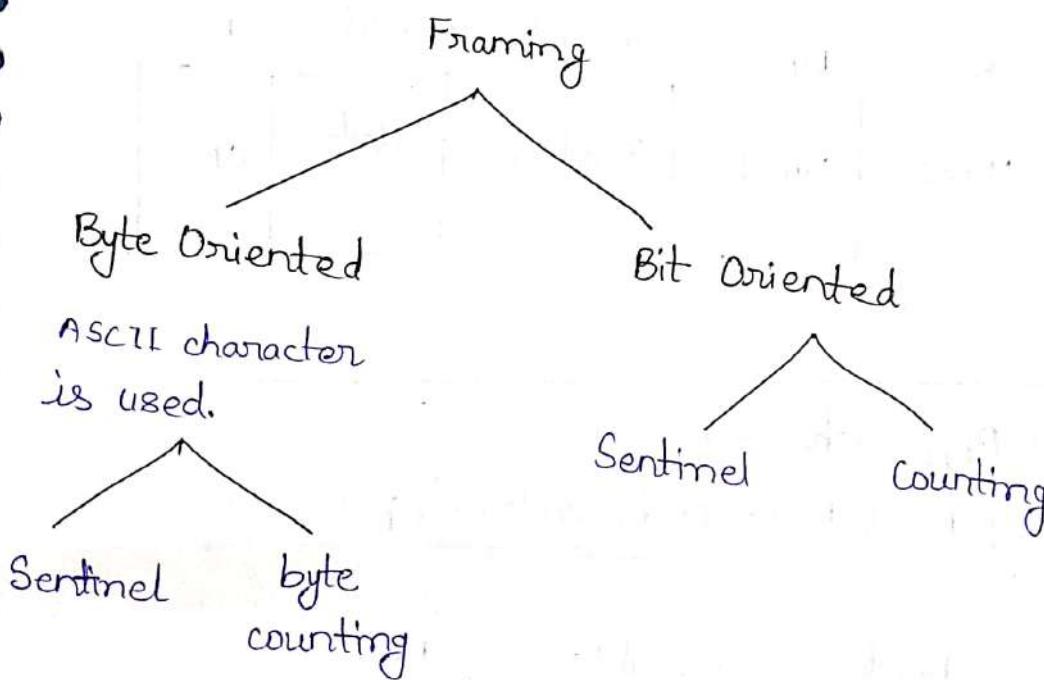
(0000, 0001, 0010, 0011, 0100, 0101, 0110, 0111,
1000, 1001, 1010, 1011, 1100, 1101, 1110, 1111)

Data Link Layer

* Framing →

Computer network operates as a packet switched network which means block of data is exchanged between the nodes.

Interfaces are transmitting and receiving stream of bits. Therefore it is required to mark the boundary of frame on the bitstreams, for this framing protocol is used.



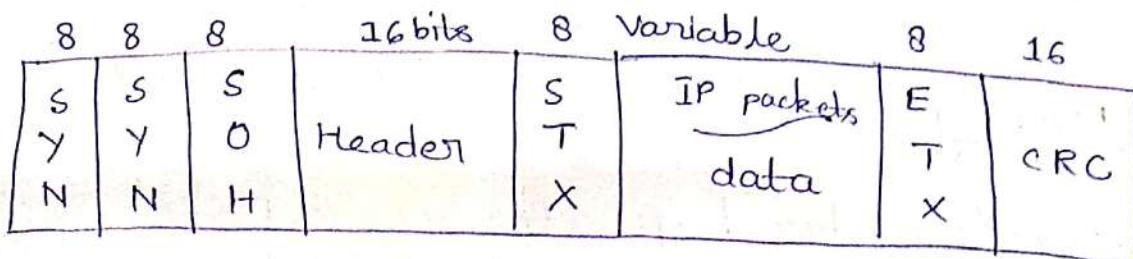
* Byte Oriented Protocol →

(BISYNC, DDCMP)

based on

Sentinel approach

BISYNC



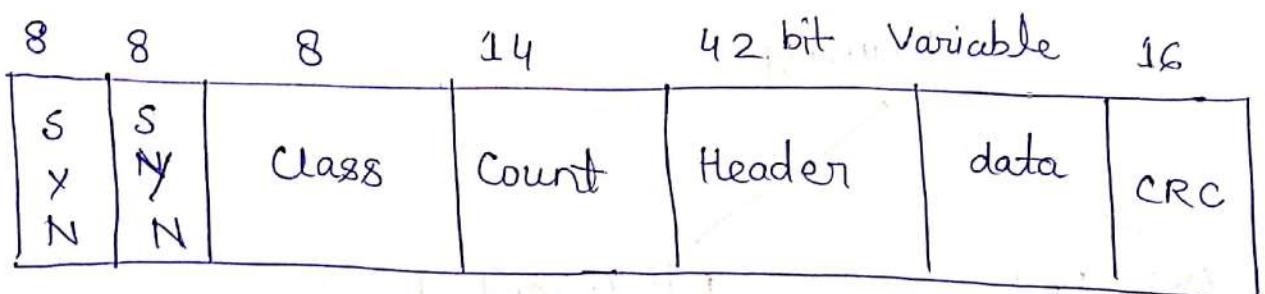
SOH = Start Of Header

→ Byte Stuffing

DLE = Data Link Escape character.

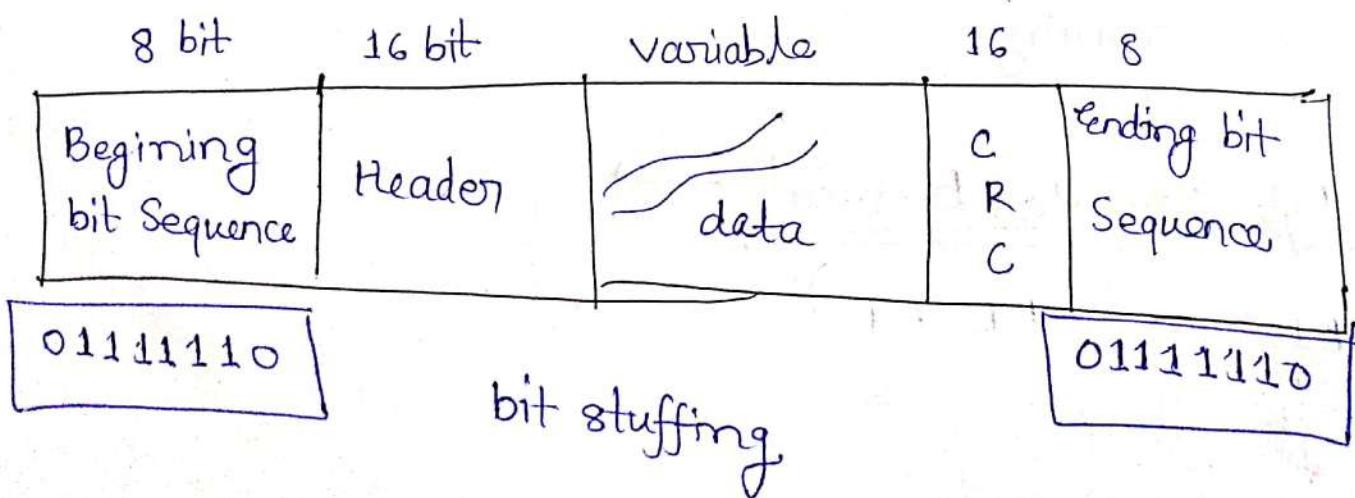
STX = Start of Transmission
ETX = End of Transmission

DDCMP (Byte Counting Approach)



Bit Oriented Approach →

HDLC (High Level Data Link Control) protocol



- * Whenever, except starting and ending bit sequence five 1's are encountered, then a zero will be stuffed.

e.g.

0 1 1 1 1 | 1 1 0 0 1 0
↓
0

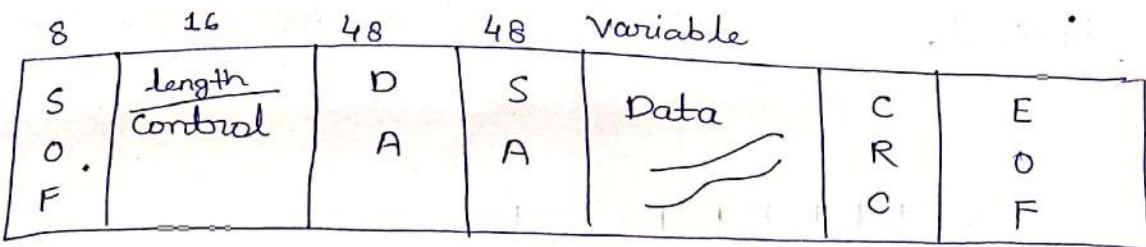
- * Receiver process will be, whenever five 1's are received, it will check next bit, if it is zero, then take out that bit and makes subsequent bit as part of bit.

If the next bit is 1 then there are two possibilities —

- * If next bit is zero, then that is boundary of frame.
- * If next bit is 1, then data received is an error.

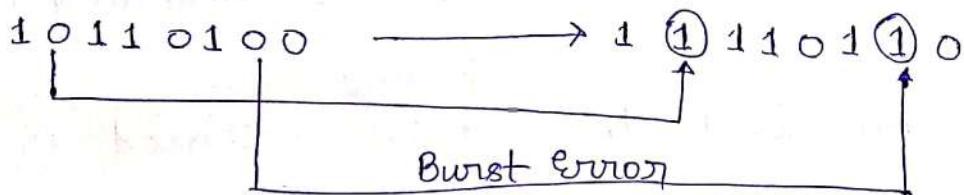
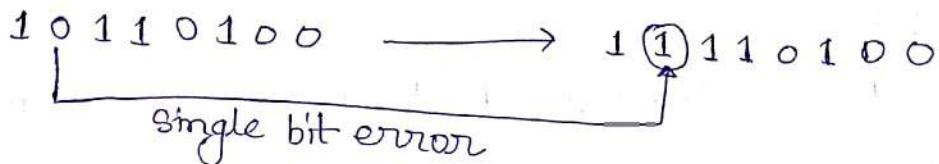
- * The above protocol discussed works of ISP.

For Ethernet (802.3)



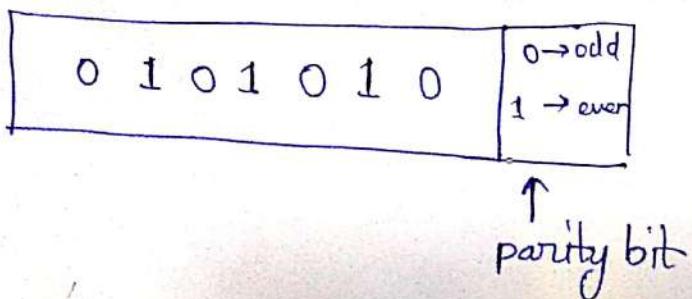
This has also Byte Oriented and Sentinel

Error Control →



Error Detection Techniques →

(1) Parity Check



→ Scalability

→ No. of extra bits required

→ power of error detection.

* Enforcement of odd parity means no. of 1's is odd, and for even, 1's should be even.

Two dimensional parity check

even parity							Row parity
1	0	0	1	1	0	0	
.	0
.	.	.	1	0	0	1	0
.	.	.	0	1	0	0	0
1	0	1	0	0	0	1	0
1	1	0	1	1	0	1	0
1	1	0	1	1	0	1	0

column parity

Checksum

- * Checksum uses 1's complement arithmetic for its implementation.

Algorithm

- * Consider a message to be made up of bytes or words.
- * Divide the message into words. Add all the words using 1's complement arithmetic to get the sum.
- * Complement the sum to get checksum.
- * Transmit checksum along with the message.

Receiver

- * Divide the received message into words. Add all the words including checksum.
- * Complement the sum.
- * All zeros implies no error, else error.
- * Power of error detection.
 - It can detect all errors where odd no. of bits are involved in error. (most of 2 bit or 4 bits error)

Date - 29/08/19

Example

$$M = \underbrace{10101001}_{W_1} \underbrace{00111001}_{W_2}$$

$$\begin{array}{r} 10101001 \\ + 00111001 \\ \hline \text{Sum} \quad 11100010 \end{array}$$

Transmit message

$$= W_1 + W_2 + W_3$$

Take complement

$$W_3 = \boxed{00011101}$$

Checksum

$$\begin{array}{r} 11100010 \\ 00011101 \\ \hline 11111111 \end{array}$$

$$\text{compliment} = 00000000$$

Let the received word is

$$\begin{array}{r} 10101111 \\ + 11111001 \\ \hline \boxed{10101000} \end{array}$$

$$\text{Sum} \quad 10101001$$

$$W_1 + W_2 + W_3 = \begin{array}{r} 10101001 \\ 00011101 \\ \hline 11000110 \end{array}$$

$$\text{compliment } \underline{\underline{00111001}} \text{ [There is an error]}$$

* CRC (Cyclic Redundancy Check) →

- * CRC is implemented using special class of polynomial arithmetic known as Polynomial Arithmetic modulo tool.
- * For implementation of CRC following properties are important

- ① Any polynomial $B(x)$ can divide a polynomial $C(x)$ if $B(x)$ is of same or higher degree than $C(x)$.
- ② The remainder obtained when $C(x)$ is divided by $B(x)$ by subtracting $B(x)$ from $C(x)$.
- ③ To subtract $B(x)$ from $C(x)$, we simply perform Ex-OR operation on the pair of matching coefficients of the polynomials and degree of polynomial is defined as -

Let $C(x) = x^3 + 1$

$$B(x) = x^3 + x^2 + 1 \quad \text{degree} = 3$$

Binary Equivalence

$$B(x) = 11.01$$

$$C(x) = 1001$$

$$\begin{array}{r}
 & 1 \\
 1101) & 1001 \\
 & 1101 \\
 & \underline{\oplus\oplus\oplus\oplus} \\
 & 100
 \end{array}$$

CRC at Sender side →

- * Let R be the degree of polynomial $C(x)$.
[Both Sender and Receiver know that . . . polynomial].
- * Append R 0 bits at the end of message bit string and call it $S(x)$.
- * Divide $S(x)$ by $C(x)$ to get the remainder $R(x)$.
- * Subtract $R(x)$ from $S(x)$ and call it $T(x)$
- * Transmit $T(x)$.

At Receiver Side

- * Let Receiver receive message and call it $M(x)$.
- * Divide $M(x)$ by $C(x)$.
- * If remainder is zero, then accept or
otherwise discard.

Example

message = 1011

$$C(x) = x^2 + 1 \quad (\text{D: } 2)$$

$$M(x) = 101100$$

$$\begin{array}{r}
 & 1001 \\
 &) 101100 \\
 101 & \downarrow \\
 & 101 \\
 & \times 001 \\
 & \hline
 & 000 \\
 & \downarrow \\
 & x 010 \\
 & 000 \\
 & \hline
 & x 100 \\
 & 101 \\
 & \hline
 & 001
 \end{array}$$

$$\begin{array}{r}
 101100 \\
 001 \\
 \hline
 \oplus \\
 \hline
 \cancel{101011} \\
 \hline
 101101
 \end{array}$$

At receiver

$$\begin{array}{r}
 & 1001 \\
 101) & 101101 \\
 & 101 \downarrow \\
 & \hline
 & X 001 \\
 & 000 \downarrow \\
 & \hline
 & X 010 \\
 & 000 \downarrow \\
 & \hline
 & X 101 \\
 & 101 \downarrow \\
 & \hline
 & 000
 \end{array}$$

Let the received bit is

101001

$$\begin{array}{r}
 & 1000 \\
 101) & 101001 \\
 & 101 \\
 \hline
 & 000 \\
 & 000 \\
 \hline
 & x 000 \\
 & 000 \\
 \hline
 & x 000 \\
 & 000 \\
 \hline
 & x 001 \\
 & 000 \\
 \hline
 & x 01
 \end{array}$$

+1 Since remainder
is not zero

hence there is an error in received message.

Power of Error detection

It can detect all errors where odd no. of bits are involved in error, 2 bit, and most of the 4 bit errors and most of burst errors.

* Error Correction Techniques

Hamming Code

$$m+r+1 \leq 2^r$$

To implement the algorithm we need to find the no. of r bits required for the implementation which can be obtained by

$$(m+r+1) \leq 2^r$$

where m = no. of message bits

r = no. of redundant bits (parity bits).

Step 1

- * bits are numbered from left to right as 1; 2, 3 ... starting from 1 to $(m+r)$.
- * Power of 2 bits positions are reserved for redundant bits.
- * Remaining positions are filled with message bits from left to right.

* Each redundant bit shall enforce parity (even or odd) on the subsets of redundant bit.

* Subset of parity bits can be determined as follows —

Example

$$M = 1011 = 4$$

$$4 + r + 1 \leq 2^3$$

$$R = 3$$

odd parity

001 010 011 100 101 110 111
1 2 3 4 5 6 7

R_1	R_2	M_3	R_4	M_5	M_6	M_7
		1		0	1	1

$(R_1, 3, 5, 7)$

1 0 1

$(R_2, 3, 6, 7)$

1 1 1

$(R_3, 5, 6, 7)$

0 1 1

$R_1 = 1$

$R_2 = 0$

$R_4 = 1$

Codeword = 1 0 1 1 0 1 1

At receiver

odd parity

Received Message is 1011010

(1, 3, 5, 7)

1100

(2, 3, 6, 7)

0110

(4, 5, 6, 7)

1010

$P_1 \quad 1 \ 1 \ 0 \ 0$

$P_2 \quad 0 \ 1 \ 1 \ 0$

$P_3 \quad 1 \ 0 \ 1 \ 0$

1 ↑

1

1

111 = 7

It means there is an error in 7th position.

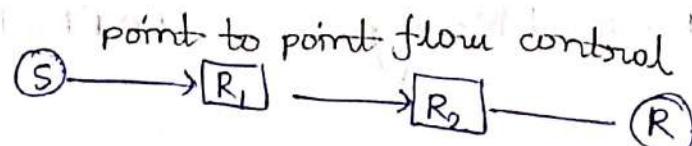
Hence the correct output will be

= 1011011

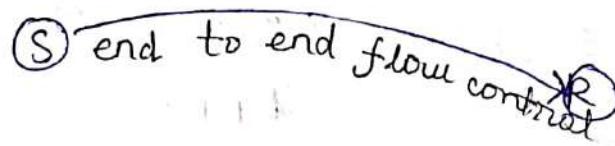
It is important to note that a forward error correction code can correct errors up to $\lfloor \frac{n-k}{2} \rfloor$ but cannot correct errors beyond this limit.

Flow Control

* Data link layer



* Transport Layer



* flow control techniques are required to deal with the problem of fast transmitter and slow receivers

* Flow control techniques are also responsible for order of data packets (Sequence of data packets)

* These techniques also cares of lost data packets.

ARQ

* Automatic Repeat Request is implemented on all the nodes on network.

(1) LLC protocols are required to deliver frames reliably and ^{need} leaves some feedback mechanism.

(2) This can be implemented by the use of ACK and timers.

Sender's process

* Sender transmits a frame and keeps a copy of it.

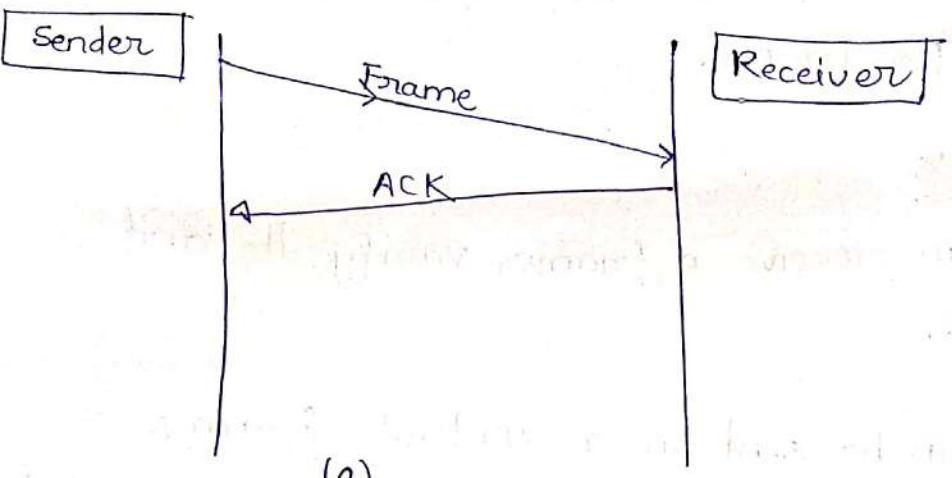
* Associates a timer with copy of frame and wait for ACK.

* If ACK is not received before time out then retransmit the frame.

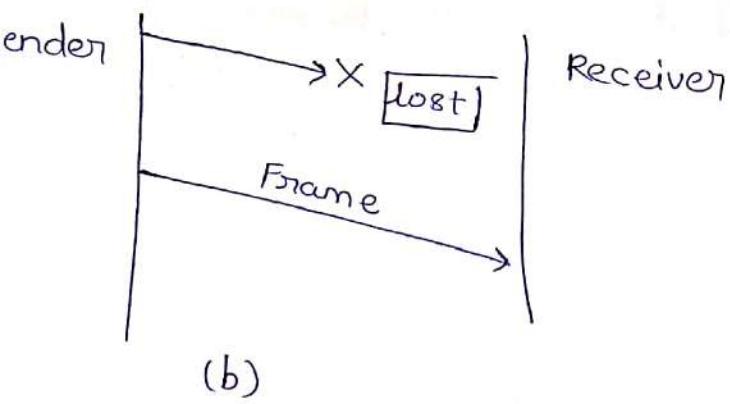
Receiver's process

* Receiver can receive a frame, verify it and send the ACK.

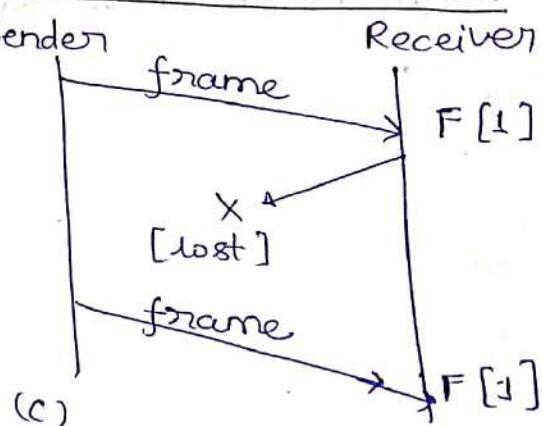
* This ACK can be sent on a control frame (Header without any data or piggyback on the data frame it is about to sent). This is known as ARQ.



(a)

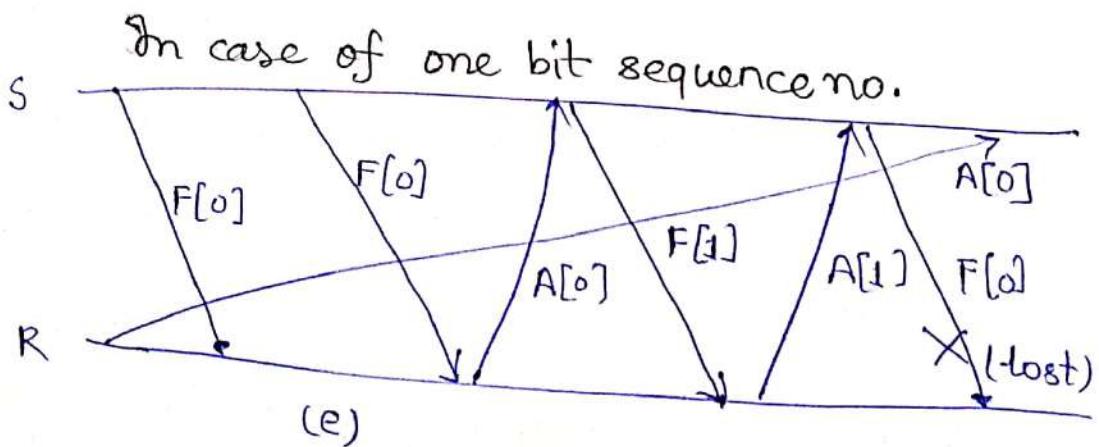
When frame has lost

(b)

When the ACK is lost

(c)

* Sequence no. is used to avoid the duplicate copies of frame



* Stop and Wait protocol can be considered as 1 bit ARQ

Problems

- * One bit sequence number is not sufficient for stop and wait protocol as shown in figure (e)
- * Only one frame can be transmitted at any point of time, which means link capacity cannot be utilized sufficiently

Ques Available link capacity is 1.5 Mbps, timer value = 45ms and frame size = 1 kB.
How many frames can be transmitted to fully utilize link capacity?

Solⁿ

$$\text{no. of frames} = \frac{1.5 \times 10^6 \times 45 \times 10^{-3}}{1 \times 10^3}$$

≤ 18.

L J

* To solve the above problems of

(1) N bits sequence no. should be used.

(2) TTL (Time to leave) constraints should be added for frame and ack.

(3) To utilize link capacity, more than one frame should be sent within the timeout value.

(4) These solutions have been encapsulated in sliding window protocol.

$$SWS = \frac{\text{link capacity} \times \text{timeout value}}{\text{frame size}}$$

(Sender Window size)

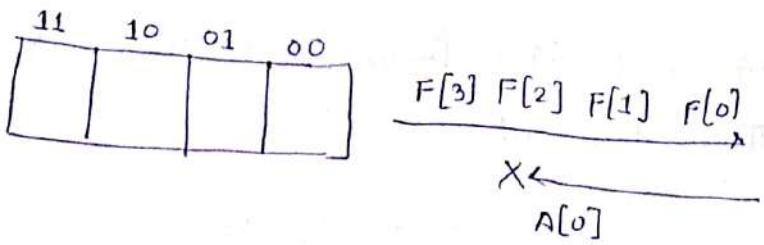
$$\text{total no. of bits in sequence} = \frac{(\text{Max. Sequence No.} + 1)}{2} \geq SWS$$

for SWS = 8

$$\Rightarrow \frac{\text{Max. Sequence No.} + 1}{2} \geq 8$$

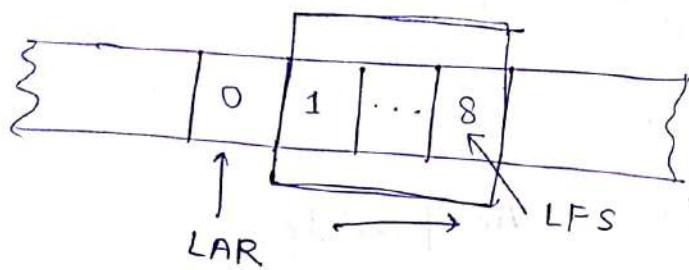
$$\Rightarrow \text{Max. Sequence No.} \geq 15$$

$$\text{no. of bits} = \log_2(15) = \log_2(\text{Max. Sequence No.})$$



~~Sliding Window Protocols~~

At Sender



Algorithm

For SWS, sender will maintain following variables -

(a) Sequence No.

(b) SWS \rightarrow Size of Sender's window (buffer)

(c) LFS (Sequence No. of Last frame Sent)

(d) LAR (Sequence No. of Last ACK received)

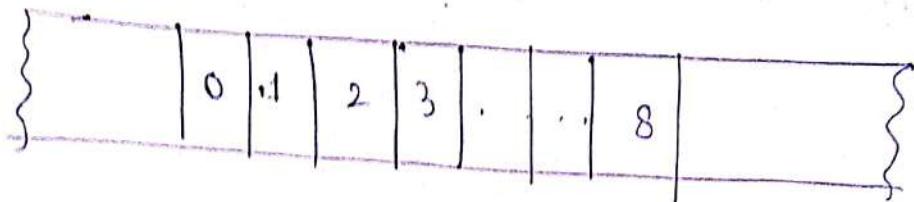
(1) Transmit frames until $(LBB - LAR) \leq SWS$.

so sender

(2) After each transmission, increment LFS, on
receipt of ACK, implement LAR.

- * Sender will retransmit the frame if ACK is not received within timeout.

At receiver



RWS = Receiver Window Size

LFS = Seq. no. of Last Frame Acceptable

NFE = Seq. no. of Next Frame Expected.

Sequence no. represented

SeqNoToAck → This variable represents seq. no. of out-of-order received.

- * Receiver can receive

$$\boxed{\text{LFA} - \text{NFE} + 1 \leq \text{RWS}}$$

Algorithm

- * Frame arrives with sequence no., if seq. no. is within receiver's window, then accept else discard. (Which means sequence no. is $\leq \text{LFA}$ and $\geq \text{NFE}$)

if (SeqNo. $\leq \text{LFA}$ & & Seq.No. $\geq \text{NFE}$)
then accept else discard.

- * If $\text{SeqNo.} = \text{NFE}$, then send ACK and set
 $\text{NFE} = \text{NFE} + 1$;
- * If $\text{SeqNo.} \neq \text{NFE}$
then set $\text{SeqNoToAck} = \text{SeqNo.}$ Stop sending ACK.

for e.g. Let $\text{NFE} = 2$ $\text{SeqNoToAck} = 5$
then $\text{SeqNo} = 5$

* Once all frame with sequence no.

$\text{SeqNo} < \text{SeqNoToAck}$ received
then Send ACK with value

SeqNoToAck (Cumulative ACK)

Set $\text{NFE} = \text{SeqNoToAck} + 1$

and $\text{LFA} = \text{SeqNoToAck} + \text{RWS}$

Advantages

- * Order of data is ensured.
- * Loss of data is cared.
- * Sliding Window is reliable of communication.
- * It will ensure the flow control.

$RWS = 1$ Stop and Wait Protocol

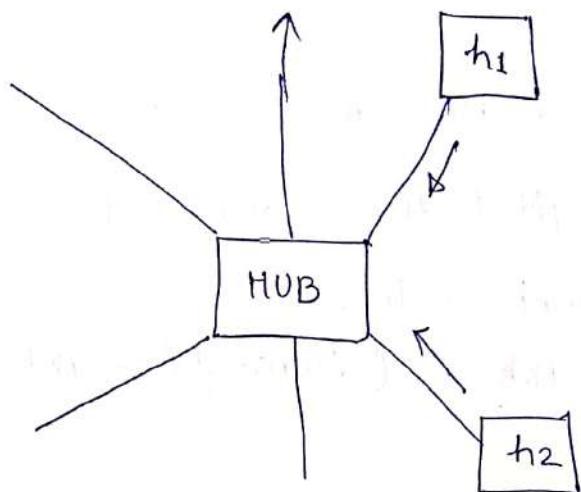
$SWS = RWS$ In general

$SWS < RWS$ No meaning

Medium Access Protocols Control

* Requires in multipoint network. (Shared Network)

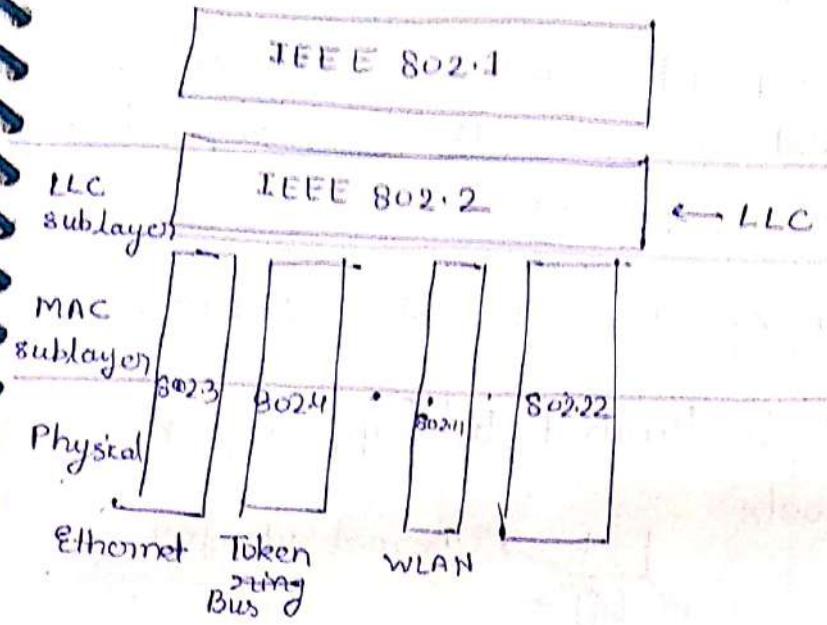
broadcast type



* MAC is nothing but Regulatory Mechanism to use the medium by host.

(1) MAC sublayer is only used in shared network
for e.g. Ethernet, WLAN, Cellular System etc.

(2) 802.1X Standard has covered the whole idea of different MAC options.



802.15 → PAN

802.16 → WiMAX

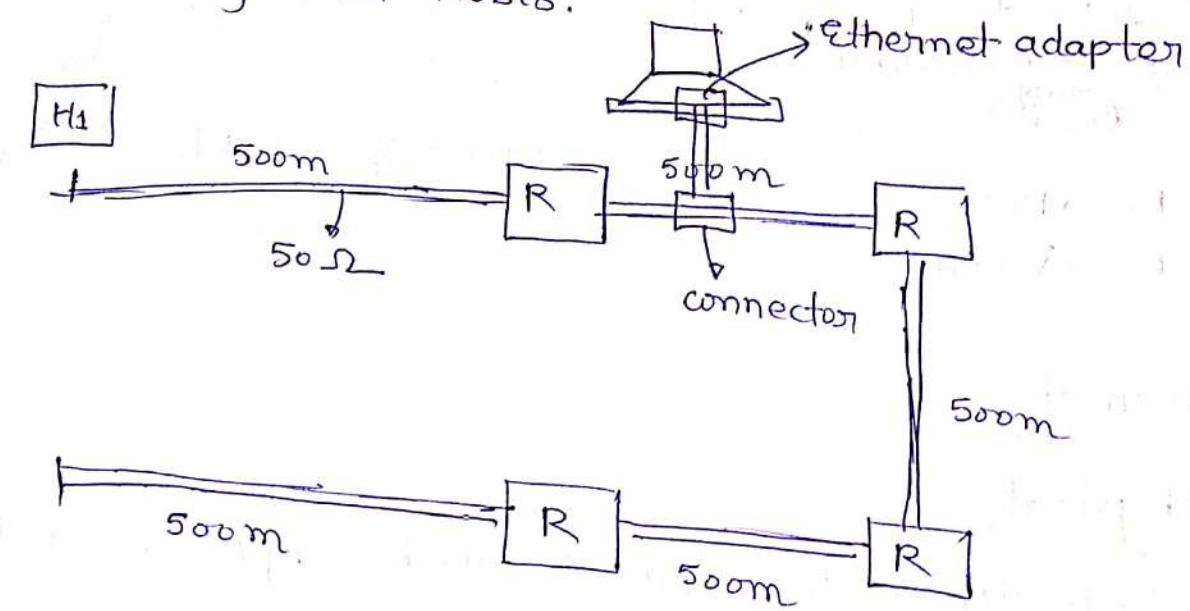
SIG = Special Interest Group.

Ethernet

(1) Physical

- * Each ethernet segment can be upto 500 m long of 50Ω coaxial cable.
- * Post tab into cable of connection and should be 2.5 m apart from each other.
- * Bus topology ~~is~~ should be used for deployment of network.
- * Transceiver should be capable of detecting whether the ~~link~~^{link} is idle or busy.
- * Transceivers are implemented in ethernet.
- * Signal placed on the link ^{are} broadcast in network.

- * Manchester encoding is used.
- * Multiple ethernet segments can be joined by the use of repeaters but not more than 4 repeaters can exist between any pair of host.
- * Limiting the span of ethernet upto 2.5 km.
- * Ethernet network is limited to support a max of 1024 hosts.

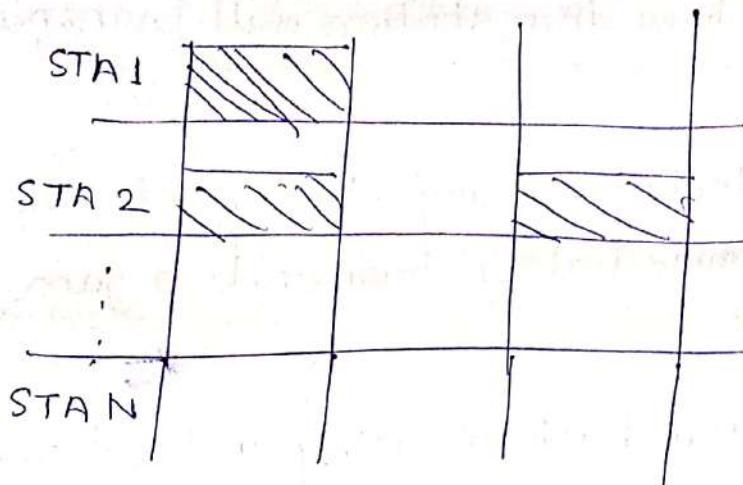


10 Base 5
↓
10 Mbps
↓
Baseband
→ 50 Ω

100 Base T
↓
100 Mbps
→ Twisted pair
Baseband

★ Slotted ALOHA →

≈ 33% bandwidth utilized bandwidth is achieved by using Slotted ALOHA.



(1) MAC protocol is known as 1-persistence CSMA/CD with binary exponential backoff where 1-persistence means

CSMA/CD

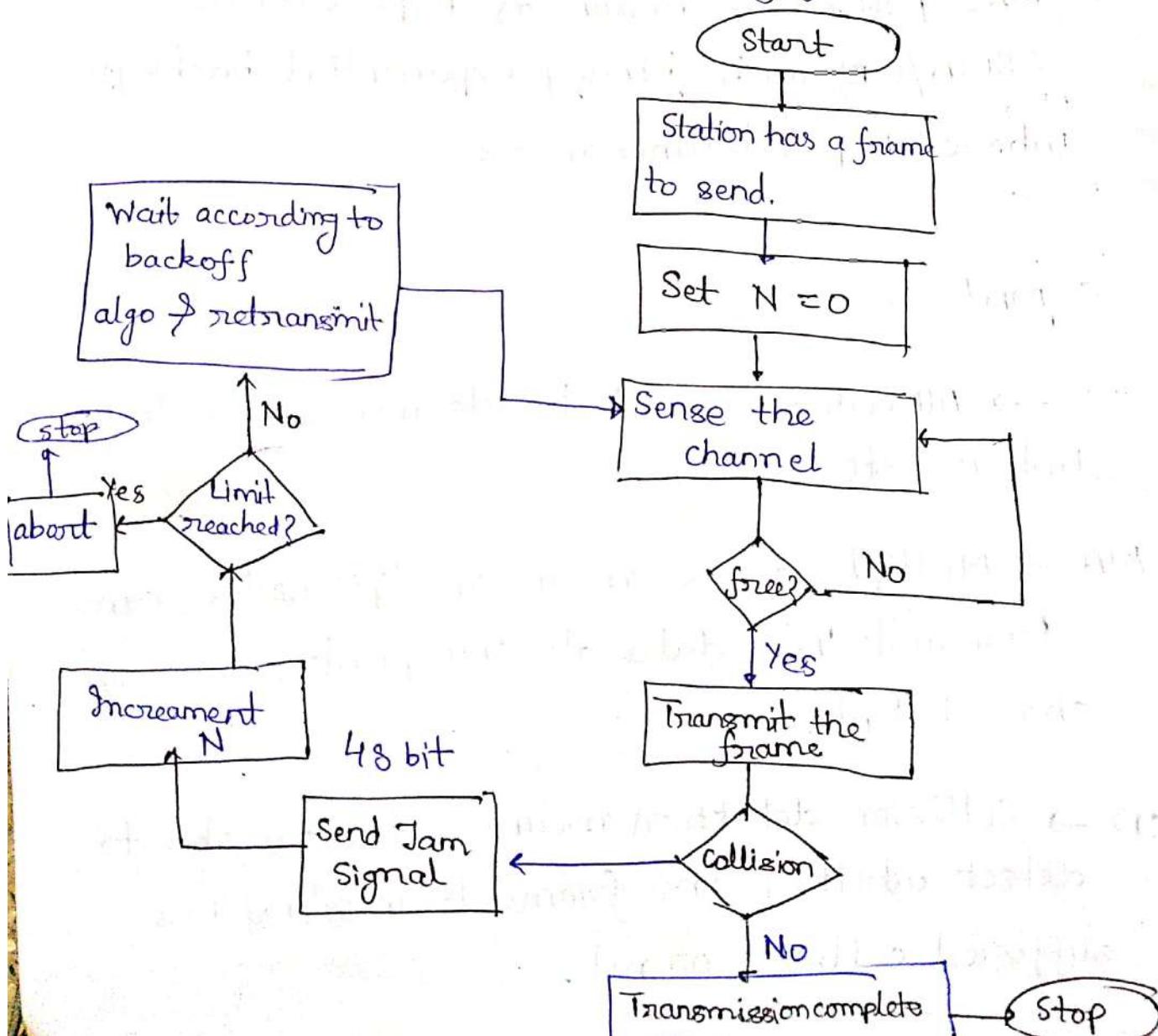
CS → All nodes are able to determine when the link is idle or busy.

MA → Multiple access means multiple nodes can transmit their data at any point of time on shared link.

CD → Collision detection means nodes are able to detect whether the frame transmitted has suffered collision or not.

MAC protocol

- * Whenever a station has frame to send, it senses the channel.
- * If the channel is free, then station will transmit otherwise wait.
- * If collision is detected by any station of the network, then it immediately transmit a jam signal.
- * On collision station backoff using a local counter and try retransmission accordingly.



$N = \text{maximum no. of retransmission.}$

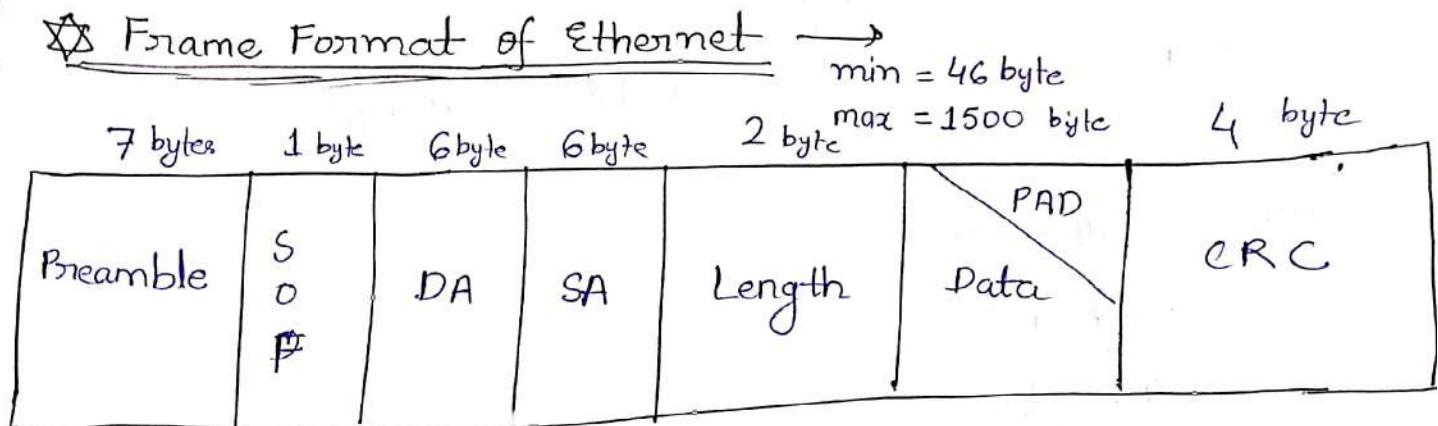
$$N_{\max} = 16$$

★ Backoff algorithm

Binary Exponential Back off algorithm

- * Upon collision, the sending station increments local counter Q . The backoff interval is randomly selected using uniform distribution over L where $L = 2^k$ slots.
- * k is initially set to 0.

★ Frame Format of Ethernet



SOF = Start of frame.

MTU = Maximum Transfer Unit ≤ 1500 byte

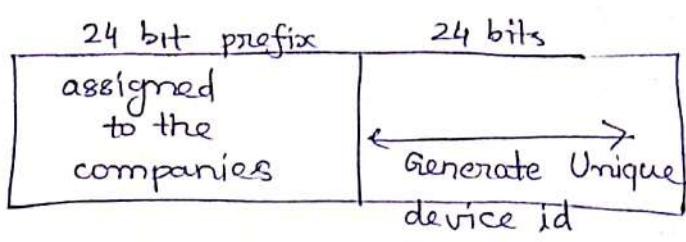
48 bit address (6 bytes) \rightarrow (Globally Unique Address)

\rightarrow Mac MAC address

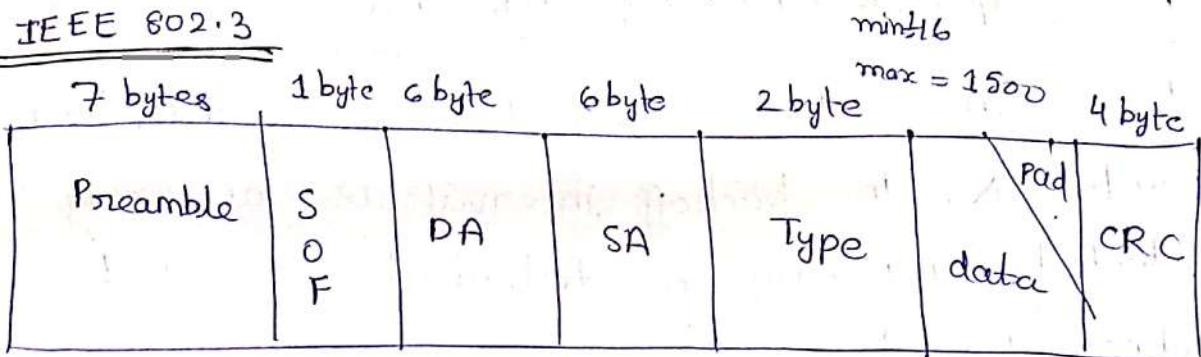
\rightarrow Physical address

\rightarrow NIC \rightarrow Network Interface Card

\rightarrow Global address



IEEE 802.3



RTT = Round Trip Time

for 10 Mbps Ethernet

$$RTT = 51.2 \mu\text{sec}$$

$$\therefore \text{Data} = 51.2 \times 10 = 512 \text{ bits} = 64 \text{ bytes.}$$

$$\text{min} = 64 - (6 + 6 + 2 + 4)$$

$$= 64 - 18$$

$$= 46 \text{ bytes}$$

MAC address

ipconfig → Windows
ifconfig → Linux

2A:46:3C:49:5E:6F

Presentation notation

0100110...01 → Network representation
 ↴
 48 bit

Network Layer

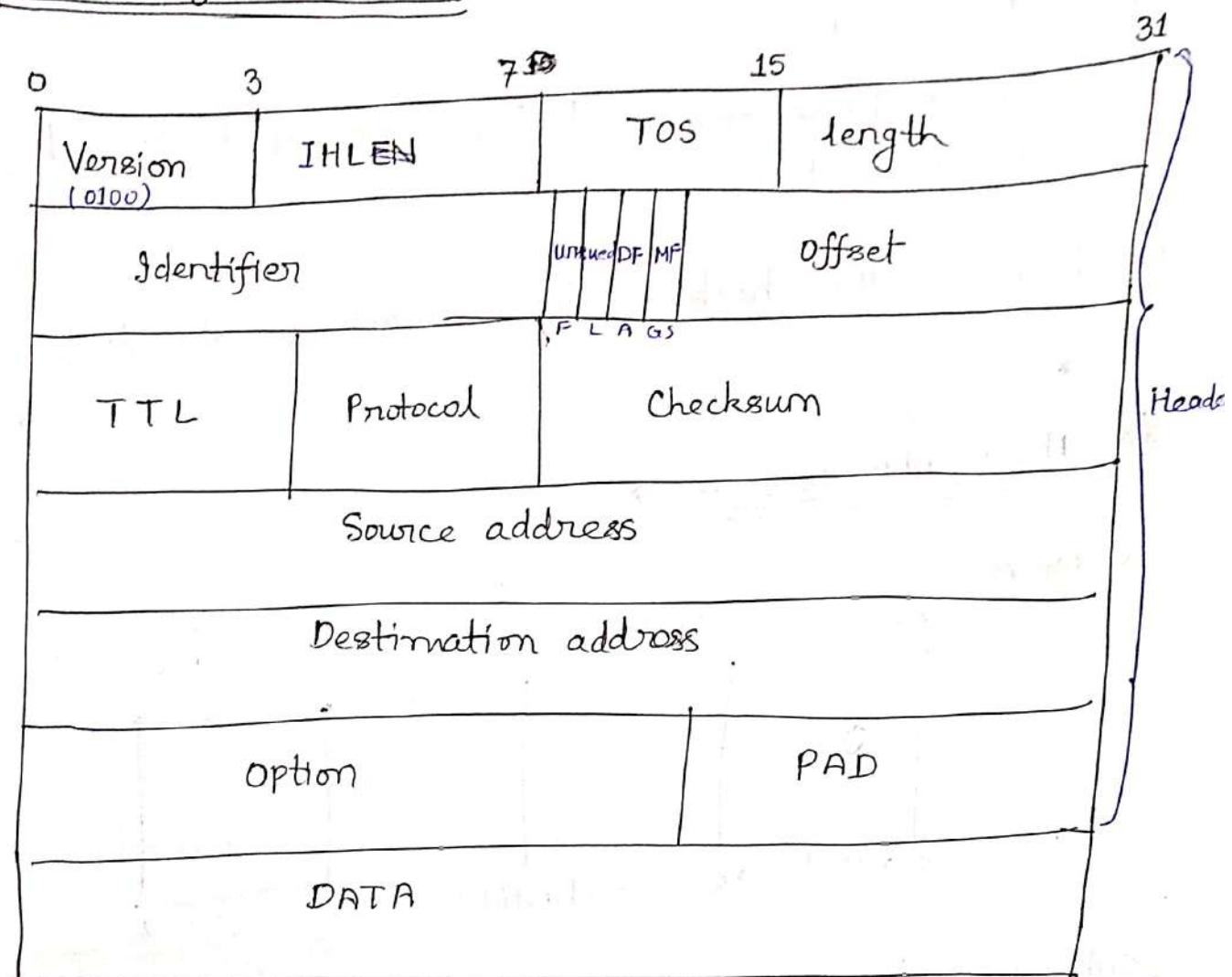
→ IPv4 → 32 bit address

→ IPv6 → 128 bit address

} logical address

* IPv4 is used to define the network and to distinguish network.

Header of IP packets →



IHL = No. of 32 bits word

in the header.

Types of services
(TOS)

- min delay
- max throughput
- reliability
- low cost

Length = No. of bytes in IP packets.

Example

A 450 bytes IP packet

F1 Id = X MF = 1 offset = 0

F2 Id = X MF = 1 offset = 1500

F3 Id = X MF = 0 offset = 3000

If DF = 1, further fragmentation is not done.

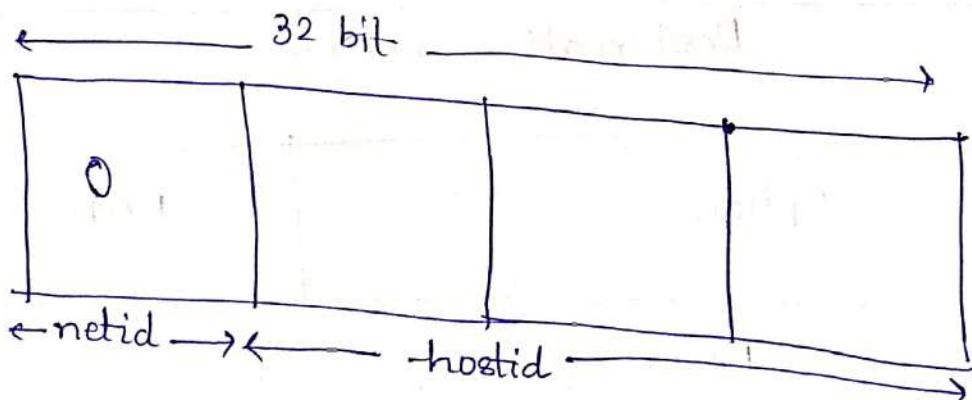
TTL = Time to Leave

= No. of hops required for a packet to reach its destination.

Checksum → This checksum is used only for header.

~~IPv4 addresses~~

Class A



subnet mask = 255.0.0.0

netid = IP & subnetmask

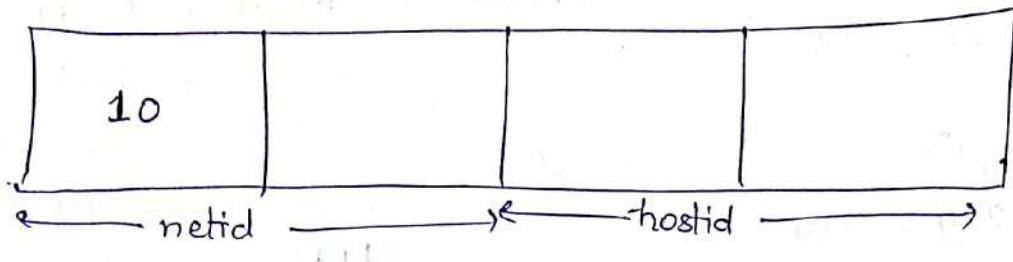
Range = 0 - 126

00000...0

01111...1

127 is defined as a special address (loopback)

Class B



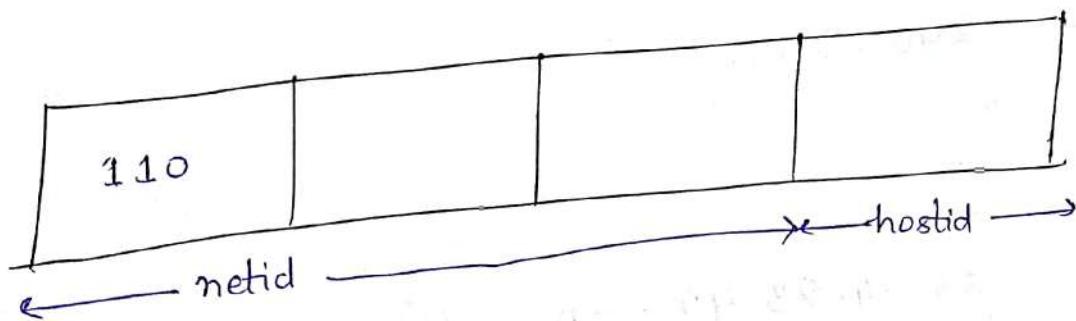
Subnet Mask = 255.255.0.0

Range = 128 - 191

100...0

1011...1

Class C



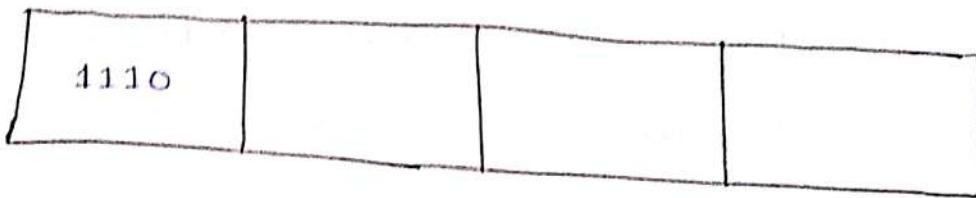
Subnet mask = 255.255.255.0

Range = 192 - 223

1100...0

11011...1

Class D



* Class D addresses are known as Multicast IP addresses.

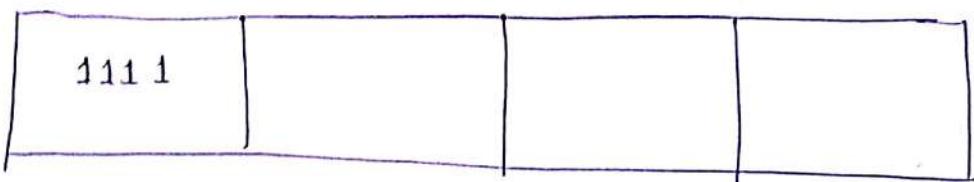
Range 224 - 239

1110 ... 0

11101 ... 1

Class E

Reserved IP addresses



Range - 240 - 255

① 63. 24. 23. 49 → A

⑤ 129. 68. 69. 27 → B

(2) 64. 25. 28. 36 → A

⑥ 129. 69. 39. 28 → B

(3) 63. 29. 28. 47 → A How many networks are there?

(4) 129. 69. 68. 29 → B

(1) 63. 24. 23. 49

$$\text{netid} = \begin{array}{c} 255. 0. 0. 0 \\ \diagdown \quad \diagup \\ 63. 24. 23. 49 \end{array} \quad 63. 0. 0. 0$$

(2) netid = 63. 0. 0. 0

(3) netid = 63. 0. 0. 0

(4) 129. 69. 0. 0

(5) netid = 129. 68. 0. 0

(6) netid = 129. 69. 0. 0

① & ③ belongs to same network as network 1.

Similarly ④ & ⑥ belongs to same network as network 3.

There are four different networks.

* $\frac{63. 255. 255. 255}{\text{destination IP}}$ → Broadcast packet

↳ Every host has to accept this packet and process it in network.

63.0.0.0 also reserved and used for management purpose.

* Loopback address →

The IP specifies a loopback network with the IPv4 address ~~172~~. 127.0.0.0/8.

The most commonly used IP address on the loopback network is 127.0.0.1 for IPv4 and ::1 for IPv6

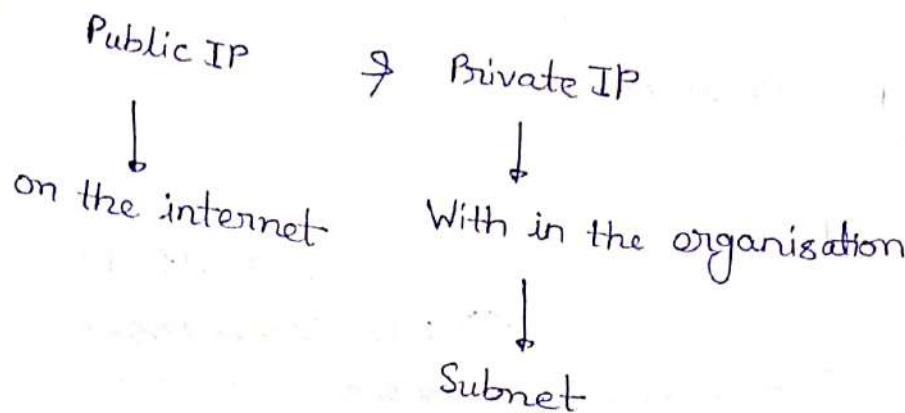
* TOS (Types of Services)

7	6	5	4	3	2	1	0
Parity	low delay	high throughput	Reliability	low cost	Reserved		

* Subnetting

(RFC 917)

(RFC 950) Variable Length Subnetting Mask



e.g.

172.16.0.0

B - class.

Default Subnet Mask = 255.255.0.0
DSM

netid = IP & DSM

000 00000 . 00

Subnetid = IP & DSM

000 11111 . 11..

Rest of the bits = hostid

Subnetid	Start IP	End IP
172.16.0.0	172.16.0.0	172.16.63.255
172.16.64.0	172.16.64.0	172.16.192.255

* Router support is required for subnetting.

example

netid = 192.27.26.0

A = 70 host

1

→

255.255.255.128

B = 64 host

00

→

255.255.255.0

C = 32 host

010

→

255

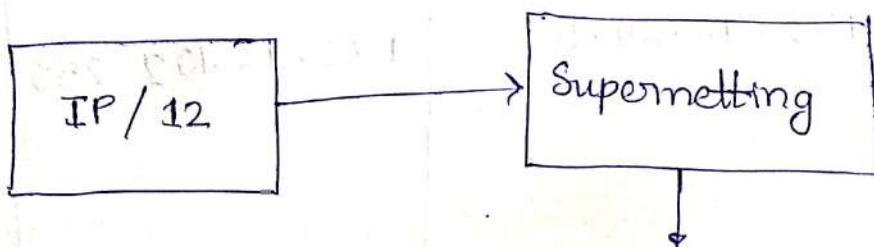
D = 28 host

011

Here VLSM is used.

Classless IP address →

These IP addresses are used in backbone network



To represent set of networks as one element

To reduce the no. of entries in the routing table

Routing table

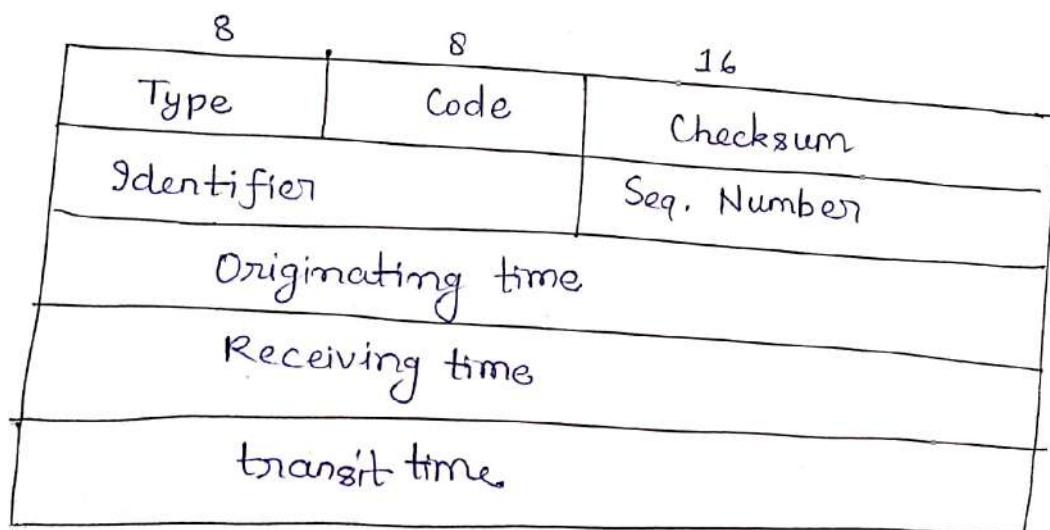
Destination net id	next-hop
172.0.0.0/10	X

ICMP

(Internet Control Messaging Protocol)

→ Also known as error reporting protocol.

Packet format



Code	meaning
0	Network unreachable
3	Port unreachable
4	Fragmentation needed if DF = 1

Type	meaning
0	echo reply
3	destination unreachable
4	source quench
11	time exceeded.
	:

~~Router~~ Router \Rightarrow Interface Id

- (1) IP is assigned to the interface and not the device.
- (2) Router is a device which is having multiple interfaces and each interface is connected to a different network.

* Routing protocol and forwarding algorithm

To create, update and maintain the routing / forwarding table

Take a packet out from the queue and forward it on appropriate interface.

forwarding Algorithm

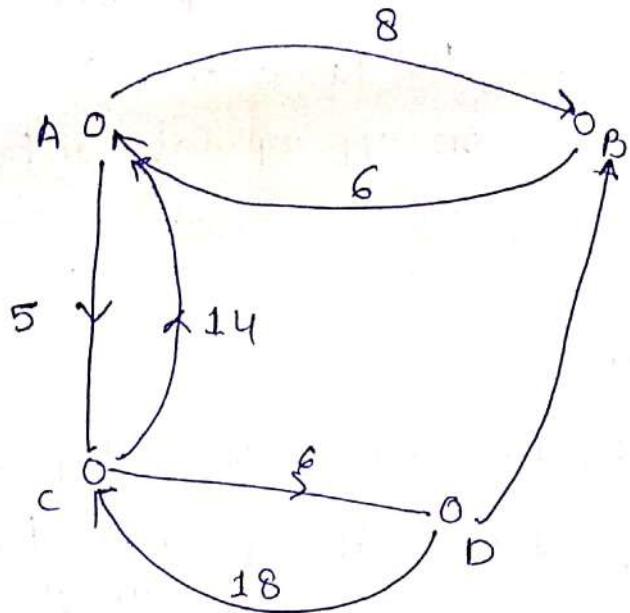
If [netid of destination equal to netid of one of the interface, then deliver the packet to the destination over that interface] ~~else if~~ netid of the destination is in the routing table then deliver the packet to the next hop else , deliver the packet to the default ~~hop~~ router.

* Routing Protocols →

* The routing protocols are representing network as graph $G(V, E)$ where routers represent set of vertices and link between routers are represented as set of edges.

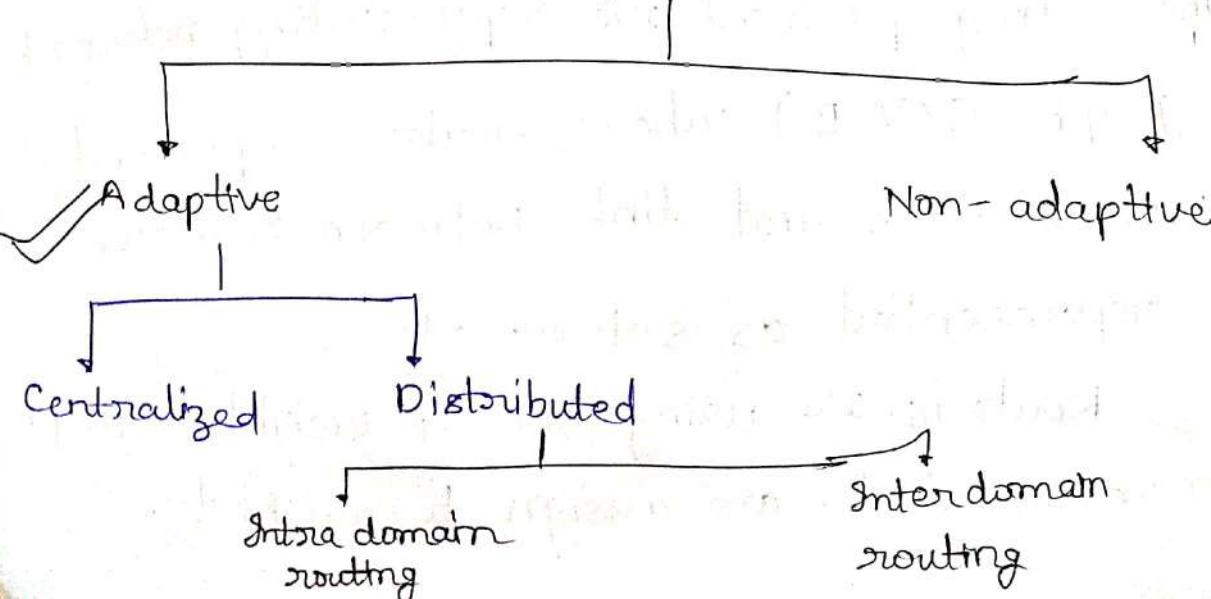
Routers are using set of weighted graph which means links are assigned to weight.

They are also using directional graph wherein weights can be different on two direction of same link.



A domain is a set of networks managed by one administrative entity. Domains can also be represented by domain id (32 bits)

Routing Protocols



Intra-domain routing

Distance Vector
(RIP)

- Routing Information Protocol $V_1 \neq V_2$
- Use distance vector routing algorithm.
- periodic update interval = 30 sec

Link State
(OSPF)

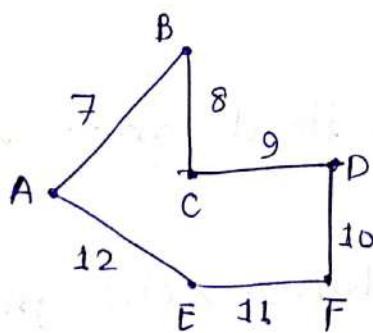
Inter domain routing

Link state
[OSPF]

Path vector
(BGP)

Distance Vector Algorithm

- (1) Historically known as old ARPANET routing algorithm, as well as Bellman - Ford algorithm.
- Each router maintains its ID (IP address).
- It also maintains list of neighbours and its cost.



for D

C	F
9	10

Distance Vector

Destination cost

A	0
B	x
C	y
D	z
:	:

Reachability to itself = 0
Unreachable nodes = ∞

Routing Table

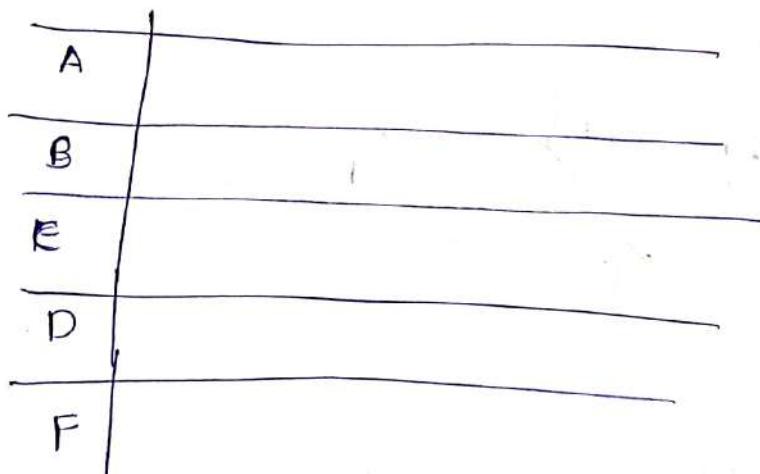
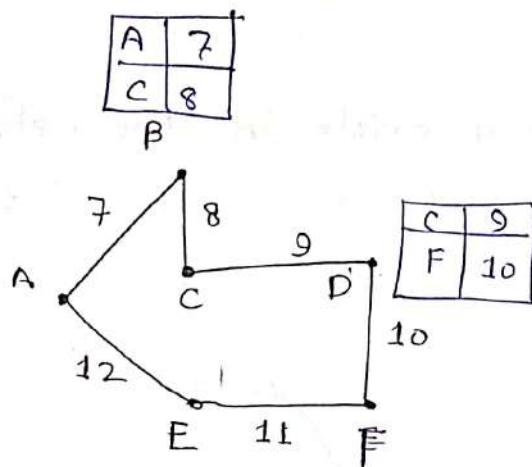
Present	cost	Next hop	timer

Algo

- Each router transmit its distance vector to its neighbours.
- Each router recalculate its distance vector when —
 - (a) It receives distance vector from its neighbour containing different information,

(b) It discover the topology change.

→ On receipt of distance vector, each router determine the min cost path for a destination and add that entry in the routing table.

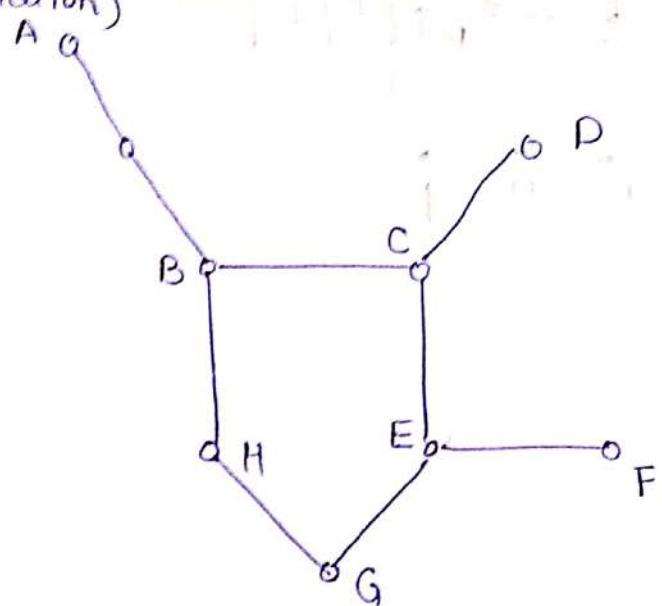


Advantages of DV →

- (1) [Redacted] * 20 bytes are allocated to create distance vector,
- (2) Overload is load. low.
- (3) Good information propagate very fast.

Disadvantages

- (1) bad information may never converge (or take too long time to converge)
(ex. count to infinity)
- (2) Routing loops may exists in the network.
(Because every node take decision based on the local information)



Solutions (Used with RIP protocol)

- (1) Max hop count = 16
- (2) Split Horizon with poison reverse.
(also eliminate two level loop)
- (3) triggered entry update.

Link State Routing Protocol

Date - 14/10/19

* In link state routing, each router do the following -

- (1) Discover its neighbours and learn their network address.
- (2) Measure the cost ^{to} of all its neighbours.
- (3) Capture Register the packet containing all information and call it LSP (link State Packet)
- (4) Send this packet to all other routers to network using reliable fading.
- (5) Every router will construct a local topology of the network using information of LSPs.
- (6) Compute shortest path to all the other destinations using local topology and apply ~~Dijkstra~~ Dijkstra algorithm for finding shortest path for each destination.
- (7) Create the routing table using next hop router in the shortest path.

Dijkstra's Algorithm

→ Initially mark all nodes except source with infinite distance and a blank.

→ Set working node = source node and sink node = destination node.

→ While

Working node \neq sink node

(1) Mark working node as permanent node

(2) Examine all adjacent nodes

If the sum of label of working node + cost to the adjacent node \geq the current label distance

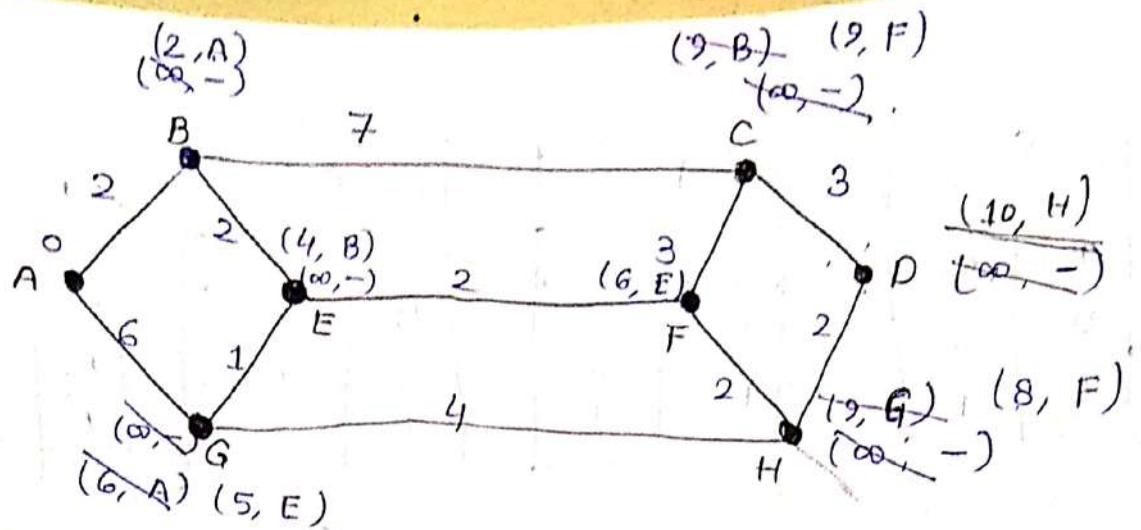
then re-label the distance on adjacent node with the cost +

(3) Examine all the nodes, except permanent nodes and mark the smallest labelled node as permanent node.

(4) Set working node = permanent node.

While

reconstruct the path back from sink to source

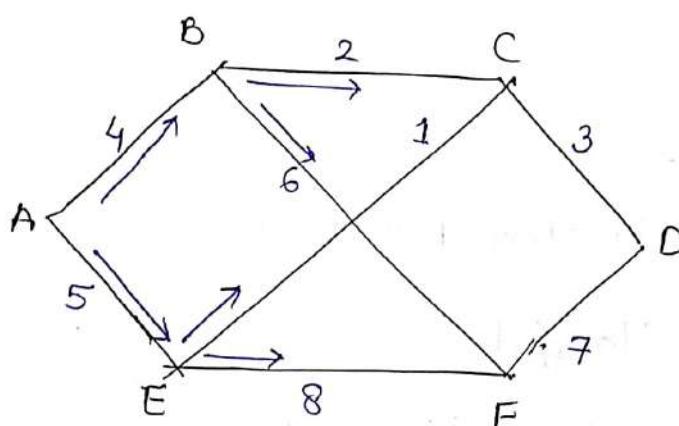


for A to D

Working Node = A, B, E, F, H
Sink Node = D

D → H → F → E → B → A

Link State Algorithm



LSP

A	
Age	
Seq No.	
B	4
E	5

(How long this information is valid)
to track recent info. and avoid duplicate

B	
Age	
Seq No.	
A	4
C	2
F	6

C	
B	2
D	3
E	1

D	
C	3
F	7

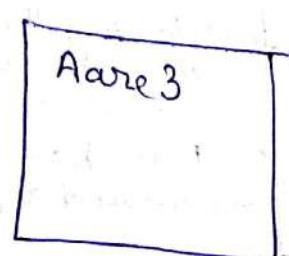
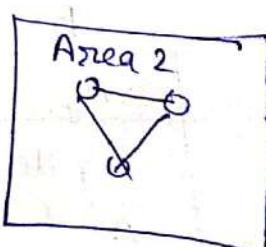
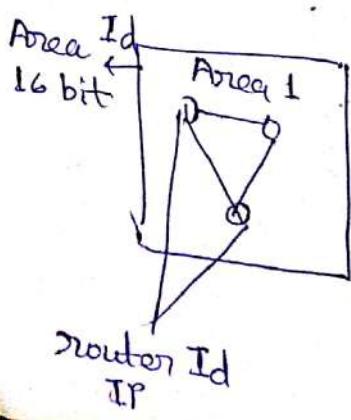
E	
A	5
C	7
F	8

F	
B	6
E	8
D	7

* OSPF (Open Shortest Path First)

(1) Classless / Classful

(2) Maintain 2 level hierarchy in the network.



- (3) Send broadcast on any change, on after 30 min.
- (4) LSP are referred as LSA (Link State Advertisement)

OSPF maintain 3 -tables

- (1) Neighbour table
- (2) topology table
- (3) Routing table.

Note → 5 different types of LSA are used by OSPF

- (1) Router LSA
- (2) Network LSA
- (3) Network summary LSA
- (4) ASBR summary LSA
(Autonomous System Boundary Router)
- (5) External LSA

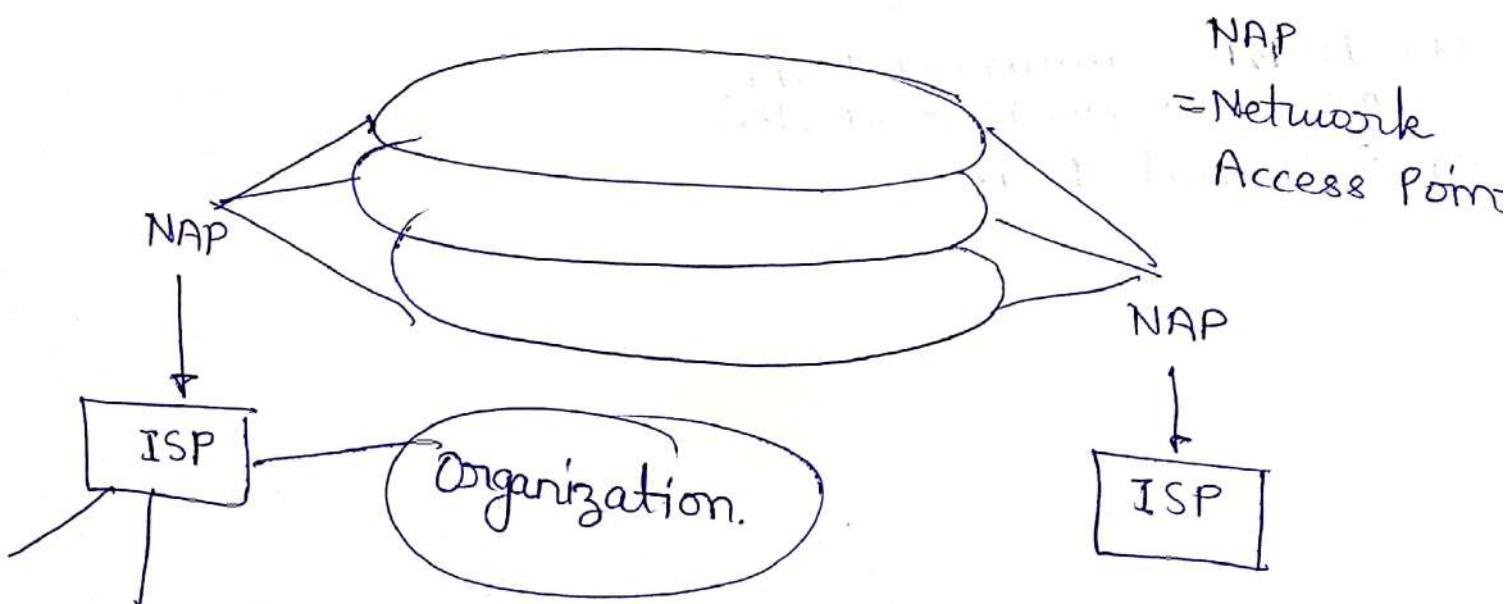
BGPv4 (Border Gateway Protocol)

- It operates as path vector routing.
- Also similar to distance vector routing.

Path vector

destination	Path	Next hop	cost

CIDR → Class less Inter Domain Routing



Transport Layer

- * It provides connection oriented services which is reliable service by TCP protocol.
- * Extends the unreliable service of the n/w layer (UDP).
- * Providing port number (16 bit)
(multiplexing and demultiplexing of data stream)
- * Every socket should provide unique port no.)
- * Congestion Control

UDP (User Datagram Protocol) →

Src. Port No.	Dest Port No.
Checksum	length
Application data	

- * It is the simplest transport layer protocol.
- * It simply extends the underlying best effort data delivery services of network layer to application layer.

- * It adds source port no. and dest. port no. in the header to recognize a socket in a host.
- * Checksum is used for error checking.
- * Length is used to determine no. of bytes in application layer.

Transport LayerTCP Header

- * TCP provides connection orientation which means it requires an explicit connection establishment and closing phase.
- * TCP is connection oriented and byte oriented protocol.
- * TCP provide full duplex communication.
- * TCP uses adaptive timeout mechanism.
- * TCP also provide congestion control.
- * TCP provides end to end flow control.

Source Port No.	Dest. Port No
Seq. Number	
Ack No.	
4 HLEN	6 bits reserved 0
6 flags.	Advertized Window.
Checksum	Urgent Pointer.
Option // pad	
	Application data

* ... and doct.

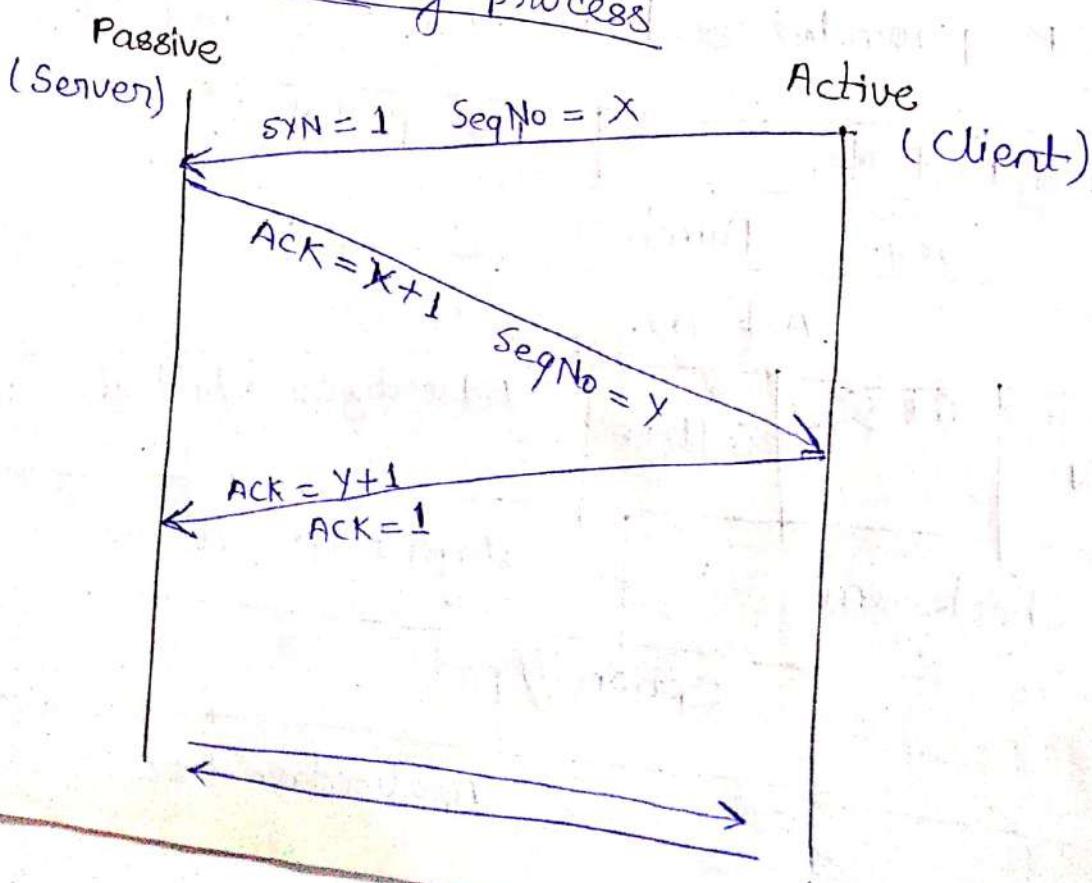
- * Seq No., Ack No., Advertise window is used to implement flow control. (Sliding Window Protocol (SWP))

* Flags

- (1) URG → Urgent Pointer
- (2) ACK → Acknowledgement.
- (3) PSH → Push
- (4) RESET
- (5) SYN SYN → Synchronization
- (6) FIN → final.
(Used for terminating the connection)

* Connection establishment in TCP

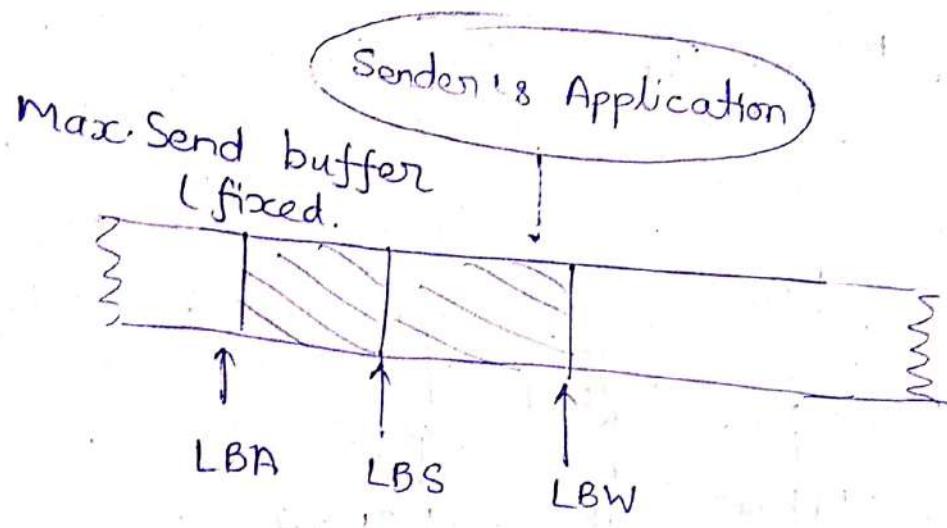
3-Way handshaking process



(Source Port No., Source IP
Dest. Port No., Dest. IP) \rightarrow provides unique
TCP connection.

* Flow control by TCP using SWP \rightarrow

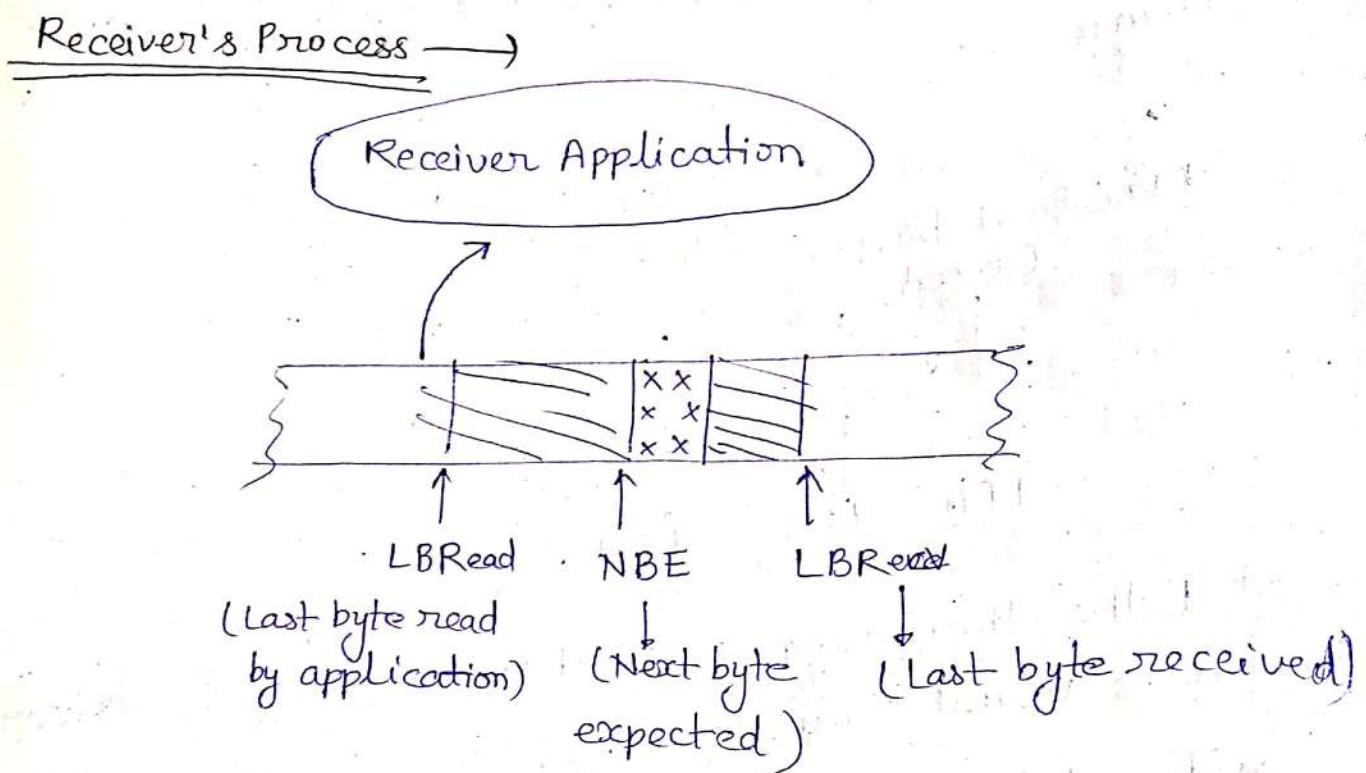
(1) The same SWP protocol as discussed in data
link layer will be used by TCP with following
changes —



- * Rather than using a fixed window size, receiver uses variable window size.
- * Two buffers are maintained
 - (a) Max send buffer
 - (b) Max receive buffer

- * It adds source port no. and dest. port no. in header
- * Sender maintains three variables
 - (a) LBA → Last byte Ack.
 - (b) LBS → Last byte Send
 - (c) LBW → Last byte written

- * Length application



- * Receiver receive data if $[LBRecv - LBRead] \leq \text{Max. receive buffer}$.
- * Then it compute advertize window.

$$\text{Advertize window} = \text{Max. receive buffer} - [LBRecv - LBRead]$$

$$\text{Effective Window} = \text{AdvertiseWindow} - (\text{LBS} - \text{LBA})$$

* Sender can subsequently send effective window amount of data.

$$\text{MSS} = \text{Max. Segment Size}$$

* TCP uses adaptive timeout for the segment it sent.

* The idea is take a running average of RTT and then compute the timeout value as the fraction of RTT.

Estimated RTT_n

$$= \alpha * \text{Estimated } \text{RTT}_{n-1} + (1-\alpha) \text{ Sampled } \text{RTT}_{n-1}$$

$$\alpha = 0.8 \text{ or } 0.9$$

Timeout for n^{th} segment

$$= 2 \cdot \text{Estimated } \text{RTT}_n$$

Case 1

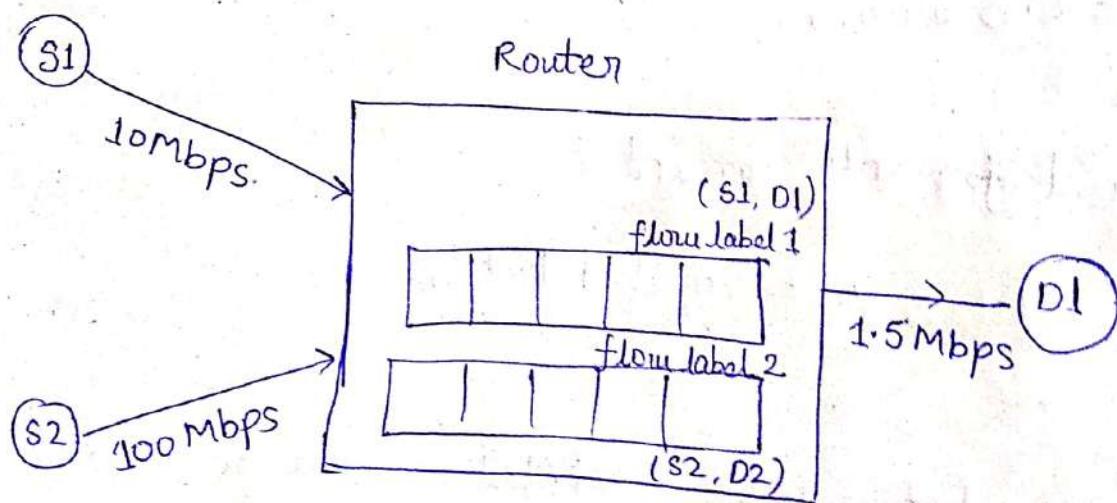
$$\text{estimated} = 10 \text{ s} \quad \text{sampled} = 8 \text{ s}, \quad \alpha = 0.8$$

$$\Rightarrow \text{Estimated } \text{RTT}_n = 9.6 \text{ s}$$

case-2

Advantages of SWP

- * SWP provides reliable and guaranteed delivery of data.
- * Seq. No. ensures order of data.
- * Advertise window enforce flow control.
- * One more window is used with TCP protocol known as congestion window which will be used for congestion control.

Resource Allocation and Congestion ControlNetwork Layer

- Weighted Fair Queuing →
- * To provide equal share of resources to every user at every round, it is required to transmit bit-by-bit data from every flow in round-robin fashion.
 - * This scheme can be simulated using weighted fair queuing algorithm.

Algorithm Step →

- (1) Suppose clock fix each time a bit is transmitted.
- (2) Let P_i denotes the length of packet i
 → S_i denotes the time at which packet P_i start getting transmitted.
 → F_i denotes the time when packet i finish its transmission.
 A_i = Arrival time

$$F_i = S_i + P_i * t \quad (\text{unit time})$$

i denotes any packet from any flow.

$i = 1, 2, 3, \dots$ for every incoming packet.

When router will start transmitting packet?

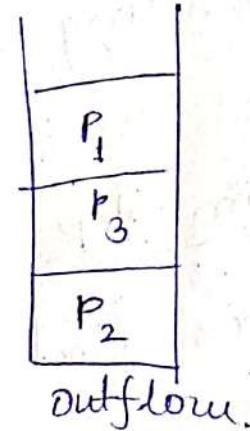
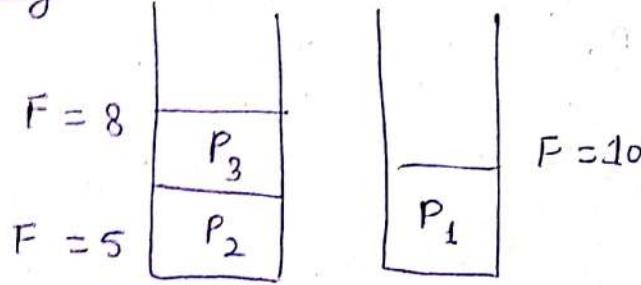
$A_i \leftarrow$ when no packet in queue

$F_{i-1} \leftarrow$ when packets are waiting in queue

then

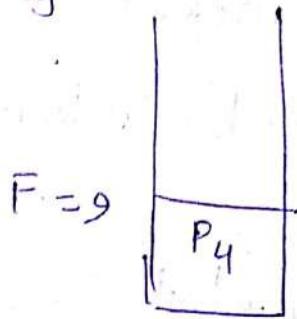
$$F_i = \text{Max}(F_{i-1}, A_i) + P_i * t$$

e.g.

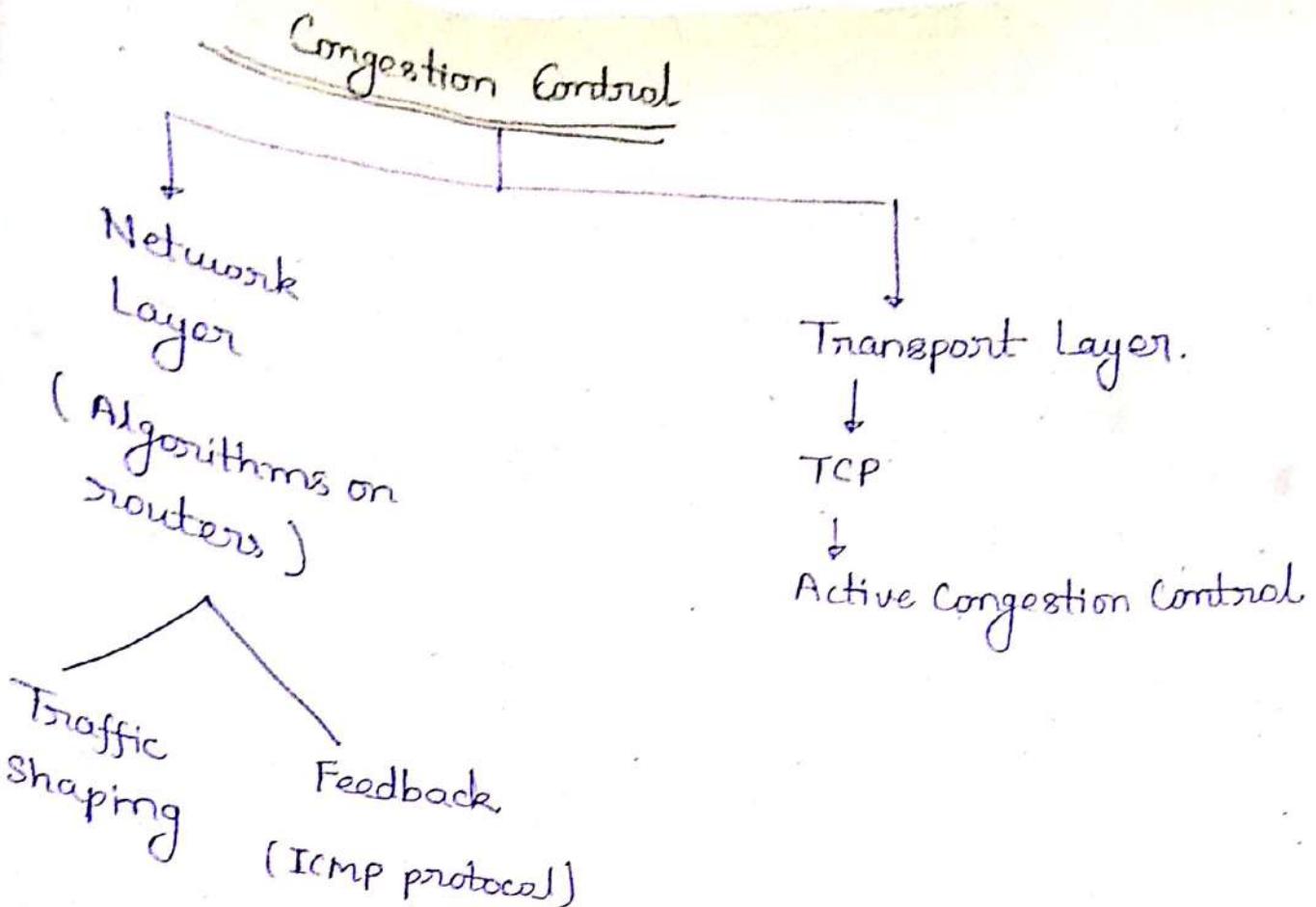


non-primitive

But if



It has to wait, and will be transmitted only after P_1 .



Congestion avoidance (Proactive)

Congestion Control (Reactive)