- voice : $f_L = 300Hz$ $f_H = 3300Hz$ $\therefore$ Bandwidth = 3000 Hz
- Every non periodic wave can be obtain by combining diff periodic waves
- Any signal that carry more info. will have more fluctuation wrt time

- Audio : $f_L = 20Hz$ $f_H = 20000 Hz$
- Info in signal $\propto$ Bandwidth of signal.
- Converting analog signal wrt time to discreate signal is called sampling

- Discreate signal are those for which we dont have signal for each instance of time
- Digital signals are discreate signal where amplitude level are predefine & finite i.e also discontinue wrt amplitude
- A subcateogary of DS is binary signal with 2 possible amplitude only

Q. Why digital is preffered over analog.
- In digital signal (binary case) we can supress noise at reciever end using repeater & it regenerate the signal removing the noise, in case of analog noise can't be removed at reciever end.
- Noise is not completely removed from digital signal.
- Error detection & Correction is better in case of digital signal.

- Transer signal from A to B is transmission & if someone meaningfull recorvered at reciever end it is communication.
- $\xrightarrow{\text{Analog}}$ $\boxed{\text{Sampling}}$ — $\boxed{\text{Quantization}}$ — $\boxed{\text{Encoding}}$

  Sampling: $2 \times f_h$ sample persecond atleast

  Quantization: levels are defined & are finite

- Info of digital signal can be calculated using bit rate & baud rate, using nyquist bit rate & Shanon Theorum
  $= 2B \log_2 L$ $= B \log_2 (1 + \frac{S}{N})$ $SNR = 10 \log_{10} (S/N)$

## Transferring file from comp.1 to comp2 →

- Impedence of insulator → ∞
- Cable designed in such a way only certain signal pass without supression else it will block it.
- Coaxial cable (TV) design in a way so they can hold high frequency.
- To avoid power loss we try to reduce resistance of wire
- General speed of light
  2/3 * speed of light

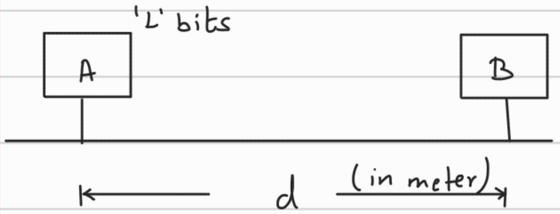
- Every channel has capacity that how much it can transfer.
- Any channel has 3 parameters which affect its capicity
  Resistance ($\Omega$), inductor (Henry) & capacitor (c).
- Combinely all factor is ampidence

$$Z = \sqrt{R^2 + (X_L - X_C)^2}$$

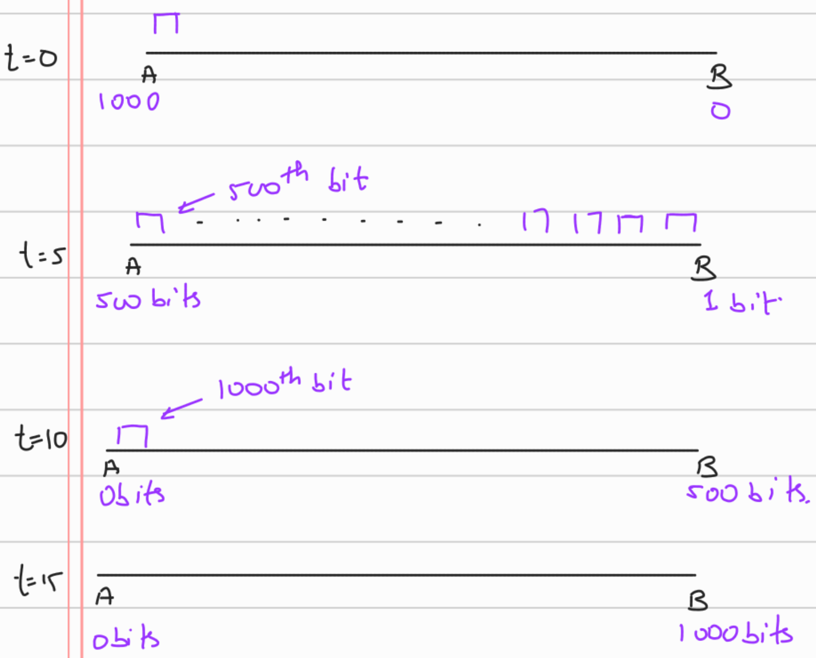
$$X_L = 2\pi f L \quad (\text{in } \Omega)$$

$$X_C = \frac{1}{2\pi f C} \quad (\text{in } \Omega)$$

- If $f \uparrow\uparrow$ for particular L & C it will block the signal.




V: Signal propogation speed (m/s)
R: Channel Capacity (in bps)
(let L = 1000, d = 1000, V = 200
R = 100)
↳ After interval of 1/100 sec it pumps up a new bit




- **Transmission time** – time required to pump a (standard size) data block into the channel.

  Transmission time = Data block size / Channel Capacity = L/R

- Propagation delay - PD b/wⁿ two points is the time taken a signal to reach from 1 point another point.

$$PD = \frac{\text{distance b/w}^n \text{ two point}}{\text{Signal prop. velocity}} = \frac{D}{V}$$

- Transfer time - Time req. to transfer datablock from A to B

$$\text{Transfer time} = Tt + PD = \frac{L}{R} + \frac{D}{V}$$

Channel utilization → $\dfrac{\text{Useful time}}{\text{Total time}} = \dfrac{\text{time req. to pump useful bit into channel}}{\text{Total time}}$

$$= \frac{\text{Transmission time}}{\text{Transfer time}}$$

$$\frac{\text{Useful time}}{\text{Useful time + Overhead time}} \Leftarrow \frac{L/R}{L/R + \underbrace{d/V}_{\text{overhead time}}}$$

Throughput → $\dfrac{\text{No. of bits (datablock) transfered}}{\text{time}}$

$$= \frac{L}{L/R + D/V}$$

∴ Channel Utilization = $\dfrac{\text{Throughput}}{\text{Channel Capacity}}$

Length of Channel →

in bits ⇒

- Time at which sys. A submits 1st bit is $= 0$
- Time when 1st bit will reach to B $= d/v$
- A is pumping 'R' bits into the channel per second
- Number of bits pumped by A into channel in time $= d/v$ sec is $R * \frac{d}{v}$ bit

length of channel in bits is $R * \frac{d}{v}$ (bits)

in term of data block (let L) $= \frac{Rd}{vL} = \frac{d/v}{R/L} = \frac{\text{Prop delay}}{\text{Transmission time}}$

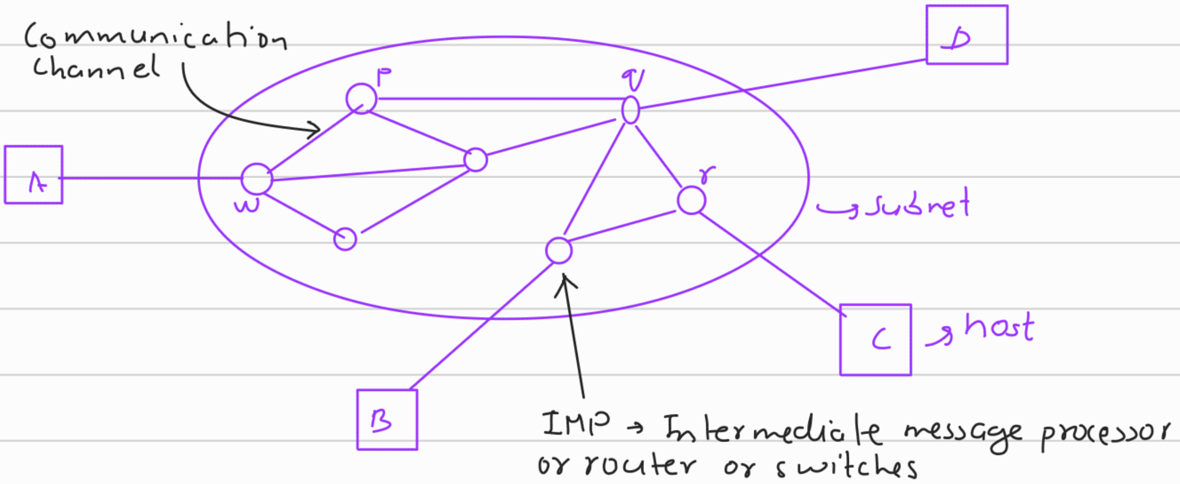
$$\text{Channel utilization} = \frac{L/R}{L/R + d/v}$$

$$= \frac{1}{1 + \frac{(d/v)}{(L/R)}}$$

$$= \frac{1}{1 + l}$$

## Structure of computer Network →

- It is interconnection of autonomus computers
  - Homogenous
  - Heterogenous (of both s/w & hardware)
- Two basic entity Hosts & Subnet, both connect to form network
- Host are comp. who are source & sink for the data.
- Subnet is like transport agency which collect data from 1 host to another host.




Ex Host are Us & Subnet is post office

- renponsibility of router is to take routing decision to select best path.
- Better path among all path will be taken is selected
- At each router, routing decision will be taken.
- LAN - local area N/w
  - In manit
  - Owned by manit

- MAN - Matropolitin area N/w
  - Manit to Aiims bhopal
  - Multiple owner

- WAN - Wide area N/w
  - Connect across cities, countries
  - Mostly we take 3rd service provider

Switching →

- Circuit switching (Connection Oriented methode) — Before actual data transer conn. need to be established
- Store and forward switching
  - Message switching
  - Packet switching
    - Datagram packet switching
    - Virtual ckt packet switching

- For voice data rate req is 64Kb/s
- Fix size data block is called packet.

packet :-
(gen. from it (let))

| DA | SA | other data |
|---|---|---|

CR packet
((conn. request)

DA : Dest. Add.
SA : Src. Add.

- Say 'w' accept req of A then this channel is reserved for them but sometime multiplexing can be used to use maximum capacity of voice.
  reserving a band instead of whole channel

- when 'w' recives it it look DA & take a routing decision say it select 'P' and reserve channel again b/w them for comm.

- when 'X' recives this 'CR' packet then channel is been created b/w 'A' & 'X'

- If 'X' is ready to recive then 'X' will send 'CC' (connection confirm) packet to source, 'CC' don't have source & dest. addres it just send some parameter or info. to source

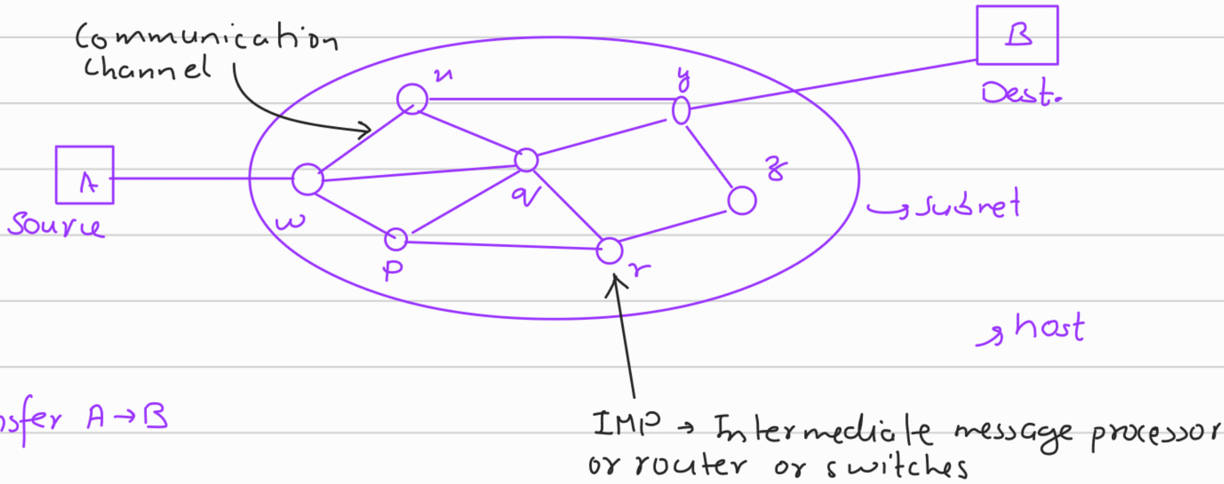
- Once 'CC' reach to 'A', 'A' knows 'X' is ready to accept data & connection is ready

- Now data is just need to pumped to channel & no need of data address or any rounter in transmission

- This is circuit switching

- Practically we are not using this in computer N/w but we use it in voice communication
- The reason is uptill conn. is their A-X channel is reserved
- Due to this channel utilization is very very poor.
- Data is not store anywhere in the channel ∴ it is good for interactive comm.

## Store & forward →




- transfer A → B

- Since channel & bandwidth utilization is poor in ckt switching ∴ we move to store & forward ∵ Channel & bandwidth is most significant resource.
- Here in S&F we don't reserve the bandwidth for comm b/w A to B
- Based on usage if someone is using it will be in router else if line is free it will forward the data.

- Channel utilization in S&F > Channel utilization in ckt switching
- prop. delay + data storage duration is delay ∴ we can't gurantee the amount of time in which data transfer from A → B
- Not good for interactive kind of communication

## Message switching →

- Same as above
- It is not practical to implement practically
- Main issue is buffer management
- To store msg. router must have that much buffer size.
- Storage can store for long time, buffer only store for a small unit of time.
- Their is no fix size of buffer it depend on msg. ∴ We can't predict
- If we keep big size buffer (∞) then cost will be very huge.
- Buffer will be under used (big msg once or twice a year)
- For transfering sms we use message switching.

↓
short msg service

- Msg switching is connection less

## Packet Switching →

- Same as message switching.
- Say A sends 11KB msg to B then in Packet switching we break msg in standard size data block for packet (say 1KB) in stand-ard packet
- Now it will send packets one by one
- Buffer is being utilized in efficient way

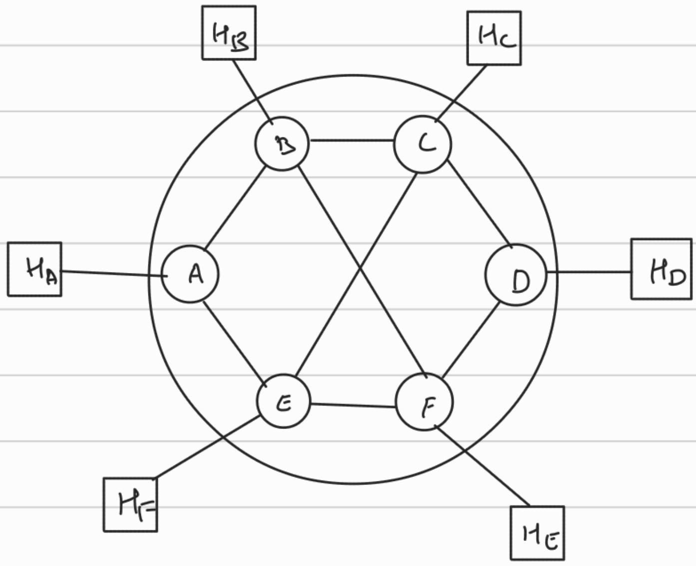
## Datagram pkt switching →

- It is connection less
- Similar to packet switching
- Each packet is send to channel with src add & dest add
- All packet may follow diff. path based on network status at this condⁿ
- This may cause disordering of data, one reach early & other with delay (diff. path)
- ∴ proper reordering is to be done ∴ we have to put some seq. no. in each & every pkt ∴ we will sort it & merge it then msg is ready.
- Each pkt is treated as diff. entity in channel.
- If some pkt is lost or damage that we will discuss later.

## Virtual ckt pkt switching →

- It is store & forward switchig
- It is connection oriented. (before transmission both party handshake)
- Initially src send CR kind of packet (same as ckt switching) additi-nally
- It is virtual connection as no physical channel is preserved

Ex-




Suppose in this N/w$^k$ following connection are to be establish one by one

- A, B, C, D
- B, C, D
- A, B, F, D
- E, A, B
- F, B, C
- D, C, E, A
- C, E, F
- E, F, D

- A, B, C, D $\quad H_A \longrightarrow H_D$

CR packet

| Dest. Add. | Src. Add. | | other imp data |
|---|---|---|---|

↑ Virtual ckt number

How router choose virtual ckt. number ?

- Virtual ckt number will always be lowest unused integer in that Link

A → B → C → D

JAB routing decision lega tab router table me entry karega

Table A

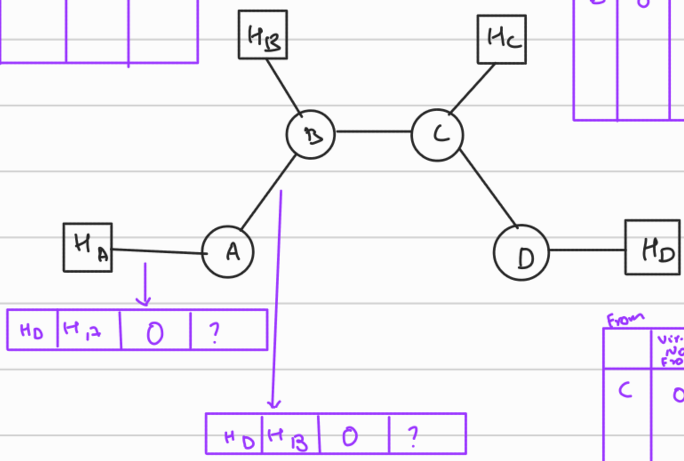
| From | Vir. pac. No. From | to | Vir. pac. No. TO |
|---|---|---|---|
| A | 0 | B | 0 |

A → B kuch use nahi ∴ Vir. pack. to = 0

| From | Vir. pac. No. From | to | Vir. pac. No. TO |
|---|---|---|---|
| B | 0 | C | 0 |

B → C koi. no. nahi ∴ V.N.T = 0

| From | Vir. pac. No. From | to | Vir. pac. No. TO |
|---|---|---|---|
| B | 0 | D | 0 |




when line is free

| From | Vir. pac. No. From | to | Vir. pac. No. TO |
|---|---|---|---|
| C | 0 | $H_D$ | 0 |

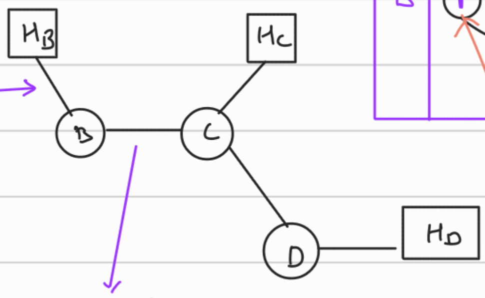
- After receiving we send confirmation 2 ways
  1- New path use karo
  2- Jis se aaya usse bhejo

- After Connection confirm A knows connection (ckt) is setup b/w $A \to D$
  i.e route is made
- Now $H_A$ will dividive msg. in fix size block called packet, in each packet it will put virtual ckt. number in each packet
- Ab routing decision nahi lena jo table bani hai wo padhega aur aage bhej dega
- Till path will be not free mention in table packet will be stored in router
- The problem of random ordering in datagram is solved as path is fix & packet will wait
- Till data is not transferred the path is not yet released

- When $2^{nd}$ path $B \to C \to D$ try to make connection.

| From | Vir. pac. No. From | to | Vir. pac. No. TO |
|---|---|---|---|
| B | 0 | C | 0 |
| $H_B$ | 0 | C | 1 |

yaha kuch nahi transfer ho ra
∴ ckt no = 0




| From | Vir. pac. No. From | to | Vir. pac. No. TO |
|---|---|---|---|
| B | 0 | D | 0 |
| B | 1 | D | 1 |

1 se aa raha

yaha to bhej rah at 0

same we kia to sab mix ho jayega

∴ 1 use hogo

| From | Vir. pac. No. From | to | Vir. pac. No. TO |
|---|---|---|---|
| C | 0 | $H_D$ | 0 |
| C | 1 | $H_D$ | 1 |

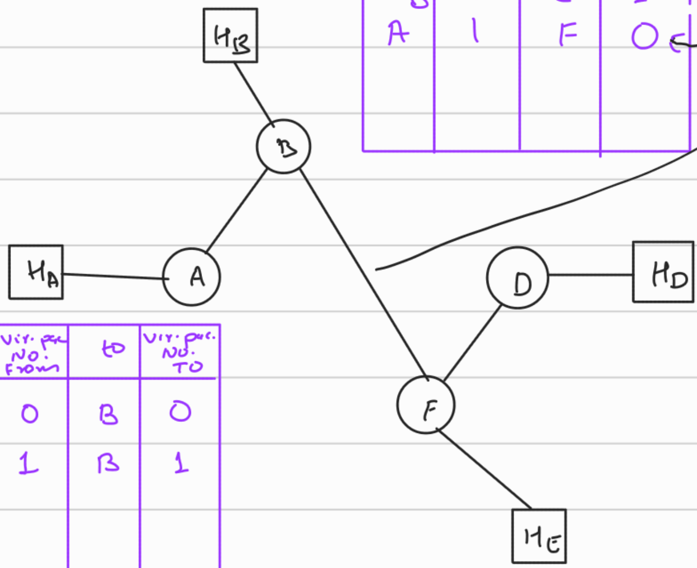
- Ab jo virtual packet no hai to sab mix nahi hoga sab seprate out hojayega

- When connection release table entry will erase
- Connection establishment is cheaper, release is much costlier

- Now make connection ABFD

| From | Vir. pac. No. From | to | Vir. pac. No. TO |
|---|---|---|---|
| B | 0 | C | 0 |
| $H_B$ | 0 | C | 1 |
| A | 1 | F | 0 |

ye to '0' hai




| From | Vir. pac. No. From | to | Vir. pac. No. TO |
|---|---|---|---|
| $H_A$ | 0 | B | 0 |
| $H_A$ | 1 | B | 1 |

| From | Vir. pac. No. From | to | Vir. pac. No. TO |
|---|---|---|---|
| C | 0 | $H_D$ | 0 |
| C | 1 | $H_D$ | 1 |
| F | 0 | $H_D$ | 2 |

| From | Vir. pac. No. From | to | Vir. pac. No. TO |
|---|---|---|---|
| B | 0 | D | 0 |

Now path E A B →

* EA and AE ko alag alag 0 start karenge

| From | Vir. pac. No. From | to | Vir. pac. No. TO |
|---|---|---|---|
| $H_E$ | 0 | A | 0 |

$H_E$

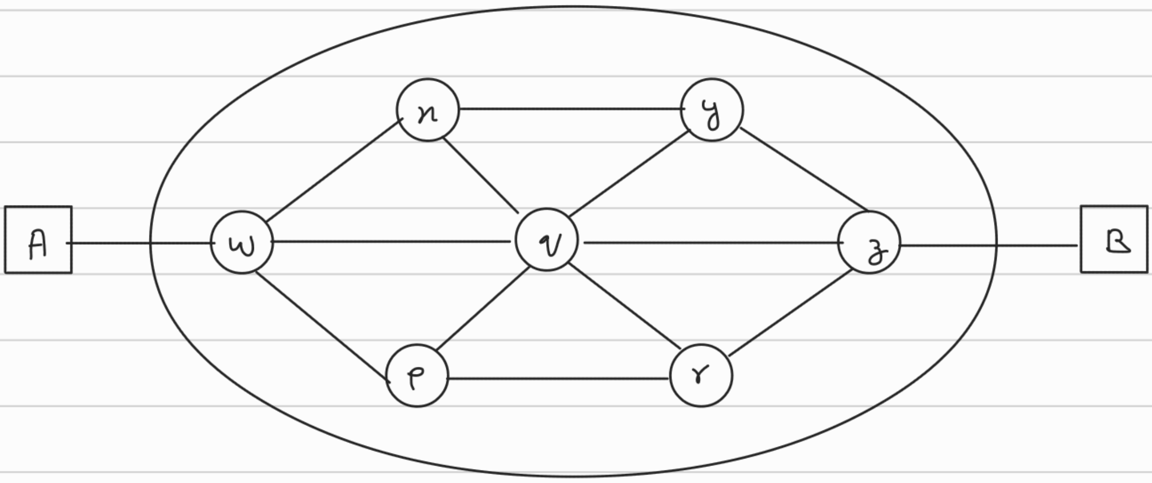
| From | Vir. pac. No. From | to | Vir. pac. No. TO |
|---|---|---|---|
| $H_A$ | 0 | B | 0 |
| $H_A$ | 1 | B | 1 |
| E | 0 | B | 2 |

A

| From | Vir. pac. No. From | to | Vir. pac. No. TO |
|---|---|---|---|
| B | 0 | C | 0 |
| $H_B$ | 0 | C | 1 |
| A | 1 | F | 0 |
| A | 2 | $H_B$ | 0 |

B

Q. Suppose 'n bit' message is to be transmitted over a 'k-hop' path using circuit switching, message switching, datagram packet switching & virtual circuit packet switching, circuit setup time is 's-seconds', propogation delay per hop is 'd-seconds', channel capacity is 'b bps', packet size is 'p bits'. Assume the network is hightly loaded hence there is no. waiting time.
Calculate data transfertime using all the mentioned switching scheme. (Msg. include All neccessary detail i.e src add, dest add, etc)




- let path to transfer msg is $A \xrightarrow{1} w \xrightarrow{2} x \xrightarrow{3} y \xrightarrow{4} z \xrightarrow{5} B$ there are 5 section that is 5 hops

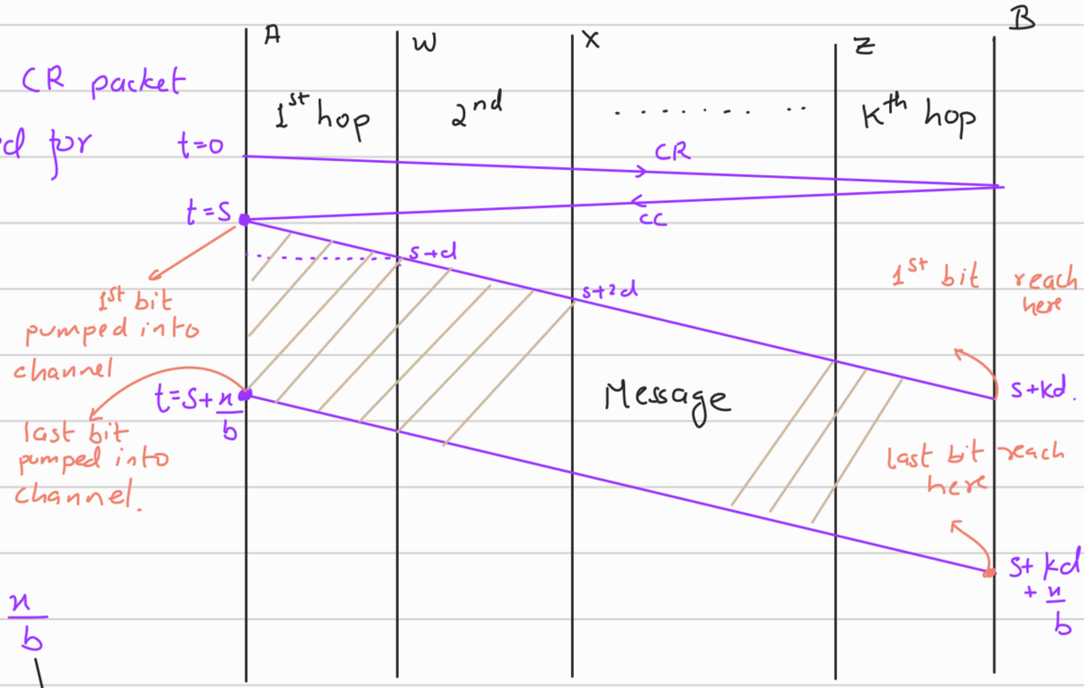
hop is also called Section

Msg size = 'n' bits
No. of hops = K
Prop. delay/hop = 'd' sec
Ch. capacity = 'b' bps
cht. setup time = 's' sec ⇒ (Src $\xrightarrow{CR}$ dest $\xrightarrow{CC}$ Src) ka time
packet size = 'p' bits

Timing diagram →

## 1. Ckt switching →

Note: In ckt switching CR packet may remain stored for some duration




$$\text{time} = S + Kd + \frac{n}{b}$$

- $S$ → Connection overhead
- $Kd$ → hop delay
- $\frac{n}{b}$ → Transmission time

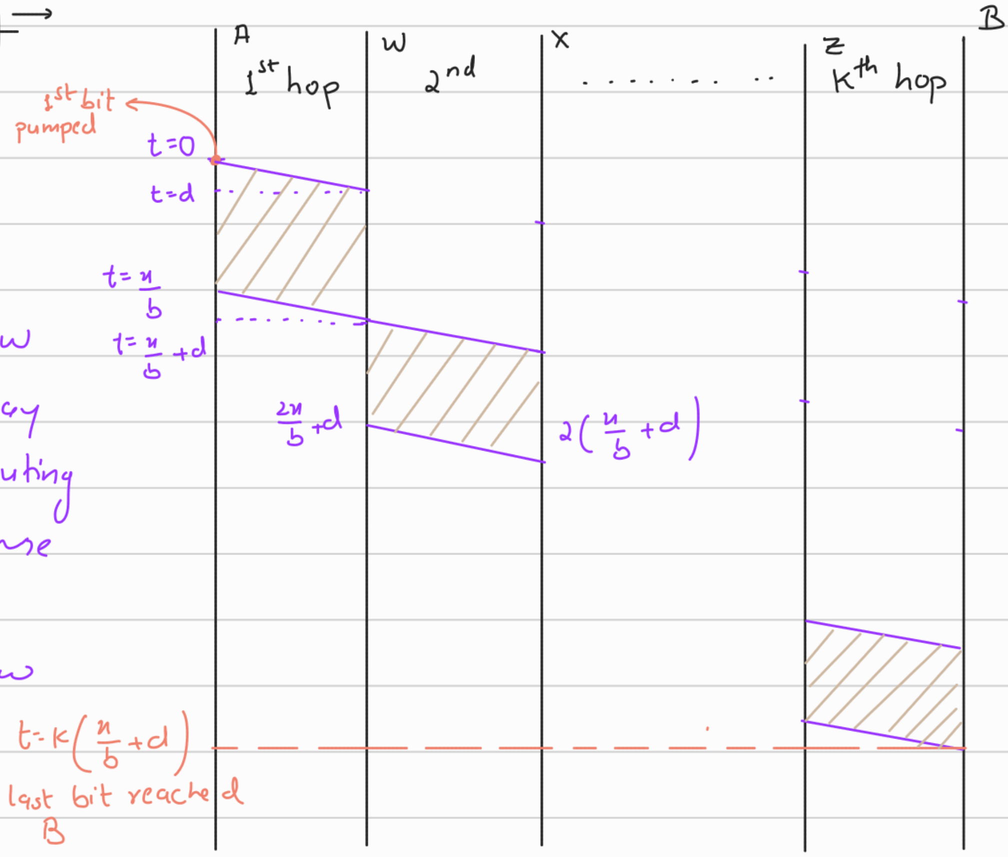
Note → In whatsapp call we use virtual circuit switching not network switching router help to remain interactive by removing delay by pusing other msg. other then calls

## 2. Message switching →

- Connection less

At $t = \frac{n}{b} + d$

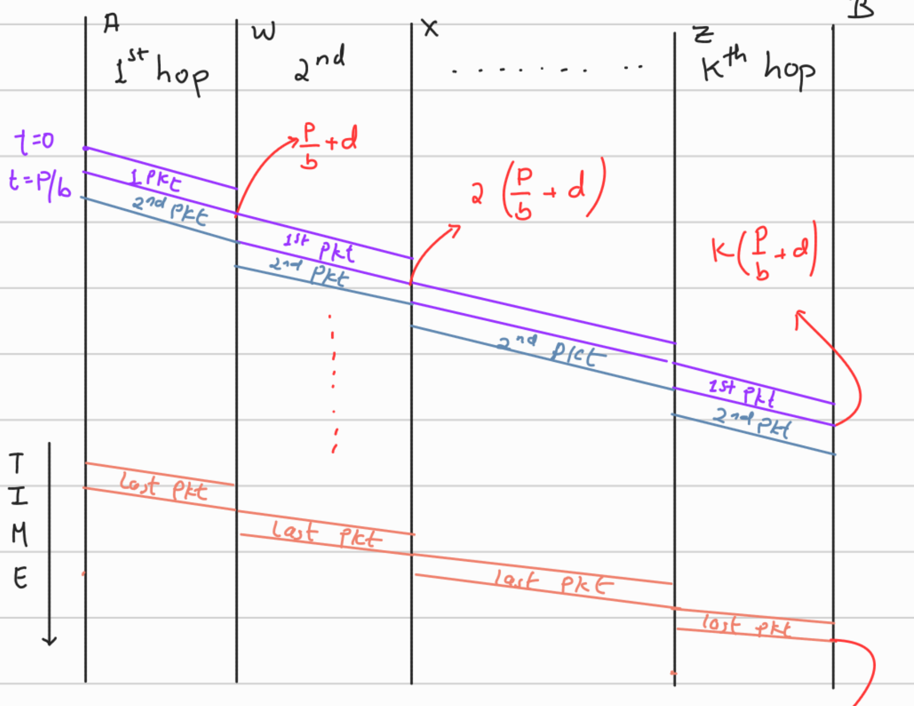
complete msg reach w after that their may be some delay in routing decision + waiting time then it against tast transmission from w



3. ## datagram pkt switching →

Transfertime =

$$k\left(\frac{P}{b}+d\right)+\frac{n-b}{P}$$




$$\left(\left\lceil\frac{n}{P}\right\rceil+k-1\right)\frac{P}{b}+kd = k\left(\frac{P}{b}+d\right)+\frac{n-b}{P}$$

4. ## Virtual circuit packet swiching →

∵ we have not taken routing in datagram

∴ it is similar we just need to add setup ckt time in datagram pkt switching

Transfertime = $S + k\left(\frac{P}{b}+d\right)+\frac{n-b}{P}$

Ckt setup time

## Protocal →

- To transfer file from comp A to comp B, we need to establish a connec-tion b/w them (let physical wiring)

- Data transmit in form of electrical signal let A do binary encoding (say 1 = 10V, 0 = -10V)


- When we transfer a file we put corrosponding signal which travel to channel & reach B

- But B can have different (say 5V = 1 & -5V = 0) encoding though we transfer successfully but B can't recognize what we are sending.

- ∴ Before transmission we need to make some rule & all connected must agree on it say all agree on ($1 = 7V$ & $0 = -7V$)
- ∴ We might need to attach a hardware to do the same to convert to required signal from generated signal or software

- Serial port → transmitt 1 bit at a time

Parallel port → k-bit parallel sys. at each instance we gave k bit.

- Protocol means set of rules & convention on which all the comm. agencies (computers) should agree to make comm.ⁿ possible.

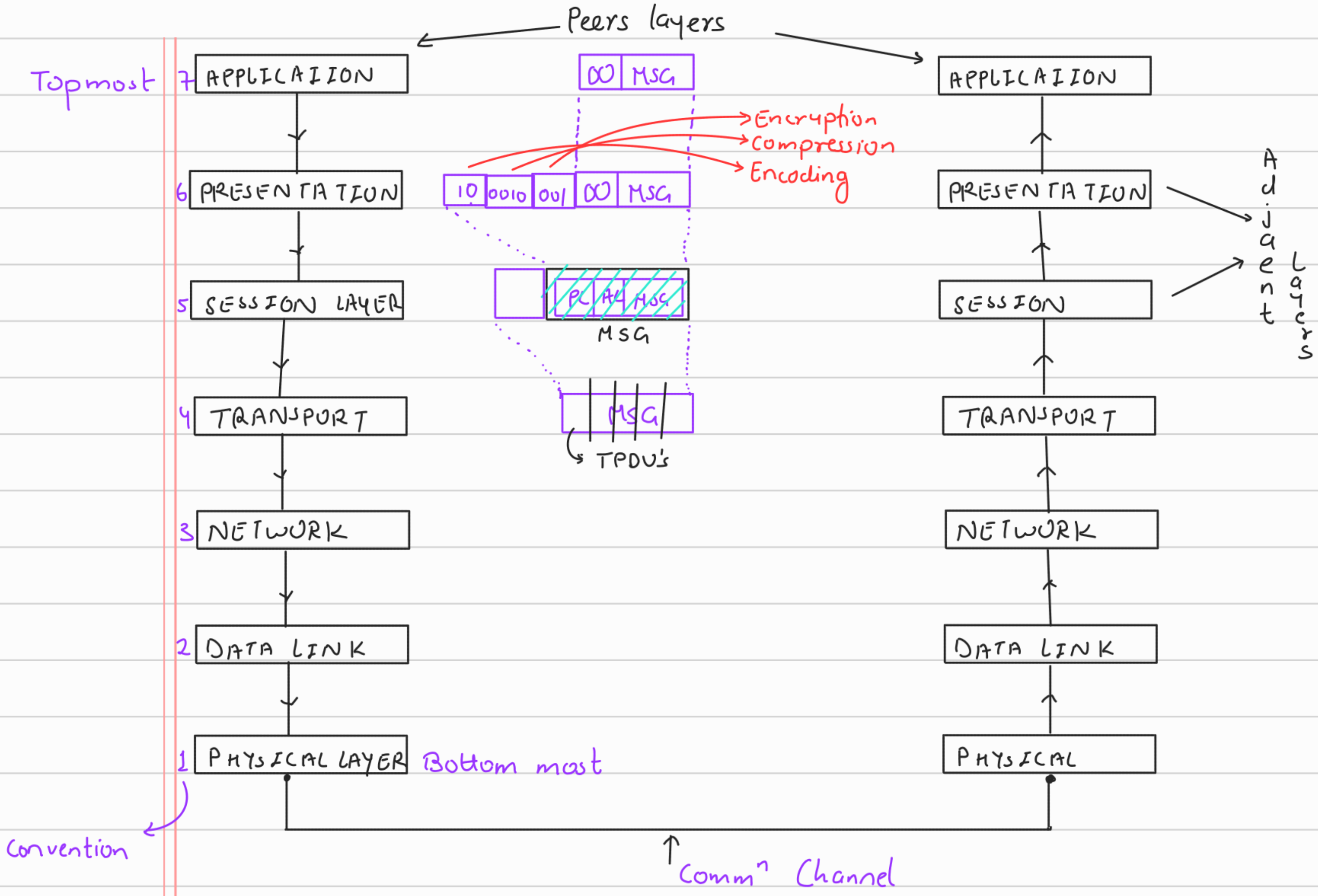
- If Everything is combined in 1 hardware then on changing 1 rule we need to replace complete hardware but if we use different modules we need to replace certain modules only ∴ rules implemented in form of certain modules
- We make stack of these modules where each layer connected with only preceding & succeding layer ∴ setting of rule easily we arrange in form of layers.

- ISO has given a model OSI (Open system interconnection) model.
- They divided OSI in 7 module / Layers which the comp. need to use while communicating
- OSI is not practically implementable.

OSI model →


- No physical exchange of bit
- Virtual comm. using header bits is done

↑




- User is accessing N/w^k through application layer
- All layers are set of rules that must be their for communication.
- Each comp. should agree & implement these layers

## Application layer →

- App. layer provides appliation S/w^r using which various info. can be generated for transmission.
- Basic app. s/w^r mean text editor kind of thing, or voice recording s/w^r or vedio generating s/w^r, etc (whatsapp)
- If 'A' has certain s/w^r then other comp. comm. must also have same s/w^r
- Along with data generated they must agree on bit representation of type of software

Ex- 00 Text 10 - video

01 Audio

- These 'k' extra bits will be appended to data by application layer to distinguish b/w^n data
- These 'k' bits are called headers

- So headers are being appended to data by application layer
- These header will help reciver to know file type
- These header + file as it is transferred to Presentation layer
- Each upper layer is getting service from connected lower layer or each lower layer give service to upper layer
- The same data is transferred from Presentation layer to app. layer of another PC

Presentation layer →

- The duty of PL is data encryption, data compression, character encoding
- Encryption is for security purpose from random access.
  - → Transposition   Jumbling of data   hello → lehlo
  - → Substitution   a → z , b → y ..... etc

- The type of encryption is represented by 'K,' header bit that will be appended to data, while reciving the P.L at reciving layer identify the type of encryption using these 'K,' bits and decrypt the msg and pass on decrypted msg to application layer at reciever end.
- Compression is used to reduce trasmission time, bandwidth and money expenditure
- ∵ In PL ∃ 3 steps & ∀ ∃ few bits of header ∃ a rule protocol for what is representing what
- ∵ The msg. is transferred in bits ∴ we need to encode and both parties need to made some agreement what bit seq. rep. what.

Session layer →

- Establishment & release of session
- To provide synchronization or check points
- Remote procedure Call (RPC)
- In 1 Connection we can have many session and vice a versa

Ex- (1 hr Class on Ms team may conn. loss for 3 time but session continue)

(Attending Conference different rights given by diff. speakers)

- Simplex, half duplex, duplex

- Synchronization point → When reciever will send acknowledgement, when he/she have to send is when they recieve correctly error can be in any segment then i need to retransmitt it & if we again transmit & we found error again we are wasting bandwidth ∴ while connection establishment we make agreement that after every 100 pages of book i have to send acknowledgement. say if 783rd page is wasted then we need to send b/w 700 – 800 only there can be minor (Article Ex) or major synchronization point.

- Client Server architecture mean that client feels every service is availible in his/her machine but actually it is on server

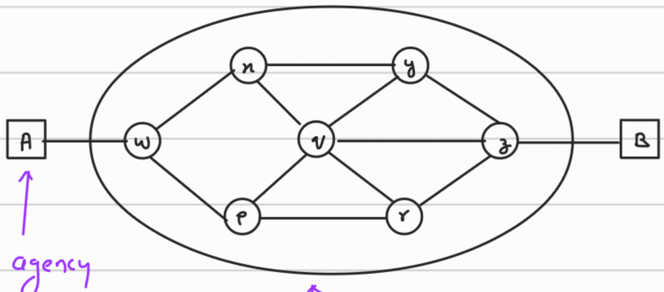
<u>Transport layer</u> →
- To divide the msg in fixed size datablock called "Trasport protocol data units" or "TPDU" or "TDU"
- Establishment & release of connection (In case of connection oriented system)
- End to End error control
- End to End flow control
- Assurance of quality of service (QOS) parameters

- Host has 4 layer of OSI

Mainly these layers are in host

| |
|---|
| App. layer |
| Pres. layer |
| Session. layer |
| Tran. layer |

It has physical layer but it is very light




1 agency (VI)

PPP-protocol

Another agency (BSNL)

- Router has 3 layer of OSI

Mainly these layers are in router

| |
|---|
| N/wk layer |
| Data Link |
| Physical |

- Transport layer is $1^{st}$ layer that divides the data ∵ we assume that mainly it is last layer at host & in subnet mainly store & forward is used ∴ to use buffer properly in router we first need to divide MSG in TPDU's

- Two application layer, session layer, transport layer, Presentation layer are end to end layer

- We keep TPDU's size fix is to use error control as to define error control effectively we need some fix block size.

- End to end flow control mean 1 host & N/w^k is fast & 1 host is old & slow, A can send TPDU fast but B need to process and assemble TPDU block symultaneously ∴ due to buffer overflow ∴ TPDU lost ∴ to avoid this we need to slow down the fast PC to cutdown flow of data.

- QOS mean based on subnet access user assure some parameter prob. of breaking conn. is __, prob. of error is __, etc

∴ Transport layer try to fix problems of subnet so that user don't get affected.

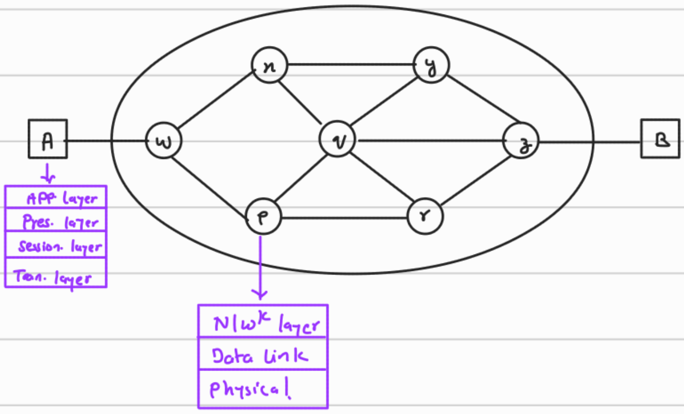
Network layer →

- Routing - kaha bheju
- Conjestion control - Capacity of subnet depending on quality of router & channel capacity
- Internetworking ??
  ↳ How to connect two or more N/w^k togather (differnt protocols)
  ∴ We need some devices and rules
- When N/w^k layer append header to TPDU's it is then called packets
- ∴ 2 N/w^k layer exchange packets not TPDU's

Jaam lag gaya N/w^k me

## Data link layer →

- Framing is done by this layer
- Error control
- Flow control
- Link management




- Error & Flow control is done in particular hop as ∃ data link layer b/w^n 2 consequtive routers (Not end to end)
- In each hop DLL works seprately
- This ensure N/w^k will have less error
- To ensure very less error error control on Transport layer is must
- To improve Error correction we need to ↑ NO of additional bits

- To repair a broken link we add some feature in this layer which exactly predict distance of breakage

Framing baad me

## Physical layer →

- It deals with electric properties like bit encoding tech., signal representation, modulation, etc.

Mechanical property → portsize, port dimension, color codes

PHYSICAL LAYER → DC se dekho

# DATA LINK LAYER →

Topology

## Point to Point

- Section of channel is controlled by 2 end points only
- Star, ring, mesh etc
- Generally duplex channel ∴ no chances collision
- Used in WAN

## Multipoint

- Section of channel is controlled by more than 2 points
- Bus, etc
- Collision can occour
- ∴ We need to decide how to acces common channel.
- Most of LAN