COMMUNICATION NETWORK NOTES

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Communication Networks Notes, First Edition

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Puffin
Date
Page

Communication Networks (CN)

↳ Set of equipment & facilities that provides service.

* Information = Data + context
* Service : transferring info. b/w users located at diff^t geographical points.
  ↳ from communic^n point :
    ↳ Internet transfer of individual block of info.
    ↳ Internet reliable transfer of stream of bytes
    ↳ Real time transfer of voice signal

* Applic^{ns} of networks :
  Email, web, fax, modems, SMS _ _
* Voice over internet protocol : VOIP

* One of the popular CN$_s$:
  OSI : Open Sys. Interconnection model (obsolete)

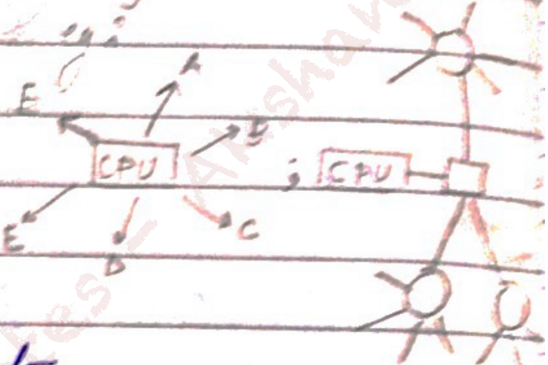
Reading assignment ① :

Q. What brought evolution (improvement) in CN? (from 10's of bps to tera bps)
Q. What are the advantages & disadvantages of a medium (Cu wire, coaxial cables, optical fibres) used in a CN?
Q. What is subnet & subnet mask?

Network Architecture evolution — Telegraph networks → Telephone → Internet, Optical & Wireless → Next Generation

* Network architecture :
  a plan that specifies how the network is built and operated.
  ↳ it divides overall CN into LAYERS
  ↳ f^{nal} areas

eg:



• Layering, or layered architecture
  ↳ hierarchy of nested black boxes
* A black box approach was thought to be used for making a model.

* OSI model (Open System Interconnection Model)
  ↳ a 7 layered structure

Sender		Reciver
Topmost layer Applicⁿ	———	Applicⁿ layer
Presentⁿ	———	Presentⁿ layer
Session	———	Session layer
Transport	———	Transport layer
* Network	———	* Network layer
* Data link layer	———	* Data link layer
base layer * Physical layer	———	* Physical layer

→ These 7 layers exist, be it any type of network
→ First three have to be there

* Protocol.

A process to establish communicⁿ b/w sender & receiver, followed with mutual understanding that both of them will communicate in that way.

eg. Telephone protocol
↳ Sender calling receiver
Sender : Hello
Receiver : Hello
} exchange of hello is a mutual understanding b/w sender & receiver to start communicⁿ.

* QOS : Quality of Service.

↳ a standard set.

eg. in communicⁿ, there should be some min. QOS to be satisfied i.e, some min. quality should be there in communicⁿ.

* ESSENTIAL NETWORK FUNCTIONS.

1 - Terminal - end sys. that connects to network
2 - Transmission of info across phy. sys.
3 - Info. representⁿ - In a format handled by sys.
4 - Switching - transferring of info flow b/w communicⁿ lines
5 - Addressing - identifying points of connection to network
6 - Traffic controls - to ensure smooth flow of info through network
7 - Congestion / Overload control : Occurs due to a surge in traffic or a fault in equipment; control needed to ensure a degree of continued operⁿ of network.
8 - Multiplexing : means of connecting multiple info. flows into shared connection lines.

Date
Page

9- Network Management : has support sys such as monitoring network performance, detecting and recovering from faults, configuring network resources, maintaining accounting info for cost and billing purposes & providing security by controlling access to info flows in the network.

Reading Assignment (2)

★ Network Architecture Evolution :

- ✓ Telegraph Networks
  - message switching & digital transmission
- ✓ Telephone Networks
  - circuit switching
  - analog to digital transmission
  - mobile communic^ns
- ✓ Internet
  - Packet switching & computer applic^ns.
- ✓ Next-Generation Internet
  - Multi-service packet switching network

★ 7-layer OSI model (details).

1) Physical layer :
- ✓ transfers bits across link.
- ✓ design & specific^n of physical aspects of a communic^ns link.
- ✓ ethernet, DSL, cable modems.
- ✓ twisted pair cable, coaxial cable.

- f^n : using a "virtual bit pipe" (copper wire, fibre optic, - - - : what cant be seen what is going through the pipe) : A virtual link for transmitting a sequence of bits b/w any 2 pair of nodes.
- ✓ It is accomplished by using physical interface (like modem) which makes bits to signals & vice versa.
- Issues : delays, error char. at physical layer

2) Data link layer :
   - transfers frames across direct connections.
   - groups bits into frames.
   - detection of bit errors, retransm^n of frames.
   - activ^n, maintenance & deactiv^n of data link connections.
   - Medium acess control for local area networks
   - flow control :

$$\text{Data link layer} \xleftrightarrow{\text{frames}} \text{Data link layer}$$
$$\text{Physical} \xleftrightarrow{\text{bits}} \text{Physical}$$

   - Purpose : convert unreliable bit pipe (at physical layer) to higher level for sending packets.
   - Done by adding extra bits at starting & end of packet, making frames longer and more detailed. These extra bits are called header/trailer (can be point to point or multi-access links)

3) Network layer:
- transfers packets across multiple links and/or multiple networks
- addressing must scale to large networks.
- nodes jointly execute routing algorithm to determine paths across the network.
- forwarding transfers packet across a node.
- congestion control to deal with traffic surges
- network layer exists at both sender & receiver side & also at all nodes.

4) Transport layer:
- transfers data end-to-end from process in a machine to process in another machine.
- reliable stream transfer or quick-and-simple single-block transfer.
- port numbers enable multiplexing.
- message separ^n & reassembly.
- connection setup, maintenance & release.

5) Session layer:
dialog management, recovery from errors...

6) Present^n layer:
machine independent represent^n of data...

Today, these layers have been incorporated in applic^n layer (is not found separately today)

7) Applic^n layer:
provides service, frequently used by applic^ns like web access, file transfer, email, etc...

Puffin
Date
Page

5) Session layer :
deals with two end points in setting up a session, while network layer handles the subnet aspects of setting up a session.
- if a user wants a service in a network, this layer provides reqd info to transport layer.
- deals with access rights in setting up sessions
- reqd of a session with session, network & transport layer varies from network to network

6) Presentn layer :
fn : - Data encryption
- Data compression → there CAN be loss of data, but should NOT be loss of inform^n.
- Code conversion
→ convert info in the format the sender can send & receiver can accept

7) Applic^n layer
deals with what is left over after other layers have performed their fns.
- each applic^n should have its own software
- lower layers : perform those parts of overall task
Applic^n layer : performs particular task.

☆ SWITCHING CONCEPTS

- Circuit Switching
- Packet Switching

* Circuit Switching
- Space division switching
- Time-division switching
- Blocking/nonblocking
- TST (time-space time)

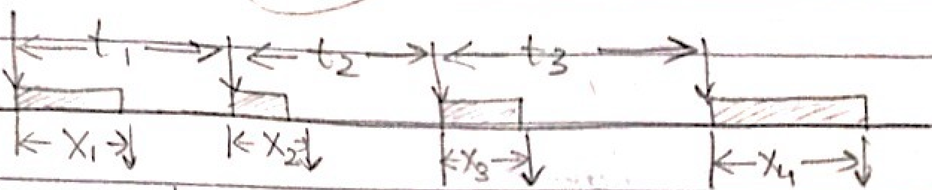
* Packet Network Technology
- Datagrams
- Virtual circuits
- connectionless switching
- store-and-forward

* Circuit switching:
- User signals for call setup and tear-down
- route selected during connection setup
- end to end connection b/w sender & receiver
- replying back to sender in case user is busy

* Transmission rate, r_s (bits/sec):
  - each communicat^n link on its path allocates a portion of r_s of its transmission capacity in the given dir^n for that session using either TDM & FDM. (This happens after path has been created b/w transmitting side & receiving side)
  - ∃ a guaranteed transm^n rate through the network

- Data link doesn't use circuit switching.

$E(t_i) = 1/\lambda$ → Expected time of arrival of i^{th} session.



Model depicting how a ~~Cus~~ session arrives & seeing how busy/vacant an exchange is for the user.

→ $t_1 = t_2 = t_3$ i.e, the same amount of time has been allocated to all users.
These days, time varies as per user usage.

→ X_1, X_2, X_3 & X_4 are sessions of diff^t widths.

→ λ : Arrival rate of a session (s)

- If $\lambda \bar{X} \ll 1$, sessions' portion of link is idle most of the time.
  - → $1/\lambda$: inter arrival time b/w messages of s.
  - → L : expected length (in bits) of messages from s & r_s in allocated bit rate of s
  - → r_s : session bit rate.

* We must have : $\bar{X} + P \leq T$
- ↳ P : Propagⁿ delay through the network
- → $\bar{X}$: Expected time until last bit of message has been sent to first link.
- ↳ T : Allowable expected delay from message arrival (at source) to delivery (destination)

* $\lambda \bar{X} < \lambda T$

↳ If $\lambda T \ll 1$ i.e, utilizing session for less time

↳ called Bursty session.

• With circuit switching → fraction of link allocated to data sessions is utilized at most 1% of the time.

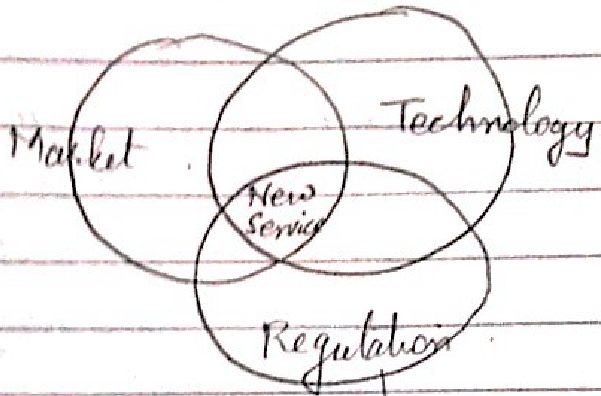
* for $\lambda T \geq 1$, switching & circuit delays maybe ignored, but queing delays can't be ignored.

Reading Assignment :- 3.
list elements of telephone network architecture

* Elements of Network Architecture :
1 • Digital transmission & switching :
  - Digital voice, Time division multiplexing
2 • Circuit switching
  - User signals for call setup & tear down
  - Route selected during connection setup
  - End-to end connection across network
  - Signalling coordinates connection setup
3 • Hierarchical network
  - Decimal numbering sys.
  - Hierarchical structure; simplified routing, scalability
4 • Signaling Network
  - Intelligence inside the network

* Success factor for the evolution of network services
  • Technology is not the only factor in success of a new service.

• 3 factors considered in new telecom services:



→ Regulatory bodies, ITU, IETF, IEEE etc.

☆ 3 similarities b/w each of the national transportⁿ networks & a communicⁿ network
- Airline network
- Highway sys.

Similarities (a few)	Transportⁿ Network	Communicⁿ Network
Entities transferred	People & goods	Info / data
Medium involved for transfer	Roads, rails, air, corridors	links: copper, air, optical fibre.
Routes info through	Stations, airports, highway interchanges	Switching points
Destinⁿ are identified by	Addresses involving geographical names	Unique no. in form of : Phone no, IP Ad.
Entities are directed through	Geographical routes	Electronic routes
Shared resources employed for transport/ multiplexing	Cars, airplanes are shared to traverse over rail/road/air corridors	Packets/streams are made to traverse over links/routes via channel through mux/demux

In addition to table :
Airline network :
In this case, passengers purchase tickets that guarantee a seat all the way to the destinⁿ, even if transfers are made at intermediate airports. This is similar to establishment of connections across a telephone network a connection oriented service.

Highway systems :
Trucks or cars enter highway without making reservⁿˢ ahead of time & without informing any central authority of their destinⁿ / route. This mode of operⁿ corresponds closely to operⁿ of connectionless packet network

☆ Is transportⁿ network more similar to telephone network or packet network?
- Transportⁿ network : neither similar nor dissimilar exactly.
- Transportⁿ sys varies in how its organised & how transfers are made.
- By combinⁿ of transportⁿ sys : Using these sys. in combinⁿ provide a high degree of connectivity b/w sources & destinⁿˢ for transfer of either people and goods OR info & data
- eg : combinⁿ of air, rail & highway transportⁿ sys.
  — used jointly for transfer of people & goods.
- Internet plays a role similar to the combined transportⁿ sys in that the internet enables

the transfer of info across multiple dissimilar networks that may differ in how they are organised & how they operate.

* Describe what a step-by-step procedure might be involved inside network, making a telephone network

Note : Telephone network :• Tells receiver address.

• 1st 2-3 digits : area code → main geograph. locⁿ.

• Next 3 digits : particular telephone office

• Final 4 digits : particular locⁿ of receiver

(S1) On dialing a no. from the line, the code is used to link origin to destⁿ.

(S2) Circuit is established b/w them

(S3) Ringing tone is applied at destⁿ to tell incoming call

(a) If destⁿ is willing to answer - call is completed through lifting of phone.

(b) If destⁿ is not ready →

(i) it can disconnect — signal can come

(ii) ask to hold

- terminate & redial the call

☆ Advantages and disadvantages of transmitting fax messages over internet instead of telephone.

Fax over internet or telephone ≡ email or telephone call

- Fax over internet or, attachment to email
  - ↳ Advantages / Disadvantages
    - Not in real time
    - Delivery - not necessarily confirmed
    - Not expensive
    - Not depending on distance
- Fax over telephone :
  - real time.
  - high certainty of delivery.
  - Costly transmission

☆ Find propagⁿ delay for signal travelling the following network at speed of light in cable (2.3×10^8 m/s)

1. Metropolitan area (100 m)
2. Circuit board (10 cm)

Idea : To find propagⁿ delay ⇒ find time

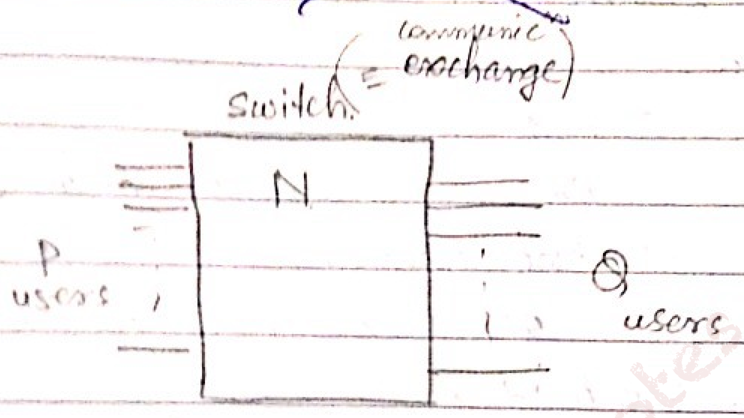
$$\text{Time} = \frac{\text{Distance}}{\text{speed}} \quad \text{Ans}$$

☆ CIRCUIT SWITCH TYPES.

1. **Space Division switches :**
   ↳ Provide separate physical connection b/w i/p & o/p.

☆Note :

☆



→ Switch is used to connect p users to q users
→ Only N lines can be entertained at a time (simultaneously). N on one side & N on the other side.
↳ If N+1th user comes on any line (on any side), he'll get a busy tone. So, switch is said to be blocking (switch capacity = N, here)

☆Blocking / non-blocking switch

→ If A ⟶ 1   A connects to 1,
B ⟶ 2.   B connects to 2,
C - - → 3 → Y   & C tries to connect to 3, it is not able to do that ∵ it is connected to "Y" through some other exchange. So, he gets ENGAGE tone.

here, user is at fault, not the switch. (user is blocking)

* Suppose

A ⟶ exchange ⟶ B

If exchange is busy, it can dynamically shift the route to some free exchange & then connect to B. This is Connectionless switching.

* Time division switching :

Using single line is used to connect to different users ⟶ how?

the pauses occured when a user talks, is used to share/let other user talk in that. ⟶ Use of MUX & DEMUX
i/p o/p.

synchronously will make transmission easy (∃ some delay).

* Suppose I use multiple voice band (frequencies) limited signals & are combined & connected to SINGLE carrier (say satellite) or, I use single frequency for multiple carriers. Space division switching is concerned with this.
Illy, we can have a mixture → TST switching (time-space-time)

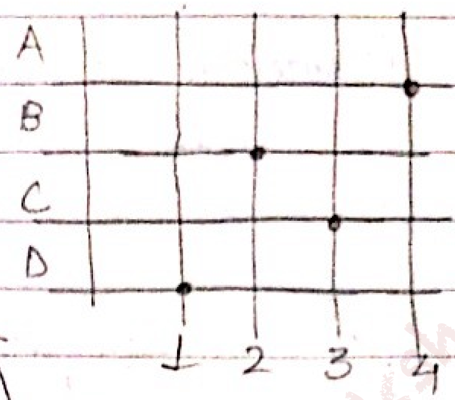
* Store & forward : At exchange, ∃ some delay. So, storing is used so that ∃ no loss in the process of exchanging/connecting to other user.

* Data grams & virtual circuits :

A unified approach of visualising the switching concepts using data grams & virtual circuits.

* Circuit switching → holding, call conferences etc is very easy to do in SPACE SWITCHES

* Crossbar Space switch



Suppose now B wants to connect to 4, but A is already talking. So, B can talk if the exchange has conference facility

Suppose A wants to connect to 4
B → 2
C → 3
D → 1

So, we close the switch
If 5th user comes, he's blocked

* For N X N cross points.

* Multistage space switch

↳ large switches built from multiple stages of small switches

* $\boxed{\text{no. of cross points} = 2\left(\frac{N}{n}\right)nk + k\left(\frac{N}{n}\right)^2}$

V.Imp.

↳ read derivation.
↳ Reading Assignment-4

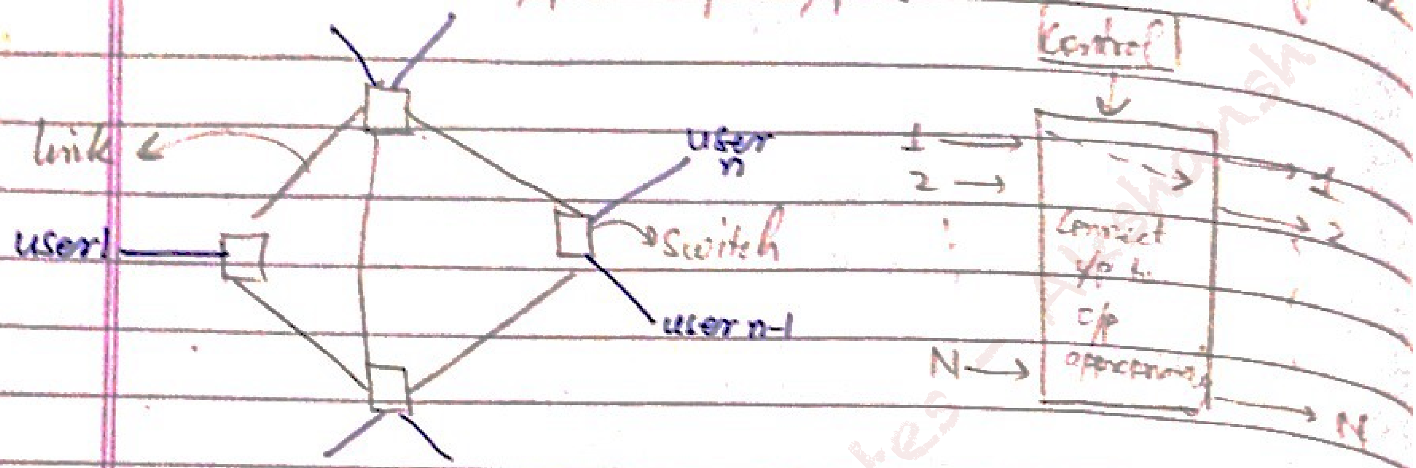
* CLOS Non-blocking condition ;

Condition : $k = 2n-1$

V. Imp * VON-NEUMAN Architecture : Standard Architecture used in ALL computers today.

* LINKS & SWITCHES.

Circuit switch : Transfers the signal arriving at a given i/p to apt. o/p.



Modern switches deal with multiplexed info. flows & each switch transfers a specific sub flow from an i/p line to a specific sub flow in a given o/p line.

Idea :

LHS RHS

A = B + C + D + E.

Given

Q B, C, D, find A

Sol^n :- Not possible : Von-Neumann architecture

Possible : Neural network architecture } name #

* We see for direct communic^n (Virtual) b/w layers. So, we see UNIFIED OSI MODEL.

* OSI Unified View : Protocols

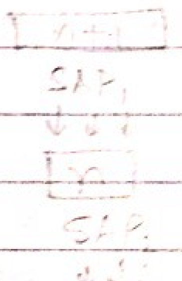
- layer n $\xrightarrow[\text{with}]{\text{connects}}$ layer n } provides service to the above layer
  layer n+1 &

hardware, software, switch, router (Physical or virtual)

- Entities comprising the corresponding layers on diff^t machines are called PEER PROCESSES.
- Processes at layer n are referred as LAYER N ENTITIES
- Layer-n peer processes communicate by exchanging Protocol Data Units (PDUs)
- Machines use a set of rules & conventions called layer-n protocol (i.e., every layer checks for the incoming info. If it belongs to that layer, it decodes & uses it, otherwise passes it on)

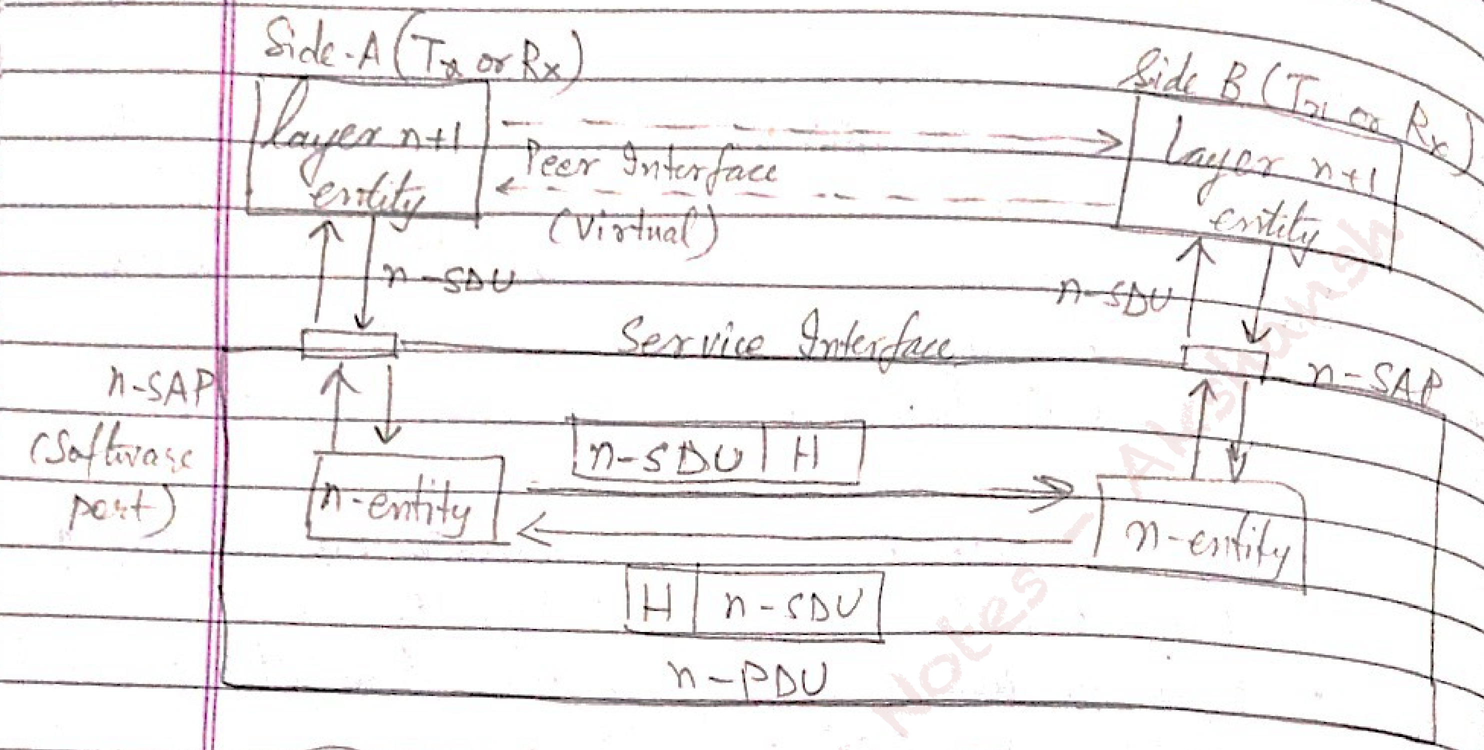
* SERVICES

- The n+1 layer transfers info. by invoking services provided by layer n. (n+1th layer can send info only that much that lower layers can take)
- Transmission of layer n+1 PDU is accomplished by passing a "block of info" from layer n+1 to layer n through software port called layer n SERVICE ACCESS POINTS (SAP's)



- Each SAP is identified by unique identifier.

- Each layer passes data & control info.



Note: ★ SDU's are encapsulated in PDU's

SDU's → Service Data Units

PDU's → Protocol Data Units

- Headers & Trailers :

Headers keep adding as we go from upper to lower layers (So that every layer can see if info is meant for it or not)

App Data

Applicⁿ layer header ← Ah + App Data

Transport layer header ← Th + Ah + App Data

Network layer header ← Nh + Th + Ah + App Data

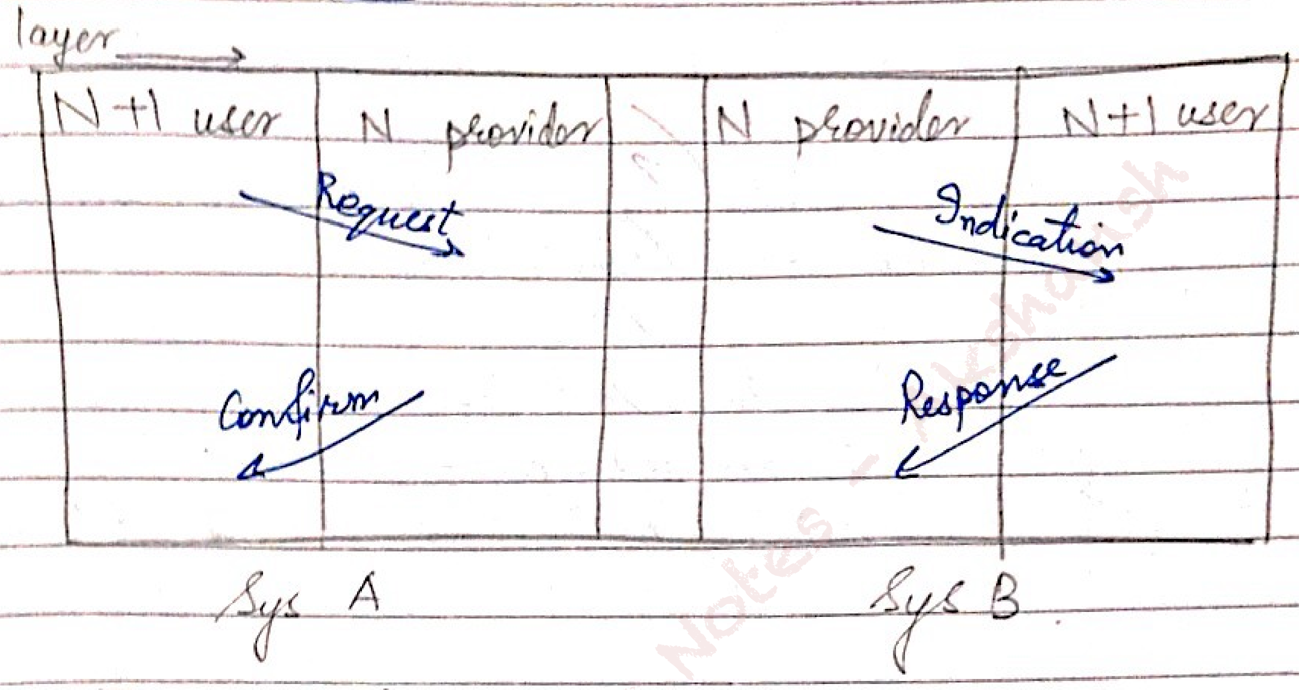
Data layer header ← Dh + Nh + Th + Ah + App Data + CRC

CRC → CRC trailer

CRC → For error correction.

★ Multiplexing: for effective use of channel capacity.

★ Interlayer Interaction



1. ∃ request from user to provider.
2. Transmit request from provider to get connected to other user
3. Then, other user responds to his provider.
4. The first provider confirms this incoming & connects the call.

★ Segmentation & Reassembly:
(sender's side) (receiver's side)

Sender's side :- SDU is segmented into multiple PDU's

Receiver's side : SDU is reassembled from sequence of PDU.

Date
Page

* Multiplexing : There needs to be a multiplexing tag so that a message (or a part) is sent at the right place

- Service provided by a layer can be connection oriented or connectionless.

- ATM : Asynchronous transfer mode.
  TCM : Transmission control protocol.

∘ Connection oriented	Connection less
✓ eg :- TCP, ATM	✓ eg : TCP operates over IP, IP operates over ATM
✓ Three phases : 1. Connection setup b/w 2 SAP, to initialize state info. 2. SDU transfer 3. Connection release	✓ Immediate SDU transfer directly from SAP to SAP - No connection setup eg : UDP, IP
	✓ layered services need not be of same type

* Reading assignment 4 : Do slides of 20.2.2014 (Slide 32 - 50)

✭ Packet Switching Network

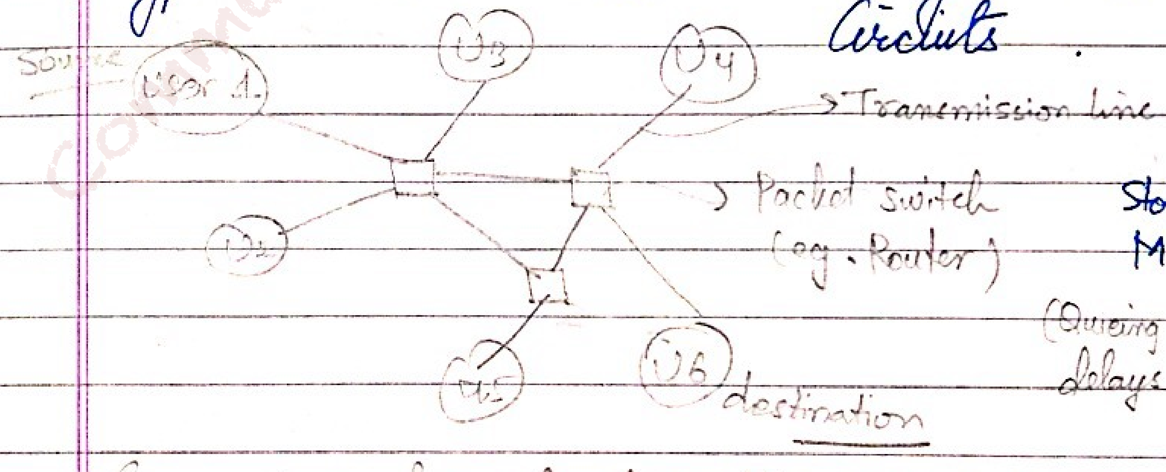
- ✓ Transfer packets b/w users
- ✓ Transmission lines + packet switches (routers)
- ✓ Origin in message switching.
- ✓ 2 modes of oper^n: Connectionless
  Virtual circuit
- ✓ There can be loss of messages if it is insufficient buffering at switch to store a message
  ↳ insufficient storage space to keep stack of info

✭ Message Switching :

- Invented for telegraphy
- Messages multiplexed onto shared lines, stored & forwarded.
- Headers for source & destin^n address.
- Routing at message switches.
- Connectionless.

Its a message. So, I'm not bothered about instantaneous transfer.
So, a switch connects across a free line to send to destin^n

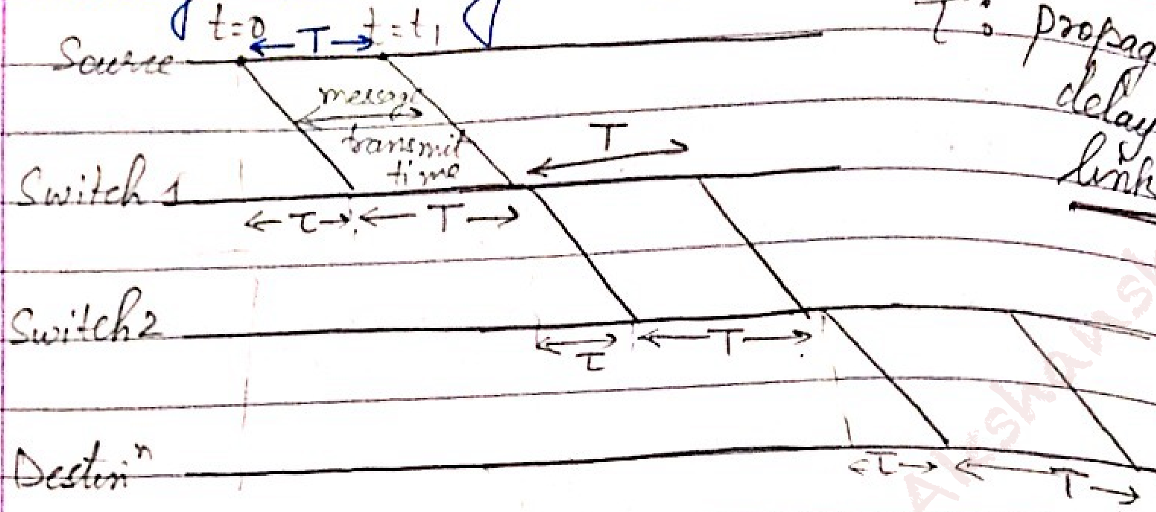
✭ Typical Switched Network – Datagrams & Virtual Circuits.



Store – Forward Mechanism.
(Queuing delays, Routing delays ... etc)

Suppose Source has info of 1000 bits. That is divided into small no. of bits (with headers & trailers) → CALLED PACKETS & received at destin^n.

☆ Message Switching Delay



τ : propagation delay of link ☆

At $t=0$, sender (source) sends a bit (or packet or message)
At $t=t_1$, 2nd sentence or message is sent
Now, at switch 1, ∃ some delay in reaching
(Propagⁿ delay : τ)

So, finally, it reaches destinⁿ in time $= 3\tau + 3T$

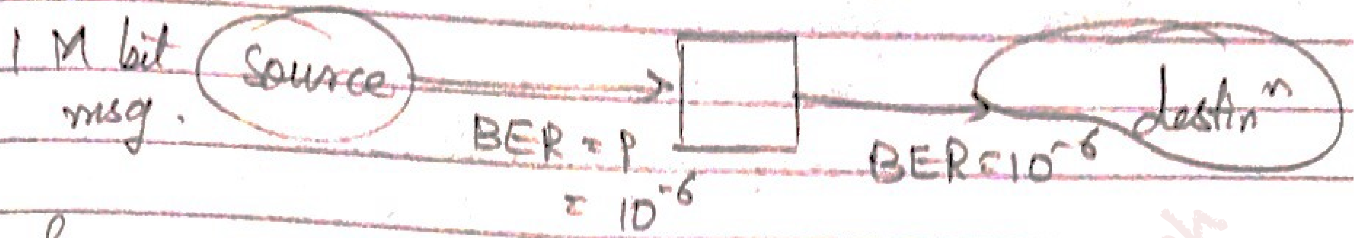
Assuming:

☆ Imp
- ☆ ∃ no queieing delay (i.e., switch immediately transmits once it gets)
- ☆ T & bit rate are same.
- ☆ Time for error checks is ignored
- ☆ Time for retransmission is ignored.
- ☆ Propagⁿ delay is assumed to be same b/w each link.

☆ BER : Bit error rate
↳ Prob. telling how many bits in an error

Puffin
Date
Page

* long Messages Vs Packets



how many bits need to be transmitted to deliver message?

Approach 1 :- Send 1 M bit msg.

Prob. that message arrives correctly,

$P_c = (1 - 10^{-6})^{10^6} \approx e^{-10^6 \cdot 10^{-6}} = e^{-1} \approx 1/3$.

So, on avg, it takes about 3 transmissions / hop.

$\Rightarrow$ total no. of bits transmitted = 6 M bits.

= (3 × 2 lines b/w source & destin^n)

Approach 2

Send 10 100 K bit packets.

Now

$P_c = (1 - 10^{-6})^{10^5} = e^{-10^5 \cdot 10^{-6}} = e^{-0.1} \approx 0.9$

So, on avg it takes about 1.1 transmissions / hop.

$\Rightarrow$ total no. of bits transmitted $\approx$ 2.2 M bits.

So, Approach 2 better than Approach 1.

So, transmit too much together, Prob. of error → high (& vice versa)

But, too small packets are costly. So, choose appropriately.

Q. It is req^d to transmit large message ($L = 10^6$) bits over 2 hops.
Assume transmission line in each hop has error rate of $p = 10^{-6}$ (BER) & that each hop does error checking & re-transm^n. How many bits need to be retransmitted using Message Switching.

Given :-
Message = 1 Mbit
no. of lines involved b/w source & destin^n = 2.
$BER = p = 10^{-6}$.

Now,
$$P_c = (1 - 10^{-6})^{10^6} \quad \text{(i.e, sending whole message at once)}$$
$$= 1/3$$

So, it takes 3 M bits transmitted in 1 hop.
So, to complete transmission (2 hops), it takes 2 × 3 M bits = 6 Mbits

If Approach 2 is used, like before, we get 2.2 M bits as no. of bits.
Ans

☆ PACKET SWITCHING DATAGRAM.

✓ Message switching is unsuitable for interactive applicⁿˢ, as very long messages of such applicⁿˢ impose very long waiting delays on other messages.

✓ Messages, if long need to be broken into smaller units (packets) especially if the transmission lines are noisy to avoid larger no./rate of re-transmⁿˢ

Consider a case :-

I am sending a message to India from a port 1. So, some Bandwidth gets allocated to me.
Now, If someone else sends a message through the same port, he wont get same BW to send message.
Ily, there can be a time when BW is totally occupied & user cant transmit

In case packets are used instead of long messages (or messages + noise), small part of BW is reqd for short amount of time. So, more users can be accomodated

* Note : In case message (packets) dont get send, ∃ retransmission in the line. That also uses some BW.

This BW is used more : for long message or message + noise

used less : packets.

Reading Assignment ✱ Regulations used in mobiles :-
- IEEE 802.II B
- IEEE 811.1 B
- IEEE 811.2 B

Idea :-

Divide message into packets
↓
packets should be divided s.t each packet has source address (2 bits) + destinⁿ address (2 bits) + packet sequence no (1 bit)

called OVERHEADS — headers & trailers

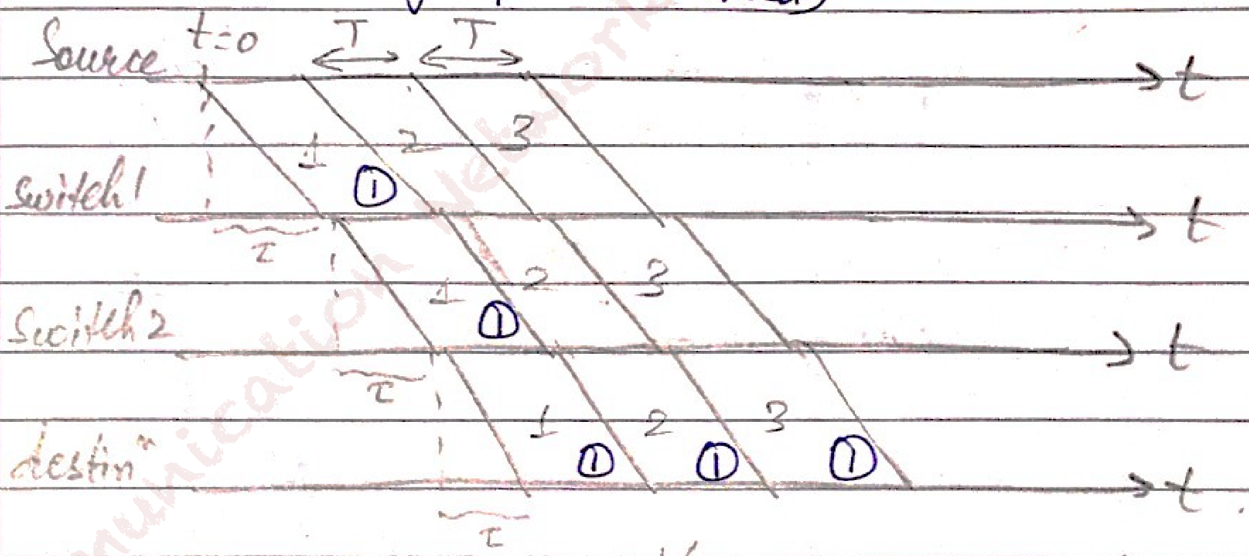
Other headers & trailers : CRC, check sum, layer headers, ...
↳ so, division of message should be done appropriately
↓
packets are recombined at destinⁿ (on the basis of labelling done to it using headers & trailers)

✱ Data + Headers + Trailers = DATAGRAM
↳ Packet

- packets may arrive out of order.
- pipelining of packets reduces delays.
- lossy packet transmission :
  ↳ ~~∃~~ enough buffer size to accomodate packets.
  eg: If a switch has capacity of 10 packet If 11th packet comes, it gets dropped.
  But, if we increase the capacity of switch, we are increasing buffer size

Suppose I go to open an NBD account (packet goes to switch)
I take token & sit in queue & will go to counter when my turn comes.
In case, no. of people are more (heavy traffic), following problems can come :
1. No. of places to sit are less (switch's buffer size is low)
2. No. of counters are less, So, it will take me more time to get my account opened (∃ delay in switching operⁿ to take place)

* **Pipelining** : Suppose a message is broken into 3 packets (single path assumed)



note : here, again, there are some assumptions (∃ no queuing delay, no error check time, no retransmission timing, T doesn't change)

$$\text{Min delay} = \underbrace{\tau + \tau + \tau}_{\text{propagⁿ delays}} + \underbrace{\left(\frac{T}{3}\right)5}_{\text{message transit time}}$$

$5 \rightarrow 1+1+1+1+1$

$$\Rightarrow \text{Min. delay} = 3\tau + 5\left(\frac{T}{3}\right)$$

↳ Assuming each packet requires $P = \frac{T}{3}$ to transmit

☆ Packet Switching : Virtual circuit

- → Packets can go in any dirⁿ. But, in virtual circuit, a pointer sets up fixed path of packet. → This is done while call is setting up.
- → All packets for connection follow same path.
- → Abbreviated header identifies connection on each link. (Abbreviated → ∵ we have fixed the path. So, don't need to allocate 2 bits for source, 2 bits → destinⁿ)

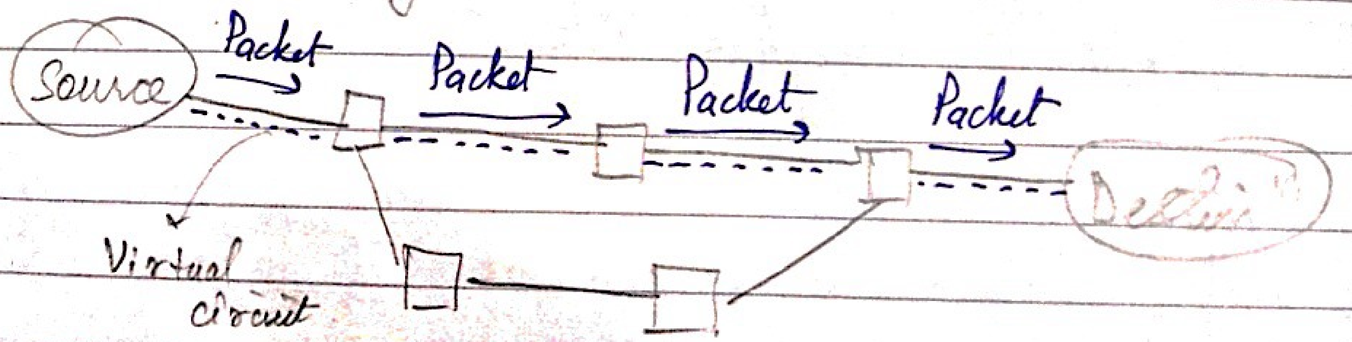
Basically
Normally :
Source bit size is 2 bits
Destinⁿ bit size is 2 bits

Now :
0 → 2 bits
1 → 2 bits
abbreviation.

- → Packets queue for transmission (∵ ∃ only one line)
- → Variable bit rates are possible, negotiated during call setup.

  negotiated → depending upon which plan I have paid for. Similarly, I'll get that QOS (Quality of Service)

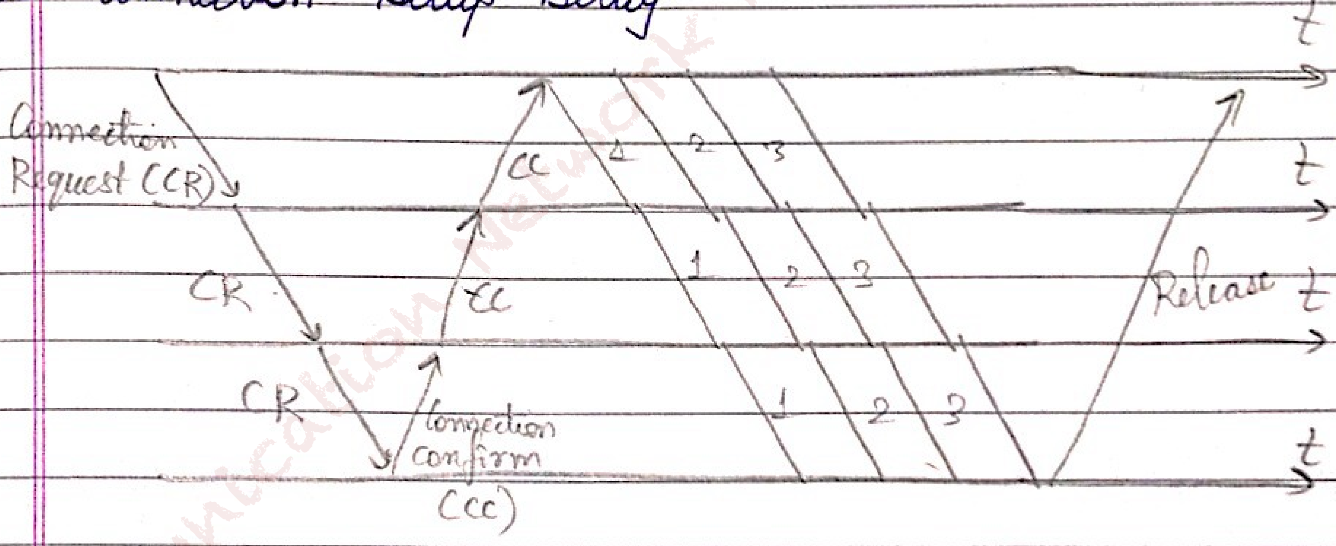
- → Delays variable, cannot be less than circuit switching



Circuit Switching: I have fixed a route. So, the delay is fixed in circuit switching
Packet Switching: Packets can go through different route. So, can have different delays.
↳ In virtual circuit, path itself can vary, leading to variable delays

* A connection is identified by a local tag, Virtual Circuit Identifier (VCI)

* Connection Setup Delay



✓ Before any packet is transmitted, ∃ connection delay
✓ Delay acceptable: large packets are transferred
not acceptable: few packets are transferred

* Delay for k-packet message over L hops.

Delay in packet switching

✓ $LT + (L-1)P$: 1st bit received
↳ $LT + (L-1)P + 1(P)$: received completely
$= LT + LP$ i.e. first bit released
✓✓ $LT + LP + (k-1)P$: last bit released, $T = kP$

Puffin
Date:
Page:

★ Pure Optical Switching :
- light-in, light out → without optical to electronic conversion.
  ↳ using light to transfer messages from one hill top to other (morse code)
- Space switching can be used to design optical switches.
- Wavelength switches.

---

eg Consider 1024 message size.
Made into 64 packets
each packet has 16 bits $\left(\frac{1024}{16} = 64 \text{ packets}\right)$

① each packet = 16 bits.
  → 2 bits source + 2 bit destⁿ
  → 14 bits header + (7×2) bit trailer.
    ↳ Layer
  → 3 bits of error checking, CRC.

Total bits per packet = 16 + 4 + 28 + 3 = 51 bits

$\frac{35}{51} \times 100 = 68\%$

So, In a packet, 32 % message
68 % overhead.

② Suppose packet = 32 bits.

$\frac{35}{67} \times 100 = 52\%$ overhead.

So, 48 % message
52 % overhead.

③ Now, if packet = 64 bits
Total size = 64 + 35 = 99
So, % overhead = $\frac{35}{99} \times 100$

= 35.3%

④ if packet = 128 bits
⇒ Total = 163
% overhead = $\frac{35}{163} \times 100$ = 21.4%

So, % message = 78.6%
% overhead = 21.4%

So, we see, when we increase packet size, overhead %age is decreasing.

☆ Consider a case when I want to send a message through a channel OPTIMALLY
↳ min. retransmission

☆ CUT THROUGH SWITCHING
Doing error checking only on the headers of incoming packets. So, packet can be forwarded as soon as header is received & processed.
So, from previous fig, Min. delay was $3\tau + 5\left(\frac{T}{3}\right)$

we have made, min. delay = $3\tau + T$

* Suppose the applicⁿ purpose is exact sending & receiving message (eg. b/w prime ministers, military commands). Then, proper headers, trailers & everything are added. Delays increase. But, exact message is sent (i.e., ∄ not even 1 bit change in message)

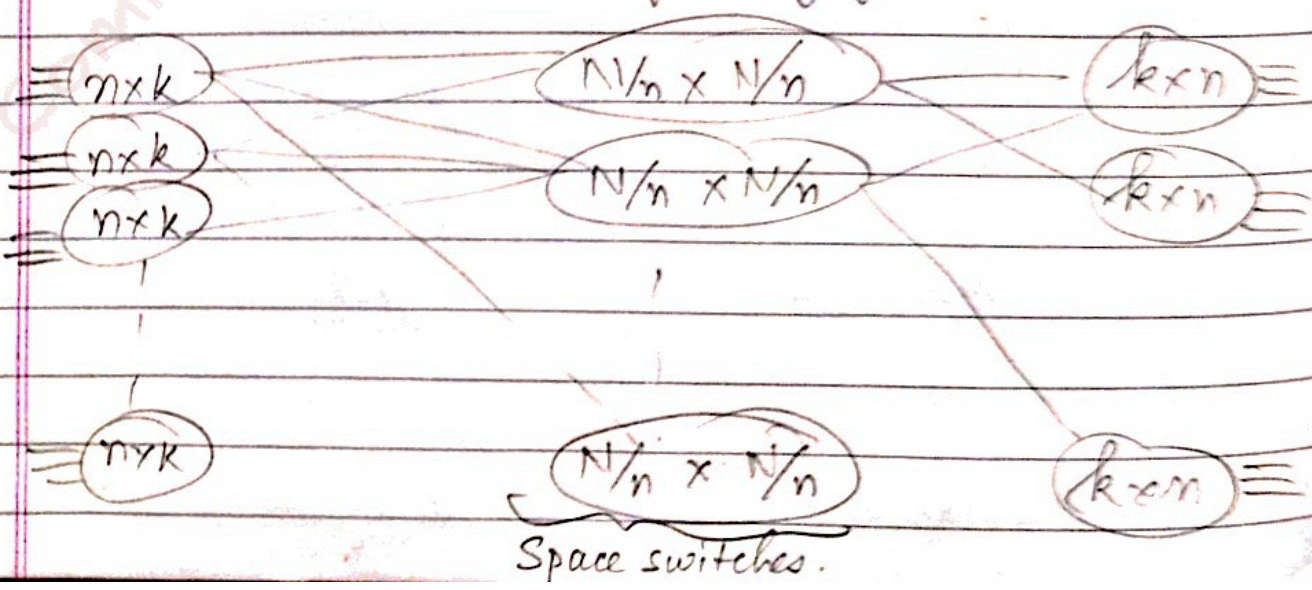
- Internet protocol (IP) uses datagram packet switching across networks.
- Hosts have 2 port IP address :-
  Network address + Host address.
- Routers do table lookup on network address
  - reducing size of routing table

I make a table telling me which route is busy when. So, at any point of time, free router can be seen from table.

•

☆ Multistage Space Switch :-

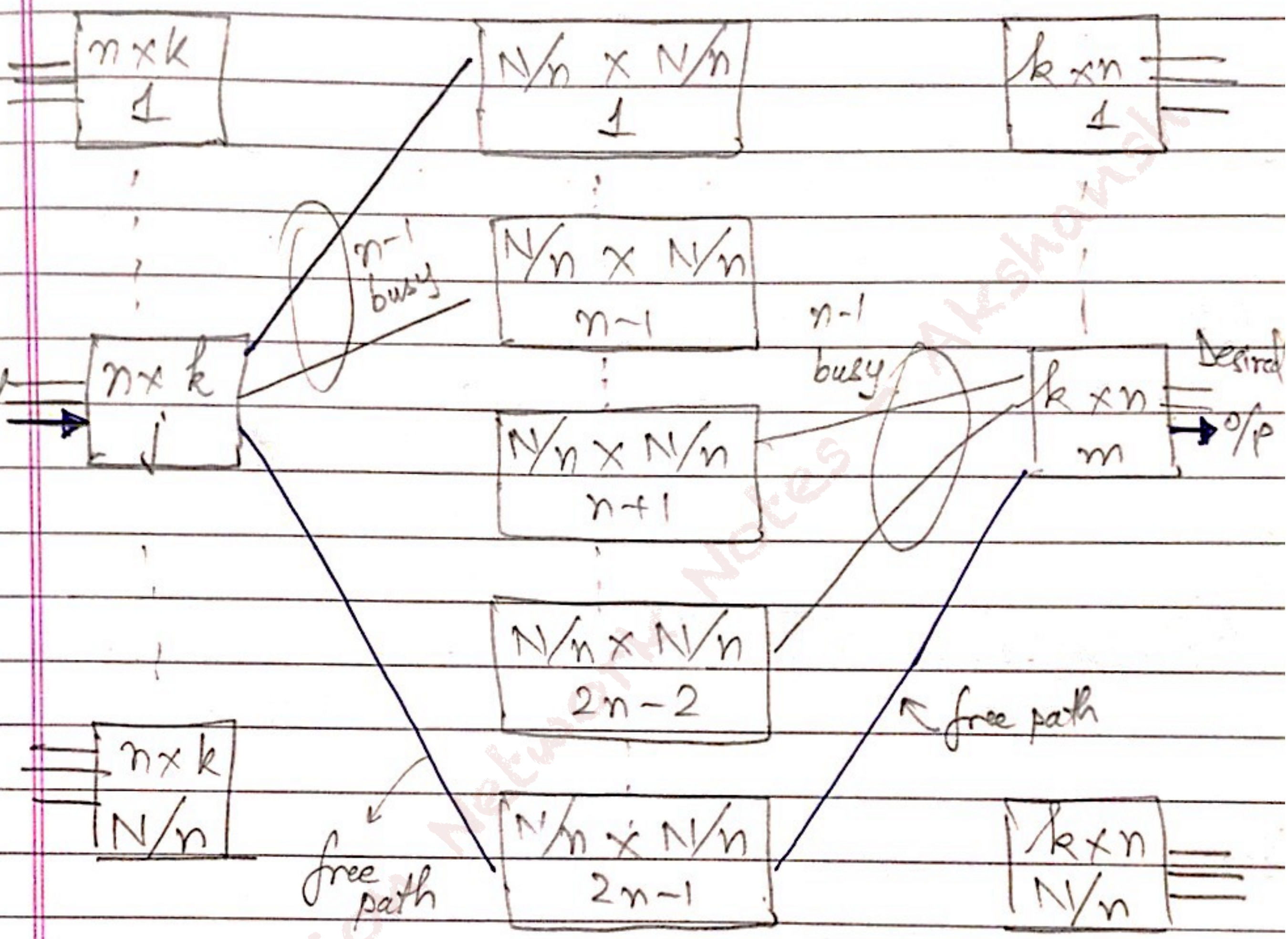
Idea : I want to make a switch non-blocking (I shouldn't get engage tone)



Proof in textbook (Leon, Gracia)

Puffin Date Page

✩ Clos Non-Blocking Condition ; $k = 2n-1$



- request connection from last i/p to i/p switch j to last o/p in o/p switch m.
- Worse case: All other i/ps have seized top n−1 middle switches AND all other o/ps have seized next n−1 middle switches
- If $k = 2n-1$, there is another path left to connect desired i/p to desired o/p.

* Note :- No. of internal links = 2 × (no. of external links)

Puffin
Date 2/3/14
Page

☆ TEST - 1 DISCUSSION (Questions + Answers)

Q1) A) Diff. b/w virtual circuit approach & datagram approach

self
- Virtual
  → Virtual connection
  → fixed path
- Datagram
  connectionless (packet switching)
  - variable path

—x—

exact
In datagram approach, each packet is treated independently with no reference to packets that have gone before.
In virtual circuit approach, a pre-planned route is established before any packets are sent & the route is followed b/w a pair of communicating parties, all through communicⁿ.

B) Packet size significance w.r.t (i) transmission time
(ii) pipelining

Given: packet switching network
it has headers
it has message bits

self
(i) Packet size $\propto \frac{1}{\text{transmission time}}$

———

exact
Significant relⁿship b/w packet size & transmission time: Smaller packet size, more efficient pipelining
Smaller transmission time, more efficient pipelining
If packet size is smaller than usual, transmission is less efficient

(C) 2 tasks performed by transport layer of OSI model:

self { ✓ multiplexing, demultiplexing
✓ sees load of packets at network layer & channelizes.

—x—

exact { > Data Reliability
> Correct sequencing

(E) Main advantage of layering approach:-
Proper sequencing, BW allocation & easy management.

—x—

Decomposes overall complexities/problems/tasks associated with any communic^{ns} architecture, including TCP/IP.

(D) What PDU consist of?
has instruction set to communicate b/w layers

—x—

PDU consists of data (from next higher communic^{n} layer) & control inf., if anything.

—x—

Q.2. end office (1) → Switch → end office (2)

For this telephone network, find max. no. of telephones that end office can support?

(PTO)

(A) Avg. 4 calls / 8 hours. Mean call duration = 6 mins
(100% utiliz^n)

Total call duration = 6 mins × 4 = 24 mins

8 hours = 8 × 60 mins = 480 mins

Total no. = $\frac{480}{24}$ = 20 telephones

— o —

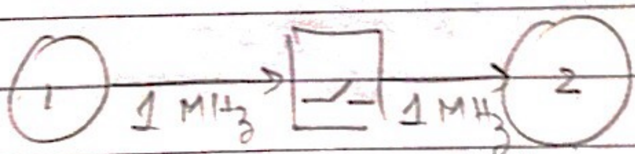
Each telephone makes 0.5 calls per hour at 6 mins each. $\left(\frac{4 \text{ calls}}{8 \text{ hour}}\right)$

A telephone occupies 3 mins/hour
(0.5 × 6 mins).

With 100% utiliz^n,

$\frac{60 \text{ mins}}{3 \text{ mins/hour}}$ = 20 telephones

(B) Given :- 10% of calls as in (A) are long distance, each utilizing 4 kHz channel for each call.
Assume 1 MHz trunk b/w each end office & switch



Total no. of calls = 4 calls × 20 telephones = 80.

So, 10% of calls = 8 calls.

So, 8 calls need 4 kHz channel

So, 32 kHz reserved for long distance

— x —

∵ 10% of calls are long distance (4 kHz channel), it takes 200 telephones to occupy a long distance of channel (FULL TIME) (20 telephones → 10% ⇒ 200 telephones = 100%)

Now, trunk has 1 MHz & 4 kHz channel.

$$\Rightarrow \frac{1\,MHz}{4\,kHz} = 250 \text{ channels}$$

Each channel supports 200 telephones.

So, total telephones $= 200 \times 250 = 50000$

Q.3 Given: No acknowledgements
No processing delay at nodes.

No. of hops, $N = 4$

Message length, $L = 3200$ bits

Data rate, $B = 9600$ bps (on all links)

Packet size, $P = 1024$ bits (fixed size)

Header (an overhead), $H = 16$ bits per packet

call setup time, $s = 0.2$ sec (wherever applicable)

Propagn delay, $D = 0.001$ sec per hop.

Compute: End to end delay for

(A) Circuit switching

(B) Datagram packet switching

(C) Virtual circuit packet switching

(A) Circuit switching :-

(bps) 9600 bits → 1 sec.

3200 bits → 0.333 sec.

$$\text{Delay} = 0.333 + \underbrace{0.2}_{s} + \underbrace{4(D)}_{4 \times \text{propagn delay}}$$

$$\text{Delay} = 0.537 \text{ sec.}$$

let C_1 : call setup time
C_2 : message delivery time
$C_1 = S = 0.2$, C_2 = Propagⁿ delay + Transmission time

$= (\text{no. of hops})_N \times D + \frac{L}{B}$

$= 4 \times 0.001 + \frac{3200}{9600} = 0.337$

$T = 0.2 + 0.337 = 0.537$

* (B) Datagram packet switching :-

$T = D_1 + D_2 + D_3 + (D_4)$

D_1 = time to transmit & deliver all packets through 1'st hop

D_2 = " " - deliver LAST packet across 2nd hop

D_3 = " " - deliver last packet across 3rd hop

D_4 = " " - 4th hop

$\therefore$ $P-H = 1024 - 16 (=1008)$ data bits / packet

Message of 3200 bits requires 4 packets
($\therefore$ 3200 bits / 1008 bits per packet
= 3.17 packets rounded up to 4 packets)

$D_1 = 4 \times t + P_0$ → propagⁿ delay through 1st hop
t → transmission time for 1st packet

$D_1 = 4 \times \left(\frac{P}{B}\right) + D = 4\left(\frac{1024}{9600}\right) + 0.001 = 0.428$

time = distance / speed

$D_2 = D_3 = D_4 = t + p = \left(\frac{P}{B}\right) + D = \frac{1024}{9600} + 0.001$
$= 0.108$

Hence, $T = 0.428 + 3(0.108)$
$T = 0.752$ sec

(C) Virtual circuit packet switching:-

$T = V_1 + V_2$

$V_1 \rightarrow$ Call setup time

$V_2 \rightarrow$ Datagram packet switching time

$= S + 0.752 = 0.2 + 0.752 = 0.952$ sec

Q.4 Consider 1 PDU in layer N is encapsulated in a PDU at layer N-1. (in OSI, say)

Now, suppose we do

(A) 1 PDU (N level) → Multiple N-1 level PDU's.
└→ Segmentation

(B) Multiple N level PDU's → 1 (N-1) level PDU.
└→ Blocking.

(A) In case of segmentⁿ, is it necessary that each (N-1) level segment contain a copy of N level header. Justify

Self {
No. Its not necessary. Specially when the bit size is large (of packets). So, keeping a copy not necessary. (Some inaccuracy)
}

B) In case of blocking, is it necessary that each N level PDU retain its own header, or can the data be consolidated into a single N-level PDU with single header.

Self — > Yes. It's necessary. To ensure that data is combined in an ordered way. (PDU's are combined)
> PDU's can be combined into single. But, that may not happen if again bit size in packets is more, making difficult to combine

— x —

(A) No. This would violate the principle of separation of layers. To layer N-1, the N layer PDU is simply data. The N-1 entity doesn't know about internal format of N level PDU. It breaks that PDU into fragments & reassembles them in proper order.

(B) Each N level PDU must retain its own header. (same as justification of (A))

Q.5 Depict a typical OSI model & tell f^n of any 5 (except transport layer)

OSI model : a chart, depicting 7 layers
(refer beginning of notes)

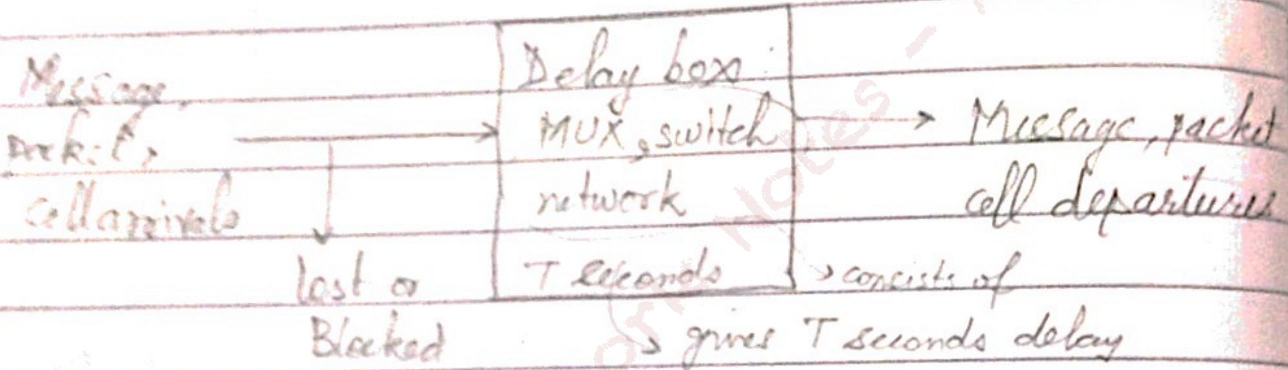
7) Application : whatever left not done in other layers.
6) Presentation layer : data encryption, compression, code conversion
5) Session : Setting up session between end points
3) Network : Sends packets & error detection & correction
2) Data link : Sends frames ; error correction & detection
1) Physical : Sends bits ; error correction

Layers	Functions
Physical	> Transfers bits across link
Data link	> Groups bits into frames & transmits/retransmits them across direct connections. > Additional f^{ns}: detects bit errors, involve in activation, maintenance & deactivation of data link connections > Medium access control for LANs, Flow control
Session layer	> Dialog management > Recovery from errors > Sets up sessions b/w 2 end pts. (while the network layer handles subnet aspects of setting up sessions) > Deals with access rights (in setting up session) *> Directs user as where a service can be accessed (like direct addressing service in telephone network)
Presentation	> Data encryption > Data compression > Machine independent represent^n of data (Code conversion req^d b/w incompatible terminals, printers, graphics ---)
Applic^n layer	> Provides services that are frequently req^d by applic^{ns} (DNS, web access, file transfer, email ---)

§ Delay & Loss Performance

* resources shared in CN.
  - Transmission BW
  - Storage
  - Processing capacity

} unscheduled demand occurs for them.



↳ packets were going from sender to receiver. So, ∃ some delay and/or loss in b/w. (Ideally, ∃ 0 delay, 0 loss ∴ not possible) ∃ diff^t combin^{ns}. We mix & match to get a configur^n s.t ∃ min. delay & loss.

* loss of message (or packet) can happen during transmission/reception or at exchange.

• Customers can be in the form of:
  - connection requests
  - individual messages
  - packets or
  - cells

• Customers arrive according to arrival pattern.

- Sys. can be :
  - An individual transmission line.
  - MUX
  - switch
  - entire network

* Let

$A(t)$: no. of arrivals at sys. from time 0 to t

$B(t)$: no. of blocked customers from time 0 to t

$D(t)$: no. of departures in same time inverval.

$N(t)$: no. of customers from 0 to t

$$N(t) = \{A(t) - B(t)\} - D(t)$$

↳ Assuming sys. is empty at $t = 0$

eg: If 100 customers arrived.
30 departed
10 blocked.

So, my customers are $100 - 30 - 10$
$= 60$.

* Arrival rate, $\lambda = \lim_{t \to \infty} \frac{A(t)}{t}$.

→ long term arrival rate
(no. of customers/sec)

* Throughput $= \lim_{t \to \infty} \frac{D(t)}{t}$

→ long term departure rate (no. of customers/sec)

* $E[N] = \lim_{t \to \infty} \frac{1}{t} \int_0^t N(t')dt'$

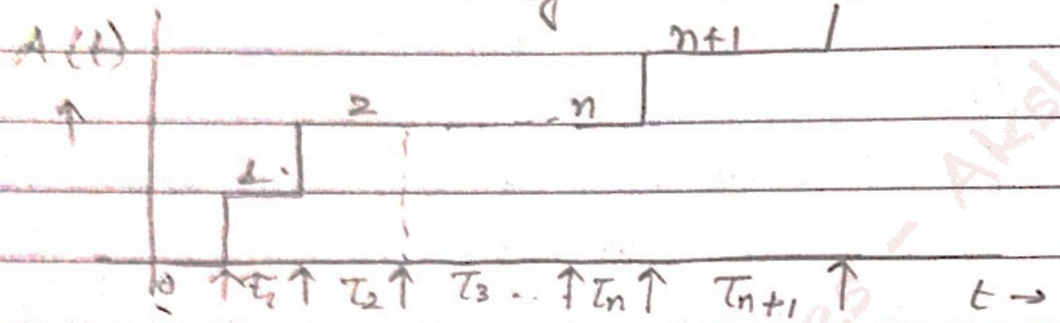
↳ estim^n

→ Avg. no. of customers in sys given by $E[N]$

* Fraction of blocked customers

$$P_b = \lim_{t \to \infty} \frac{B(t)}{A(t)}$$

* Consider an arrival sys : Plot of time vs arrivals



Time of n^{th} interval $= \tau_1 + \tau_2 + \dots + \tau_n$

So,

arrival rate upto time when n^{th} customer arrives is given by λ

$$\lambda = \lim_{n \to \infty} \frac{n \text{ arrivals}}{\tau_1 + \tau_2 + \dots \tau_n \text{ sec.}} = \lim_{n \to \infty} \frac{1}{(\tau_1 + \tau_2 + \dots + \tau_n)/n}$$

$$\Rightarrow \lambda \longrightarrow \frac{1}{E(\tau)}$$

* Note : in the above, inter arrival times are statistically independent & have same probability distribⁿ & their avg. or expected value is given by $E[\tau]$

The avg. arrival rate $= \dfrac{1}{\text{mean interarrival time}}$.

Puffin
Date
Page

eg: I am at war (communicⁿ network). I have warships (channels) with me. I want to send soldiers (packets/messages) to fight (transmit).

What to see ① how to manage soldiers; their no. to be sent across warships so that they go efficiently across the sea (buffer capacity)

② Check if enemy (receiver) is there in the first place or not. Otherwise, there is no need to send soldiers to fight (send packets/messages)

* Home Assignment ⑤: Data analysis fnˢ of MS excel

Take min. no. of sample points 20 or above (for Gaussian)

① See diffᵗ data analysis tools. Tell brief of that.

② See the distribⁿˢ in Excel (like Gaussian distribⁿ)

See what are the tests for these distribⁿˢ (how to check if Gaussian or not) → like Chi-square test

* CN theory mainly follows Poisson Distribⁿ.
(Others used: Weibull, Gaussian)

✡ LITTLE's FORMULA : (Proof: self)

→ Avg. no. of customers in the sys $E[N]$ = Arrival rate, λ times.

$A(t) \rightarrow$ | Delay Box. No. in sys., $N(t)$ Varies as $A(t) - D(t)$ [as $B(t) = 0$] | $\rightarrow D(t)$

$$E[N] = \lambda E[T]$$

$E[N]$ → no. of packets

λ → packet arrival rate

$E[T]$ → time spent by packet in network

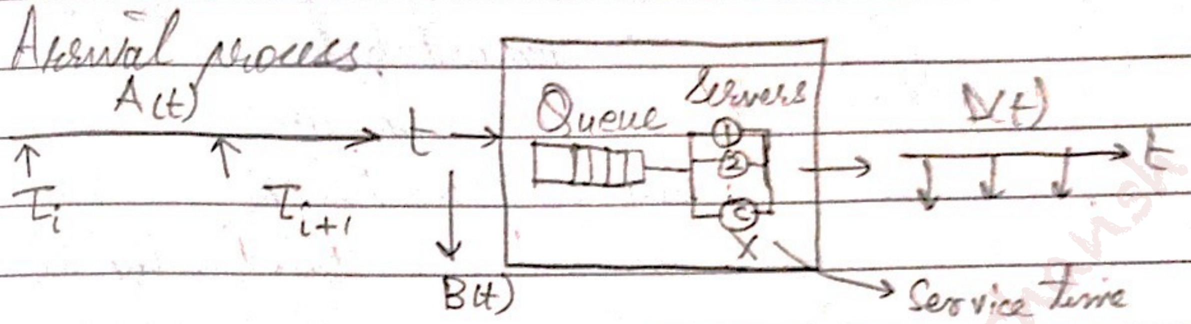
- Applicⁿ of Little's formula :
  - an individual transmission line, MUX, switch, network.
  - find avg. delay experienced by a packet in traversing a packet switching network
    - $E[N_{nt}] = \lambda_{nt} E[T_{nt}]$

$$\Rightarrow E[T_{nt}] = E[N_{nt}] / \lambda$$

* Consider a packet switch as a set of MUX. (assume)
A packet is routed into & placed in MUX.

* Observations based on Little's formula :
- Avg. network delay depends on:
  - overall arrival delay
  - arrival rate to individual MUXs
  - delay at each MUX.
- Arrival rate at each MUX is determined by routing algorithm.
- Delay in MUX depends on :
  - arrival rate
  - rate at which associated transmission line can transmit packets.
- Expression for $E[T_{nt}]$ is used to design and management of packet-switching networks.
- To obtain $E[T_{nt}]$, its necessary to analyse delay performance of each MUX, which can be done using ~~Queing~~ Queuing models.

✳ Basic Queuing models :



✳ Queuing model : given by Erlang
↳ given models for resource sharing systems

Assume In the above model :

- Interarrival time (time b/w 2 arrivals) are independent random variables with same distribⁿ
- Arrivals are known to come after a particular time (deterministic)
  T : interarrival time = const
- If arrival are T's exponential with mean $E[T] = 1/\lambda$, then, $P[T > t]$:
  - Exponential arrival times leads to tractable analytical results.
  - Arrivals in t is given by POISSON random variable with mean
    $$\underbrace{E[A(t)]}_{\text{expected arrival time}} = \lambda t$$

Prob. of interarrival time is greater than some time, t

✳ If ∃ exponential interarrival times (T), its called Poisson arrival process.

Processing capacity (μ) of single server
$= \frac{\text{max departure throughput rate}}{} = \frac{1}{E[\tau]}$ customers/sec

X : time reqd to service a customer
1, 2, 3, ... c : Servers, used to denote resources.
- Service discipline : FIFO, LIFO, Priority service, random order service.
  - usually FIFO (first in first out)

K : max. no. of customers allowed in queuing sys
So, $N(t) = \underbrace{N_q(t)}_{\substack{\text{queuing} \\ \text{(in sys)}}} + \underbrace{N_s(t)}_{\substack{\text{service} \\ \text{(receiving service)}}}$
- when $N(t) = k$: new customer arrivals are blocked.

★ Queuing Systems ~~Classf~~ Classification.
Arrival process / Service time / Servers / Max. Occupancy.

i.e, we write it as ( ) / ( ) / ( ) / ( )

Arrival process : M : exponential.
D : Deterministic
G : General
Arrival rate :
$\lambda = 1/E[\tau]$

Service time : M : exponential.
D : deterministic
G : General.
Service rate : $\mu = 1/E[X]$.

Servers : 1 server ;
c servers ;
infinite .

Max. occupancy : K customers ; unspecified if unlimited .

* Suppose I say a process as M/M/1/K
⇒ the arrival process is exponential with service process as exponential using 1 server and K customers can be accomodated

* If K is unspecified ⇒ ∃ no limit to the no. of customers allowed in the sys (M/M/1)

~

* Reading Assignment :
Study Markov Process & its applications .

~

* T : total delay — N(t) : no. in sys.
W : waiting time — $N_q(t)$: no. in queue
X : service time — $N_s(t)$: no. in service .

$T = W + X$ $\qquad$ $N(t) = N_q(t) + N_s(t)$

P_b : blocking probability

• Actual arrival rate into sys. is given by $\lambda(1-P_b)$

Avg. occupancy :
Avg delay { avg.
performance

$$E[N] = \lambda(1-P_b)E[T]$$
$$E[N_q] = \lambda(1-P_b)\,E[W]$$
$$E[N_s] = \lambda(1-P_b)\,E[X]$$

☆ Definitions

☆ Traffic offered / load : rate at which 'work' arrives at the sys.

$$a = \{\lambda \text{ customers / sec}\} \cdot \{E[X] \text{ seconds/customer}\}$$

$$\Rightarrow a = \frac{\lambda}{\mu} \text{ Erlangs}$$

☆ Carried load : avg. rate at which sys. does work

$$E[X] \cdot \lambda(1-P_b)$$

$$\Rightarrow \text{Carried load} = a(1-P_b)$$

☆ Utilization (ρ) : avg. fraction of servers in use

$$\rho = E[N_s]/c = \lambda(1-P_b)/c\mu.$$

Idea What are we doing?

Considering that I go to a currency exchange.
If a lot of rush → work delays.
counters free → work gets done
So, the exchange person has to see how many counters have to be made or, how the rows etc have to be designed so that rush is managed AND $\exists$ no wastage in off peak hours.
This same is seen in telephone networks

☆ M/M/1 : Basic multiplexer model.

☆ Time = $\frac{\text{Distance}}{\text{speed}}$

$$\Rightarrow \text{Avg. transmission time } E[X] = \frac{E[L]}{R}$$

avg packet length

speed of transmission line.

Idea of using MUX

select lines change at every fixed interval of time (say, connected to counter). So, a particular i/p is taken & given out to o/p based on select line i/ps.

* Self Imp • Service Time Variability and Delay

→ what will be the inter arrivals, capacities & efficiencies in diff^t queuing systems (M/D/1, M/E/R/1, M/M/1, M/H/1)

→ Interarrivals : Constt, Erlang, Exponential, Hyperexponential

* Reading Assignment : Pg → 157 – 162 (ex. 3.1 to 3.7)

Q. Consider a network gateway. Measurements show that packets arrive at mean rate of 125 pkts/sec & gateway takes 2 ms to forward them. Using M/M/1 model, analyse the gateway →

(1) Min. no. of packets in gateway ?

(2) Mean time spent in gateway ?

(3) Prob. of buffer overflow if gateway had only 13 buffers

(4) How many buffers we need to keep s.t packet loss is below 1 packet per million ?

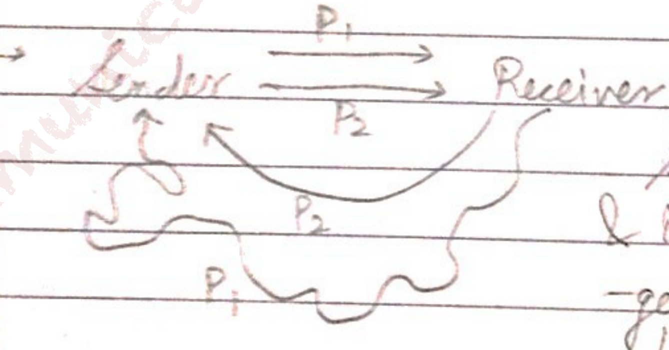
Chapter - 5.
PEER-TO-PEER PROTOCOLS AND DATA LINK LAYER

* ARQ : Automatic Repeat Request
  ↳ Purpose : ensure that packets are delivered in order without errors & duplic^ns despite transmission errors & losses.

look at

Stop & Wait ARQ's   Go back N-ARQ   Selective Repeat ARQ.

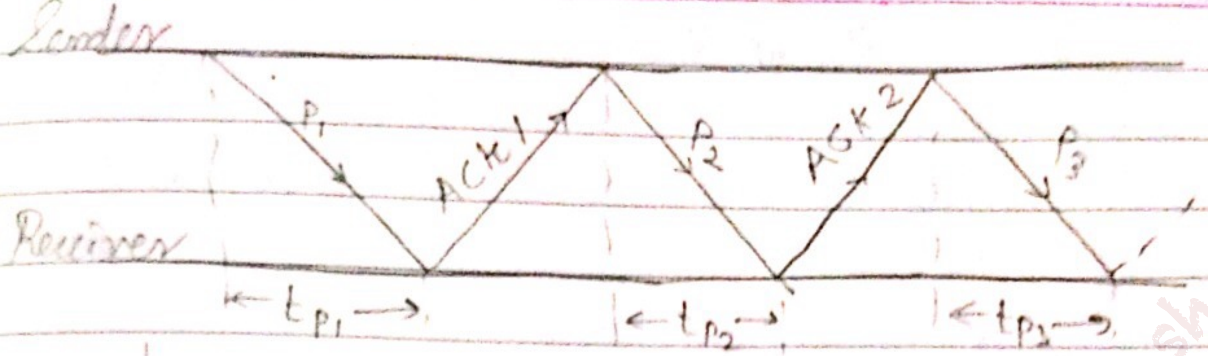
• Basic elements of ARQ.
  ✓ Error detecting code with high error coverage.
  ✓ ACK; (+ve acknowledgement)
  ✓ NAK; (-ve acknowledgements)
  ✓ Timeout mechanism.



Sender sends P_1 & P_2 & is waiting for acknowledgement from receiver. He gets P_2 quickly & mistakes it to be an acknowledgement of P_1 (∵ P_1 gets delayed). So, its -ve acknowledgement

→ I haven't send anything, still I get an acknowledgement

① Stop & Wait ARQ:



↳ Wait for acknowledgement for each & every packet.

Suppose $\exists$ N packets

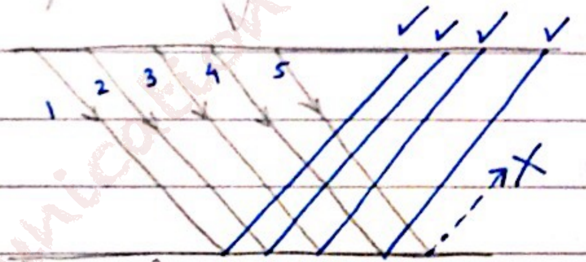
& $t_{P_1} = t_{P_2} = t_{P_3} = \dots t_{P_N} = t_P$

So, total delay =

$$N(t_P + t_{ACK})$$

② Go Back N-ARQ

Suppose we dont wait for acknowledgement & keep sending:



Suppose I miss out on getting 5th acknowledgement

What to do then?

(M1) Stop transmission process. Then, again start from P5 and send all again.

Idea: P5 is sent, P6 is sent, P7 is sent, ...

acknowledgement for ... P4 is received, P5 not received

Automatic request for retransmission

Stop → Send P5 again, P6 sent again, P7 sent again ...

So, **Duplications come** → due to loss of message or acknowledgement.

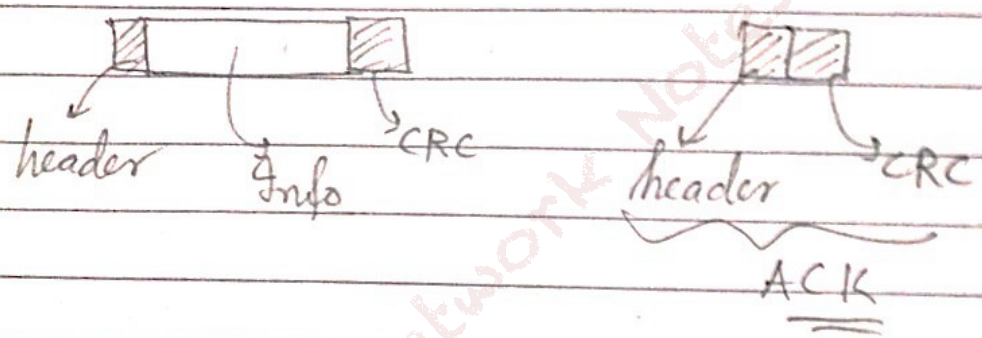
③ Selective Repeat ARQ

In previous case, P5's acknowledgement wasn't received. So, I sent all the next packets.

But now, I only send P5 (∵ P5's acknowledgement wasn't received).

But, ∃ more processing time.

① Information frame   Control frame



✓ We can also send sequence no. to the frames so that if we miss out on receiving, we get to know error in frame/acknowledgement.

✓ To remove errors, we can increase overhead, i.e., by telling the details of next incoming frame with every packet.

✓ Stop & wait algorithm :

Transmitter	Receiver.
- Ready state	- Always in ready state.
- Wait state	

Numerical type

☆ Rate of speed transfer (bps) ≡ speed

bit frame size ≡ distance.

Time taken to transmit ≡ time.

So, for 10000 bit frame at 1 Mbps,

time $= \frac{10^6}{10^4} = 10\text{ ms.}$ (usually taken time)

Numerical type. ☆

$$\text{Efficiency } (\eta) = \frac{\text{Time taken to transmit}}{\text{Time taken to transmit} + (\text{Acknowledgement time})}$$

waiting time.

→ If ACK = 1 ms,

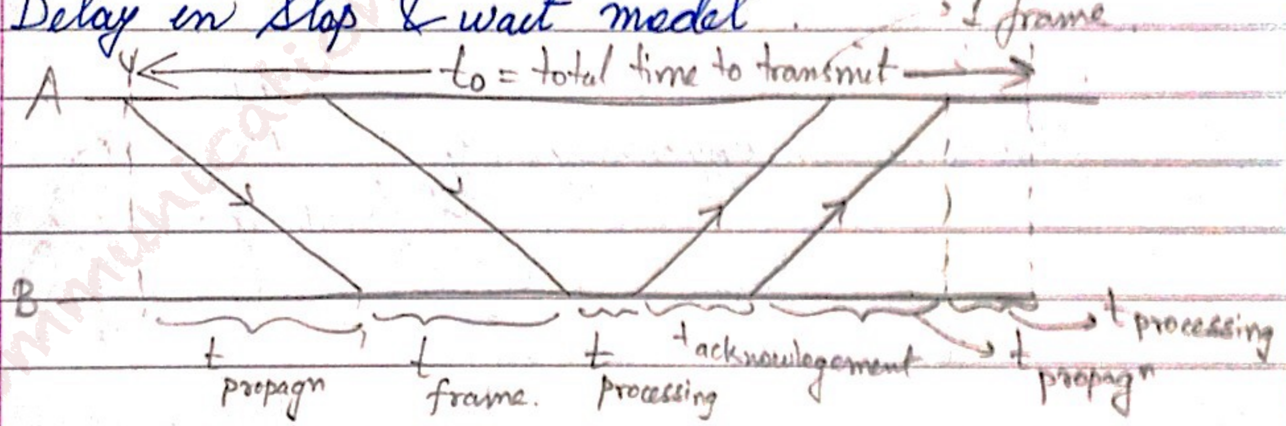
$$\eta = \frac{10}{10+1} = 91\%$$

→ If ACK = 20 ms.

$$\eta = \frac{10}{10+20} = 33\%$$

☆ Increase waiting time ⇒ decrease efficiency.

☆ Delay in Stop & wait model.



$$t_0 = t_{prop} + t_{frame} + t_{proc} + t_{ack} + t_{prop} + t_{proc}.$$

$$= 2t_{prop} + 2t_{proc} + \frac{n_f}{R} + \frac{n_a}{R}$$

n_a → bits / ack frame

n_f → bits / info frame

R → channel transmission rate

* CRC :- Error DETECTING Codes
check - Single bit detection

* What is t_{proc}?

↳ Data link does a lot of work (transferring bits across frames, error correction, ...). All the time taken to process that is t_{proc}

* Stop & wait efficiency on error free channel :-

Effective transmission rate, R^{0}_{eff} — error free, effective

$$= \frac{\text{no. of info. bits delivered to destin}^n}{\text{total time req}^d \text{ to deliver info bits}}$$

(Ignoring header & CRC)

$$= \frac{n_f - n_o}{t_o}$$

Transmission efficiency, (η_0) without error

Effect of frame overhead

Effect of delay-BW product

$$\eta_0 = \frac{R_{eff}}{R} = \frac{\left(\frac{n_f - n_o}{t_o}\right)}{R} = \frac{\left(1 - \frac{n_o}{n_f}\right)}{1 + \frac{n_a}{n_f} + \frac{2(t_{prop} + t_{proc})R}{n_f}}$$

effect of ACK frame

Assignment * What are specific applic^ns of error d/c (detecting / correcting) codes

* Self :- * What are error correcting codes?

* List error correcting codes studied as on date (in other courses) and detecting codes

* Compare the diff^t error detecting & correcting codes.

- capabilities of code
- how many bit error,
- how does it detect/correct

n_o : no. of bits in overhead
n_a : " " " acknowledgement
n_f : " " " frame.

eg : Delay-BW product :

Consider: $n_f = 1250$ bytes $= 10000$ bits
$n_a = n_o = 25$ bytes $= 200$ bits.

2x Delay x BW efficiency.	1 ms 200 km	10 ms 2000 km	1000ms 20000 km	1s 200000 km
1 Mbps.	10^3 88%.	10^4 49%.	10^5 9%.	10^6 1%.
1 Gbps	10^6 1%.	10^7 0.1%	10^8 0.01%.	10^9 0.001%

↳ Stop & wait algorithm doesn't work well for very high speeds or long propag^n delay.

Type of question : Compare the delay-BW product efficiencies & conclude results.

* Stop & wait efficiency in channel with errors.

↳ $1 - P_f$ = probability that frame arrives without errors

$$\eta_o = \frac{\left(\dfrac{n_f - n_o}{t_o}\right)}{\left(\dfrac{1-P_f}{R}\right)} = \frac{\left(1 - \dfrac{n_o}{n_f}\right) \times (1 - P_f)}{\left(1 + \dfrac{n_a}{n_f} + \dfrac{2(t_{prop} + t_{proc})R}{n_f}\right)}$$

eg: Impact bit error rate :-

$n_f = 1250$ bytes $= 10000$ bits, $n_a = n_o = 25$ bytes $= 200$ bits

Find efficiency for random bit errors with $p = 0, 10^{-6}, 10^{-5}, 10^{-4}$

$$1 - P_f = (1-p)^{n_f} \simeq e^{-n_f p}$$

↳ for large n_f & small p

$1 - P_f$ efficiency	0	10^{-6}	10^{-5}	10^{-4}
1 Mbps. & 1 ms	1 88%	0.99 86.6%	0.905 79.2%	0.368 32.2%

↳ bit errors impact performance as $n_f p$ approaches 1

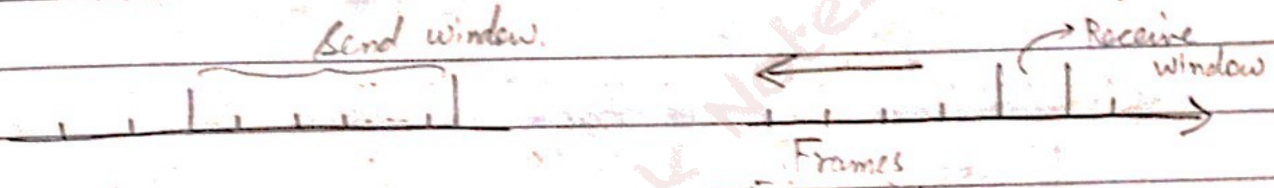
② Go-Back-N ARQ (detailed)

- Improve stop & wait by not waiting
- Keep channel busy by continuing to send frames
- Use m bit sequence numbering
- Allow a window of up to W_s outstanding frames.
- If ACK for oldest frame arrives before window is exhausted, we can continue transmitting
- If window is exhausted, pull back & retransmit all outstanding frames.

Alternative : use timeout

- ✓ frame transmission are pipelined to keep channel busy.
- ✓ If the ACK is not received, I retransmit that bit (& all others after that). But, if I don't have any more bit to transmit of the last bit, ∃ no retransmission & hence, I lose that data.

Transmitter Receiver

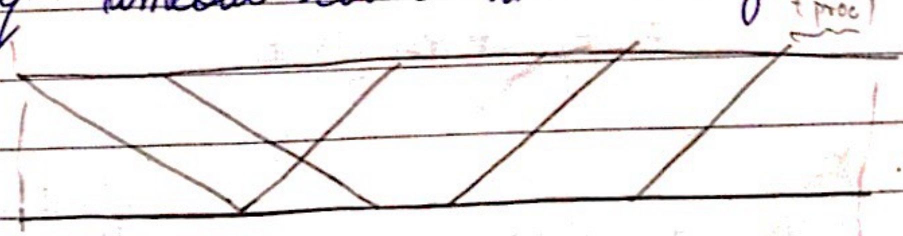


Idea of window is to see if that bit is sent (or received) & then move to next bit

* Concept of using R^A_{next}, S^A_{next}, PIGGYBANKING

Uses : In HDLC (High level Data Link control)
V.42 modem : error control over telephone window

- Reqd timeout and window size



Puffin
Date
Page

☆ Reqd Window size for delay BW product.

Frame = 1250 bytes = 10,000 bits, R = 1 Mbps

$2(t_{prop} + t_{proc})$	$2 \times$ Delay $\times$ BW	Window
1 ms	1000 bits	1
10 ms	10000 bits	2
100 ms	100,000 bits	11
1 sec	1,000,000 bits	101

☆ Efficiency for Go-Back N (GBN)

↳ GBN is completely efficient if W_s is large enough to keep channel busy & channel is error free.

P_f : frame loss prob.

t_f : time to deliver frame if frame transmission succeeds $(1-P_f)$

$T_F + W_s t_f / (1-P_f)$ if 1st transmission doesn't succeed (P_f)

$$t_{GBN} = t_f(1-P_f) + P_f\left\{t_f + \frac{W_s t_f}{1-P_f}\right\}$$

$$= t_f + P_f \frac{W_s t_f}{1-P_f}$$

&

$$\eta_{GBN} = \frac{n_f - n_o}{\frac{t_{GBN}}{R}} = \frac{1 - \frac{n_o}{n_f}}{1 + (W_s - 1) P_f} \times (1-P_f)$$

→ Delay×BW determines

Puffin
Date
Page

eg Impact BER on GBN.

$n_f = 1250$ bytes $= 10000$ bits, $n_a = n_o = 25$ bytes $= 200$ bits

Compare S&W with GBN efficiency for random bit errors with $p = 0, 10^{-6}, 10^{-4}$ and $R = 1$ Mbps & 100ms

1 Mbps × 100ms = 100000 bits = 10 frames → Use $W_s = 11$

Efficiency	0	10^{-6}	10^{-5}	10^{-4}
S&W	8.9%	8.8%	8.0%	3.3%
GBN	98%	88.2%	45.4%	4.9%

$$GBN \rightarrow \eta_{GBN} = \left(\frac{1250 - 25}{\frac{t_f}{10^6}} \right) = \frac{1 - \frac{25}{1250}}{1 + (11-1)(0)} (1-0)$$

$$= 1 - \frac{1}{50}$$

$$= 1 - 0.02 = 0.98$$

$$\Rightarrow \eta_{GBN} = 98\% \text{ for } P_f = 0.$$

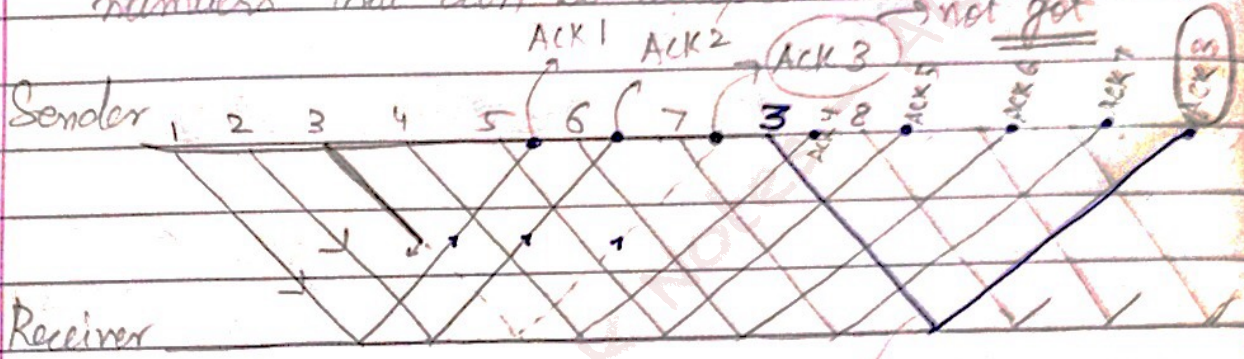
$$S\&W \rightarrow \eta_o = \frac{\left(1 - \frac{25}{1250}\right)}{1 + \frac{25}{1250} + \frac{2(t_{prop} + t_{proc})10^6}{1250}}$$

(3) ★ Selective Repeat ARQ.

↳ Transmits only an individual frame.

- ↳ Timeout causes individual corresponding frame to be present
- ↳ NAK causes retransmission of oldest un-acked frame

★ Receiver maintains a receive window of sequence numbers that can be accepted



frame no. 3 sent again

- ∃ send & receive windows.
- We will have more processing time here
  - ↳ If any frame's ACK is not got on time, we send again. Now, when we receive the ACK, we rearrange it to its expected place before giving to user ⇒ Processing

$$\eta_{SR} = \frac{\frac{n_f - n_o}{t_f/(1-P_f)}}{R} = \left(1 - \frac{n_o}{n_f}\right)(1-P_f)$$

↳ P_f : frame loss Prob.

Puffin
Date 6/4/14
Page

8) MEDIUM ACCESS CONTROL PROTOCOLS:

* In telecommunicⁿ, medium is shared (most of the time)
  ↳ Dedicated medium - very rare.
* Shared media is inexpensive - like radio, cables.
* No. of users are ↑↑ → Users communicating by broadcasting into medium.

* **Broadcasting**
  - Many listeners, one speaker. But, in CN, what happens is, packet is broadcasted, but just one person takes the info

* **Broadcalling**
  - Many speakers, one listener.

* **Approaches to media sharing**
  Medium Sharing Techniques

  - **Static/Semi-Static Channelizⁿ.**
    - ✓ Partition medium.
    - ✓ Dedicated allocⁿ to users
    - ✓ Satellite transmⁿ
    - ✓ Cellular telephone

  - **Dynamic medium access control.**
    - **Scheduling**
      - ✓ Polling: take turns
      - ✓ Request for slot in transmission schedule
      - ✓ Token ring
      - ✓ Wireless LANs.
    - **Random Access**
      - ✓ Loose Coordinⁿ
      - ✓ Send, wait, retry if necessary
      - ✓ Aloha
      - ✓ Ethernet

★ Scheduling : Polling

A host comp. sends (trying to communicate with station 1, say) poll data with some station 1. Station 1 responds. Then, data is sent to the particular station (once all stations' poll is taken)

eg:

I go to a restaurant with my friends (total 5 people ≡ 5 stations) The waiter (host computer) comes to take order. Waiter goes to each and every person to take order. Once the order comes, he (computer) brings dish 1 (poll) and goes to each person (station) asking if they ordered for that dish (taking poll) When anyone says yes (confirmation), the dish is given to that person (data transfer) This same process goes for all dishes coming.

Note: Once order is placed & we all had food, waiter (host computer) comes again to ask for desserts (any further data to be sent by any sent by station to host computer)

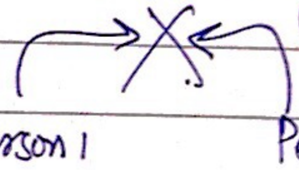
★ Scheduling: Token-Passing

∃ a token, it is put on the channel. Whosoever picks the token gets the access to the channel & can ~~st~~ send data.

✓ Ring topology.

Random Access:

Channel is accessed at random. No token, no polling. So, people sending through the channel at the same time might have a crash (collapse)



So, ARQ strategies are seen in this.

Wireless LAN

→ you get network when you need. Not always.

AdHOC : station-to-station

~~Infrastructure~~:

Selecting a Medium Access Control

✓ Applic^ns

- What type of traffic?
- You accept delay / not.
- More/less data is being sent?

✓ Scale

- how much traffic can be carried?
- how many users can be supported?

✓ Current examples

- Design MAC to provide wireless DSL-equivalent access to rural communities.
- Design MAC to provide wireless LAN-equivalent access to mobile users (user in car, traveling at 30 km/hr, say).

✯ Delay - Bandwidth Product

- Key Parameter:
  - coordinⁿ in sharing medium involves using bandwidth (explicitly or implicitly)
  - Difficulty of coordinⁿ commensurate with delay-BW product.

- Simple two station example
  - Station with frame to send, listens to medium and transmits if medium found idle.
  - Station monitors medium to detect collision.
  - If collision occurs, station that begin transmitting earlier, retransmits (propagⁿ time is known)

eg 2 station MAC example.

Consider 2 stations, d metres apart

$t_{prop} = \frac{d}{v}$ sec.

Case (1):- B does not transmit before $t = t_{prop}$. So, A captures channel (So, no collision)

Case (2):- B transmits, exactly at, or, before $t = t_{prop}$ & detects collision soon thereafter (at $t = t_{prop}$)

Each frame transmission requires $2t_{prop}$ of quiet time
- ✓ Station B needs to be quiet t_{prop} before & after time when Station A transmits. So, waiting time = $2t_{prop}$
- ✓ R: transmission bit rate (diff^t from R_{eff})
- ✓ L bits/frame; sending station requires $X = L/R$ sec for retransmitting its frame.
- ✓ Effective Throughput (R_{eff}) is actual rate (bps) at which info sent over channel

* Efficiency : Normalized max. throughput (R_{eff}/R)
* Normalised delay-BW product (a) : ratio of 1-way delay b/w product to avg. frame length.

Formulas for efficiency, a, ...

Inference from example :
- If $a \ll 1$ → efficiency close to 100%
- If $a > 1$ → efficiency decreases.

Two station example :

$$\text{efficiency} = \frac{1}{1+2a}$$

Carrier Sense Multiple access collision detection (CSMA-CD) Ethernet Protocol :-

$$\text{efficiency} = \frac{1}{1+6.44a}$$

Token ring network

$$\text{efficiency} = \frac{1}{1+a}$$

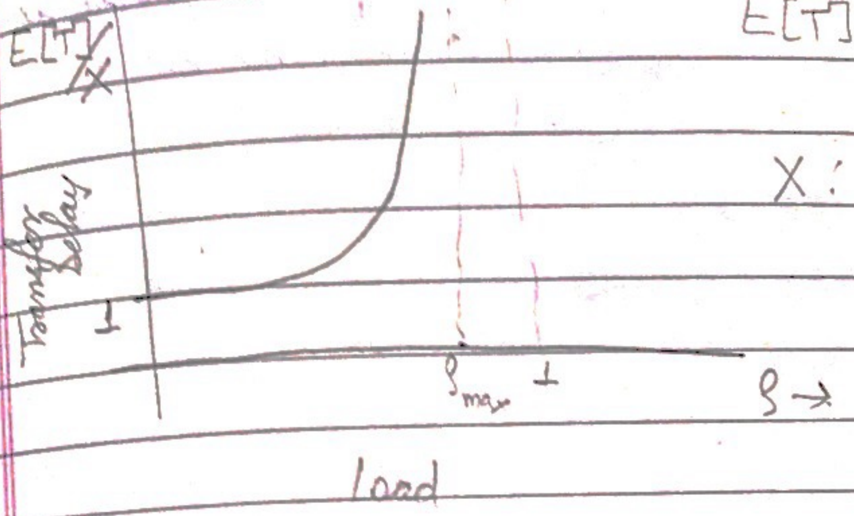
→ latency of ring bits/avg. frame length.

* Desk area network : Connect systems in a room from desk to desk, through ethernet

* MAC Protocol features
  - Delay-BW product
  - Efficiency
  - Transfer delay
  - Reliability
  - Capability to carry diff^t types of traffic.
  - ~~QS~~ QOS.
  - Cost

* MAC Delay Performance
  To evaluate MAC delay performance, look into:
  ① Frame transfer delay.
    - From 1st bit of frame arrives at source MAC
    - To last bit of frame delivered at destin^n MAC
  ② Throughput : Actual transfer rate through shared medium (in frames/sec or bits/sec)
  ③ Parameters
    - R bits/sec & L bits/frame.
    - $X = L/R$ seconds/frame
    - λ frames/second average arrival rate.
    - Load $S = \lambda X$, rate at which "work" arrives
    - Max. throughput (at 100% efficiency) : R/L frames/sec

* Normalized Delay Vs Load



E[T]: avg frame transfer delay.

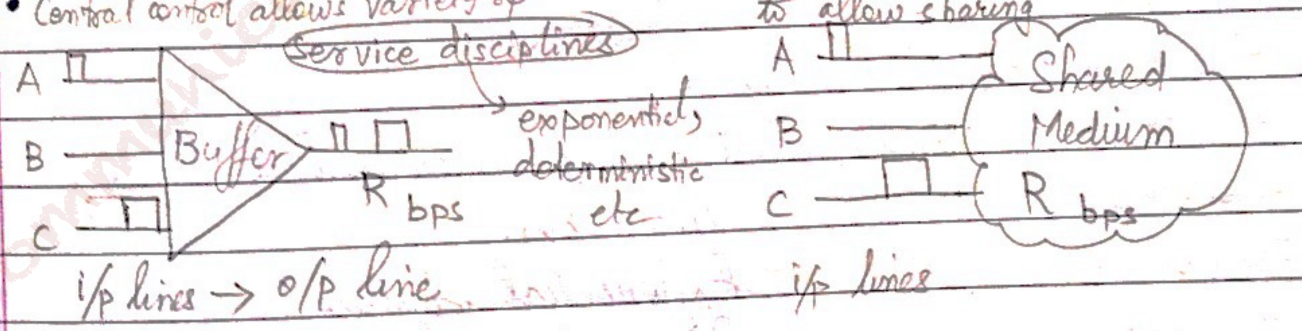
X: avg. frame transmission time

* Statistical Multiplexing and Random Access
↳ Delay Performance of Channeliz^n schemes.

* Channeliz^n: Protocols & terms we adopt to gain access in a communic^n channel

- Multiplexing concentrates bursty traffic into a shared line
- Central control allows variety of service disciplines

- MAC allows sharing of a broadcast medium
- Decentralized control "wastes" BW to allow sharing



* There is a trade-off between statistical multiplexing & MAC (or other methods). Decide what is your purpose & choose.

- Packets are encapsulated in frames & queued in a buffer, prior to transmission

- Packets are encapsulated in frames and queued at station, prior to transmission

☆ Statistical Multiplexing and Multiple Access Control

☆ Applicⁿ Properties

- how often are packets generated?
- how long are packets?
- what are loss & delay requirements

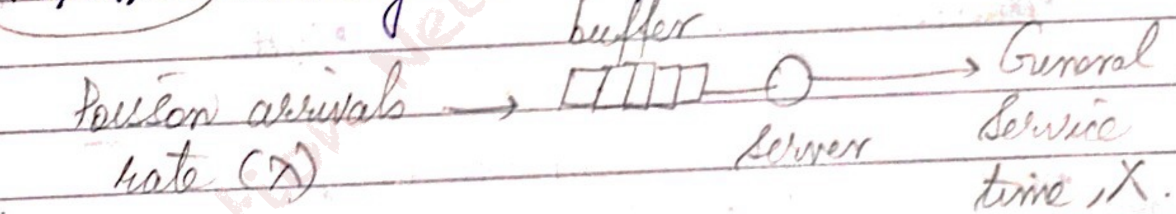
☆ Sys. Performance

- Transfer delay.
- Packet/frame loss
- efficiency & throughput
- Priority, scheduling & QOS

Basically, in both the models, I am seeing what exactly do we look at, in terms of applicⁿ properties & sys. performance

☆ M/G/1 Queuing model for Statistical Multiplexer

Exponential arrival, General service time, 1 server, Infinite customers.



- Arrival mode.
  - Independent frame interarrival times.
    - Avg 1/λ.
    - Exponential distribⁿ
    - Poisson Arrivals.
- Infinite Buffer
  - No Blocking.
- Frame Length model
  - Independent frame transmission times X
    - Avg E[X] = 1/μ.
    - General distribⁿ — Constt, exponential, ...

Puffin
Date
Page

- Load, $\rho = \lambda/\mu$.
  - Stability Condition : $\rho < 1$.

Inference :

1. M/G/1 model as baseline for MAC performance.

Channelization Approaches.

- Frequency Division Multiple Access (FDMA)
- Time " " " (TDMA)
- Code " " " (CDMA)

→ each user is allocated a band. eg in FM radio, television (everyone gets some band), internet phones etc.
→ Periodic time slots allocated to users eg: Main backbone of telephone working, GSM digital cellular phone
→ Code is allocated to users. Info has the CODE TAG through whichever medium it goes eg: Cellular phones, 3G phones

To take care of :

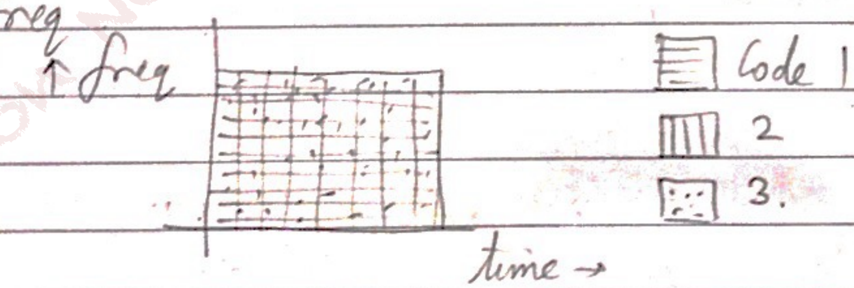
FDMA	TDMA	~~CDMA~~
freq. bands must be non overlapping.	Stations must be synchronized for common clock.	

Q. Compare FDMA & TDMA in terms of their ability to handle groups of stations that produce info. flows that are produced at constt. but diff^t bit rates

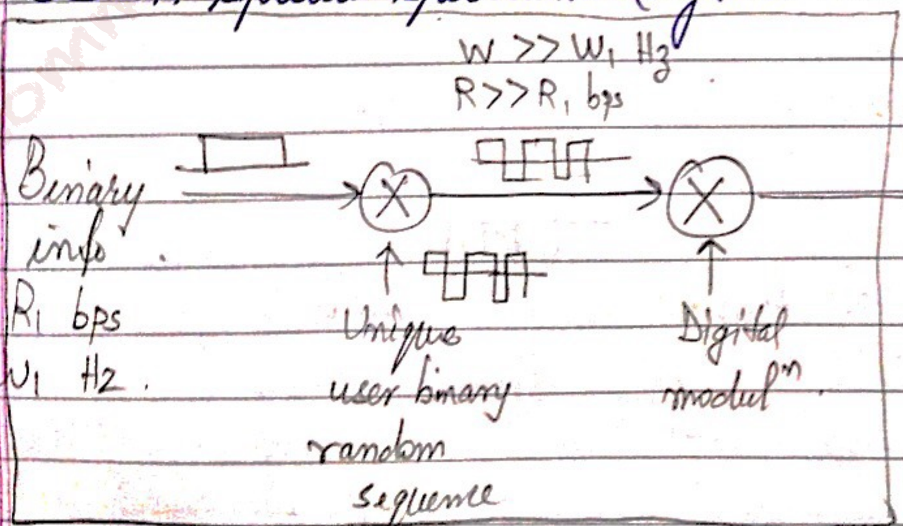
Ans:- let bit rate $= R$.

FDMA	TDMA
- Better efficiency (relatively)	- Not as good as FDMA
- Lesser flexibility	- More flexible than FDMA
- Transmission rate $= \frac{R}{M}$, M: no. of stations	- Avg. bit rate $= \frac{R}{M}$, M: no. of time slots

- Channeliz^n using CDMA:
    - Stations transmit over entire freq. band all the time.
    - $\exists$ diff^t codes all, existing at all times & freq



- CDMA Spread Spectrum Signal.



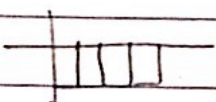
Date
Page

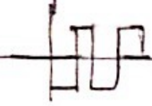
- User info is mapped into +1 & -1 for T seconds.
- Many codes (set of Gi) are added to it.
- Reverse process in Demodulation

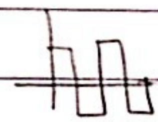
self : * Pseudorandom codes.

eg: 3 Users are there. Implement CDMA
Each user uses a 4-bit orthogonal code
That code is multiplied by info. bits & given to receiver.

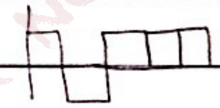
Codes

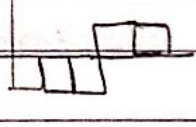
User (1) : Code : {-1, -1, -1, -1} = 

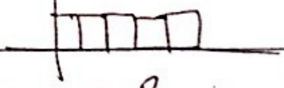
User (2) : Code : {-1, 1, -1, 1} = 

User (3) : Code : {+1, +1, -1, -1} = 

Info

User (1) :- 

User (2) :- 

User (3) :- 

* Coding happens as :

Code X Info. for User 1

+

Code X Info for User 2

+

Code X Info for User 3.

Self : Decoding

✯ WALSH Functions:

- Provide orthogonal code sequences by mapping 0 to -1 and 1 to +1.
- Walsh matrix provides orthogonal spreading sequences of length $n = 2^m$
- Matrix

$$W_{2n} = \begin{bmatrix} W_n & W_n \\ W_n & W_n^c \end{bmatrix}$$

Q. Suppose that a 1 MHz channel can support a 1 Mbps transmission rate. The channel is to be shared by 10 stations. Each station receives frames with exponential inter-arrivals and rate $\lambda = 50$ frames/sec & frames are of constt length $L = 1000$ bits.
- Compare the total frame delay of a sys. that uses FDMA to a sys. that uses TDMA

Solⁿ Given :- $R = 1$ Mbps.

$L = 1000$ bits.

$M = 10$ stations.

$X = L/R = 10^{-3}$ sec.

Load at one station, $\rho = (\lambda/M)(X \times M)$

$= 50(10 \times 10^{-3}) = 0.5$

FDMA

Total packet delay $= [\rho M / 2(1-\rho) + M/2 + M] E[X]$

$= [(0.5)(10)/2(1-0.5) + 10/2 + 10][10^{-3}]$

$= [20][10^{-3}] = 0.02$

TDMA

Total packet delay $= [\rho M / 2(1-\rho) + M/2 + 1] E[X]$

$= [(0.5)(10)/2(1-0.5) + 10/2 + 1][10^{-3}]$

$= [11][10^{-3}]$

$= 0.011$

- Both TDMA & FDMA are sensitive to no. of stations. We can observe that TDMA outperforms FDMA ∵ of the fastest frame transmission line

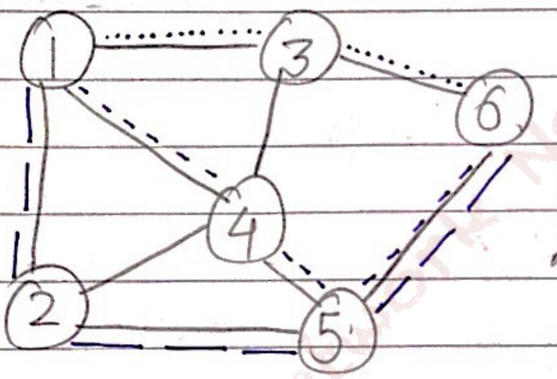
Date 14/4/14

Chapter - 7

PACKET-SWITCHING NETWORKS

* Routing in packet networks -

Consider a network of routing



We see 3 routes from 1 to 6.
• 1-3-6, 1-4-5-6, 1-2-5-6.

Problems Note : We can't say that 1-3-6 is shortest & best. We need to know a few things
So, Assume : (i) Distances are scaled to all routes (min hop)
(Min delay) { (ii) Traffic congestion is same in all routes.
(iii) BER of a link (stations are connected through links). If BER is high in a link, ∃ more errors in message. This arises need of retransmission. Hence, chances of delay are more.

(iv) Max. Bandwidth.
eg: 1-3-6 : Copper link
1-2-5-6 : Optical fibre
Whichever has higher BW, data is got faster

(v) Min. cost

(vi) Max. reliability

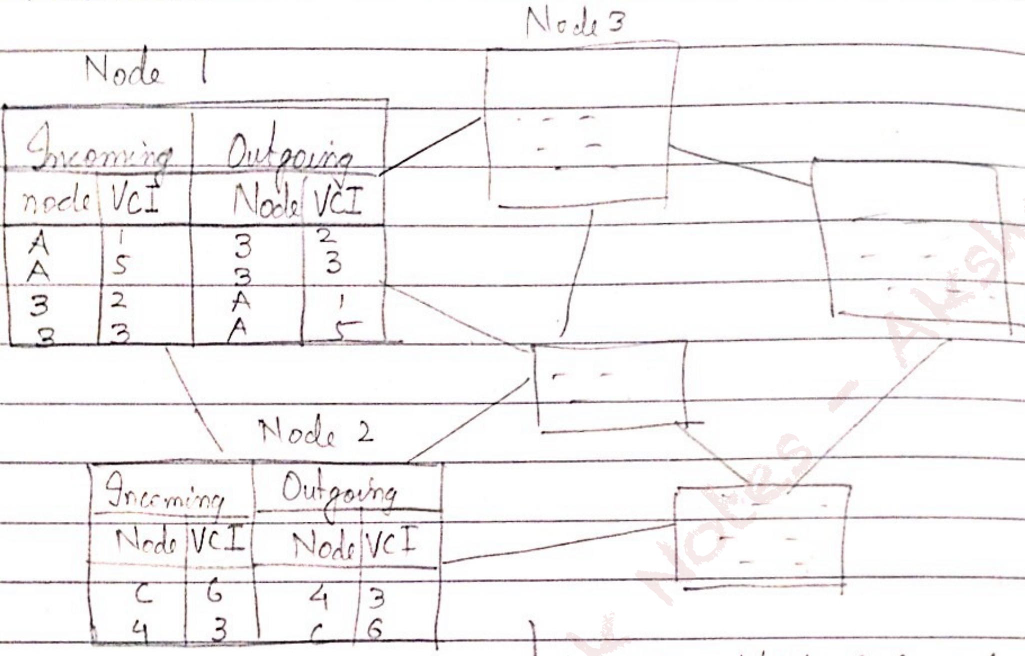
Solutions Usually done through Optimiz^n

1) Creating Routing tables
Create routing table at every node. A ready reckoner. ∴ See the problem points & design the table

* Routing Algorithm Requirements.
  - Responsiveness to changes.
  - Optimality
  - Robustness.
  - Simplicity.

self : TB for details

☆ Routing Tables in Virtual Circuit Packet Networks.



Node 1

Incoming		Outgoing	
node	VCI	Node	VCI
A	1	3	2
A	5	3	3
3	2	A	1
3	3	A	5

Node 2

Incoming		Outgoing	
Node	VCI	Node	VCI
C	6	4	3
4	3	C	6

↳ VCI : Virtual Circuit Interface

• eg : I am driving on a road. I see traffic ahead. Now,

Option 1 : Keep the same road & delay.

Option 2 : See the total route options that I have, seeing the conditions like lower fuel consumption, faster moving traffic etc. (i.e, Analysing Routing Table)

Any route I choose is based on my priority (like, I wont choose a longer & faster route if my fuel tank is nearly empty).

Self Reading * Traveling salesman problem

Puffin
Date
Page

✱ Hierarchical Addresses and Routing

↳ Prioritize your customers based on the plan they have chosen

* Specialized Routing

(1) Flooding

↳ i.e, I don't bother if everyone is getting packets I have just flooded the channel with packets let it go the way it likes.

→ useful in starting up network

→ useful in propagating info. to all nodes

(2) Deflection Routing

↳ i.e, deflecting from one node to another to transfer to receivers. So, routing tables are used.

✱ Shortest paths & routing

→ selection of path to be used to accomplish a given transfer

geographically shortest, assumed.

• typically, its possible to attach a ~~cost~~ cost or distance to a link connecting two nodes.

↳ So, basically, Routing ⇒ shortest path problem.

• Routing Metrics

↳ means for measuring desirability of path.

• path length = sum of costs or distances.

• Possible metrics

sum of delays along path

rough measure of resources used ← > Hop count, Reliability, Delay, BW, load, Cost

link availability, BER, link & router utiliz^n ← along path

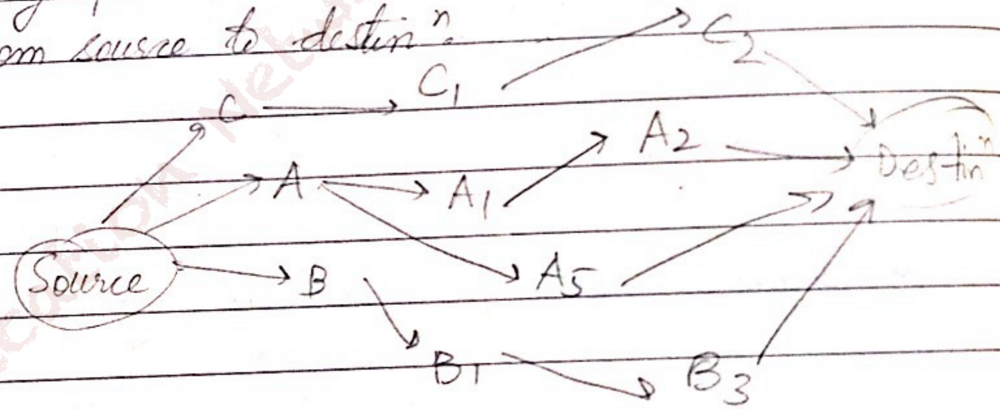
* Shortest Path Approaches

(i) Distance vector protocols.
Neighbors exchange list of distances to destin^{ns}

(ii) Link state protocols.
Info about link is provided to all — Routers (all) have complete topology info.
* Dijkstra (centralized) shortest distance algorithm.

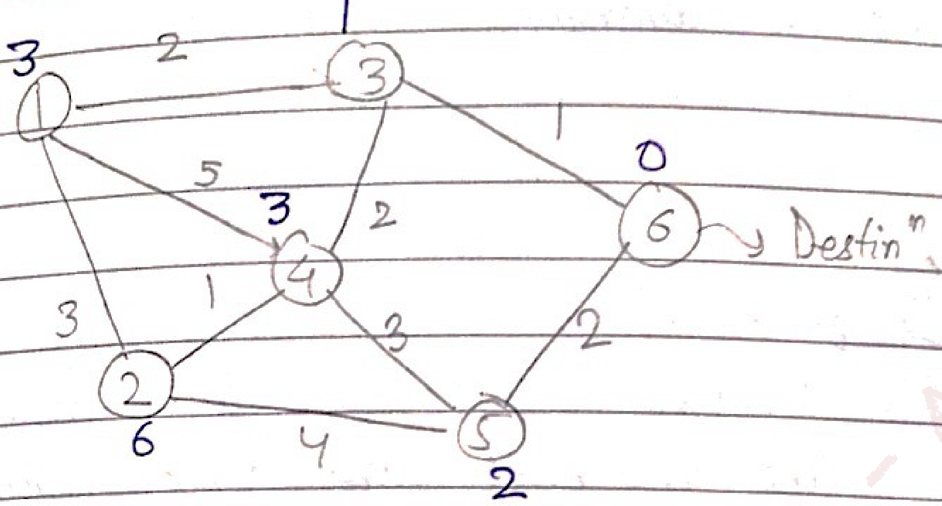
→ If I am going from source to destin^n, while making distance vector, I see for various factors

→ One way of communic^n network is, I go from source to destin^n.



Source doesn't know complete path, only its neighbours. A knows only its neighbours.....

V.Imp. • Bellman-Ford Algorithm
✓ An algorithm kept at each node, for finding neighbours
✓ Used to find shortest path by seeing routing tables



I don't know my neighbours

I don't know my path

Iteration	Node 1	Node 2	Node 3	Node 4	Node 5
Initial	$(-1, \infty)$	$(-1, \infty)$	$(-1, \infty)$	$(-1, \infty)$	$(-1, \infty)$
1	$(-1, \infty)$	$(-1, \infty)$	$(6, 1)$	$(-1, \infty)$	$(6, 2)$
2	$(3, 3)$	$(5, 6)$	$(6, 1)$	$(3, 3)$	$(6, 2)$
3	$(3, 3)$	$(4, 4)$	$(6, 1)$	$(3, 3)$	$(6, 2)$

Type of Q: If any point in the Bellman-Ford Algorithm is changed, find the routing table
If any link is broken, find routing table.

Puffin
Date
Page

* Problems in Communic^n Channels (Networks)
↳ like, broken links, etc.
The info. about it travels slowly
Remedies:
① Split Horizon
✓ Do not report route to a destin^n to the neighbour from which route was learned.
② Poisoned reverse
✓ Report route to a destin^n to the neighbour from which route was learned, but with infinite distance.
✓ Breaks erroneous direct loops immediately
✓ Doesn't work on some indirect loops.
cannot be done always. I can do the remedy at 1-2 places. Not infinite!
* We increase buffer size & set priority

* Link State Algorithm
• Each source node gets a map of all nodes and link metrics (link state) of entire network
• Find shortest path on the map from the source node to all destin^n nodes

* Dijkstra Algorithm: Finding shortest path in order.
• We are evaluating all possible destin^{ns} from a source

Initializ^n ✓ Start with source node s.
Step A ✓ find next closest node i
Step B ✓ Update min. costs.

V.Imp. • Do : Execution of Dijkstra's algorithm

§ ☆ Integrated Services Vision

- how telephone network changed from analog to digital.
- Vision is : ✓ network should be digital end to end.
  ✓ network should support all services : telephone, data, video.

• Internet came in 2000's
* ISDN : Integrated Services Digital Network.
* BISDN : Broadband ISDN.

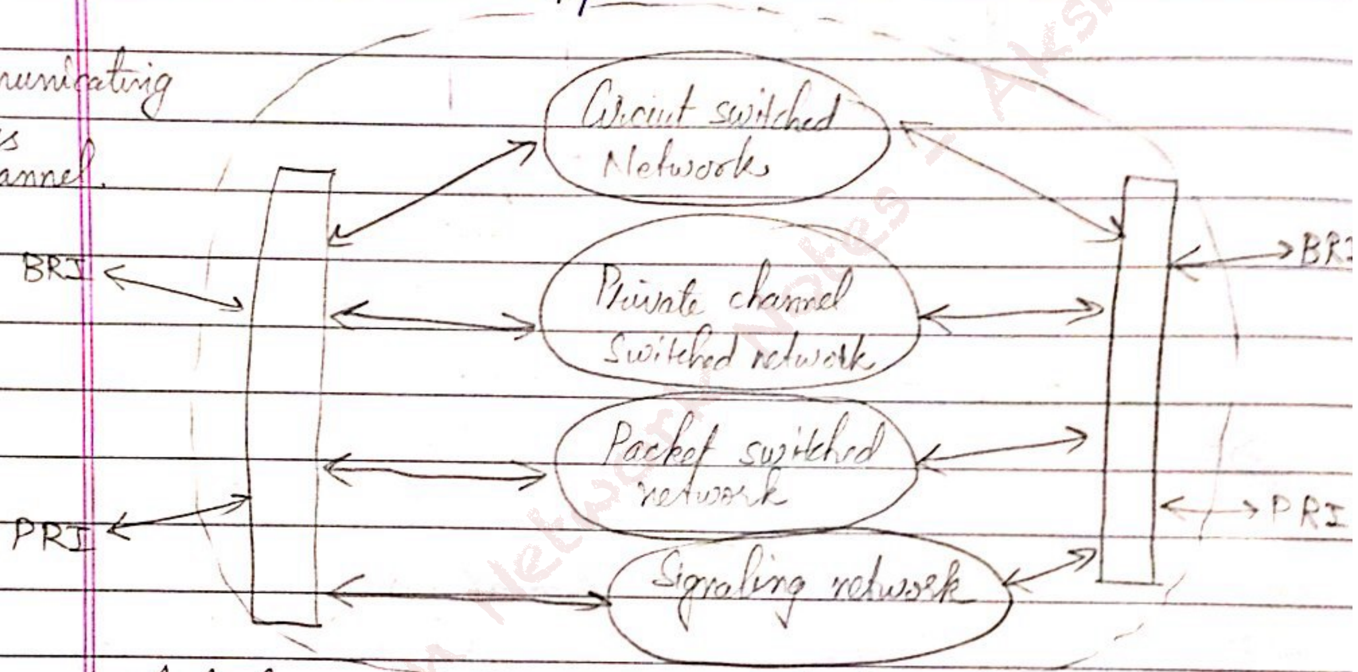
— x —

✲ ISDN

↳ It gave interface to a network.

- BRI : Bit rate interface. (2B + D)
- PRI : Primary rate interface. (23B + D)
- B channel : 64 kbps.
- D channel : 16 kbps

rate at which you'll be communicating across channel.



✲ Interface groups various signals (noise, packets ...) into predefined groups & transfers them through apt paths.

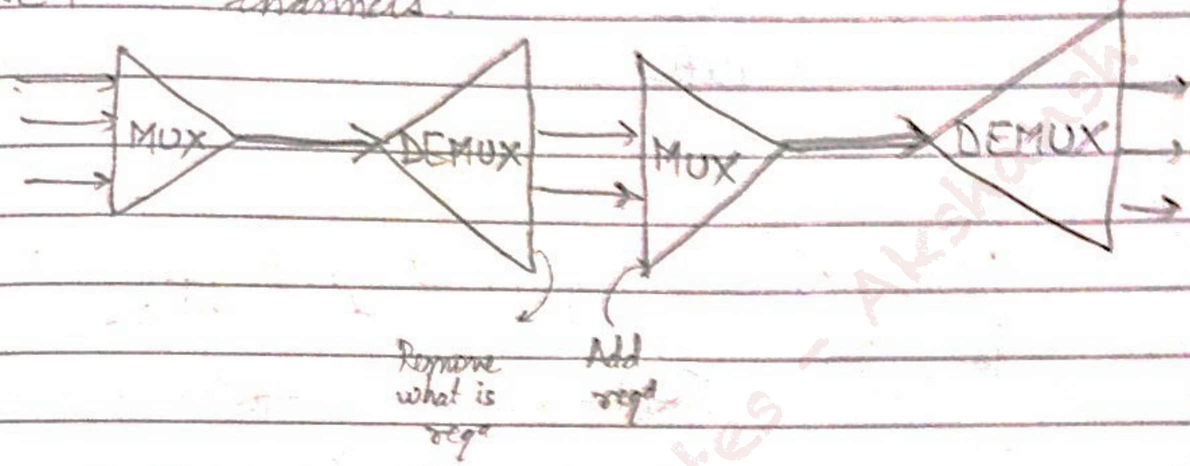
✲ SONET

↳ Synchronous Optical NETwork.

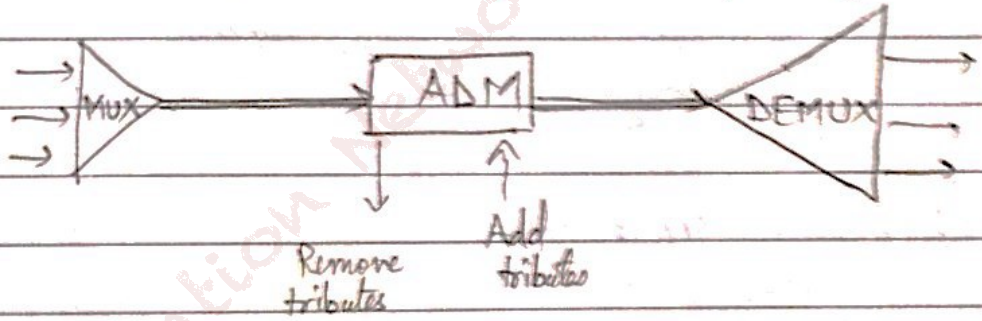
- ISDN had a vision. That needed a large BW.
- This BW is provided by SONET.
- Its a TDM network using Optical fibre
- 8000 frames/sec ($T_{frame} = 125\,\mu sec$)
- SDH (Synchronous Digital Hierarchy).

• ~~Sonet~~ SONET simplifies multiplexing

Not using SONET : Pulse stuffing req^d demultiplexing all channels.



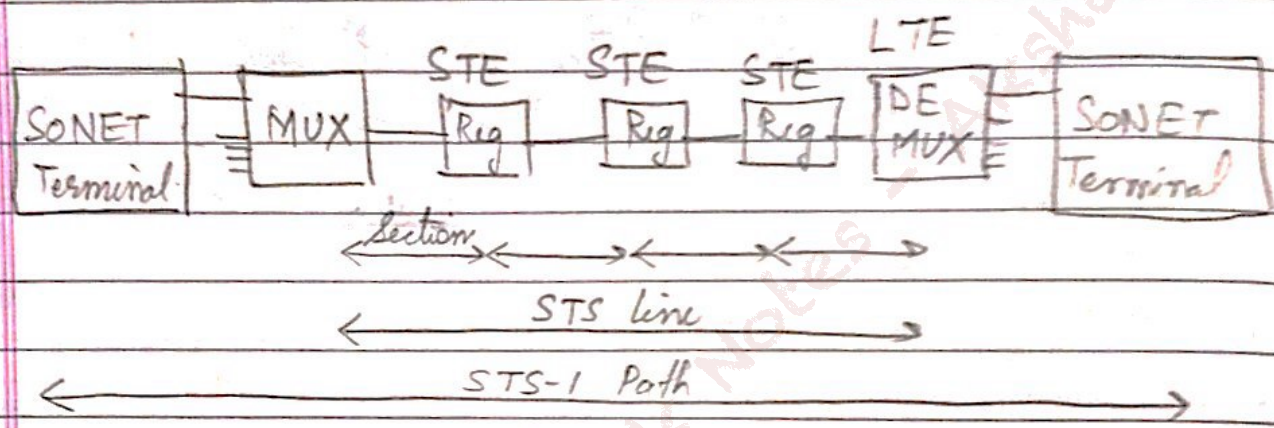
Using SONET : Allows taking individual channels in and out, without full demultiplexing.



• Normal : Only Electrical.
SONET : Electrical & Optical parts are there

• Functionality:
  ✓ Dropping & adding of tributaries can be done at ADMs.
  ✓ Digital signal can be regenerated at regenerators
  ✓ ∃ cross connects to interconnect SONET streams.

- Basic SONET equipments : For Signaling b/w elements :
    - Section Terminating Equipment (STE)
    - line " " (LTE)
    - Path " " (PTE)



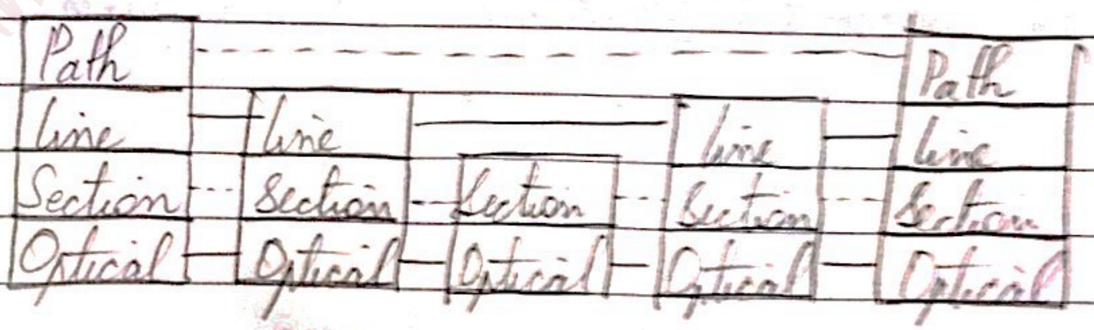
STS : Section terminating equipment (eg: repeaters, regenerators)

PTE : At ends

LTE : In the middle

* Electrical : 7 layers : OSI

✲ OPTICAL NETWORK



* Each layer has its own protocol

LNA : Low Noise Amplifier

- SONET STS Frame :
  ∃ 2 types of overhead :
  ✓ Path Overhead (POH)
  ✓ Transport Overhead (TOH)

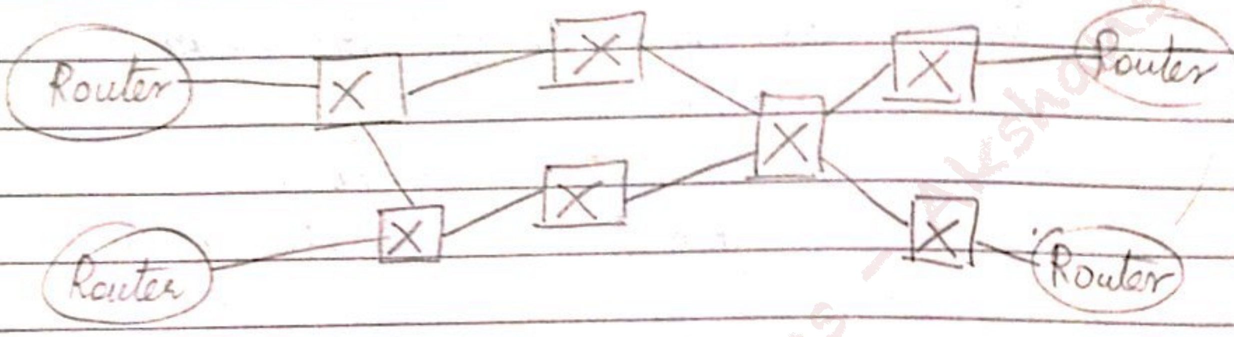
- SONET can have various topologies (eg: Ring
  basically, each topology will have ADM's connected)
  - Multiple options in Ring network (or topology)
    ✓ 2 vs 4 fibre ring network
    ✓ Path vs link protection
    ✓ Unidirectional vs bidirectional transmission.

Read : Topologies of SONET Ring :
  ✓ 2 fibre unidirectional
  ✓ 4 fibre bidirectional switched ring (4-BLSR)

* Only network changes : But : Routing algorithms, Retransmission (ARQ) strategies remain same as before

* Problems with SONET ring network.
  - Managing BW can be complex.
  - Increasing transmission rate in one span affects all equipment in ring.
  - Introducing Wavelength Division Multiplexing (WDM) means stacking SONET ADMs to build parallel rings.
  * Rings have to be interconnected for long distance. Managing one ring is simple, but, managing multiple rings is very complex.

* SONET Mesh Topology using cross connects.
  - Cross connects are nxn switches.
  - More flexible & efficient than rings.
  - Need mesh protection & restoration.



Assignment 5(a): Communication Network Topologies:

DEADLINE (6.5.14 8:25am)
5M: Submit
15M: Test

Identify various ≠ topologies employed in TODAY'S CN. Compare ~~&~~ and contrast the same with conventional topologies employed in CN. like, ring, mesh... networks

What to answer:

(a) Conventional topologies
↳ list, describe, highlight advantages & disadvantages in brief.

(b) Compare & contrast conventional topologies in the form of a table
↳ features that are common ∀ topologies (ring, mesh...), features that distinguish diff^t topologies, applic^ns (where are the topologies used)

(c) Today's topologies
↳ talk just like in parts (a) & (b).

Puffin
Date
Page

* WDM: Predominantly used in clouds these days.

SONET	WDM.
• combines multiple SPEs into high speed digital stream.	• Combines multiple wavelengths into common fibre
• SPE paths between clients from logical topology.	• All optical backbone networks will provide end to end wavelength connections
• High reliability through protection switching.	• Protection schemes for recovering from failures are being developed to provide high reliability in all-optical networks.

* FTTH : Fibre-to-the-Home (Today's technology)
  FTTC : Fibre-to-the-Curve.

* Fibre connection to home provides huge amount of BW, but cost of optical modems still comparatively high.
* Fibre to curve (pedestal) with shorter distance from pedestal to home can provide high speeds over copper pairs.

• Mobile telephone service were introduced in 1940's

* Cellular Communicn
  – Two basic concepts :
    ① Freq. reuse. → region partitioned into cells, each cell covered by base station
    ② Handoff. → procedure to ensure continuity of call as user moves from one cell to another.

- Components of Cellular Networks
  - ACC : Authenticⁿ Centre.
  - BSS : Base station subsys.
  - EIR : Equipment identity register.
  - HLR : home locⁿ register
  - MSC : Mobile switching centre
  - PSTN : Public switching telephone network
  - STP : Signal transfer point
  - VLR : Visitor Location Register

Self Reading

▷ Incoming call to mobile unit.
  - MSC sends call request to all BSS.
  - - - - - -

Other protocols like how connections are done with mobiles, - - - -

* GSM Signalling Standard
  - BTS : Base Transceiver Station.

* Cellular Network Protocol Stack

Just like OSI model, ∃ diffᵗ layers as we go from Mobile Station → Base Tranceiver Station → Base Station Controller → MSC

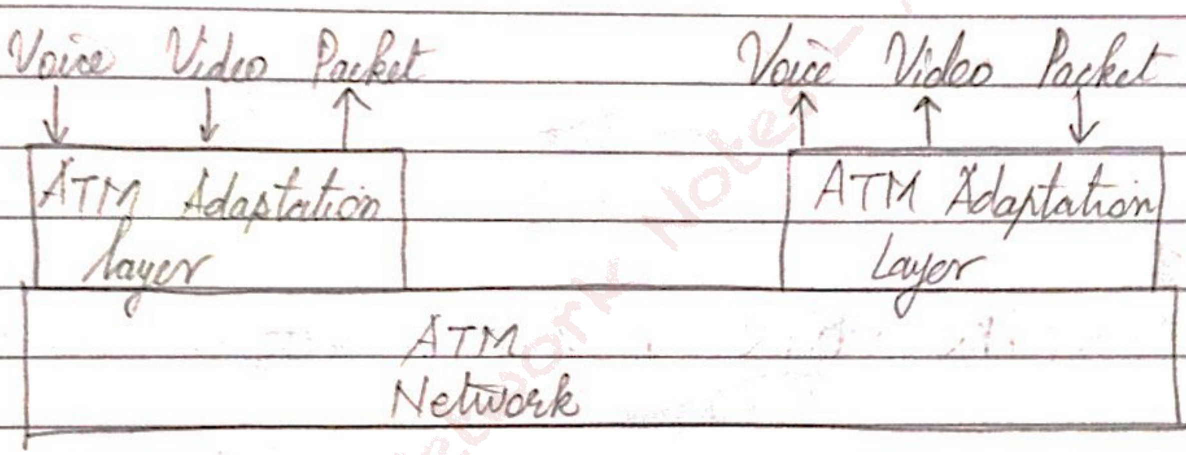
Group Activity

Self : See the linking of layers of OSI model & diffᵗ Networks (like telephone, telegraph, satellite - - -)

* Asynchronous Transfer Mode (ATM) Networks

✓ Packet multiplexing & switching.
✓ Conceived as end-to-end product (than localized)

- As it doesn't depend on clock, so, real time transmissions take place in this
- BW guaranteed



* TDM vs Packet Multiplexing

	Variable bit rate	Delay	Burst traffic	Processing
TDM	Multirate only	low, fixed	Inefficient	Minimal, very high speeds.
Packet	Easily handled	Variable	Efficient	Header & packet processing reqd

* TDM: If I have 4 sources to send, every source is assigned some time of channel. So, if ∃ nothing to send from any source, ∃ wastage of BW. (Synchronous)
* ATM: Time is alloted when data is there (Asynchronous)

* ATM Switching
  Switch carries out table translation & routing

* ATM Virtual Connections
  Virtual connections can be set up across network

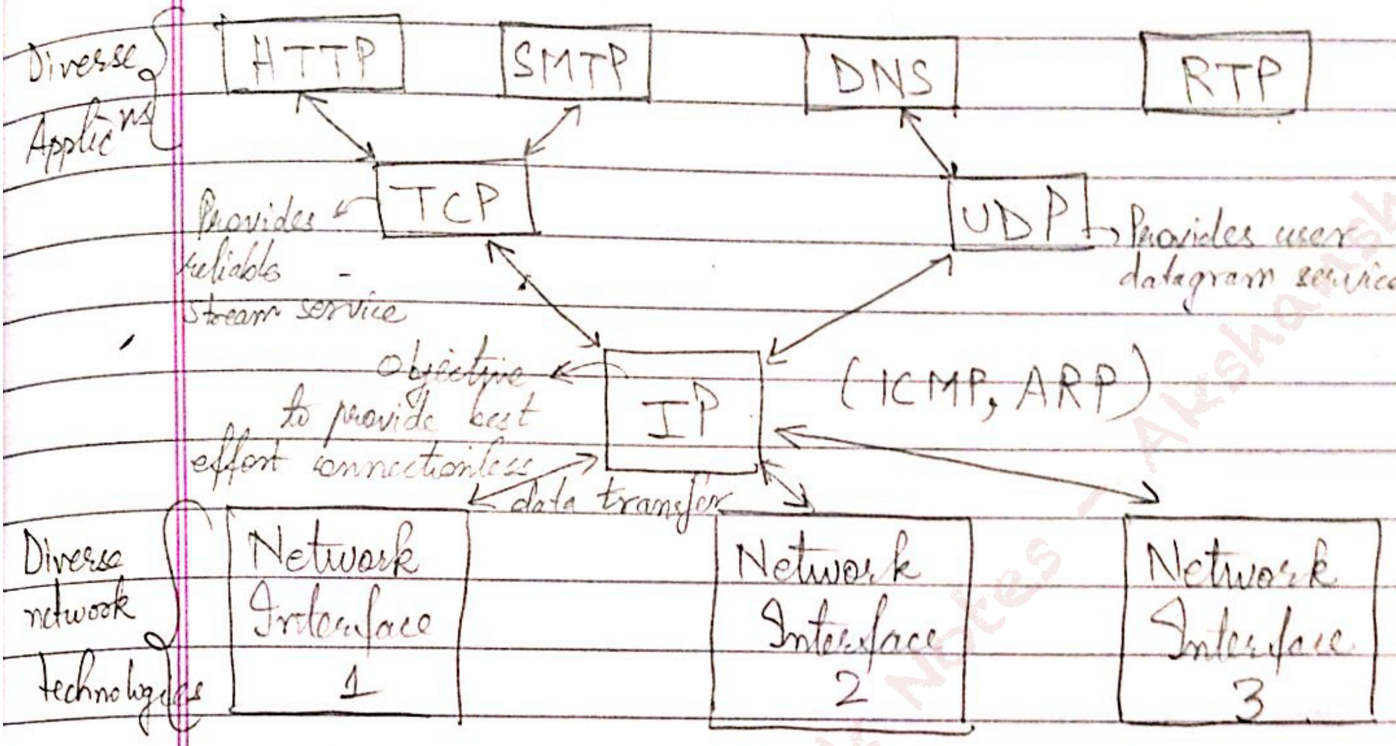
INTERNETWORKING
↳ to build network of networks or internet

* Why internetworking?
  - ✓ To provide universal communicⁿ services (eg: If I have a mac, someone else has windows, both should be able to receive an email in the same way)
  - ✓ Rapid devt. of new applicⁿˢ
    ↳ Emails etc

• Internet Protocol Approach
  • IP packets transfer info. across internet
  HOST A IP → router → router ... → router → HOST B IP

V.V.Imp

* TCP/IP Protocol Suite



This diagram has protocols that operate over TCP. It also shows DNS & real time protocol (RTP) which operate over UDP.

Transport layer protocols TCP & UDP on the other hand, operate over IP.

Many network interfaces are defined to support Internet Protocol (IP)

Note: All higher layer protocols of fig. shown access the network interfaces through IP. This feature provides capability to operate over multiple networks

The IP protocol is complemented by add^nal protocols as in fig. (ICMP, ARP) that are reqd to operate on internet.

(Refer: Ch-8 Leon Garcia for details on ICMP, ARP)

The operⁿ of single protocol (IP) over various networks, provides independence ~~for~~ from underlined network techs. While communicⁿ services TCP & UDP provide a network independent platform on which applicⁿˢ can be developed; thus, allowing multiple network technologies to co-exist, internet is capable to provide bit connectivity & to achieve enormous economies of scale

---

* Internet name & Addresses.
  - eg → IP name = f2011300@dubai.bits-pilani.ac.in
    IP address = 128.100.12.3 (32 bit)
  - → has rules to write it.
    * ∃ network id (netid) & (hostid)

* Physical Addresses
  - → eg: Ethernet uses 48 bits
    1st 24 : manufacturer
    next 24 : Serial no

$$\underbrace{00:97:28}_{\text{Intel}}:\underbrace{12:34:56}_{\text{serial no.}}$$

* Encapsulⁿ

the lects

IP header	IP Payload

↓

Ethernet header	IP header	IP Payload	FCS

* Self : Role of NIC in delivering packets from source to destin" (deliver across workstations, servers & PCs)

* Imp : Do Back Questions : Leon Garcia

IEEE 802 STANDARDS.

Standard	Deals with
802.1	Bridging & Management
802.2	Logical Link Control
802.3	Ethernet – CSMA/CD Access Method
802.4	Token ~~Ring~~ Passing/Bus Access Method
802.5	~~Distributed~~ Token Ring Access Method
802.6	Distributed Queue Dual Bus Access Method
802.7	Broadband LAN
802.8	Fibre Optic.
802.9	Integrated Services LAN
802.10	Security
* 802.11	Wireless LAN
802.12	Demand Priority Access
802.14	Medium Access Control
802.15	Wireless Personal Area Networks
802.16	Broadband wireless metro area networks
802.17	Resistent Packet Ring

* IEEE 802 standards works till layer 2 (OSI)
802.2 : Layer 2 (Logical link Control, LLC)
802.3, 802.4, 802.5, 802.11 : Layer 1
- Seeing what this standard consists of

§ * IEEE 802.11 Wireless LAN
- ✓ Stimulated by availability of unlicensed spectrum
- ✓ Operational freq. is fixed
  - Industrial : 902-928 MHz
  - Scientific : 2.4-2.4835 GHz
  - Medical : 5.725-5.85 GHz

  ISM bands
- ✓ They operate @ 20 Mbps speed

- Definitions:
  1) Basic Service Set (BSS): Group of stations that coordinate their access using a given instant of MAC
  2) Extended Service Set (ESS)
     - ✓ Based on Access Points
     - ✓ Used today

- Infrastructure
  Servers, gateways & base Stations are connected in a network (internet)
  * Stations within BSS can communicate directly to each other.

Application questions from here

- Infrastructure Services
  - ✓ Association, reassociation, disassociation, authentication with access point

- MAC sublayer responsibilities.
- MAC security service & options.
- MAC management services.

- Contention service:
  When two people ask access for channel. Who should be given?
  This is contention.
- MAC Services.
  ✓ MAC can alternate b/w Contention Periods (CPs) & Contention Free Periods (CFPs)
  ↳ Contention free Service: time slots are allotted to each station so that noone comes simultaneously.

- Distributed Coordinⁿ Fⁿ (DCF)
- How carrier sensing is done in 802.11.
  ↳ Physical & Virtual.
- Collisions, Losses & Errors.
  ↳ Collision avoidance.
  ↳ Receiving stations of error free frames send ACK

✩ Frame Types
↳ Management frame
↳ Data frame
↳ ACK frame.

☆ Even within 802.11, the frequency band, bit rate & modulⁿ schemes differ for 802.11, 802.11b, 802.11g, 802.11a ...

☆

Q. LAN DESIGN.

Consider an "open concept" office where 64 carrels are organized in an 8×8 square array of 3m × 3m space per carrel with 2m alley b/w office rows. Suppose a conduit runs in the floor below each alley & provides a wiring for a LAN to each carrel.

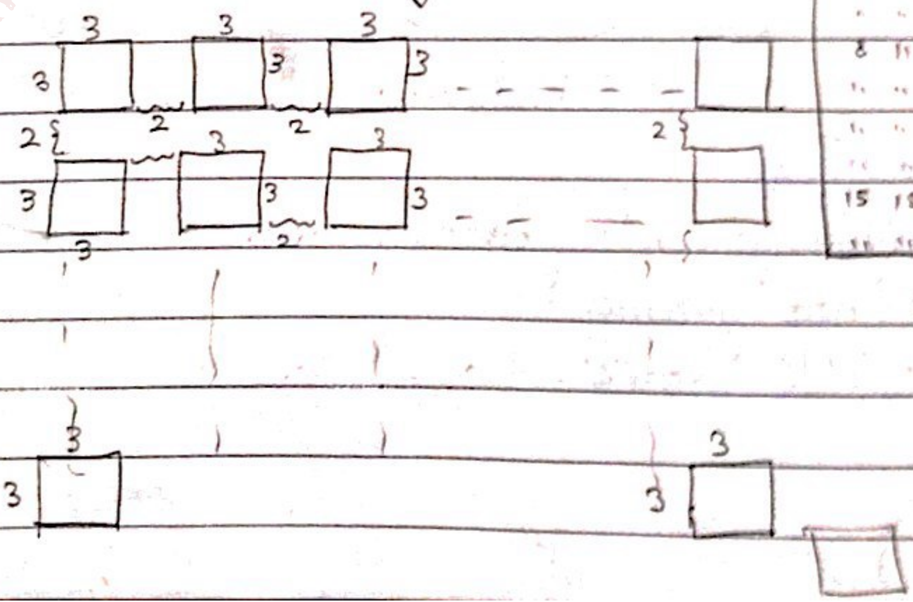
(1) Estimate the distance from each carrel to a wiring closet situated at the side of the square office so that distance are minimized.

(2) Does it matter whether LAN is token ring or Ethernet? Explain.

(3) Discuss the merits of using wireless LAN in this setting.

Ans :-

15	18	21	24	27	30	33	36
"	"	"	"	"	"	"	"
8	11	14	17	20	23	26	29
"	"	"	"	"	"	"	"
"	"	"	"	"	"	"	"
"	"	"	"	"	"	"	"
15	18	21	24	27	30	33	36
"	"	"	"	"	"	"	"



to wiring room

(2) Distances are < 100 m, so, Ethernet is fine. Token ring LAN requires a wire to & from the wiring closet for each station. Length of the ring is then the sum of twice the sum of all distances from stations to the closet: 2816 m. It's clear that token ring better

(3) Wireless LAN will obviously save a lot of wiring.

✓ Overall cost reduced.

✓ However, BW will be shared b/w users. So, slower speed

Sol^n: Switched Ethernet : to reduce amount of BW sharing by introducing separate collision domains

- continued on next page

Extra :

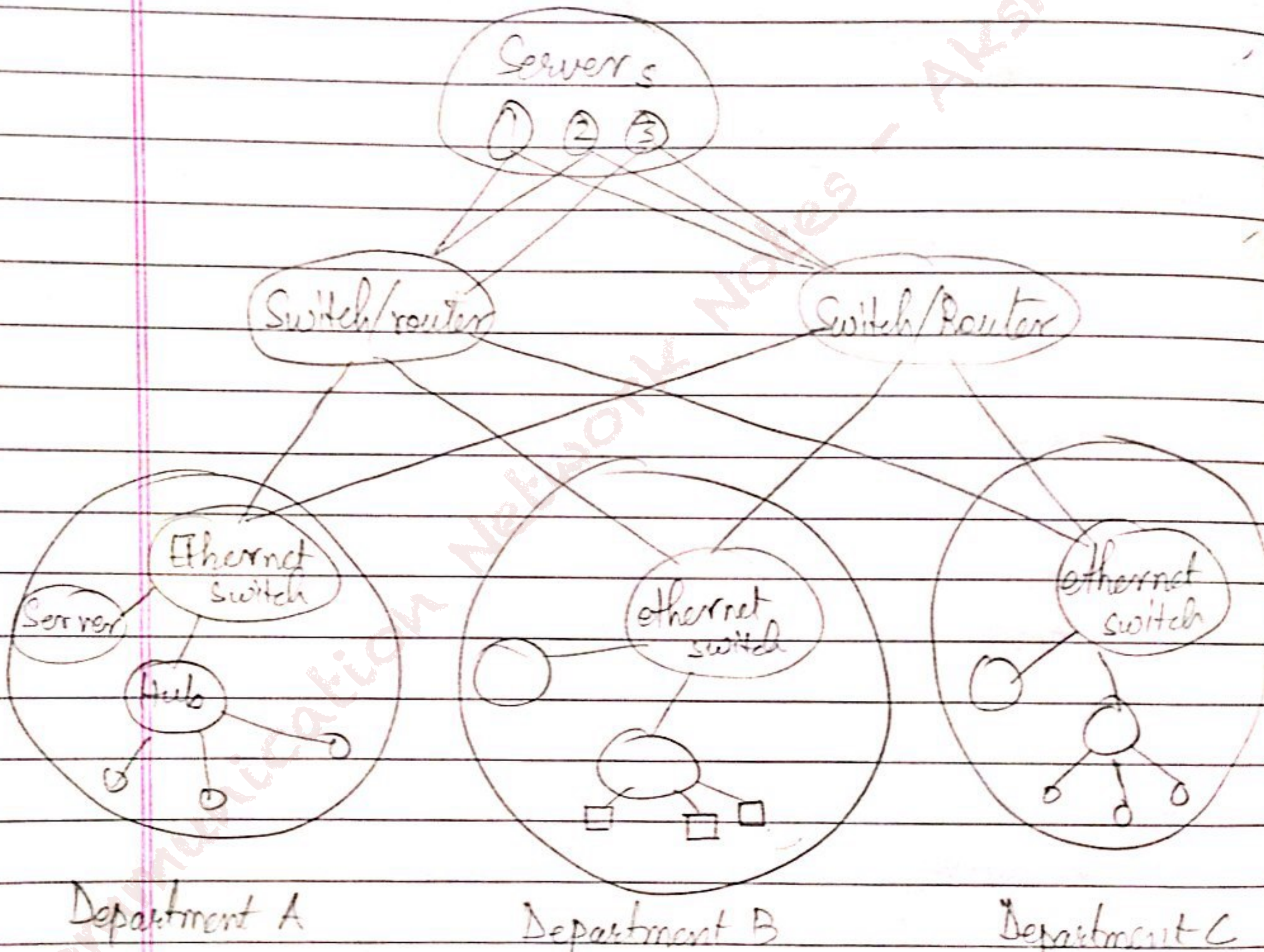
* Advantage of Star Topology :
  Traffic from those heavily used computers can be separated from the rest or dispersed throughout for a more even flow of traffic.
* Why terminate bus topology?
  All buses implemented using coaxial cable & ends of cable must be terminated with terminating resistor that matches impedance of cable. Terminating resistor prevents data reflections from coming at the end.

... LAN Design continued

ch-6 Leon Garcia → ethernet standards

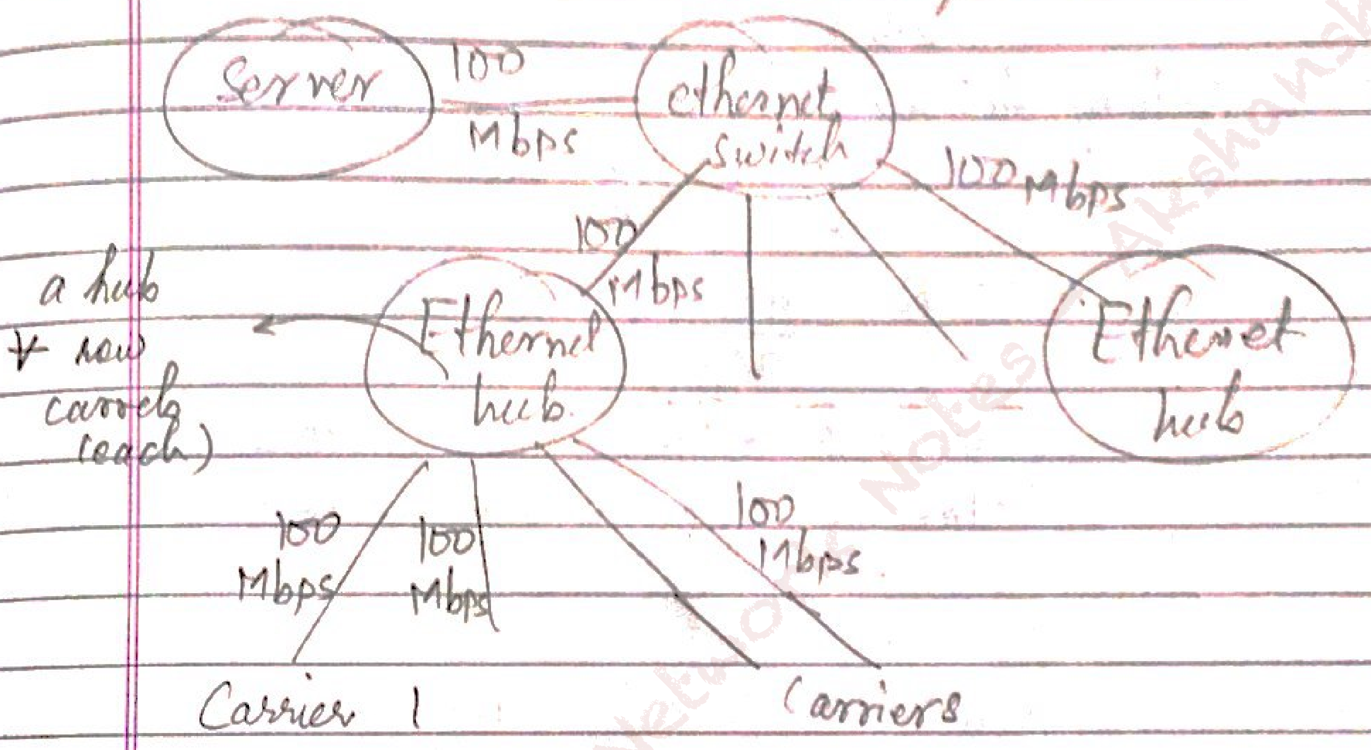
(4) Can the requirements of one row of carrels be met by 10 Mbps ethernet hub? By a 10 Mbps ethernet switch

(5) Can the requirements of office be met by hierarchical arrangement of ethernet switches or as shown?



Ans 4) Bit requirement for each worker is 33.6 Mbps (1 Mbyte file retrieved in 250 ms)
Max rates of both Ethernet hub and switch are both 10Mbps. At 10Mbps, file transfer would take 800 ms.

For the arrangement shown below, 8 users in a row share 100 Mbps
3 users can download a file, still giving 250 ms file download requirement



★ FDDI : Fibre Distributed Data Interface.

- → uses token ring protocol.
- → gives 100 Mbps on optical fibre.
- → Can cover upto 200 km diameter, 500 stations
- → max frame size that can travel = 40,000 bits
- → 500 stations @ 200 km gives ring latency of 105,000 bits.
- → can operate in multi-token mode

Imp. for Numericals

time reqd for signal to propagate once around the ring

* Ring Latency & Ring Reinsertion

- M stations
- b : bit delay at each station
  B = 2.5 bits (using Manchester coding)
- Ring Latency :
  - $\tau' = \frac{d}{V} + \frac{Mb}{R}$ sec.
  - $\tau' R = \frac{dR}{V} + Mb$ bits.

eg : Case ① :- R = 4 Mbps, M = 20, 100 m separation,

- Latency = $\frac{20 \times 100 \times 4 \times 10^6}{2 \times 10^8} + 20 \times 2.5$

= 90 bits

Case ② : R = 16 Mbps, M = 80

- Latency = 840 bits

* Total ring latency (in bits) i.e $\tau' R$ =

No. of stations (M) × Latency (bit delay) for each station (b)

+

$\frac{\text{length of ring (d)}}{\text{speed of transmission (V)}}$ × Rate of data transmission (R)

So,

$$\tau' R = Mb + \left(\frac{d}{V}\right) R \text{ bits}$$

eg : For FDDI, M = 500 (approx), b = 10 bits, R = 100 Mbps, d = 200 km, V = 2×10^8 m/s

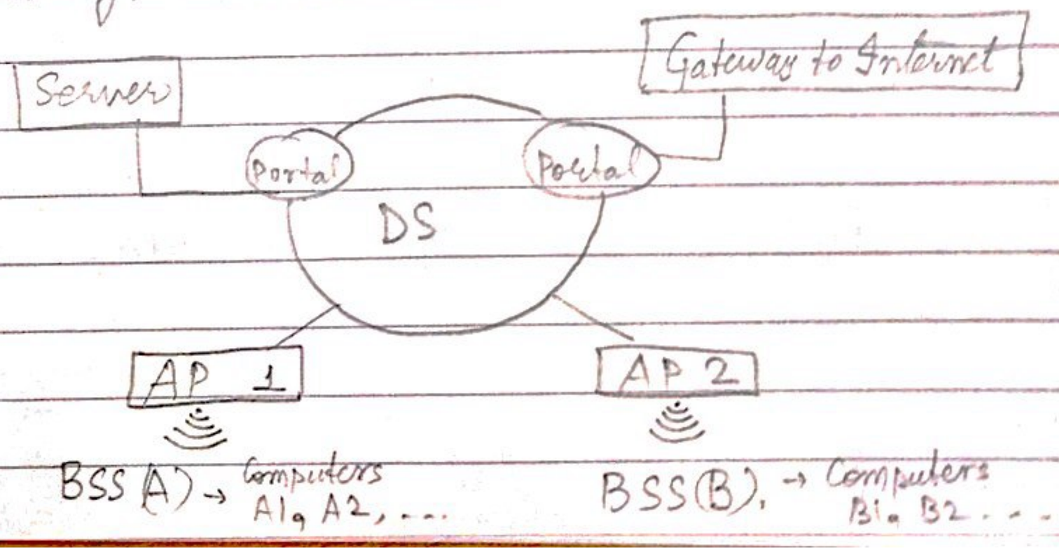
$\Rightarrow \tau' R = (500 \times 10) + \left(\frac{200 \times 10^3}{2 \times 10^8}\right) \times 100 \times 10^6 = 105,000$ bits

✲ IEEE 802.11 Standard → WIRELESS LAN.

This standard talks about a system of wireless local area network. This network consists of :

- Basic service set (BSS) which is basically a group of stations located closely and are interconnected.
- Access Points (AP.) : A BSS has one access point that it uses to link to other access points in the network.
- Distribution System (DS) : A system to link multiple access points. When BSS is connected with DS, it forms Extended Service Set (ESS).
- Portal : These are the gateways for making an interconnection, i.e, allowing outside servers, wireless networks to connect to the DS.

This standard says that if any control goes from one part to another, in a network, then how will the bits be arranged, and what will each bit stand for.



Puffin
Date
Page

* Frames supported by IEEE 802.11 Standard

1) Management frames :
Frames that convey how the user (station) stays connected to a DS; like, moving from one AP to another, so, association & dissassociation with the APs. Also, synchroniz^n, authentic^n & deauthentic^n in case the stations moves across secure APs.

2) Control frames :
Frames that convey whether data was received or not; like talking about +ve & -ve acknowledgements.

3) Data frames :
Frames that contain the data to be transmitted.

* Frame Structure

Frame

MAC header	Frame Body	CRC

✓ Frame Control : Protocol version, Type of frame, Subtype of frame, To DS, From DS, Power management, WEP...

To DS	From DS	Result
0	0	Transfer within BSS
1	1	Internet (b/w APs)

✓ Duration
✓ Part to store source & destin^n addresses
✓ Sequence control.

- Protocol : A step by step procedure that an engineer at an exchange follows to fulfil any communicⁿ.

Q. What is a protocol analyser ?
Gives a way to compare various features of a protocol
- time taken to connect
- reliability
- delays involved
- in case of traffic, do the protocols have alternatives to punch through a connection

Q. • What are the protocols it supports ?

Q. Analyse these protocols from the textbook & see its applications

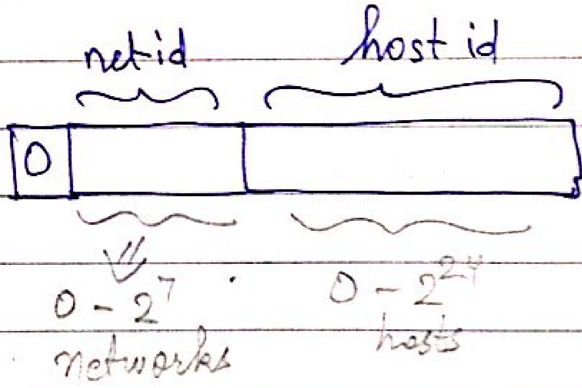
Q. Conclude :- how do you use a protocol analyser ?
- What are the aspects of a protocol can be analysed using protocol analyser ?

☆ TTL : Time to live : No. of hops that your packet is going to take

☆ Types of IP Addresses :

Class A :

1st bit 0.
Next 7 bits Netid
Last 24 bits Hostid



⇒ IP addresses from 1.0.0.0 to 127.255.255.255

Class B :-

1st bit 1
2nd bit 0
Next 14 bits netid
Last 16 bits host id

		netid	host id
1	0		
		$0-2^{14}$ networks	$0-2^{16}$ hosts

Class C :-

			22 netid bits	8 bits for host id.
1	1	0		

Class D :-

↳ for Multicast addressing: i.e, sending to a group. 28 bits of multicast address

				28 bits of multicast address
1	1	1	0	

Class E :

1	1	1	1	28 bits of ~~multicast~~ reserved bits

eg Identify the address class of following IP Address :
(a) 200. 58. 20. 165
(b) 128. 167. 23. 20
(c) 16. 196. 128. 50
(d) 50. 156. 10. 10
(e) 250. 10. 24. 96

Idea : Convert each number to binary equivalent (Resolving for 8 bits) & then match with the classes.

(a) 200 = 11001000
58 = 00111010
20 = 00010100
165 = 10100101 ⇒ Class C

⇒ 200.58.20.165 = 11001000.00111010.00010100.10100101

(b) 128.167.23.20
= 1000000 ⇒ Class B

(c) 16.196.128.50 = Class A
(d) 50.156.10.10 = Class A
(e) 250.10.24.96 = Class E

11111010
2+8+16+32+64+128

☆ UDP : User Datagram Protocol.
↳ ∃ no handshake. ⇒ ~~Only~~ acknowledgements are not taken or sent.

* Basically, we cannot keep track if reception went well

§

TCP

✓ full duplex connection b/w applicⁿ layer
→ goes both ways.

✓ Provides flow control
→ telling that buffer is full, & send remaining later

* TCP Segment format
↳ Contains sequence no. : if a message was divided in 4 parts, so, rearranging at receiver side. So, sequence no. is used

* PORT Numbers :
Well known ports :
HTTP : 80
DNS : 23

* 3-way hand shake in TCP. → If both A & B can send data

* WITH DIFFERENT SEQUENCE NO.

Host A | Host B

Can I start connection? → SYN
alongwith sequence no. → x

request acknowledged → x+1
start & send me sequence no. → y
because B wants to send data, if at all

Okay. Your sequence y's request acknowledged & now I'm sending my data. Take it

Date
Page

WITH SAME SEQUENCE NO → If only A can send data

eg. Host B advertizes : window size of 2048 & sequence no. of 2000.

Host A | Host B

till 2000, all other packets have been acknowledged

Seq. no = 1, Ack no = 2000.
WIN = 2048,

So, after 2000, 2048 more bytes can be taken

Seq no = 2000, Ack = 1, Win = 1024
Data sent = 2000 – 3023

We have sent till 2000. & A wants to send 1024
So, 2000 + 1023 = 3023

Seq = 3024, Ack = 1, Win = 1024
Data = 3024 – 4047

B said to A : You can send me 2048.
So, A sends 1024, twice

Seq = 1, Ack = 4048, Win = 512
Data = 1 – 128 → Data transmitted back to B.

→ B had to acknowledge 4047's data that it got. It also had to send its own data. So, it does that in one step

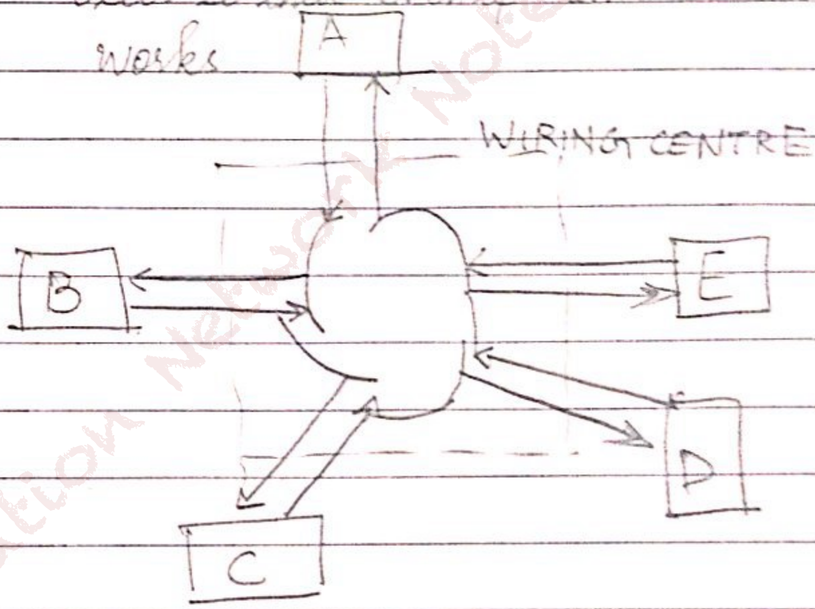
Seq = 4048, Ack = 129, Win = 1024
Data = 4048 – 4559

✯ IEEE 802.5 RING LAN

- Unidirectional Ring Network.
- Uses Differential Manchester line coding.

* Star Topology Ring LAN

↳ better than simple Ring LAN
Used so that even if link breaks, circuit works



end of course.