

A decorative banner featuring five large, bold, red letters arranged horizontally: 'I', 'N', 'D', 'E', and 'X'. Each letter is enclosed in a white square frame with a thin black border. The letters are slightly tilted, giving them a dynamic appearance.

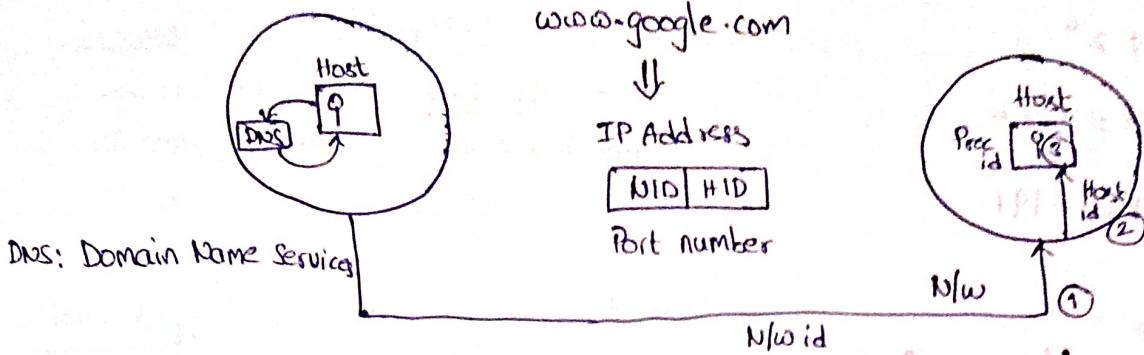
NAME: K. Growtham

STD.: _____ SEC.: _____ ROLL NO.: _____ SUB.: _____

02/12/20

Computer Networks

Introduction:



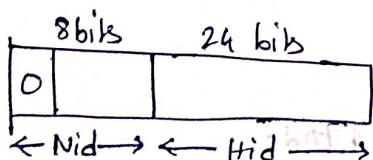
- Port number is used to identify a process in host. Well known services have predefined fixed port numbers.
- The overhead involved in translating domain name to IP address using DNS is called DNS overhead.

Class full IP address classification:

Class A: 0 ____ ; Class B: 10 ____ ; Class C: 110 ____ ; Class D: 1110 ____ ; Class E: 1111 ____

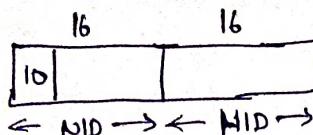
- Size of IPv4 address is 32 bits.
- It is represented using dotted decimal notation where every 8 bit part of the address is represented in decimal form.

Class A:



- No of networks = ~~128~~ 126
- No of IP add per n/w = 2^{24}
- No of hosts per n/w = $2^{24} - 2$
- Out of 128 N/w's we don't use starting ~~all 0's~~ (all 0's) and last N/w (all 1's) ⇒ Range: 1. ____ to 126. ____

Class B:



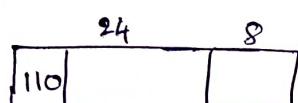
→ NWs: 2^{14}

→ IP/NW: 2^{16}

→ Hosts/NW = $2^{16} - 2$

→ Range: 128 - 191

Class C:



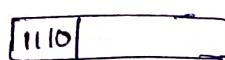
→ NWs: 2^2

→ IP/NW: 2^8

→ Hosts/NW: $2^8 - 2$

→ Range: 192 - 223

Class D:



→ Here addresses are not classified into Nid and Hid.

→ This class of addresses are used for multicasting.

→ No of addresses: 2^{28}

→ This class is a drawback because 2^{28} is a large number for multicasting.

Range: 224 - 239

Class E:



→ These are also not divided into Nid and Hid.

→ These are reserved for special purpose.

→ Range: 240 - 255.

Note: Under every NW, 1st & last IP addresses are used for special purposes (i.e., for network id & ~~multicasting~~ Direct broadcast)

Caching:

unicast: sending packet from one host to other host

broadcast: sending a packet from one host to many hosts.

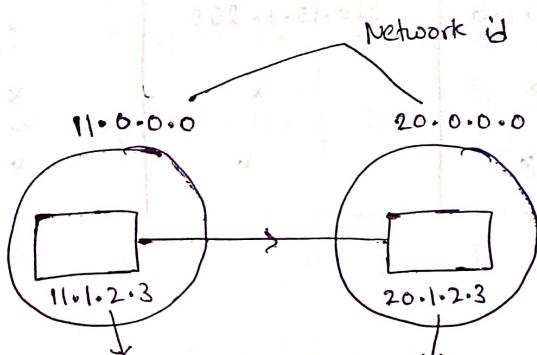
Limited

- send pkt to all hosts of same N/w

Directed

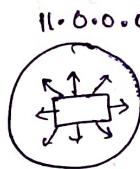
- send pkt to all host of diff N/w

Unicasting:



Source Add Destination add spread the transmission
no of hosts less SH DA destination address must be 0000...0
all hosts equal packet to receive msg at different

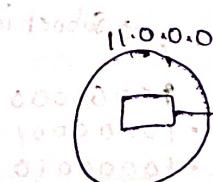
Limited Broadcasting:



Source Add: 11.1.2.3

Destination Add: 255.255.255.255

Directed Broadcast:



Source Add: 11.1.2.3

Destination Add: 20.255.255.255

Note:

Limited broadcast add: 255.255.255.255

Directed broadcast add: Nid.1111....1

Network id: Nid.000...0

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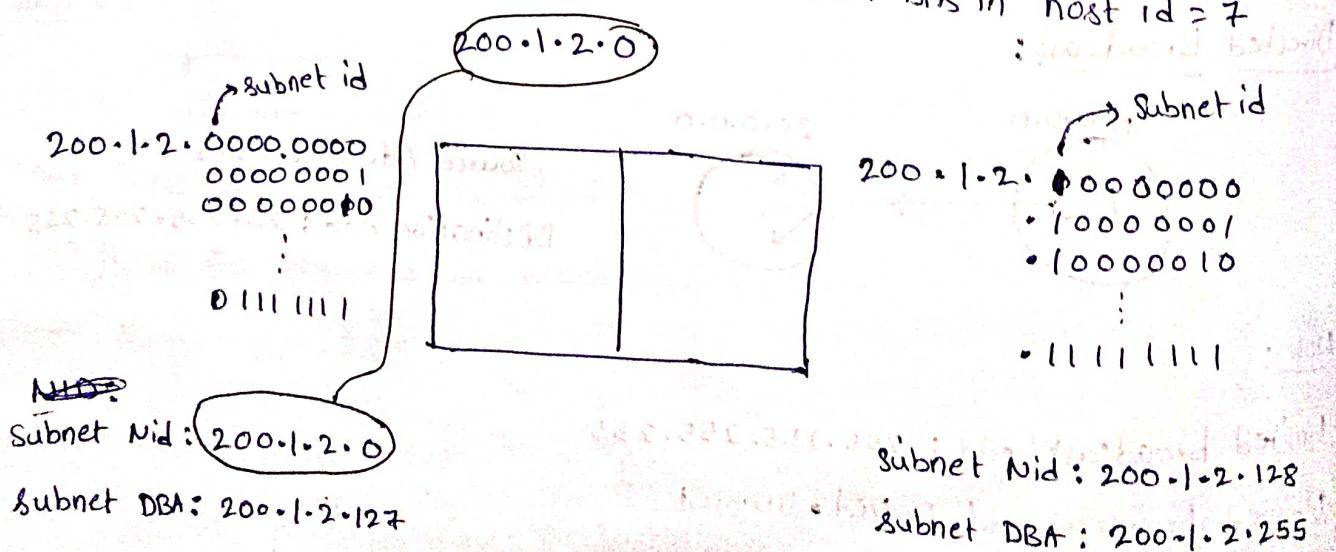
IP Add	Class	Nid	DBA	LBA
1.2.3.4	A	1.0.0.0	1.255.255.255	2.55.255.255.255
10.15.20.60	B A	10.0.0.0	10.255.255.255	"
130.1.2.3	B	130.1.0.0	130.1.255.255	"
150.0.150.150	B	150.0.0.0	150.0.255.255	"
200.1.10.100	C	200.1.10.0	200.1.10.255	"
220.15.1.10	C	220.15.1.0	220.15.1.255	"
250.0.1.2	E	x	x	x
530.1.2.3	invalid	x	x	x

Subnets:

- Maintenance of large Nws is difficult and also within a Nw we need small subnetworks (for industrial needs).
- Subnetting is process of dividing large Nw into smaller networks.
- But subnetting incurs more time in identifying host.

Eg: Consider class C Nw 200.1.2.0

Now lets divide it into 2 subnets \Rightarrow no of bits in subnet id = 1
no of bits in host id = 7



→ Now there is a confusion that whether

200.1.2.0 is Nid or subnet Nid

200.1.2.255 is DBA of subnet or DBA of whole N/w.

→

Sender	Address	Interpretation
within N/w	200.1.2.0	subnet N/w id
outside	200.1.2.0	N/w id
inside	200.1.2.255	DBA of subnet
outside	200.1.2.255	DBA of N/w

→ Now no of hosts possible over this organization is

$$(128-2) + (128-2) = 252$$

→ Now consider dividing the same network into 8 subnets.

⇒ no of bits in subnet id: 2

no of bits in host id: 6

S1	S2
200.1.2.0 to 200.1.2.63	200.1.2.64 to 200.1.2.127
200.1.2.128 to 200.1.2.191	200.1.2.192 to 200.1.2.255

S3 S4

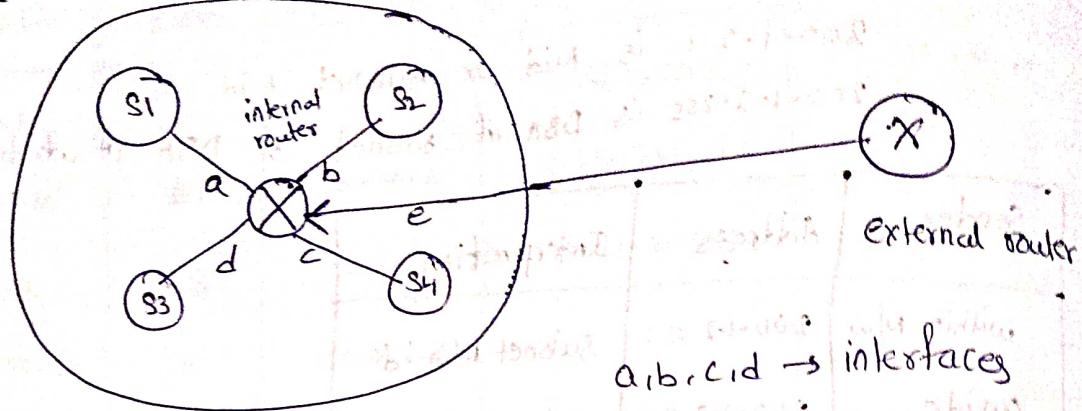
no of hosts possible: $256 - (2 \times \text{no of subnets})$

$$= 256 - (8) = 248$$

→ Generally router gets the packets and forwards to the

Correct host and makes correct interpretation (in the case when there is confusion)

Routing :



Subnet mask: It is a 32-bit number, used by router to identify the subnet to which a given address belongs. In subnet mask, there are two parts:

Eg: For current example subnet mask is

Subnet mask is also known as network mask

1111111	1111111	1111111	11000000
---------	---------	---------	----------

$$\cong 255.255.255.192$$

We obtain subnet or network id by performing bitwise AND with given IP address.

For example consider ip add 200.1.2.130

Subnet mask: 11111111 11111111 11111111 11000000

ip add : 11001000 00000001 00000010 10000010

bitwise and : 11001000 0000001 00000010 10000000

i.e., 200.1.2.128 (Subnet id)

→ whenever router gets an address, it performs bit wise 'and' b/w ip add & subnet mask.

After this it finds subnet network id and forwards the packet to the correct interface accordingly. This process is done using routing table.

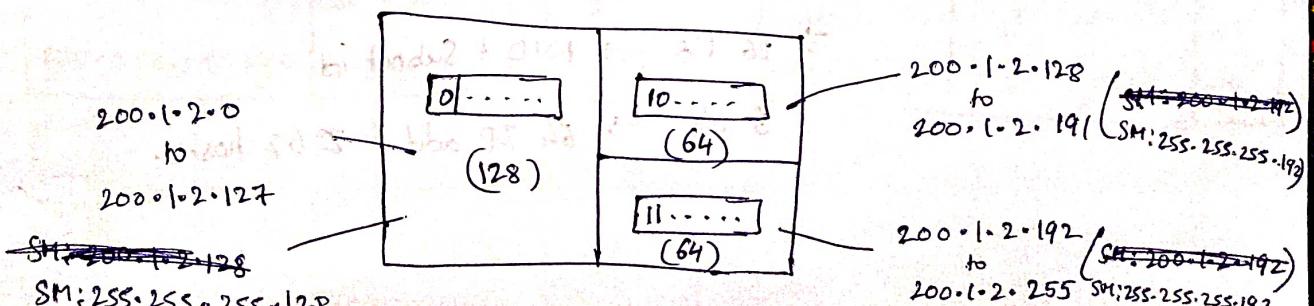
→ Routing table:

Subnet id	Subnet mask	Interface
200.1.2.0	255.255.255.192 255.255.255.192	a
200.1.2.64	255.255.255.192	b
200.1.2.128	255.255.255.192	c
200.1.2.192	255.255.255.192	d
0.0.0.0	0.0.0.0	e

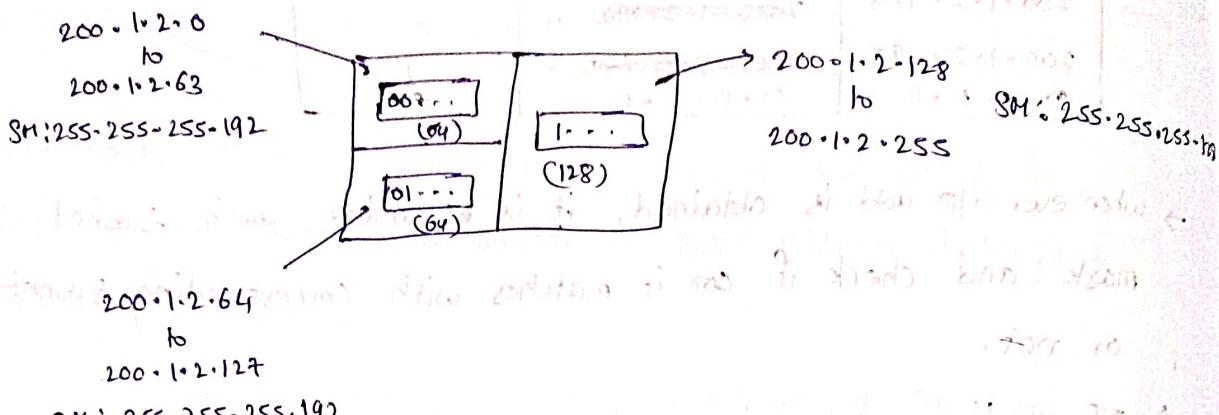
- whenever IP add is obtained, it is ~~compared~~ with subnet mask and check if ~~or~~ it matches with corresponding subnet id or not.
- If all the entries doesn't match, then default entry (0.0.0.0) is used.
- If an address match with more than one pair then choose the pair with longest subnet mask (i.e., the one with more no of 1's)
- when sizes of all the subnets are same, it is called fixed length Subnet Masking (FLSM):
- when sizes of subnets differ, it is called Variable Length Subnet Masking (VLSM)

→ Now consider subnetting of previous example into 3 subnets with 128, 64, 64 IP addresses.

~~zero bits in subnet id = 2~~

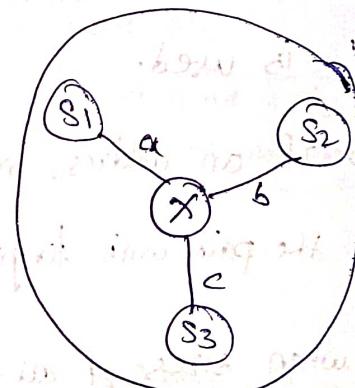


- Subnets of equal sizes will have same value of subnet mask.
- Smaller Larger the size of subnet, smaller the size of subnet mask is.
- Another variant can also be used where ~~larger subnets gets larger values of subnet~~ ~~smaller subnets gets smaller values of subnet~~



Routing table:

Subnet No/ id	SM	Interface
200.1.2.0	255.255.255.128	a
200.1.2.64	255.255.255.192	b
200.1.2.128	255.255.255.192	c
0.0.0.0	0.0.0.0	d



Q3/12/20

- Given a subnet mask we can find no of IP add. in the subnet.

Eg: Consider 255.255.255.192

11111111.11111111.11111111.11000000

⇒ 26 1's → NID + Subnet id

6 0's → 64 IP add & 62 hosts.

1	255
2	255
3	255
4	255
5	255
6	255
7	255
8	255
9	255

If class of IP add is given then we can find subnet id also

Class A:

NID: 8 bits Sid: 18

Class B:

NID: 16 Sid: 10

Class C: NID: 24 Sid: 2

→ Practically MSB bits of NID are only used for subnet id.

But theoretically any bits of NID can be used.

→ If LSB bits are used for subnet id then we can't get proper ranges

Eg: Consider dividing 200.1.2.0 into 2 subnets. If LSB^{bit} is used as subnet id then one subnet will have all even IP addresses and other will have all odd IP addresses.

(Q2)

	Subnet Mask	No of hosts per subnet	Subnets in class A	Subnets in class B	Subnets in class C	Subnet if no of bits in NID > 10
1	255.0.0.0	$2^{24}-2$	1	-	-	-
2	255.128.0.0	$2^{23}-2$	2	-	-	-
3	255.192.0.0	$2^{22}-2$	2 ²	-	-	1
4	255.240.0.0	2^{23} 2^0-2	2^4	-	-	2 ²
5	255.255.0.0	$2^{16}-2$	(2¹⁶ 2 ⁸) diagonal 1st address	-	-	2 ⁶
6	255.255.254.0	2^9-2	2^{15}	-	-	2 ¹³
7	255.255.255.0	2^8-2	2^{16}	2^8	1	2 ¹⁴
8	255.255.255.224	2^5-2	2^{19}	2^{11}	2^3	2 ⁷
9	225.255.255.240	2^4-2	2^{20}	2^{12}	2^4	2 ⁸

1) 255.0.0.0

NID + SID : 8 bits

HID : 24 bits \Rightarrow no. of hosts $= 2^{24-2}$

Class A : NID + SID ≥ 8

8 + SID ≥ 8

SID ≥ 0

i.e., no subnets.

Entire network is one subnet with net掩码

Class B : NID + SID ≥ 16 (not possible)

NID of Class B: 16 bits

Class C: not possible

Class C: not possible

Here also not possible

2) 255.128.0.0

\Rightarrow NID + SID ≥ 9 bits

Class A : $2^{23-2} \Rightarrow 12$ bits for SID

Class B : not possible

Class C : not possible

CIDR : (Classless Inter Domain Routing)

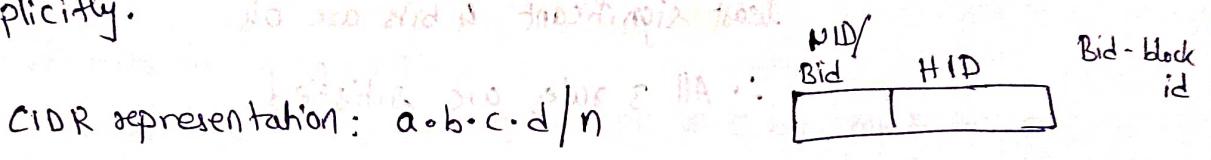
→ The problem with classful classification ~~that~~ is that it's not flexible for assigning IP addresses.

→ So we use ~~can~~ CIDR ~~to~~ technique which doesn't have any ~~any~~ classes.

→ Whenever a set of IP addresses are needed, the available IP address set is ~~all~~ assigned. This set of IP addresses is called block.

→ In classful, given an IP address we can determine ~~no of~~ network id to which the address belongs to (find in class of address) 13

But in CIDR we need to mention no of bits in NID field explicitly.



$n \rightarrow$ no of bits in NID id part

$32-n \rightarrow$ no of bits in host id

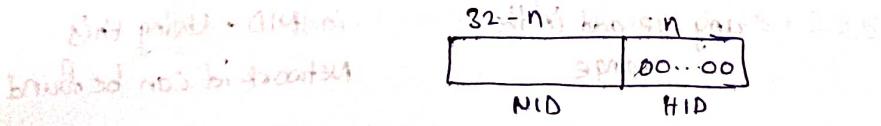
$$\text{no of IP addresses} = 2^{32-n}$$

Rules for CIDR blocks:

- All the IP addresses should be contiguous.
- Block size is always in power of 2.
- First IP address in the block should be divisible by size of the block.

i.e., ~~the~~ starting address should have all 0's in HID.

For example if block size is 2^n then least significant n bits should be all 0's.



This lets IP addresses of block range from $\text{NID} \cdot 00..0$ to $\text{NID} \cdot 11..1$ with 2^n steps.

Eg: Check whether below set of IP addresses form a valid CIDR block or not.

100.1.0.2-32

100.1.1.2-33

100.1.2.47

All the addresses are contiguous

Block size of block = $16 = 2^4$

Given : Starting address : $100 \cdot 1 \cdot 2 \cdot \boxed{00100000}$

least significant 4 bits are 0s.

\therefore All 3 rules are satisfied

\therefore It is a valid CIDR block.

Eg : $20 \cdot 10 \cdot 30 \cdot 32$
 $20 \cdot 10 \cdot 30 \cdot 33$
.....
 $20 \cdot 10 \cdot 30 \cdot 63$

Contiguous

Size = $2^5 = 32$

Starting host id : $\boxed{00100000}$

\therefore valid CIDR block.

Eg : $150 \cdot 10 \cdot 20 \cdot 64$
 $150 \cdot 10 \cdot 20 \cdot 65$
.....
 $150 \cdot 10 \cdot 20 \cdot 127$

Contiguous

Size : $64 : 2^6$

$\boxed{01000000}$

6 bits

\therefore valid CIDR block

This is represented as : $150 \cdot 10 \cdot 20 \cdot 70 / 26$

This can be

any IP add in the range

No of bits

in NID. Using this Network id can be found

→ Here also, the first address of block is used to represent block id or network id.

The last address of the block is used for as directed broadcast address.

Eg: CIDR

→ Find the 7th block of addresses using below CIDR representation.

$$20 \cdot 10 \cdot 30 \cdot 35 / 27$$

Sol:

No of bits in NID = 27

No of bits in host id = 5 \Rightarrow LSR 5 bits must be 0

No of IP add: $2^5 = 32$ for starting add

$$20 \cdot 10 \cdot 30 \cdot [0010000]$$

= 20 · 10 · 30 · 32 is that starting address

Range of addresses is: $20 \cdot 10 \cdot 30 \cdot 32$ to $20 \cdot 10 \cdot 30 \cdot 63$

$$20 \cdot 10 \cdot 30 \cdot 63$$

Eg:

$$100 \cdot 1 \cdot 2 \cdot 35 / 28$$

$$100 \cdot 1 \cdot 2 \cdot [0010001]$$

Range of add: $100 \cdot 1 \cdot 2 \cdot 0010 [0000]$

$$100 \cdot 1 \cdot 2 \cdot 0010 [1111]$$

$$\approx 100 \cdot 1 \cdot 2 \cdot 32$$

$$\approx 100 \cdot 1 \cdot 2 \cdot 47$$

$$\text{Eg: } 100 \cdot 1 \cdot 2 \cdot 35 / 20$$

$$\text{Ans: } 100 \cdot 1 \cdot 0 \cdot 0 \text{ to } 100 \cdot 1 \cdot 15 \cdot 255$$

Subnetting in CIDR:

→ Consider forming 2 subnets from block $20 \cdot 30 \cdot 40 \cdot 10 / 25$

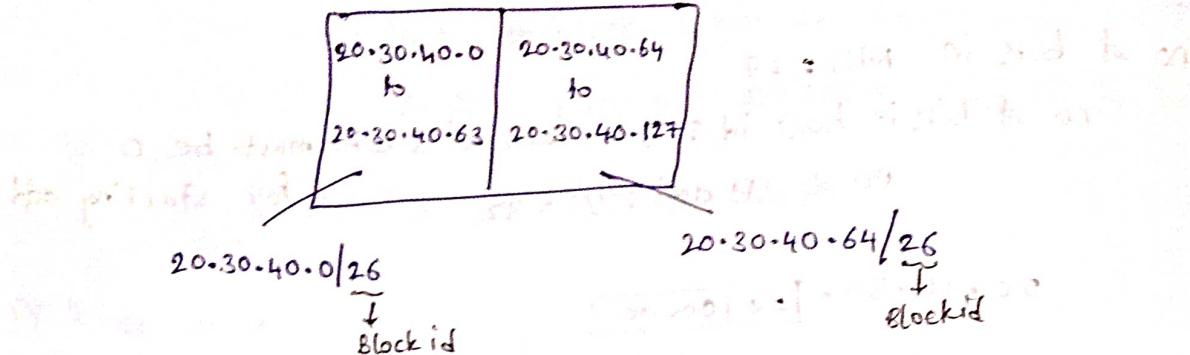
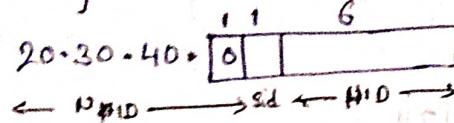
\Rightarrow No of bits in host id = 7

~~$20 \cdot 30 \cdot 40 \cdot [00001010]$~~

$\Rightarrow \text{Net id: } 20 \cdot 30 \cdot 40 \cdot [00000000]$

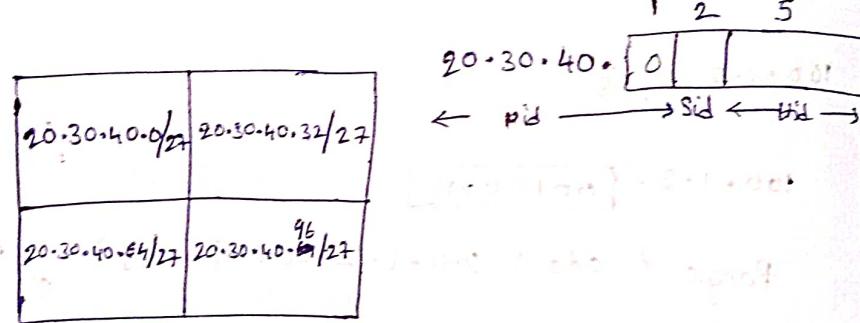
Host id

Now by subnetting we have



In CIDR representation ~~are~~ no bits in block id of subnet are represented. Also these bits represent ~~not~~ subnet mask also.

Similarly we can divide it into 4 subnets

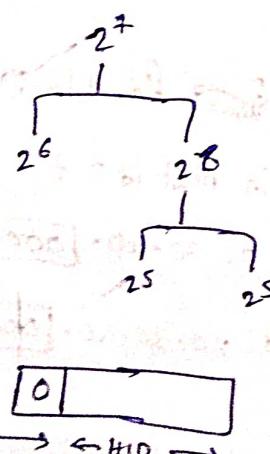


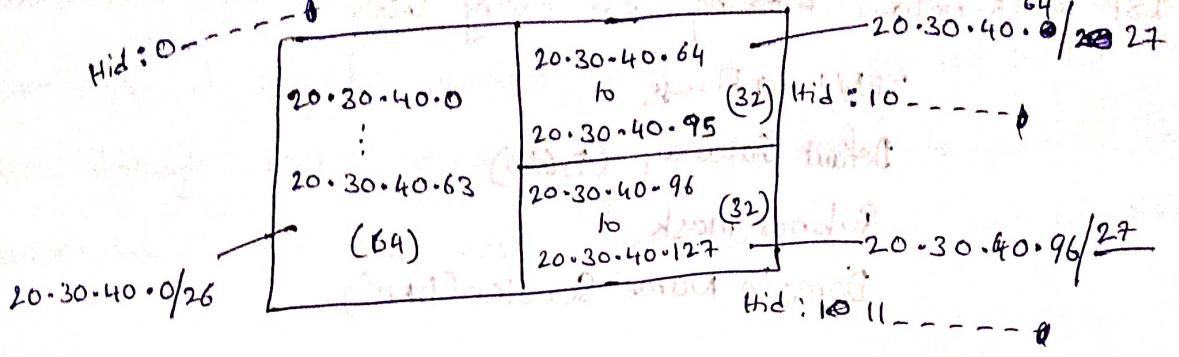
VLSM in CIDR Blocks

→ Consider $20 \cdot 30 \cdot 40 \cdot 10/25$

⇒ no of hosts: 2^7

Consider following division





Another way of subnetting is

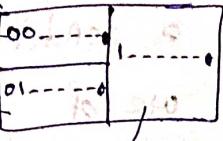
by adding 1 to the number of subnet bits.

Example of subnetting is shown below:

$20.30.40.0/27$

$20.30.40.32/27$

$20.30.40.64/26$



Eg: Consider $40.30.10.10/20$

base of 20 bits for no. of hosts needed, 20 bits address

$40.30. \quad \boxed{00001010} \quad \boxed{00001010}$

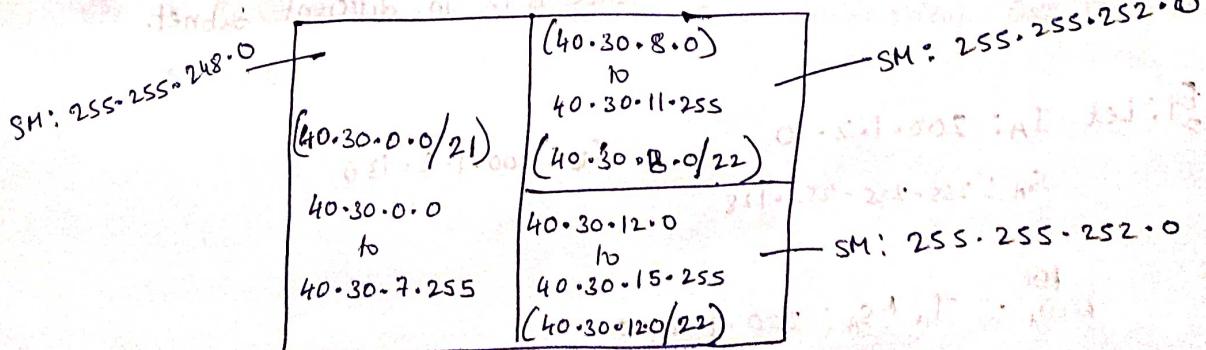
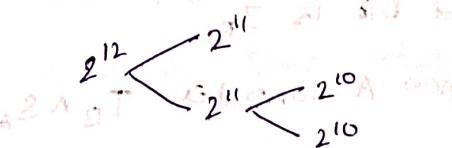
HID

$\therefore \text{NID: } 40.30.0.0$

No of IP add: 2^{12}

Range of add: $40.30.0.0$ to $40.30.15.255$

Consider subnetting this into 3 subnets as shown below



→ ISP gives below 4 things for every user

IPV4 address

Default Gateway (D.G.W)

Subnet mask

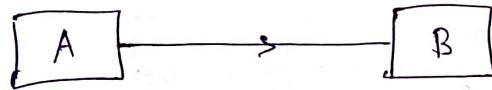
Domain Name Service (DNS)

→ Using subnet mask if destination is in same N/w then

a sender can directly send packet to destination without use of router. Otherwise sender sends the packet to router.

For this purpose of identifying whether destination is within same network subnet or not, the host is provided subnet mask.

→ Consider



let IP add of A is I_A and subnet mask be S_A and ~~Network id~~ be Nid_A

$$Nid_A = I_A \wedge S_A$$

let IP add of B is I_B .

now A computes $I_B \wedge S_A$

if $Nid_A = I_B \wedge S_A$ then B is in same subnet

else B is in different subnet.

Eg: Let $I_A: 200.1.2.10$

$$S_A: 255.255.255.128$$

$$Nid_A = I_A \wedge S_A: 200.1.2.0$$

$$I_B \wedge S_B: 200.1.2.128$$

$$Nid_A \neq I_B \wedge S_B$$

∴ B is in diff subnet. So A forwards pckt to D.G.W.

→ Subnet mask given to a host by ISP may or may not be correct.
Sometimes it may be needed that a pkts has to be sent through router even if the destination is within the same subnet. In that case false subnet mask will be given.

Eg: Consider

$I_A: 200.1.2.10$ & $I_B: 200.1.2.69$, $S_A: 255.255.255.128$ & $S_B: 255.255.255.192$

Here, A knows that A & B are in same network.

But for B, it gets an illusion that A is in different subnet

If we assume the address belong to class C

According to A, subnetting is



According to B, subnetting is



→ Thus using subnet mask, security is provided

i.e., some host must send data through router even if the destination is in the same network.

→ When subnet mask is 255.255.255.255,

↳ a host must send data to any destination ~~only~~ only through a router

Supernetting or aggregation:

- In ~~the~~ a routing table, no of entries = no of networks connected to the router.
- For large no of networks, size of routing table grows exponentially.
- So we perform supernetting using which create an illusion to router that all the small networks belong to one network.
- Supernetting doesn't really combine NWS. It is an illusion.

Rules for aggregation:

- Addresses & subnets ~~are~~ must be contiguous.
i.e., contiguous blocks can only be aggregated.
- Size of blocks must be same (sum of all block sizes ~~is~~ is power 2 provided given blocks are valid)
- The first network id should be divisible by size of total aggregated block.

Eg: Consider supernetting of below 4 networks

200.1.0.0/24

200.1.1.0/24

200.1.2.0/24

200.1.3.0/24

Sol:

Addresses are contiguous

size of all blocks are same i.e., 2⁸

∴ size of total aggregated block = $2^8 \times 2^2 = 2^{10}$

∴ starting IP add 200.1.0.0 is divisible by 2¹⁰
i.e., least significant 10 bits are 0's.

∴ supernetting is possible.

Now after supernetting the block is represented as

Supernet id : $200 \cdot 1 \cdot 0 \cdot 0 / 22$ ~~$\rightarrow 255.255.252.0$~~

Now the router will have only one entry instead of 4 entries.

For this block we have supernet mask

$$\text{Supernet Mask} : \cancel{255.255.255.} \cdot \boxed{1111100} \cdot \boxed{0000000} \\ = 255.255.252.0$$

→ The internal router will have 4 entries

external router will have 1 entry

Eg: $200 \cdot 1 \cdot 32 \cdot 0 / 24$

$200 \cdot 1 \cdot 33 \cdot 0 / 24$

$200 \cdot 1 \cdot 47 \cdot 0 / 24$

(i) contiguous

(ii) sizes are same, i.e., $2^8 \times 16 = 2^{12}$

(iii) CIDR of 1st network

$200 \cdot 1 \cdot \boxed{00100000} \cdot \boxed{00000000}$

size of supernet: 2^{12}

No CIDR 1st network is divisible by 2^{12}

∴ Supernetting is possible.

∴ Supernet id: $200 \cdot 1 \cdot 32 \cdot 0 / 20$

$$\rightarrow 24 - \log_2(16) = 24 - 4 = 20$$

Supernet mask: $200 \cdot 1 \cdot \cancel{255.255.} \cdot \boxed{11110000} \cdot \boxed{00000000}$

$$255.255.240.0$$

*

Eg: $100 \cdot 1 \cdot 2 \cdot 0 / 25$

* $100 \cdot 1 \cdot 2 \cdot 128 / 26$

$100 \cdot 1 \cdot 2 \cdot 192 / 26$

(i) contiguous

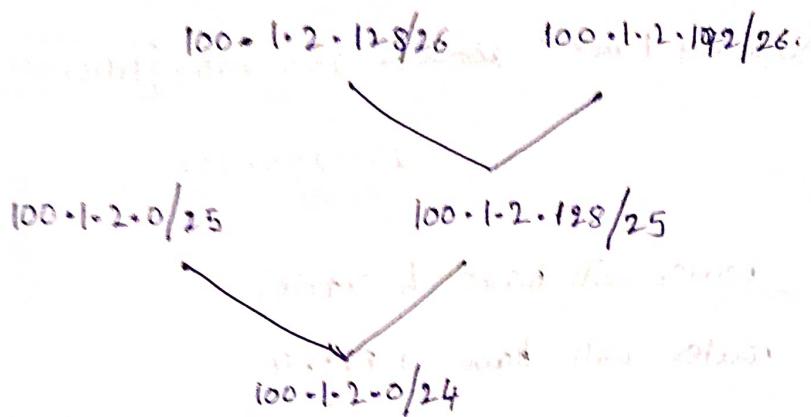
* (ii) Though size is not same, we can first supernet the 2 small networks and then supernet with the larger one.

(iii) 1st size of supernet

$$2^4 + 2^6 + 2^6 = 2^{10} \rightarrow 1024 \text{ hosts}$$

∴ if net id is divisible by 2^8 then it will be

∴ Supernetting is possible



Supernet id: 100.1.2.0/24

Supernet mask: 255.255.255.0

Q3/6-12) An Internet service provider has the following chunk...
S1:

The two variants possible are:

(i) 245.248.128.0/21 → A

(ii) 245.248.136.0/22

(iii) 245.248.140.0/22

(i) 245.248.128.0/22

(ii) 245.248.132.0/22

(iii) 245.248.136.0/21 → A

→ 245.248.132.0/22

Flow Control Method

Delays in Computer Networks:

Transmission Delay (T_t)

It is time taken by the host to put that data packet on outgoing link.

Eg: If bandwidth = 1 bps

then for 10 bit data, transmission delay = 10 sec

If Data is L bits and bandwidth is B bits/sec then

$$\boxed{\text{transmission delay} = \frac{L}{B}}$$

Note:

Whenever k is given with data, it is 1024.

Whenever k is given with bandwidth, it is 1000

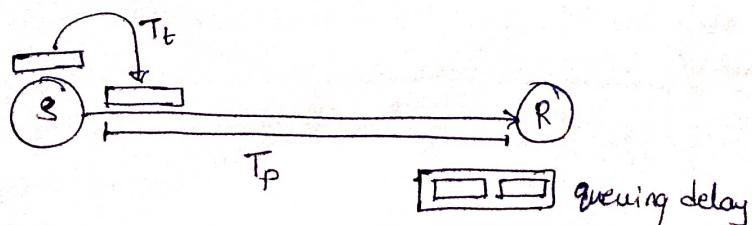
Eg:illy for M & G

Eg: L=1000 bits BW=1 kbps | L=k kb BW=1 kbps

$$\Rightarrow T_t = \frac{1000}{1000} = 1 \text{ sec} \quad \Rightarrow T_t = \frac{1024}{1000} = 1.024 \text{ sec}$$

Propagation Delay:

It is amount of time required for data to travel from source to destination after data being placed on the link.



If d is the distance b/w the laser hosts and v is the velocity of signal then

$$T_p = d/v$$

Generally velocity of signal in optical fibre is 70% of velocity of light

$$\text{Let } d = 10 \text{ m} \Rightarrow v = 2.1 \times 10^8 \text{ m/s}$$

Total time to send

a packet = transmission delay + propagation delay

$$\text{and we know that } T_t + T_p \text{ is about } 3 \mu\text{s}$$

Queuing delay:

The packets that are sent are not processed immediately by the receiver. The sent packet will wait in a queue. The amount of time the packets wait in the queue, before they are processed, is called queuing delay.

Processing delay:

The amount of time required by the receiver to process the packet is called processing delay.

→ Queuing delay & processing delay depend on the speed of the receiver. In numerical, if not given, queuing delay and processing delay are considered 0.

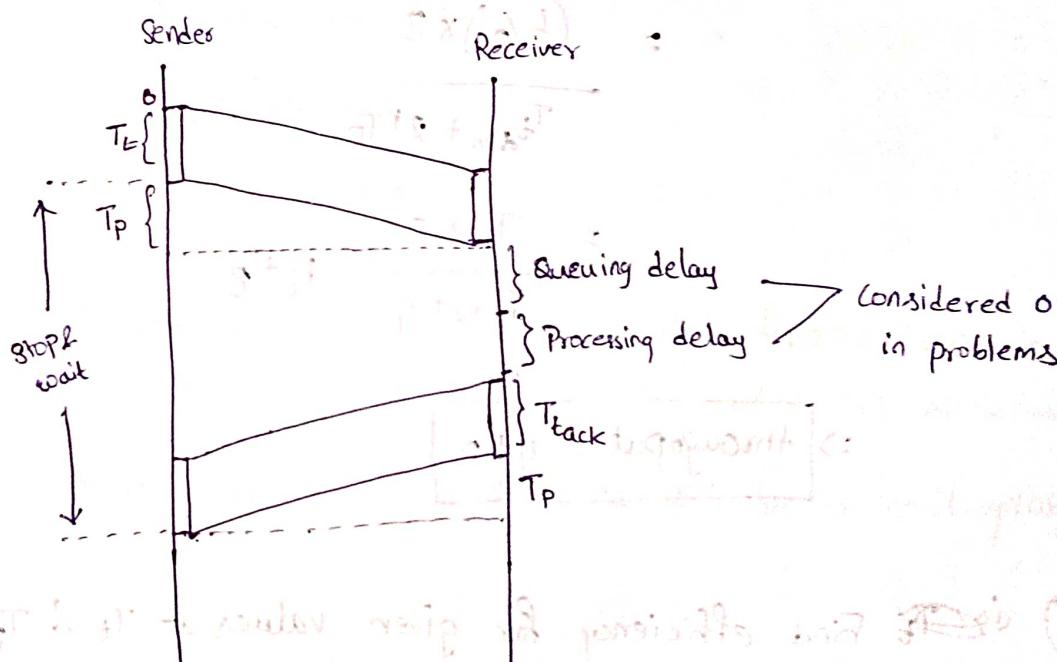
Flow Control Methods:

→ If receiver is slow and sender is fast, then packets sent by sender may get lost.

So the speed at which sender has to send the data depends on receiver. This is known as closed loop protocol.

Stop & Wait Protocol:

→ Here sender sends a packet and waits for the acknowledgment from receiver.



∴ Total time to send one packet

$$= T_{t\text{data}} + T_p + T_{t\text{ack}} + T_p$$

$$\therefore \text{Total time in stop & wait} = T_{t\text{data}} + T_{t\text{ack}} + 2 * T_p \approx T_{t\text{data}} + 2 * T_p$$

→ negligible compared to $T_{t\text{data}}$ (without piggybacking)

$$= T_{t\text{data}} + 2 * T_p$$

$$\text{efficiency} = \frac{\text{useful time}}{\text{total cycle time}} = \frac{T_{t\text{data}}}{T_{t\text{data}} + 2 * T_p} = \frac{1}{1 + 2\left(\frac{T_p}{T_{t\text{data}}}\right)}$$

$$= \frac{1}{1+2a} \quad \left(a = \frac{T_p}{T_{t\text{data}}} \text{, This just for simplicity} \right)$$

throughput: no of bits (effective) sent per second.

(also known as effective bandwidth / bandwidth utilization.)

$$\text{throughput} = \frac{L}{T_{t\text{data}} + 2 * T_p}$$

L is size of data

$$= \frac{(L/B) * B}{T_{t\text{data}} + 2 * T_p}$$

$$= \frac{T_f * B}{T_f + 2 * T_p} = n * B$$

$$\Rightarrow \text{throughput} = n * B$$

Q4) ~~Q4~~ Find efficiency for given values of T_f & T_p

$$\text{i) } T_f = 1\text{ms} \quad T_p = 1\text{ms}$$

$$\text{Ans: } 1/3 \Rightarrow 33.33\%$$

$$\text{ii) } T_f = 2\text{ms} \quad T_p = 1\text{ms}$$

$$\text{Ans: } 1/2 \Rightarrow 50\%$$

Q5) Find relation b/w T_p & T_f if min efficiency is 0.5

$$\eta \geq 0.5$$

$$\frac{T_f}{T_f + 2 * T_p} \geq \frac{1}{2}$$

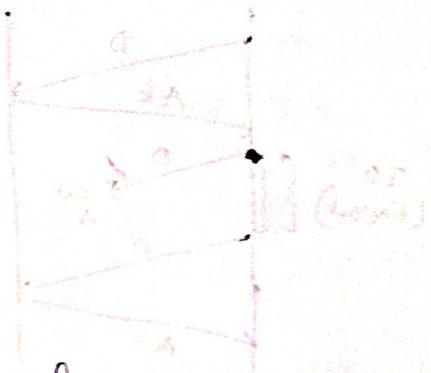
$$2T_t \geq T_t + 2T_p$$

$$T_t \geq 2T_p$$

Factors affecting efficiency in stop & wait:

$$\eta = \frac{1}{1 + 2\left(\frac{T_p}{T_t}\right)}$$

$$\eta = \frac{1}{1 + 2\left(\frac{d}{v}\right) + \left(\frac{B}{L}\right)}$$



for a given medium v & B are fixed

so η depends on d & L

(as v & B are constant, so η varies with d & L)

distance, $d \uparrow \Rightarrow \eta \downarrow$

size of packet, $L \uparrow \Rightarrow \eta \uparrow$

advantages of stop & wait

Stop & wait is good for shorter distance transmissions

It is good for networks like LANs but not for WANs

Also stop & wait is good ~~while~~ for transmitting larger packets.

Note: note not good for WANs as we making side effects of it

→ If sender doesn't receive ~~a packet~~ acknowledgement it may be due to 2 reasons

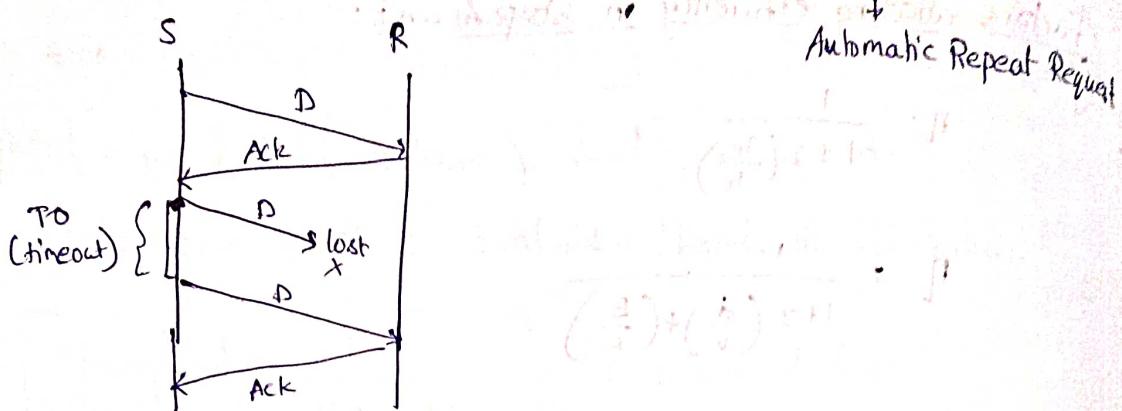
- (i) Data is lost
- (ii) Ack is lost

Data packet lost problem:

Assume a case where sent packet is lost. Then receiver won't send ack and sender assumes data is lost.

So sender waits for a fixed amount of time; timeout and resends the same data.

This variant of stop & wait is called Stop & Wait ARQ.



→ without using this variant, the sender & the receiver may go to deadlock (sender waits for ack & receiver waits for data)

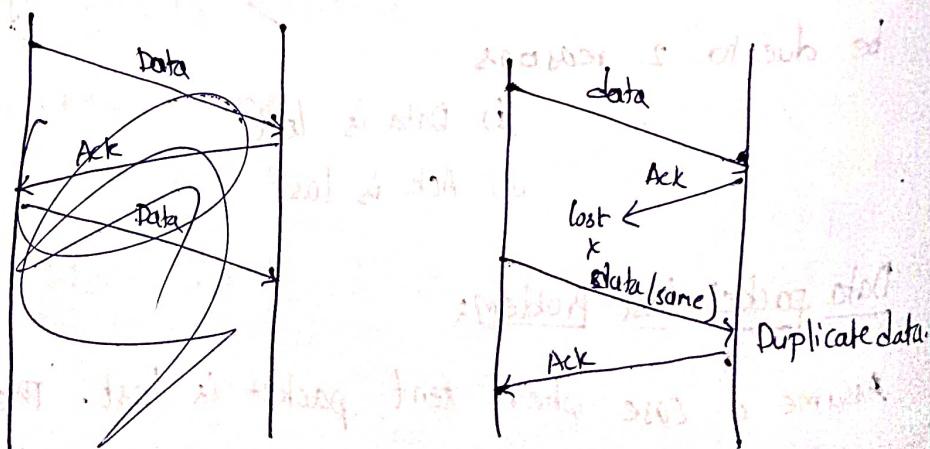
Duplicate Packet Problem:

→ This problem occurs when ack sent by receiver is lost.

→ In this case, sender assumes data is lost and resends it.

→ Now receiver receives the same packet again. This problem is called duplicate packet problem.

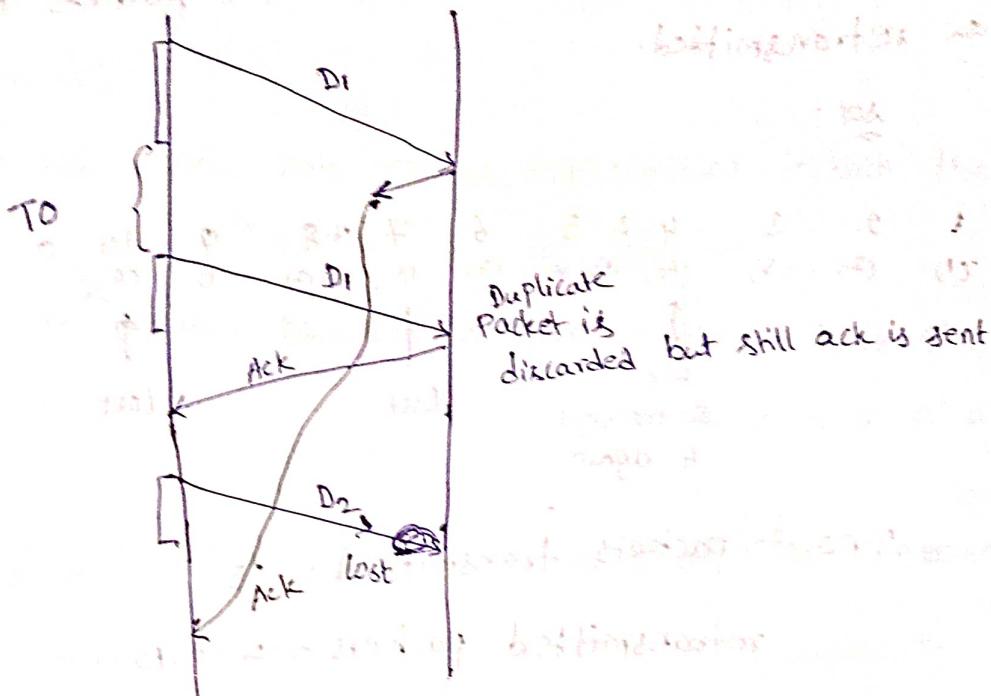
→ To solve this problem we use sequence number for every packet.



→ So here we have timeout along with sequence number.

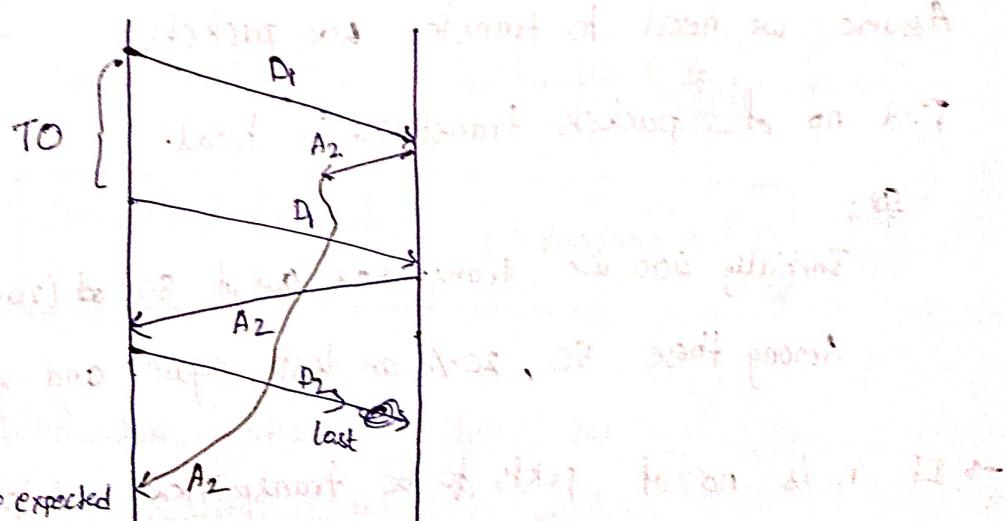
Delayed Acknowledgment (Missing packet problem):

→ Here ack is not lost but it is delayed such that timeout occurs and sender retransmits the data.



→ Here even though D_2 is lost, sender gets the ack of D_1 and thus sender thinks that D_2 is received by the receiver.

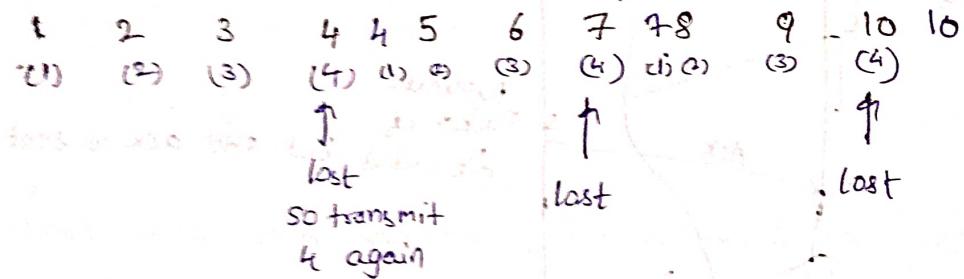
→ To avoid this problem we give sequence number for ack also.



Received. So sender will understand that D_2 is lost.

(Q5) Consider using stop & wait protocol and transmitting 10 packets. Assume every 4th packet that is transmitted is lost. Find total no of packets that are transmitted and also find the packets that are retransmitted.

Sol:



$$\therefore \text{no of packets transmitted} = 13$$

$$\text{retransmitted packets} = 4, 7, 10$$

Note:

→ Consider a channel b/w source & destination in which

error probability of channel = 0.2

Assume we need to transfer 400 packets.

Find no of packets transferred in total.

Sol:

Initially 400 are transmitted out of 80 (20%) are lost.

Among these 80, 20% are lost again and so on...

→ If n is no of pkts to be transmitted and ' p ' be error probability of channel,

$$\text{no of pkts transmitted} = n + np + np^2 + \dots$$

$$= n \left(\frac{1}{1-p} \right)$$

Here $n=400$; $p=0.2$

$$\text{no of pkts transmitted} = 400 \left(\frac{1}{1-0.2} \right) = 500 \text{ packets.}$$

Capacity of channel/wire/link:

types of channels:

- Full duplex: Data can be transmitted in both the directions at the same time.
- Half duplex: Data can be transmitted in both the directions but only in one direction at a time.

→ Capacity of a channel is no of bits present on the channel when the channel is operating at its maximum capacity.

→ After a bit is placed on the channel it takes T_p time (propagation delay) to reach the other end. That means all the bits placed on the channel during this time T_p are present on the channel.

$$\therefore \text{Capacity of channel} = \text{Bandwidth} * T_p$$

(half duplex)

$$\text{Capacity of full duplex} = 2 * \text{Bandwidth} * T_p$$

Channel

Eg: If bandwidth = 1 bps and $T_p = 10 \text{ sec}$
 $\text{Capacity} = 10 \text{ bits (half duplex)}$

→ Channels with high capacity are called thick channel/thick pipe

- One more disadvantage of Stop & wait, it is not good with thick channels. But it is good with thin pipes
- ∴ Stop & thick channels → less efficiency
 Stop & wait, thin pipe → high efficiency.

$$\eta = \frac{1}{1 + 2 + \frac{T_p}{T_t}} = \frac{1}{1 + 2 \times \frac{T_p}{T_t} + \frac{1}{L}} = \frac{1}{1 + \frac{\text{Capacity}}{L}}$$

all we want is capacity ↑ ⇒ efficiency ↓

Pipelining / Sliding Window Protocol:

- This is a technique used to increase efficiency.

$$\eta = \frac{1}{1 + 2a}$$

This means only 1 unit of time is used in $(1+2a)$ units.

which means we can increase efficiency by transmitting packets in that $2a$ units of time also.

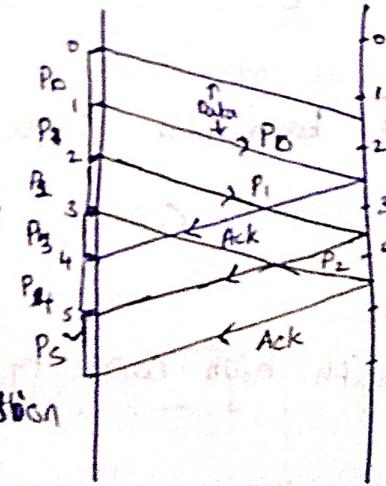
This is known as pipelining.

Eg: Consider $T_t = 1\text{ ms}$, $T_p = 1.5\text{ ms}$.

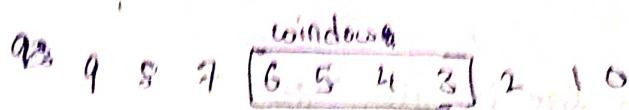
$$\eta = \frac{1}{1 + 2a} = \frac{1}{1 + 2 \left(\frac{1.5}{1} \right)} = \frac{1}{4}$$

Here ack of P_0 is received at ~~end~~ after 4 units of time.

so sender has to leave P_0 until that time so that retransmission can be done if P_1 is lost.



- Q. minimum windows size required = 4
- ↳ group of packets received by sender so that retransmission can be done if needed.
 - Packets in windows are called outstanding packets.
 - After n units of time, packet p_0 will be discarded from buffer (window) and p_n will be stored.
 - Packets in the windows are those that are waiting for ack.



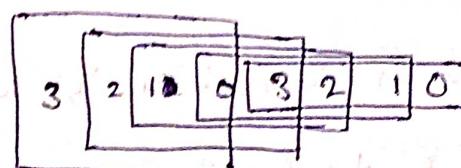
As the time progresses, the

window slides towards left, i.e. RE

- * Minimum size of windows required = $1 + 2a = \frac{1}{n}$ packets (ceil if not integer)
- Sequence number has to be stored in the header of packet.
- So we cannot use arbitrarily large no of bits for seq. num.

Initially packets in window are P_3, P_2, P_1, P_0

Later when P_0 is discarded, we can again use sequence number 0.



$$+ \text{ Thus no of bits needed for representing seq. num} = \lceil \log_2 (1 + 2a) \rceil$$

- (Q) $T_f = 1\text{ms}; T_p = 44.5\text{ms};$ find min window size & min no of bits req in seq. num.

Ans: 100 & 7

→ But sometimes due to protocol overhead, max no. of bits in seq num field may be less. i.e., for example in Q7 no. of bits may be just 6, but not 7. in this case we won't get more efficiency. Then what?

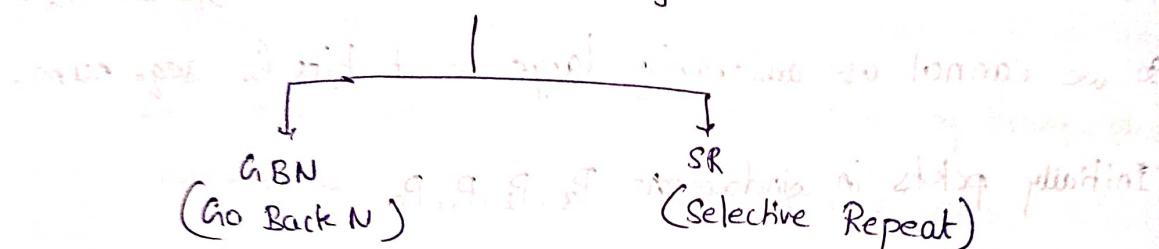
So here efficiency = $\frac{\text{no of pkts sent}}{\text{max possible no. of pkts that can be sent}} \times 100 = \frac{64}{100} \approx 64\%$

So here window size is also 64.

→ If size of seq. numbers is 2^n bits, then

$$\text{window size} = \min(1+2a, 2^n)$$

→ Sliding window protocol is a theoretical concept. It can be implemented in 2 ways:



Go Back N: ($N > 1$)

i) Sender window size in Go Back N is $\leq N$ and $N > 1$

If $N=1$ it is just stop & wait.

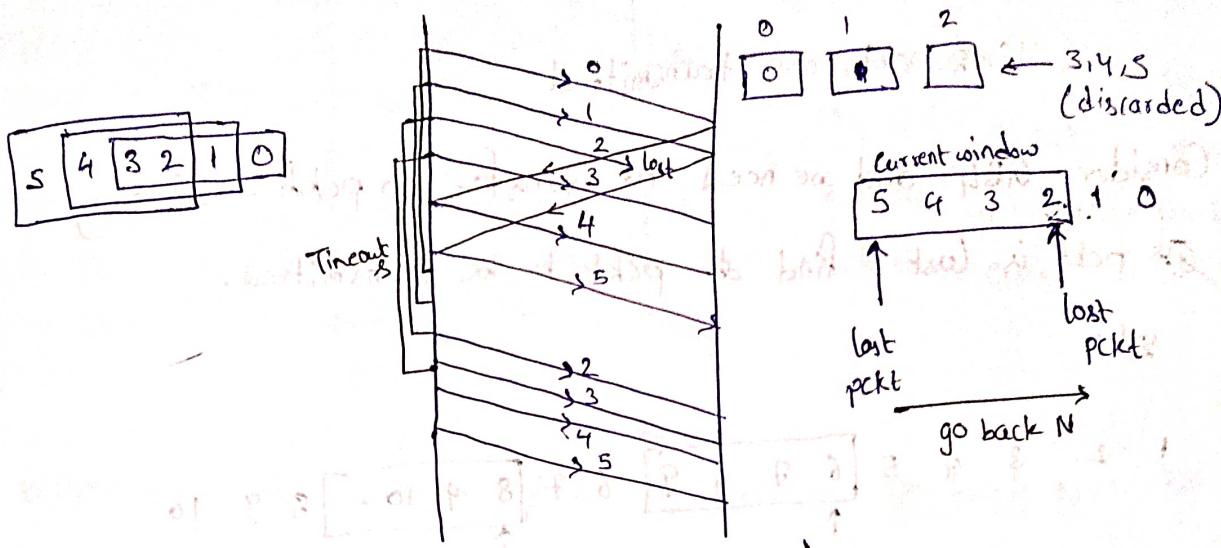
Eg: Consider using GBN and $T_f = 1ms$ & $T_p = 49.5ms$. Find η

$$\eta = \frac{\text{no of pkts sent}}{\text{max possible}} = \frac{10}{1+2a} = \frac{10}{100} \Rightarrow \text{i.e., } 10\%$$

$$\text{throughput} = \eta * B = \frac{10}{100} \times 40 = 4 \text{ Mbps}$$

iii) Receiver window size = 1

Eg: Consider below situation & let sender window size = 4



→ Consider a situation where receiver receives pkts 0,1 but pkt 2 is lost. By the time timeout of Packet 2 occurs sender transmit pkts 3,4,5.

~~After the timeout~~

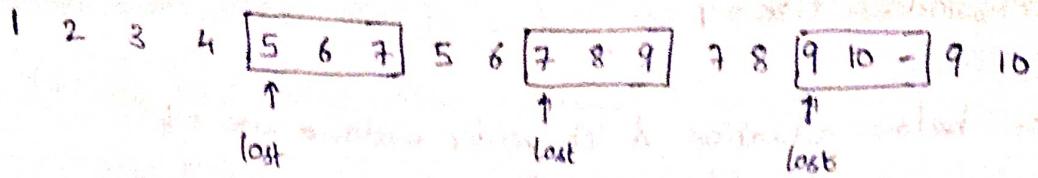
→ Here receiver first receives pkts 0,1 and waits for pkt 2. Pkt 2 is lost and receiver will reject pkts 3,4,5.

→ So considering this, after timeout of pkt 2, sender again sends pkts 2,3,4,5 by going back 4-packets.

→ Here the window slides once the ack of oldest pkt of window is received

*Eg: In GB3, if every 5th packet that is being transmitted is lost and we have to send 10 packets, then how many transmissions are required.

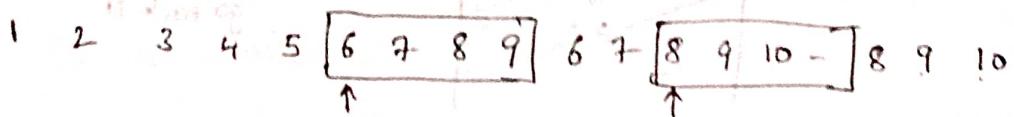
Sol:



$\therefore 18$ pkts are transmitted.

- (Q6) Consider GBN and we need to transfer 10 pkts. If every 6th pkt is lost, find # of pkts to be transmitted.

Sol:



$\therefore 17$ pkts are to be transmitted.

- (iii) Go Back N uses cumulative acknowledgement.

Ack is of two types:

$\begin{cases} \text{cumulative ack: One ack is sent for group of packets} \\ \text{adv: less traffic} \\ \text{disadv: less reliable (even if one pkt is lost, ack is not sent and all has to be retransmitted)} \end{cases}$

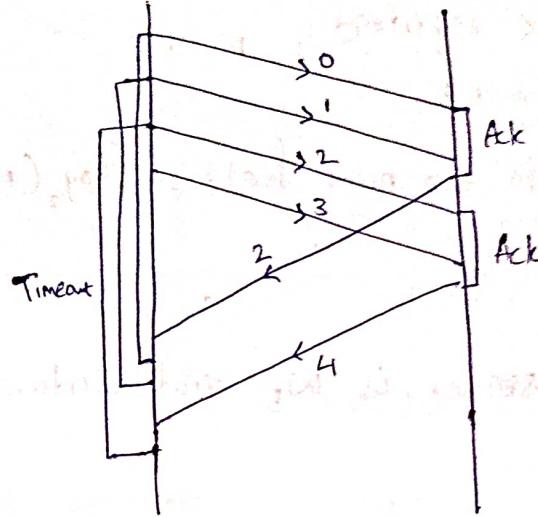
 $\begin{cases} \text{independent ack: one ack is sent for each packet.} \\ \text{adv: more traffic} \\ \text{disadv: more reliable} \end{cases}$

\rightarrow Receiver has an acknowledgement timer. Once a pkt is received the receiver starts timer and once the timer expires receiver send ack for all the pkts received in that time.

\rightarrow Ack timer should be so large that timeout occurs on sender size.

\therefore timeout timer $>$ ack timer.

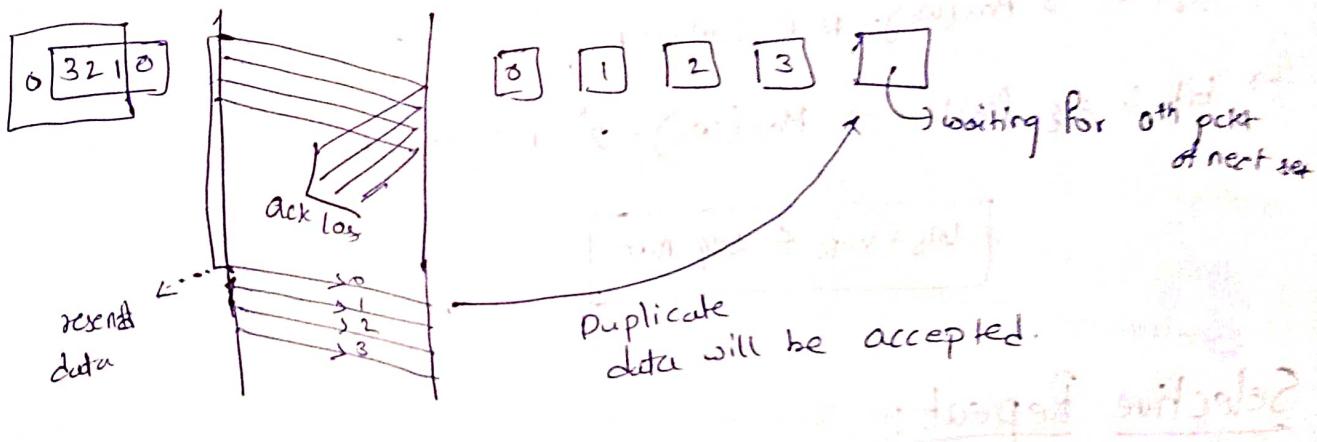
→ Also ack times should not be too small.



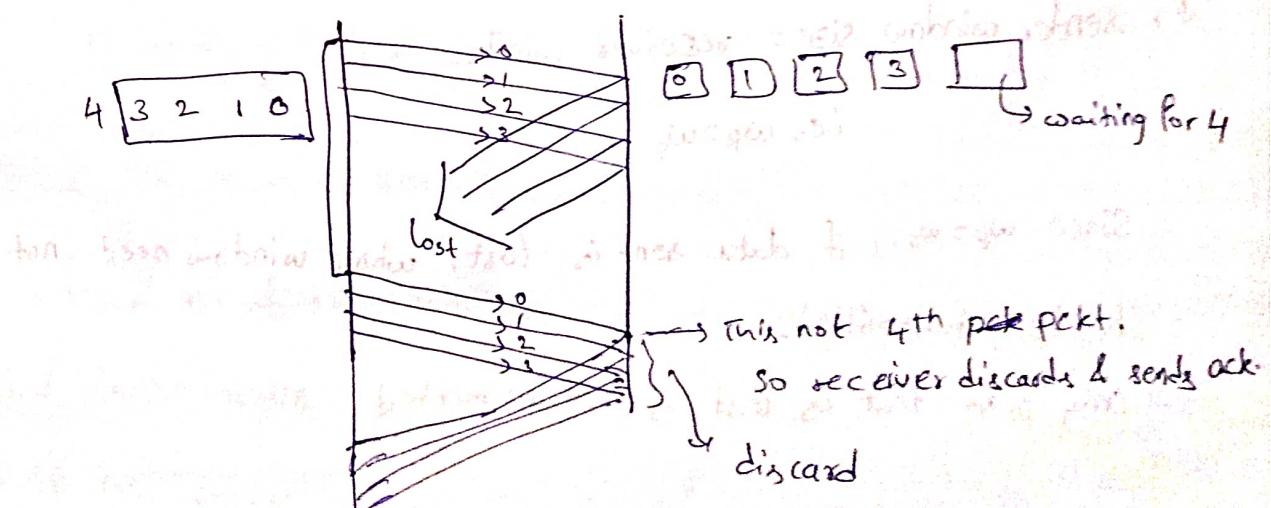
Ack time \rightarrow new ack times starts
only after a pkt is received
but not after the previous
ack timer ends

Relationship b/w window size & seq num in go back N:

Consider GB4, and let no of seq# numbers available = 4.



But if take ~~any~~ no of seq numbers = 5 then we will not have this problem



so here no of seq numbers required $\geq N+1$

in go back N

no of bits in seq num field = $\lceil \log_2(N+1) \rceil$

Note: if window size of sender is w_s and window size of receiver is w_r ,

no of seq numbers req = $w_s + w_r$

no of bits in seq num field = $\lceil \log_2(w_s + w_r) \rceil$

Note:

In GBN, max seq number is $w_s + w_r - 1$

$$\rightarrow w_s = N \quad w_r = 1 \Rightarrow \text{seq} = N+1$$

$$\rightarrow \text{seq} = N \Rightarrow \text{Max}(w_s) = N-1; w_r = 1$$

$$\rightarrow \text{bits in seq field} = k \Rightarrow \text{Max}(w_s) = 2^k - 1; w_r = 1$$

$$w_s + w_r \leq \text{seq num}$$

Selective Repeat:

i) Sender window size > 1

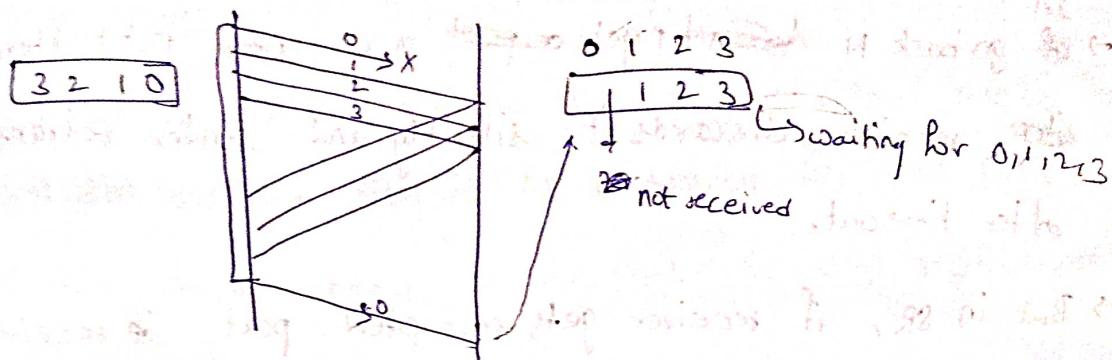
ii) Sender window size = receiver window size

$$\text{i.e., } w_r = w_s$$

Since $w_r = w_s$, if data sent is lost, whole window need not be retransmitted.

only pkt that is lost is retransmitted.

Eg: let $w_R = w_S = 4$



Assume data pckt 0 fist lost.

Still receive can accept pckts 1, 2, 3 but it has sufficient window.

So here no of retransmission req will be less.

Eg: Consider we need send 10 pckts where every 5th pckt that is transmitted is lost. let $w_S = w_R = 3$

1 2 3 4 5 6 7 8 9 10
↑ ↑

\therefore no of transmission = 12

but go back N required 18 transmission.

(This may not be the exact order in which transmission is done)

Eg: Consider 10 packets to be transmitted and every 4th pckt is lost.

1 2 3 4 5 6 7 8 9 10
↑ ↑ ↑

Here no of transmissions = 13

→ In terms of no of transmissions SR and stop & wait have same no of transmissions

→ So SR has advantages of Stop & wait & go Back N.
But there are some disadvantages too.

- 3) SR uses independent acknowledgment
- In
 \rightarrow go back N if receiver gets a corrupted pkts then,
 receiver discards it silently and sender retransmits after timeout.
- \rightarrow But in SR, if receiver gets corrupted pkts, receiver sends negative ack to sender so that sender ~~forgets~~ retransmits the corrupted pkts even before the timeout.

Comparison b/w the protocols:

	Stop & Wait	GBN	SR
Efficiency	$\frac{1}{1+2a}$	$\frac{N}{1+2a}$	$\frac{w_s}{1+2a}$
Buffer req (sender+receiver)	$1+1$	$N+1$	$w_s + w_d$
Min seq num required	$1+1=2$	$N+1$	$w_s + w_d$
Retransmission Per pkts loss	1	N	1
BW req / BW consumption	low	high	moderate
CPU requirement	low	moderate	high
Implementation difficulty	low	moderate	high
ack	independent	cumulative	independent

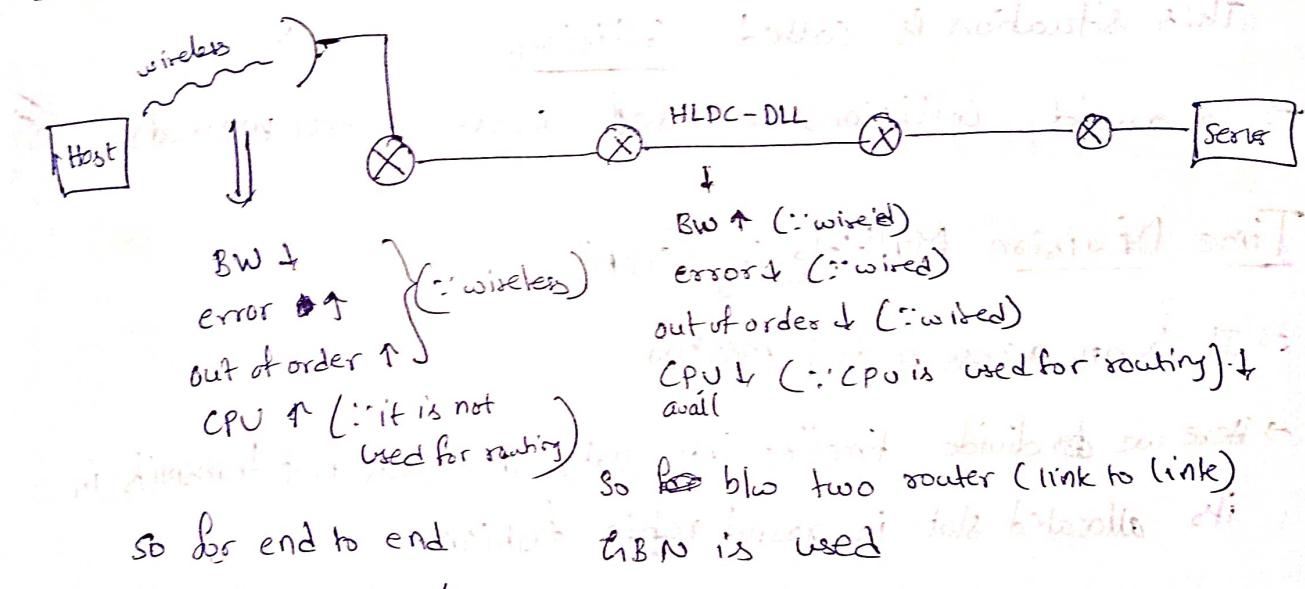
→ BW requirement is high for GBN, ~~because of more no of retransmissions~~

→ In SR, out of order pkts are also accepted, so it needs to perform ~~etc~~ sorting. Also it has to search for lost pkt. But GBN doesn't need to do this. So CPU requirement in SR is high.

→ If BW is ~~less~~ high, CPU is less efficient, buffers are less then GBN is used

If BW is moderate, CPU is more available, buffers are more then SR is used

Example: end-to-end link layer protocols



→ A system may have several processes running.

Each process has its own buffer. These buffers are operated

with SR protocol.

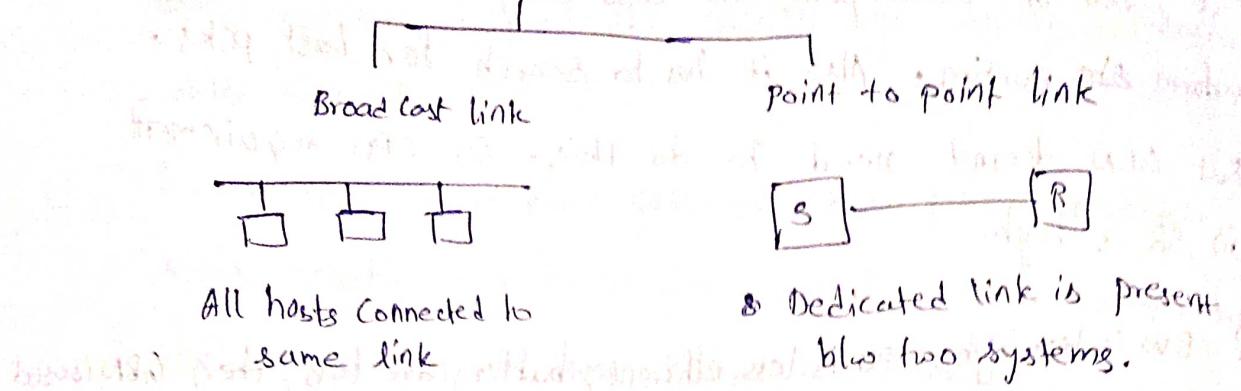
All the packets (from diff processes) will be sent to DLL. At this layer, the buffers are operated with GBN.

so for higher level layers SR is used and for lower level layers GBN is used.

TCP → (SR (75%), GBN (25%))

Access Control Methods:

→ Links are of 2 types



→ In broadcast link, there may be a chance that multiple station may transmit data at the same time. This situation is called Collision.

→ To avoid collisions we used access control methods.

Time Division Multiplexing (TDM):

→ It is an access control method.

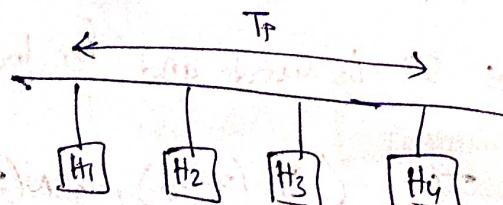
→ Here we divide timeline into slots and each host transmits in its allocated slot. in round robin fashion.

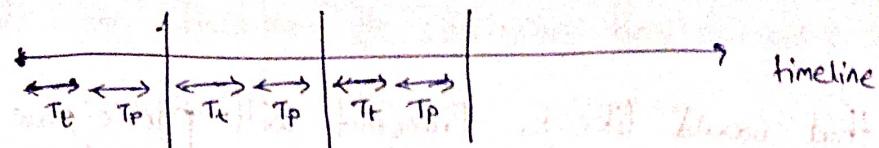
→ Allocated slot should be large enough that a host can transmit data.

$$\text{i.e., time slot} = T_e + T_p$$

Assuming that
every station send
Same amount of data

Propagate delay in the worst case. i.e., when sender and destination are at two different ends of the channel





$$\Rightarrow \text{efficiency}, \eta = \frac{\text{useful time}}{\text{cycle time}} = \frac{T_F}{T_E + T_P} = \frac{1}{1+a}, a = \frac{T_P}{T_E}$$

Ex: $T_E = 1\text{ms}$, $T_P = 1\text{ms}$ & $BW = 4\text{Mbps}$. Find, η & effective BW (throughput)

$$\eta = \frac{1}{1+\frac{1}{4}} = \frac{1}{\frac{5}{4}} = 0.8 = 80\%$$

$$EBW = \eta \times BW = 2\text{Mbps}$$

Ex: Let BW of channel be 2Mbps . Assume each host needs BW of 2kbytes . Then find max no of hosts possible satisfying this requirement using TDM.

Sol:

$$N * 2\text{kbytes} = 2\text{Mbps}$$

$$\Rightarrow N = 1000$$

Disadvantages: slot left unused when no data is transmitted.

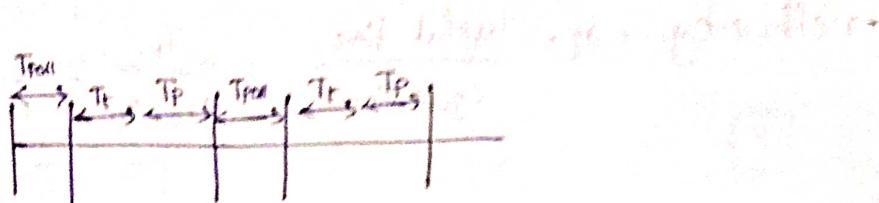
→ If a station doesn't want to transmit data, then the slot reserved for that station is wasted.

→ This disadvantage is overcome by using polling.

Polling:

- The system that would like to transmit will participate in polling.
- From those one will be selected and will be given time to transmit.

transmit:



$$D = \frac{T_d}{T_{poll} + T_d + T_p}$$

(taken as 0 if it is negligible.)

Demand:

- Time for polling is a waste.

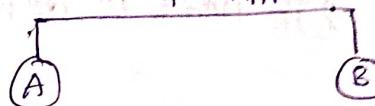
CSMA/CD: (Carrier Sense Multiple Access Collision Detection)

- Every host, before sending, senses the carrier and if it is free then it transmits data.
- However a host can sense only the part that it is connected to the channel.
- Also two stations may sense that the channel is empty and initiate transmission at the same time.
- CSMA/CD doesn't use acknowledgement since it is mostly used in LANs. So we need to detect that collision has occurred without using acknowledgement.
- Collision produces a signal with different frequency from data. So a device can detect collision if it sees this signal.

- A device thinks a collision has occurred and its data is corrupted only if it senses collision signal while it is transmitting data.
- So we make sure that every device transmits long enough that even in the worst case the collision signal will reach the device.

④ For example consider

consider transmission time $T_p = 1\text{ hr}$



let $t = 10:00 \text{ AM}$

let A & B both sense that channel is ~~empty~~ free and started sending data.

Then collision occurs at 10:30

and both A, B will detect ^{collision} signal at 11:00 AM

so this collision can be detected if $T_d > 1\text{ hr}$.

However this not the worst case.

consider A has transmitted data to B and when its just about to reach B (after 1hr) B has started transmission

Then this collision will be detected by A and by B after 1hr.

④ A transmission initiation: 10:00 AM

Collision detection: 11:00 AM

detection by A: 12:00 PM

∴ To detect any collision, $T_d \geq 2T_p$

i.e., min data to be transferred, $L \geq 2 \times T_p \times B$

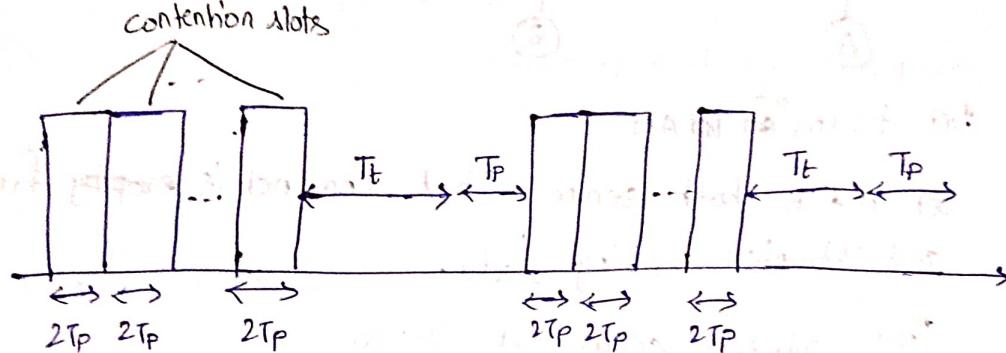
→ If we don't have sufficient amount of data to transfer then we need to pad extra bits.

Efficiency of CSMA/CD:

↳ (Ethernet)

→ Before a successful transmission, a series of collisions may occur. (we don't know the no of collisions)

In the worst case it takes $2T_p$ time to detect collision contention slots



$$\text{efficiency} = \frac{T_e}{C * (2T_p) + T_t + T_p}$$

$C \rightarrow$ no of collision slots in each cycle.

↳ Finding C :

Assume we have n stations & every station transmits with probability p .

So there will be no collision if only one station transmits the data.

$$\Rightarrow P_{\text{success}} = n \cdot p \times (1-p)^{n-1} = np(1-p)^{n-1}$$

$$\frac{dP}{dp} = np(n-1)(1-p)^{n-2}(-1) + n(1-p)^{n-1}$$

$$= n(1-p)^{n-1} - n(n-1)p(1-p)^{n-2} = 0$$

$$= n(1-p)^{n-2} [(1-p) - (n-1)p] = 0$$

$$\Rightarrow (1-p) - (n-1)p = 0$$

$$\cancel{\text{np} - p - 1 + p = 0 \Rightarrow p \cancel{= 1}} \boxed{p = 1/n}$$

$$\Rightarrow P_{\max} = n \times \frac{1}{n} \times \left(1 - \frac{1}{n}\right)^{n-1}$$

$$P_{\max} = \left(1 - \frac{1}{n}\right)^{n-1}$$

If n is sufficiently large

$$P_{\max} = \lim_{n \rightarrow \infty} \left(1 - \frac{1}{n}\right)^{n-1}$$

$$\boxed{P_{\max} = 1/e}$$

\Rightarrow no of times we should try (collisions) before getting the first success

$$\text{first success} = \frac{1}{P_{\max}} = e$$

i.e., e collisions and $2e+1$ attempts will do.

\therefore value of $c = e$

$$\therefore \text{Efficiency, } \eta = \frac{T_t}{(2e+1)T_p + T_t} = \frac{1}{1 + 6.44 \alpha}$$

$$\eta = \frac{1}{1 + 6.44 \left(\frac{d}{v}\right) \left(\frac{B}{L}\right)}$$

$d \uparrow \Rightarrow \eta \downarrow$

\therefore CSMA/CD is good for ~~small~~ LANs

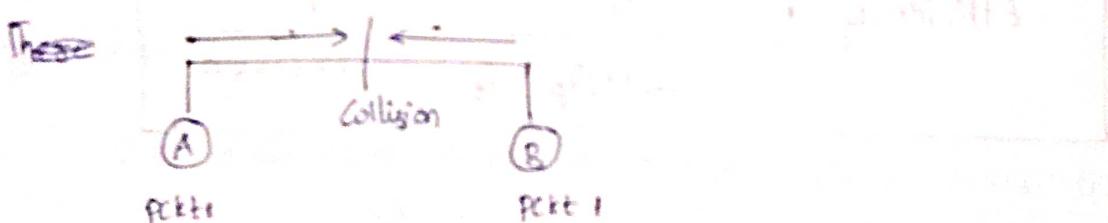
$L \uparrow \Rightarrow \eta \uparrow$

\therefore CSMA/CD has good efficiency for larger packets.

Backoff Algorithm:

- This algorithm is used to give waiting time (back off time) to stations that involved in collision.
- This is also known as binary exponential back off algorithm.
- This algorithm applies only for 2 stations.
- Every packet is given a collision number.
i.e., the number of collisions the packet has involved in.
- After a packet gets collided, its collision number is incremented.
- ~~After a collision~~ After this both the hosts selects a number in the range $[0, 2^n - 1]$
where n is collision number.
Let the selected number be k.
Then waiting time = $(k * \text{time slot})$

Eg: Consider a case where two stations A & B wants to transmit data.



Collision number
for $\text{pkt} 1 = 1$

Collision number
for $\text{pkt} 1 = 1$

choose a num b/w $(0, 2^1 - 1)$ i.e. $(0, 1)$
 $\approx (0, 1)$

choose a num
b/w $(0, 1)$

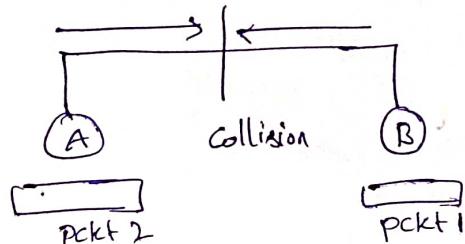
A	B	
0	0	→ Collision
0	1	→ A transmits
1	0	→ B transmits
1	1	→ Collision

Let $P(A)$, $P(B)$, $P(\text{collision})$ be probabilities that A transmits,

B transmits, collision occurs

$$P(A) = 1/2; P(B) = 1/2; P(\text{collision}) = 2/4$$

Assume A has transmitted here and it is ready with packet 2
and B is ready with packet 1

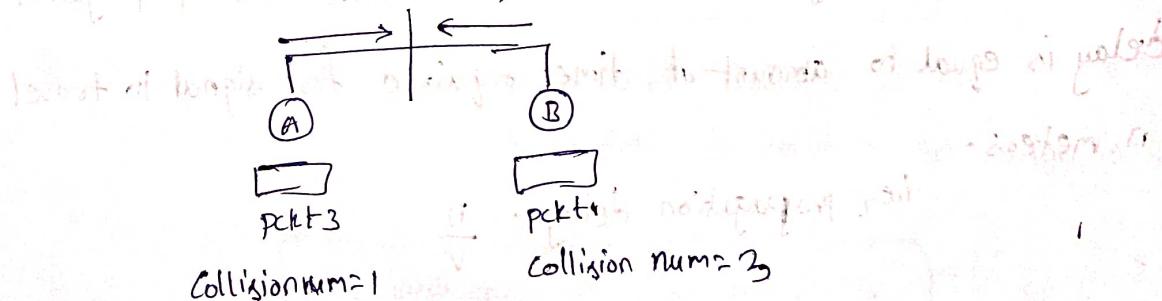


Collision num = 1 Collision num = 2

Choose k b/w (0,1) Choose k b/w (0,1,2,3)

<u>A</u>	<u>B</u>	
0	0 → collision	$\Rightarrow P(A) = 5/8$
0	1 → A	$P(B) = 1/8$
0	2 → A	$P(\text{collision}) = 2/8$
0	3 → A	
1	0 → B	
1	1 → collision	
1	2 → A	
1	3 → A	

Now, if A is successful again, we have below scenario



$$\Rightarrow P(A) = 13/16 \quad P(B) = 1/16 \quad P(\text{collision}) = 2/16$$

Observations:

- As collision number increases, the probability of collision is decreasing exponentially

→ If a system wins in transmitting successfully, then its probability of winning for further transmission increases exponentially, while the probability of other system decreases exponentially. This is a disadvantage.

This is known as capturing effect.

Token Passing:

Note:

→ In some cases propagation delay is given in bits.

If propagation delay is n bits, it means that ~~it is~~

~~amount~~ do, propagation delay is equal to transmission delay of n bits.

$$\text{i.e., propagation delay of } n \text{ bits} = \frac{n}{B} \text{ sec}$$

↳ Bandwidth.

→ In some cases, propagation delay is given in meters.

If propagation delay is n metres, it means that propagation delay is equal to amount of time required for signal to travel n meters.

$$\text{i.e., propagation delay} = \frac{n}{v} \text{ sec}$$

$$\begin{array}{ccc} *v & & *B \\ \text{metres} & \xrightarrow{\frac{1}{v}} & \text{Sec} \\ & & \xleftarrow{\frac{1}{B}} \text{bits} \end{array}$$

Token
→ It is
→ All
A +
and
will

Ring

Token Passing:

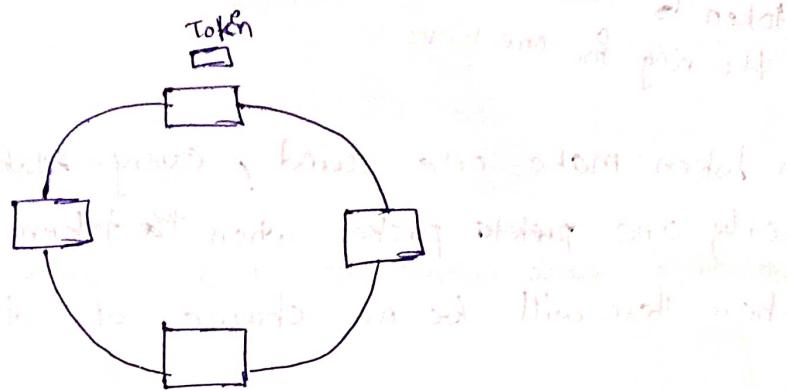
→ It is an access control method.

→ All the hosts are connected in a ring topology.

A token is present and this token moves ~~around~~ around the ring.

This is known as token ring.

whichever host has the ~~token~~ can transmit the data.



Ring latency: it amount of time taken for a bit to travel around the ring and come back to the same point.

A bit while travelling around the ring, the bit has to be read by every station.

$$\therefore \text{Ring latency} = \frac{d}{v} \text{ sec} + (N * b) \text{ bits}$$

where $N \rightarrow$ no of stations

$b \rightarrow$ delay at each station in bits

$$\Rightarrow \text{ring latency} = \left(\frac{d}{v} + \frac{N * b}{B} \right) \text{ seconds}$$

↳ Bandwidth

$$\text{ring latency} = \left(\frac{d}{v} * B + N * b \right) \text{ bits}$$

Token holding time: It is amount of time for which a token is held at any station.

$$\text{Cycle time} = \left(\frac{d}{v} + N * THT \right)$$

↓
 T_p

time req for a token to travel around the ring for one time

→ when token make one round, every station transmits exactly one packet when the token comes to it. So here there will be no chance of collisions.

Efficiency, $\eta_L = \frac{\text{useful time}}{\text{cycle time}}$

$$\boxed{\eta_L = \frac{N * T_t}{T_p + N * THT}}$$

→ There are 2 mechanisms for token passing.

i) Delayed token reinsertion

ii) Early token reinsertion → Default consideration

i) Delayed token reinsertion:

After getting the token, system transmits data and waits until data returns to the same system. Once data comes back, token is passed to next ~~next~~ station.

Here $THT = T_t + \text{ring latency}$

$$= T_t + T_p + N * b$$

$$= T_t + T_p \text{ (assuming } b=0)$$

So in this strategy

$$\text{efficiency, } \eta = \frac{N + T_E}{T_E + N(T_E + T_P)}$$

$$\boxed{\eta = \frac{1}{1 + \left(\frac{N+1}{N}\right)a}, a = T_P/T_E}$$

(ii) Early Token Reininsertion:

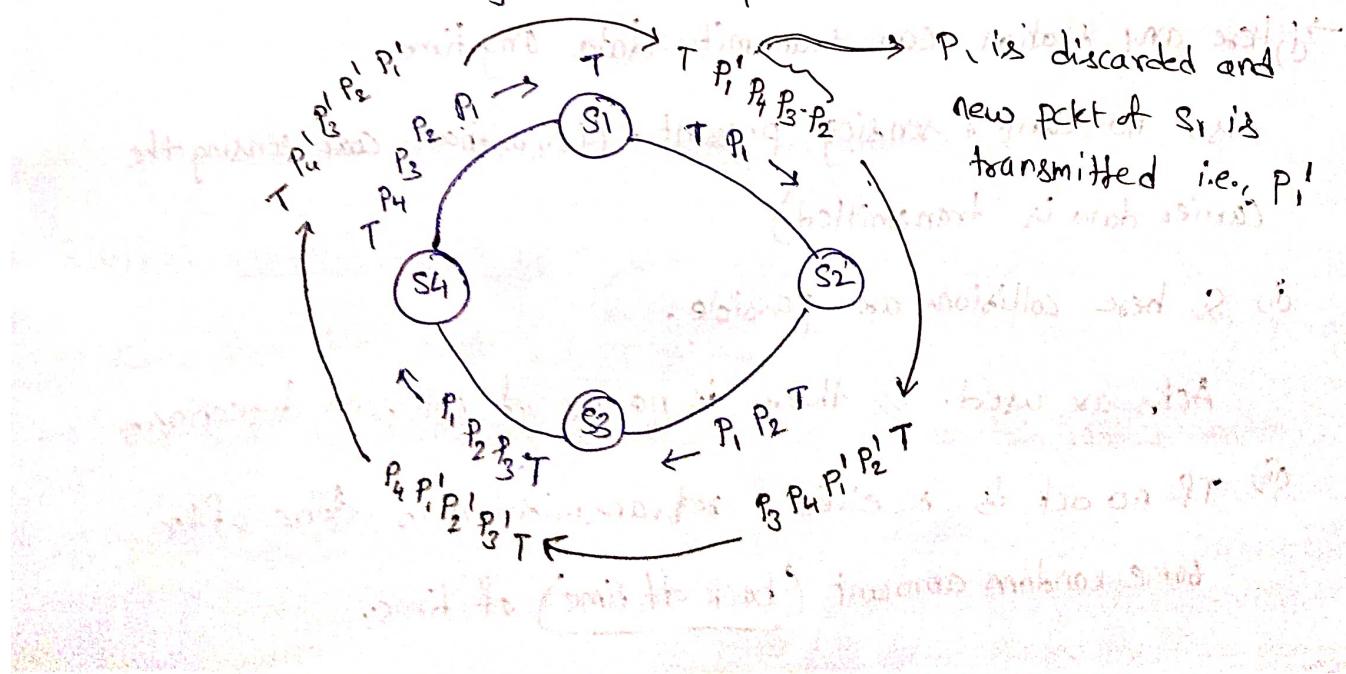
As soon as packet is sent, the station sends ~~that token~~ passes the token to the next station.

→ Assume token is with S1. So S1 sends P_1 and then places the token.

Now S2 after receiving P_1 & T , S2 forwards P_1 , then its sends its data P_2 and then finally forwards T .

Silly S3 forwards P_1 , then P_2 , then its pckt P_3 , then the token T .

Also, in token ring there ~~is~~ sender has to remove the pckt from ring once the pckt comes back to the sender.



So here, $THT = T_t$ (i.e., token is held until packet is transmitted)

So efficiency, $\eta = \frac{N * T_t}{T_p + N * T_t}$

$$\boxed{\eta = \frac{1}{1 + \frac{a}{N}}, a = T_p / T_t}$$

- For efficiency it is clear that early token reinsertion is good
- ETR is efficient but less reliable (\because more than 1 pckt is present at a time on the channel)
- DTR is less efficient but more reliable
- In ETR, at any given time n number pckts are present in the ring.

In DTR, at any given time, only one pckt is present. ~~at any given time~~

ALOHA:

number of stations present is not pair related in a slot

→ It is another access control method.

→ i) Here any station can transmit data any time.

Also no carrier sensing present. (i.e., without ~~carrier~~ sensing the carrier data is transmitted)

ii) So here collisions are possible.

Acks are used. So there is no use of collision detection.

iii) If no ack is received, retransmission is done after some random amount (back off time) of time.

ALOHA is of 2 types:

- Pure aloha
- slotted aloha

Pure aloha: It consists of 2 stations who want to transmit.

→ All the 3 rules apply here. This will be the efficient.

→ If a transmission has to be done, no collision, no one should ~~transmit~~ have transmitted before T_t time and no one should transmit for next T_t time.
 This is known as vulnerable time.

$$\text{Here vulnerable time} = 2T_t \quad (\text{neglecting } T_p)$$

$$\text{Efficiency, } \eta = G * e^{-2G}$$

where $G \rightarrow$ no. of stations who wants to transmit in T_t slot.

$$\frac{d\eta}{dG} = 0 \Rightarrow G = 1/2 \rightarrow \text{i.e., only 1 station transmits in vulnerable time.}$$

$$\Rightarrow \eta_{\max} = \frac{1}{2} * e^{-1}$$

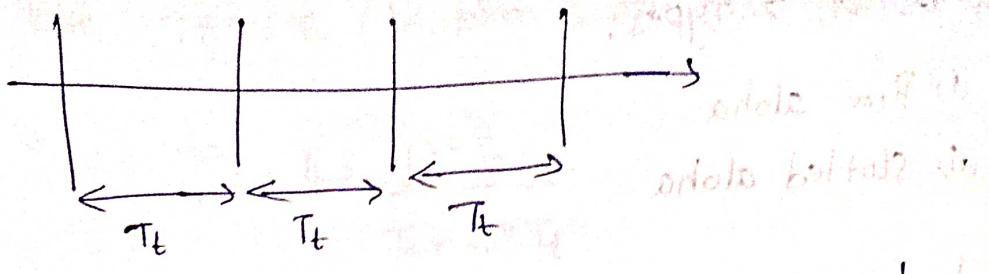
$$\therefore \eta_{\max} = \frac{1}{2e} \approx 0.184$$

∴ Max efficiency of pure aloha = 18.4%.

Slotted ALOHA:

→ Here also the same 3 rule apply.

→ But here the timeline is divided in time slots where each slot is T_t .



Here every station is allowed to transmit only at the beginning of a time slot.

∴ vulnerable time = T_t

$$\text{efficiency}, n = G * e^{-G}$$

$$\frac{dn}{dG} = 0 \Rightarrow G = 1$$

$$n_{\max} = e^{-1} = 0.368$$

∴ Max efficiency in slotted ALOHA = 36.8%

~~slot~~ slot duration

Error Control Methods:



Error Detection

- Detects & req for retransmission

Eg: (D+D), parity check

send data twice 2 times.

and if they are same then no error

thus double data unit is required

CRC, Checksum

Error Correction

- Error is detected and corrected

Eg: Hamming Code

In Hamming code, if d data has

to be sent then $d/2$ more data

has to be sent. All it requires lot of computation. (not used practically)

Cyclic Redundancy Check (CRC)

- Here both sender and receiver will have a common number called CRC generation.
- ~~If~~ Initially, send pads $n-1$ bits to data (on right end) if CRC is n -bits.
- Now perform XOR as shown below.

Let data on sender video belt 1011011000

let CRC generator be 1101

CRC generator is 4 bits.

So pad 3 bits to data

1011011000

Now perform XOR as shown below

$1101)1011011000($

$\underline{1101 \downarrow \downarrow \downarrow \downarrow}$

0110011000

1101

$\underline{000111000}$

1101

$\underline{001100}$

1101

$\underline{0001}$

start XOR
again from 1st
from left
side

CRC no. of bits
so append it
to data

Here we stop the procedure
cuz we have only one bit

∴ Data to be transmitted = data CRC

After obtaining data, receiver does the same procedure with received data. If data received is Crt, CRC will be 0.

1101) 1011011001(

1101

0110011001

1101

000111001

1101

001101001

1101

000000001

CRC=0

∴ Correct data is received.

Assume data has got corrupted and received as 1001011001
I error

1101) 1001011001(

1101

0100011001

1101

010111001

1101

01101001

1101

0000001

∴ remainder ≠ 0

∴ There is an error. So req for retransmission.

→ Sometime CRC generator will be represented as a polynomial.

If degree of CRC generator polynomial = $n-1$, then CRC generator is of n bits and CRC is of $n-1$ bits.

Eg: CRC generator: $x^3 + x + 1 \equiv 1011$

Checksum:

- If we need to compute n-bit checksum, then data is divided into parts each of size n-bits.
- Now convert each part into its decimal equivalent and add all these values.
- Now negate the result obtained and place it in the checksum field.

Note:

while performing above steps, if data part has checksum then we take its decimal equivalent as 0.

Eg: Consider below 8-bit checksum

8	8	8	8	8	8	8	Checksum
5	6	2	1	3	4	$25 - 21$	

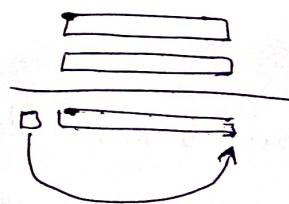
$$\text{checksum} = 5 + 6 + 2 + 1 + 3 + 4 = 21$$

$$\therefore \text{checksum} = -1$$

- Now if the data received by receiver is correct then checksum sum calculated by receiver is zero.

Note:

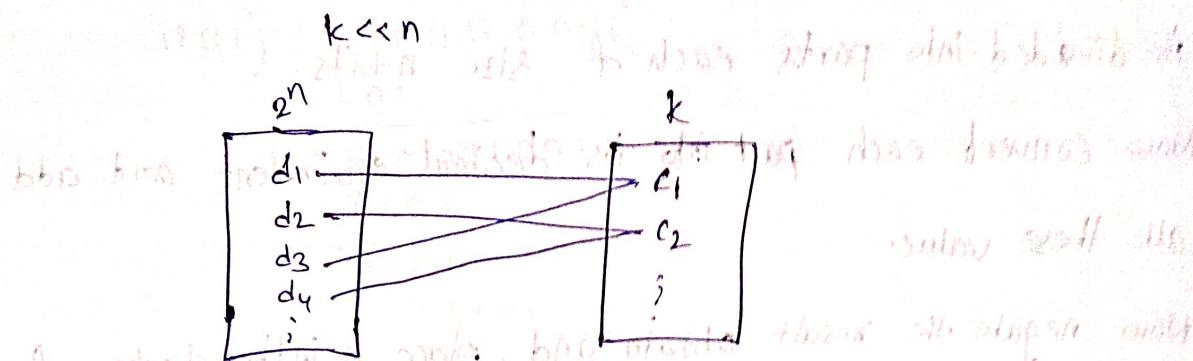
Also while adding two numbers, if there is an overflow, then wrap around carry and add it.



Meaningful Errors: Sometimes errors are possible in such a way that corrupted data appears as a valid data. i.e., both data & detection bits are corrupted in such a way that they form a valid combination.

Note:

Let data be of size of n bits and CRC/checksum is k bits.



d_1 & d_3 produces same CRC/checksum.

So if d_1 is corrupted to d_3 , then this error won't be detected.

→ This problem can be minimized by increasing value of k .

If $k=n$ then the mapping will be one-to-one.

~~still errors are~~,

Even if mapping is $1:1$, meaningful errors are possible.

ISO/OSI Stack

Functions provided by Computer networks

Mandatory	Optional
1. Error Control	1. Encryption
2. Flow Control	2. Checkpointing (while transferring large files checkpoints are used)
3. Access Control	3. Routing
4. Multiplexing & Demultiplexing	
5. Addressing etc.	

These functions are implemented by ISO-OSI, TCP/IP, ATM, ... etc.

ISO-OSI Layers

- Application layer
 - Presentation layer
 - Session layer
 - Transport layer - More functionalities we can test only a specified layer
 - Network layer - Complex functionalities are provided by Network layer (Routing)
 - Datalink layer - Contains H/w & S/w
 - Physical layer - Contains H/w
- i) By dividing all functions into layers, we achieve Encapsulation & abstraction
- ii) Dividing layers makes testing easy because we can test only a specified layer

→ A host need not to contain network layer. Network layer is generally implemented on routers.

But transport layer is present on every host.

Physical Layer:

Physical layer deals with electrical, mechanical, functional and procedural characteristics of physical links.

→ Depending on propagation medium signals has to be converted appropriately i.e., Copper - Electrical signals ; optical - light ; wireless - electro magnetic waves

→ Functional properties means types of transmission modes.

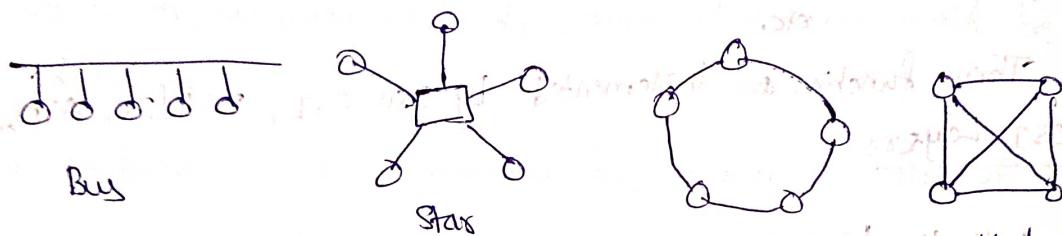
Simplex: Communication is one way

half duplex: Communication is possible in both way, but only one a time.

full duplex: Communication is possible in both ways at same time.

(In gate for all questions we consider full duplex by default)

→ Physical layer also deals with topologies



see original which contains no printed title

periferia de Madrid y de la capital, que se extiende por el norte y el este de la ciudad.

Hybrid is a word used to represent frequency distribution which is a combination of two or more different distributions.

→ Physical layer also does the job of encoding:

The ~~co~~ i.e., converting bits into electrical signals.

we have 2 types of encodings:

iii) Manchester encoding

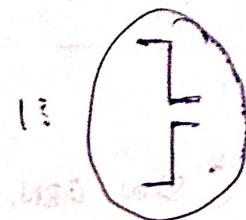
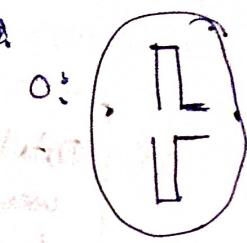
(ii) Differential Manchester encoding.

i) Manchester Encoding :

ben kent hand gerichteten Abstecher alias Clark royal typograph

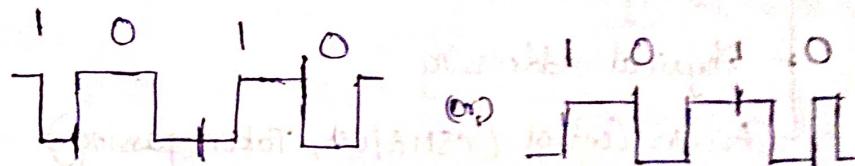
1: [05] *california state report*

iii) Differential Manchester Encoding



Here we have 2 representations for 0 & 1.

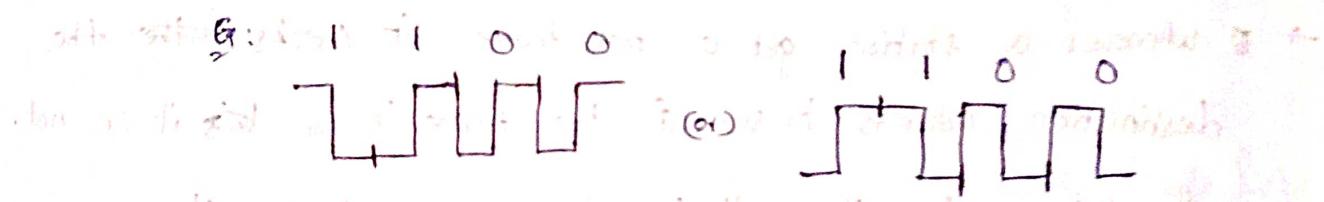
Eg:



1 0 1 0 1 0

0 1 0 1 0 1

against 3 the voltages are positive. This is followed by



so after a 1 there will be no voltage. In this bit 0 0

Based on the representation we use for the 1st bit we get 2 different encodings. In the next part we will discuss with which bit state of voltage it helps. Now we have seen

$$\text{Baud rate} = 2 \times \text{Bitrate}$$

Baud rate is no. of voltages sent per second.

For each bit 2 voltages are sent.

→ PL also provides bit synchronization, bit rate control.

Data Link Layer:

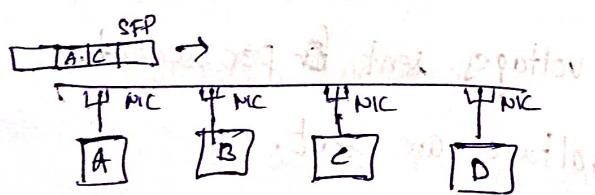
functionalities

- Flow Control (S & W, GBN, SR)
- Error Control (CRC, checksum) (Used in TCP/IP)
- Framing
- Physical Addressing
- Access Control (CSMA/CD, Token passing).

Data link layer uses CRC as it is more efficient to implement at HW level

Framing:

- Generally in LANs stations are connected with bus topology.
- whenever a station gets a ~~pet~~ frame, it checks ~~for~~ the destination address to see if the frame is for ~~this~~ it or not.
- So for this, rather than all the stations monitoring the channel, we used SFD (Starting Frame Delimiter) at the beginning of ~~for~~ every frame and it will be detected by NIC and NIC will alert to station to check the destination address of the frame



SFD is generally of for $(10)^*$ 11

There will be no problem if data match with SFD because one SFD is found frame is read till end

- Also there may be multiple frames on a channel at a time. So a station should read only the frame meant for it. For this purpose we should be able to detect the ending of frame. This detection is based on the framing technique we use.

Framing is of 2 types:

Fixed length framing

- Here, knowing the starting delimiter (SFD) frame size is enough. (Ending delimiter (ED) is known implicitly)

- Disadv: If data to send is less than the frame size then we need to pad dummy bits.

- In variable length framing,

- the problem with length field is that, if the length field gets corrupted, then the end will be identified incorrectly.

However in LANs error probability is less and hence we can use it.

- the problem with End delimiter is that, there is a chance that data pattern may match ED. This problem is dealt in two ways:

i) Character stuffing / Byte stuffing

ii) Bit stuffing

Character Stuffing:

- For every pattern in data, that matches with ED, we add a null character as a prefix to it.

Variable length framing

- Here starting is identified using SFD. Also we need to know the ending. This can be done in 2 ways:

Using length field
Using end delimiter (ED)

ED [frame] SFD

- This is the mostly used framing

~~Ex~~ If ED is \$ and data has \$; then sender will add null and receiver will remove the null.



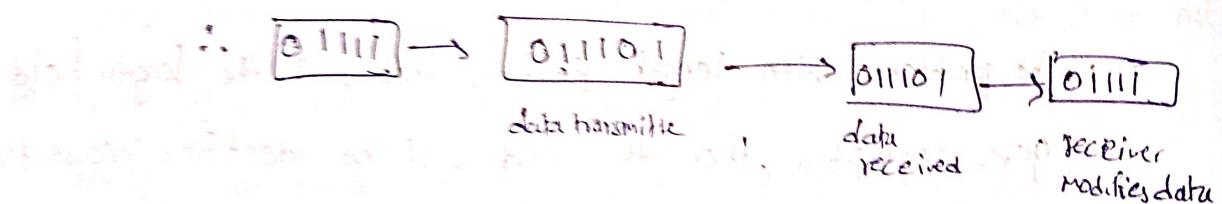
→ If data has \$ & N\$ then sender will change it to N\$
i.e., for every null add a null and for every \$ add a null.

Disadv: Adding character requires more space.

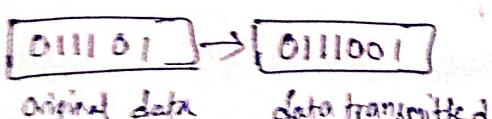
Bit Stuffing:

Let ED be 01111.

If ~~only~~ 01111 is found in data then sender will add 0 in ~~this~~ position.



But consider the case where data transmitted itself is 011101. In this case also receiver removes the 0. So to deal all possible cases, we come up with a strategy where we ~~will~~ add a zero after 0111 ~~and~~ and receiver removes zero after 0111 is found. So here sender adds 0 even in the cases where there is no ED pattern in data.



→ Similarly if ED is 011111, then we add 0 after every 011111.

Eg: If ED = 01111 and data is 011100011110

Find data that should be transmitted.

01110 000111010

↑ ↑

Hex bit is

stuffed even

without match

Eg: If ED is 10000, then we add 1 after every 1000

Physical Addressing:

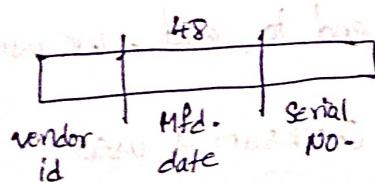
logical address: It ~~is address unique~~ It is an address which is unique across the world

physical address: It is an address which is unique within a network.
i.e., two system in two different NW may have same physical address.

→ IP address is a logical address. It is a SW number.

→ MAC address is a logical address. It is a H/w number.

MAC address is 48-bit address



Even though MAC is unique, IP add is used in communication.

Because IP add has fields (PID, SID, HAD) that help in routing.

→ Physical address is constant and doesn't change.

Generally MAC is used as physical address even though it is a logical address.

→ DLL is further divided into 2 sublayers

i) Logical Link ~~Layer~~ Control (Flow control)

ii) Medium Access Control (Framing, access control, physical addressing)

Error Control

Transport Layer:

Functions:

- End to End Connection: It is used to establish end to end connection.

connection, i.e., process to process



It uses port numbers for this. This is known as SAM & SPAM.

Service Point Addressing

- Flow Control: Since it is end to end, we use SR protocol.

- Error Control: Mainly checksum is used.

- Segmentation & Assembly: Data received from upper layers may of different sizes. This is ~~changed~~ segmented into different sizes.

Multiplexing & Demultiplexing

Transport layer on sender side has to take

Network

Functions:

Ho

Session

Functions:

Co

message from many processes and send it through a single channel.

Also TL on the order side take data from one channel and give it to different processes.

Congestion Control

Network Layer:

Functions:

Host to Host Connectivity: It transmits data from host to host (but not from process to process)



Logical Addressing: Identifying hosts

Switching: Switching is process of connecting various networks.

Routing: Routing is building routing tables.

Later using this routing table is called switching

Congestion Control:

Fragmentation: Different Networks may support different maximum allowable data size.

So in this case fragmentation is used.

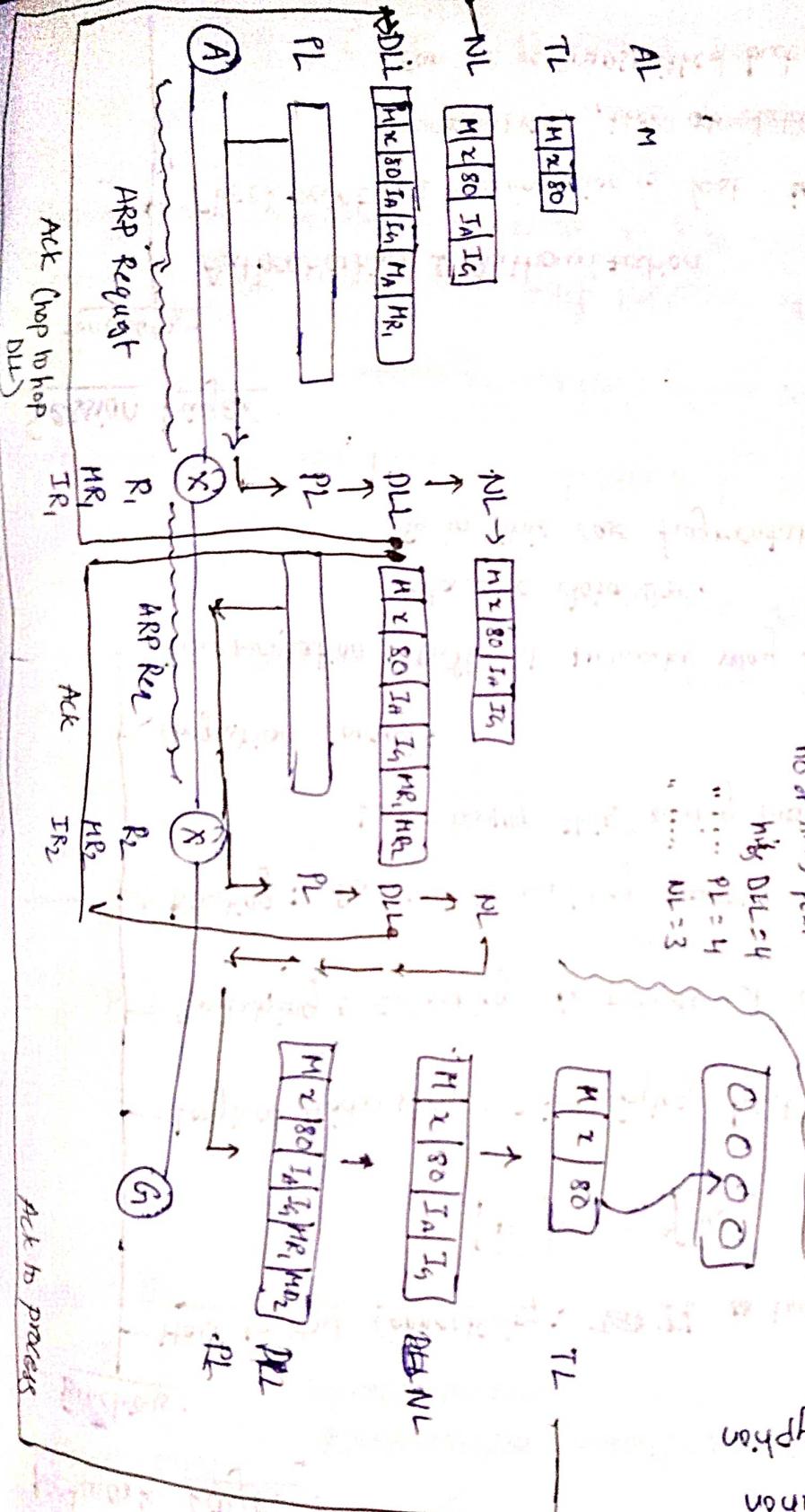
Session Layer:

Functions:

Authentication & Authorization

Checkpoint: If connection is lost while data is being transmitted, then data from previous checkpoint can be retransmitted but not all.

How all the layers work



On sender side:

In TL, port num of source & destination are added
↳ well known port.

In NL, source IP & Destination IP are added
↳ found using DNS

In DLL, source MAC & Destination MAC are added
↳ found using ARP
MAC of the default gateway.

Now this moved to PL and from there it will be sent to default gateway router

At Router R_i:

- Routers have only 3 layers: PL, DLL, NL
- Based on the type of N/wc: router is connected - it may have several DLLs and several PLs but only one NL
- Before router forwards to next router, it changes source MAC & destination MAC.
- But source IP and destination IP are kept same.
- Also DLL of R_i sends ack to DLL of host

At G:

- NL uses UDP protocol. So no ack will be sent.
- TL sends ack to the sender.
- Using port number the pckt is sent to appropriate process.

Ethernet (IEEE 802.3)

It is one of the LAN technologies.

→ Topology : Bus

→ Access Control : CSMA/CD

→ No acks are used.

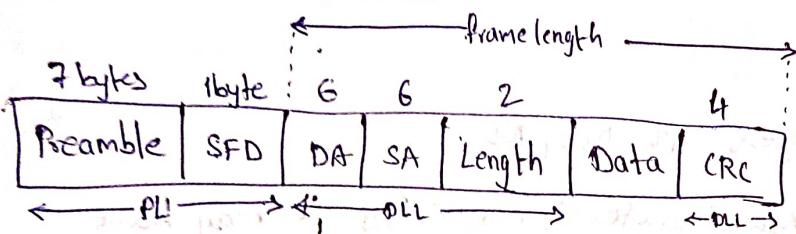
However if an application needs ack then it may send ack as if it is data.

→ Data rate/Band width : 10 Mbps, 100 Mbps, 1 Gbps

Fast ethernet

Gigabit ethernet

→ Encoding technique ; Manchester

Ethernet Frame Format:

Preamble : 10101010...1010

SFD : 10101011

Data at diff layers:

AL - Message

TL - segment

NL - Datagram

DLL - Frame

PL - Single Protocol

- Data Unit (IPDU)

• Preamble & SFD are not added to the frame.

PL just sends preamble & SFD just before sending it.

• Preamble & SFD alerts other stations indicating start of the frame.

• Preamble & SFD also provides synchronization.

• DA & SA are MAC address of destination & source.

∴ It is 6 bytes

Types of MAC addresses:

i) Unicast: ~~Least significant bit of 1st byte is 0.~~

SA is always a unicast address. Otherwise the frame is invalid

ii) Multicast address: Least significant bit of 1st byte is 1.

For multicasting, we ~~go~~ assign MCA (Multicast Address) to a set of stations of the group.

Now all stations after reading DA; if the DA matches with their unicast or multicast address, then they will read the data.

iii) Broadcast address: All the bits are 1's excepting all 0's.

When a station sees all bits as 1's, then it will accept it.

→ Length field is used to tell the length of the frame. ~~It is 2bytes.~~

Ethernet uses variable frame length. So we used length field.

Even though frame length is variable, since ethernet uses CSMA/CD, the frame must be of some minimum size.

$$\Rightarrow T_f \geq 2 T_p$$

$$L \geq 2 T_p * BW$$

$$L \geq 64 \text{ bytes} \quad (\text{from standard value of ethernet})$$

↳ from D.A to CRC

Also maximum data (not frame) that can be sent is 1500.

This restriction is made so that one user won't be sending large frames and use the line.

i.e., to avoid monopolization.

	Min	Max
Data	46B	1500B
Frame	64B	1518B

→ CRC is 4 bytes i.e. 32 bits

∴ we use ~~32~~ CRC polynomial first degree 32

Disadvantages: (possible) ATM up to now, disadvantages

→ It is not applicable for real time application.

→ i.e. in applications where sending data within time is important.

→ The minimum size of data to be sent is 46B.

so it is not applicable for interactive applications.

→ No priorities are present among stations.

∴ it is not good for client-server application where server has to be given more priority.

Token Ring (IEEE 802.5)

→ It is another LAN technology that overcomes the drawbacks of Ethernet. But it has its own disadvantages.

→ Topology: Ring

→ Access Control: Token passing

(Grants to active hosts next to offend host)

→ Data sent: unidirectional (clockwise or anticlockwise)
∴ links used are simplex

→ Data rate: 4Mbps, 6Mbps (smart token ring maximum data)

→ Piggy back acknowledgments are used.

→ Encoding: Differential Manchester encoding.

Sender side Problems:

In token ring, once data is placed on the link, it is removed by the sender only. Due to this there are 2 problems possible.

i) Orphan packet Problem:

If a packet is placed on link and if the sender goes down by the time the packet comes back, the packet will never be removed from the link.

ii) Stray packet Problem:

By the ~~sent~~ time sent packet comes back to the sender,

if the pkct is corrupted enough that sender cannot

recognize it, then the packet will remain on the link.

To solve the above two problems we introduce new station called monitor in the ring. Monitor is ~~one~~ one of stations in the ring.

~~If sender doesn't remove~~

~~stray~~

i) A monitor bit is maintained, which is initially 0.

When this pkct passes through the ~~monitor~~ monitor, it sets the

bit to 1. When pkct comes to the monitor for second time, the pkct will be removed.

ii) CRC is present in every pkct. Whenever the packet is corrupted monitor can detect it. ~~CRC will be changed~~. Monitor removes the packet if it detects this change.

Problems on Destination:

Top

i) Destination is down:

In this case sender should ~~not~~ resend.

ii) Destination is busy:

In this case destination buffers may be full or destination may be ~~very~~ busy. So ~~the~~ sender has to resend the exam.

iii) Packet corrupted:

In this case pckt is corrupted & sender has to resend.

iv) Copied:

Here destination receives pckt and hence there is no problem

To overcome these problems we use 3 special bits ~~A, C, E~~ in header

A - Available (availability of destination)

C - Copied

E - Error

A	C
---	---

0 0

— Initial

1 1

— Destination available & pckt is copied

1 0

— $E=0$, Destination busy
 $E=1$, pckt is erroneous

0 0

— Destination is down

0 1

— Invalid

→ This A & C are kind of acknowledgments.

So we say it is piggy backing acknowledgment.

→ Also while retransmitting, sender has to reset monitor bit otherwise monitor will remove the pckt.

→ A & C are used ~~not~~ only for unicasting.

Token Problem

i) A station send large amount of data without releasing the token. This will lead to a problem of monopolization.

This is known as Captured token.

So to overcome this, we impose restriction on maximum THT.

Default value of max THT = 10 ms

BW = 4 Mbps

→ Maximum frame size that can be sent with ETR

$$\begin{aligned} \text{Max data sent} &= \text{THT} \times \text{BW} \\ &= 10 \times 10^{-3} \times 4 \times 10^6 \\ &= 40000 \text{ bits} \end{aligned}$$

→ Maximum frame size that can be sent with DTR

$$\begin{aligned} \text{let THT} &= 10 \text{ms} \quad \text{BW} = 4 \text{Mbps} \\ \text{RL} &= 9 \text{ms} \end{aligned}$$

within THT (10ms) transmitting
and getting the pkt back should
happen.

$$\begin{aligned} \text{Overall round trip time of } 23 \text{ms} \text{ is not possible} \\ \therefore \text{THT} &= T_f + RL \end{aligned}$$

$$\Rightarrow T_f = 10 - 9 = 1 \text{ ms}$$

$$\begin{aligned} \therefore \text{Max data sent} &= 10^{-3} \times 4 \times 10^6 \\ &= 4000 \text{ bits} \end{aligned}$$

∴ In ETR, transmission can be done

$$\therefore \text{In ETR, } T_f = \text{THT}$$

$$\text{In DTR, } T_f = \text{THT} - RL$$

while solving problems, take
ring latency as ~~approx~~ as
 $RL = T_p + N \frac{b}{R}$
If bit delay is not
given then take $RL = T_p$

(ii) Token is Lost:

Assume a case where a station takes a token and the station is went down.

In this case monitor will ~~not~~ generate a new token.

For that monitor has to be sure that token is ~~lost~~ but not delayed.

In best case

$$\text{minimum token return time} = RL$$

In worst case

$$\text{maximum token return time} = RL + NTHT$$

RTT (Round Trip Time (TRT))

\therefore If token doesn't return to the monitor within $RL + NTHT$, monitor will generate new token.

Eg: $RL = 10ms$; $NTHT = 10ms$; $N = 10$;

$\text{Max TRT} = 10 + 10 \times 10$

$= 110ms$

(iii) Token is Corrupted:

In general, token size is 3 bytes. So since it is small we don't use any CRC for it.

So if a corrupted token comes to monitor, monitor assumes it's some data which has split from some data and monitor discard.

In this case also monitor generates token after maximum token return time.

Monitor Problems

i) If monitor goes down, there will be lot of problems with stray pkts, orphan pkts etc.

Due to this communication will halt at some point.

To overcome this problem, monitor sends heart beat message periodically saying that it is present.

This heart beat messages are implemented using

AMP frames

↳ Active Monitor Presence

Every station expects AMP frames in regular interval of time.

If monitor goes down now, using polling, one of the existing hosts will be selected as monitor.

(ii) Monitor is hacked (Monitor Malfunction)

In this case monitor is present but as it hacked it sends AMP frames without doing any work.

The only ~~problem~~ solution for this problem is human verification.

If communication is found to be halt then we will check if monitor is malfunctioning.

Advantages of Token ring:

→ It is applicable for real time applications since there are no collisions.

- It is also suitable for interactive applications as there is no limitation on minimum size of data that can be sent
- It is also applicable for client-server applications by giving priorities to stations.

Token Ring Frame Format:

Data frame:

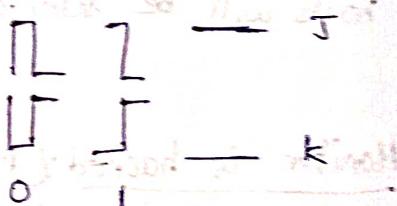
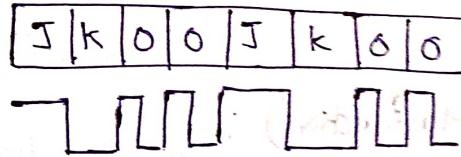
1	1	1	6	6	4	1	1	(in bytes)
SD	AC	FC	DA	SA	Data	CRC	ED	FS

Token:

1	1	1
SD	AC	ED

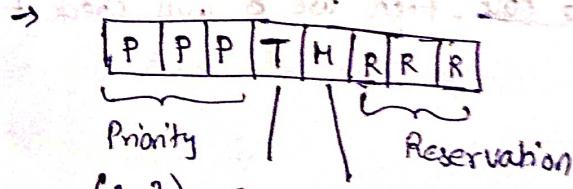
Want to avoid collisions so if any station violates priority

SD: Used for alerting that frame is being transferred
and also performs synchronization
(Start Delimiter)



Access Control:

→ This field is of 1 byte size

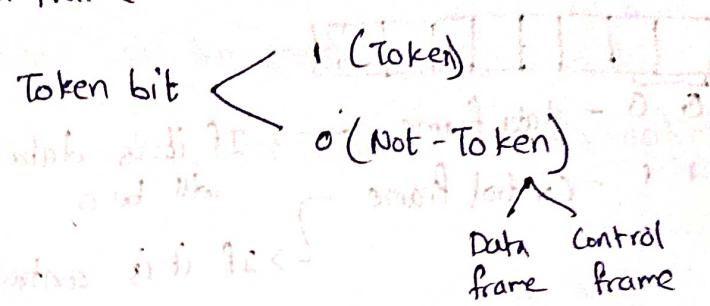


(0-7) Token Monitor Bit bit

* 0-7, eight levels of priorities are possible.

J & k doesn't represent any data. They are just invalid patterns. So using it in SD, we can make sure that SD never occurs in data.

→ Token bit is used to say whether the frame is token frame or data frame



Priorities & Reservations:

Using these we implement priorities.

→ Every host is assigned a priority.

→ If a host has to transmit it will put its priority in reservation field - saying that it needs that token.

A host can update reservation field only if its priority is more than the content in reservation field.

→ Assume DTR is being used.

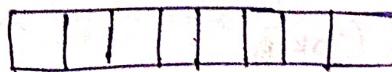
sender sends data packet and waits until it comes back. As data pkct passes through all the hosts each host if needed will modify the reservation field.

when sender gets back the data field, it copies the reservation field of AC to reservation field of token.

Now this token can "be held" by a host only if its priority is greater than or equal to the content in the reservation field.

→ In Client server applications, Server will be given priority 7 and all other host will be given a priority of 0.

Frame Control:



0 0 - data frame → If it is data frame then all fields will be 0

1 1 - Control frame →

→ If it is control frame then the remaining 6 bit will say which type of control frame it is.

Examples of control frames:

• AMP

Polling - used at the time of polling

Purge - used when some maintenance is going on so that no one will send data

Beacon - It is used find if there is any cut in the frame.

DA & SA:

byte

6 MAC addresses

(one) bits How many bits frame can contain

CRC:

16 bits of 4 bytes are used to detect error

• CRC polynomial of degree 32

End Delimiter: ends off that stop sending data

Used to detect end of the frame.



Information Error
bit

→ Information bit = 1 → there is some of data following.

As there is restriction on max THT, a station may not be able

to send all the data in one THT.

So it uses information bit to tell the destination that some more data will be sent.

→ If destination detects any error then destination will set E.

Frame Status:

A	C	O	O	A	C	O	O
---	---	---	---	---	---	---	---

→ why 2 copies of AC?

* CRC is computed only on field that are left to FS but not on FS & ED.

It is because AC bit will be changed on receiver side.

So to have more reliability we maintain 2 copies of AC.

→ Max size of data depends on max THT with priority bits.

→ If ETR is used then priorities are implemented differently as

token has to be released as soon as sender sends the data.

So here based on the latest pkt (pkt in last cycle) seen

by the sender, sender gets the reservation details and puts it on the AC field of the token.

Minimum length of token Ring:

Assume a case where all the hosts are down.

In this case, monitor places token and receiving it quickly. So there is a chance that monitor gets 1st bit of token back before it finishes the transmission.

In this case collision is possible.

To overcome this we impose restriction on min length of ring. The min framed sent is taken. Its size is 24 bits.

Propagation delay \geq transmission

\Rightarrow Min propagation time of 24 bits

$$\Rightarrow \frac{d}{V} \geq \frac{24}{B}$$

$$\Rightarrow N \geq \frac{24 \times V}{B}$$

Capacity ≥ 24

$$T_p + BW \geq 24 \text{ (for simplex)}$$

$$\frac{d}{V} + BW \geq 24$$

$$\frac{d}{V} \geq \frac{24 \times V}{BW}$$

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S
/S

Ex: If speed of both bits in fiber is 200 m/sec & $BW = 4 \text{ Mbps}$; $V = 2 \times 10^8 \text{ m/sec}$

Min length of ring \geq 1.2 km

$$\Rightarrow \text{Min length of ring} \geq \frac{24 \times 2 \times 10^8}{4 \times 10^6} = 1200 \text{ m} = 1.2 \text{ km}$$

But having this large link may not be possible.

In that assume we have 1km link. In this case we add a delay device.

The delay produce by this device should be equal to delay produced by a 200m link.

$$\therefore \text{delay req} = 200 \text{ m}$$

$$= 200 / V \text{ sec}$$

$$= 200 / 2 \times 10^8 \text{ sec}$$

$$= 10^{-10}$$

$$= \frac{200}{2 \times 10^8} \text{ sec}$$

$$= \frac{200}{2 \times 10^8} \times BW \text{ bits} = \frac{200 \times 4 \times 10^6}{2 \times 10^8} \text{ bits}$$

$$= 4 \text{ bit times}$$

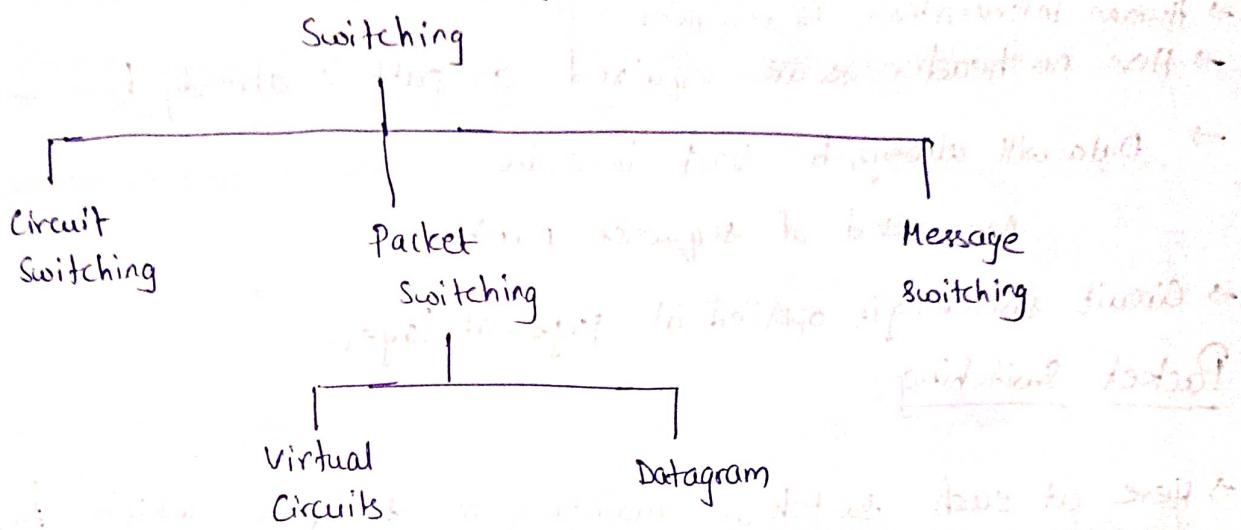
. now assume that every station adds 1 bit delay, then

min no of stations to be present in the ring = 4

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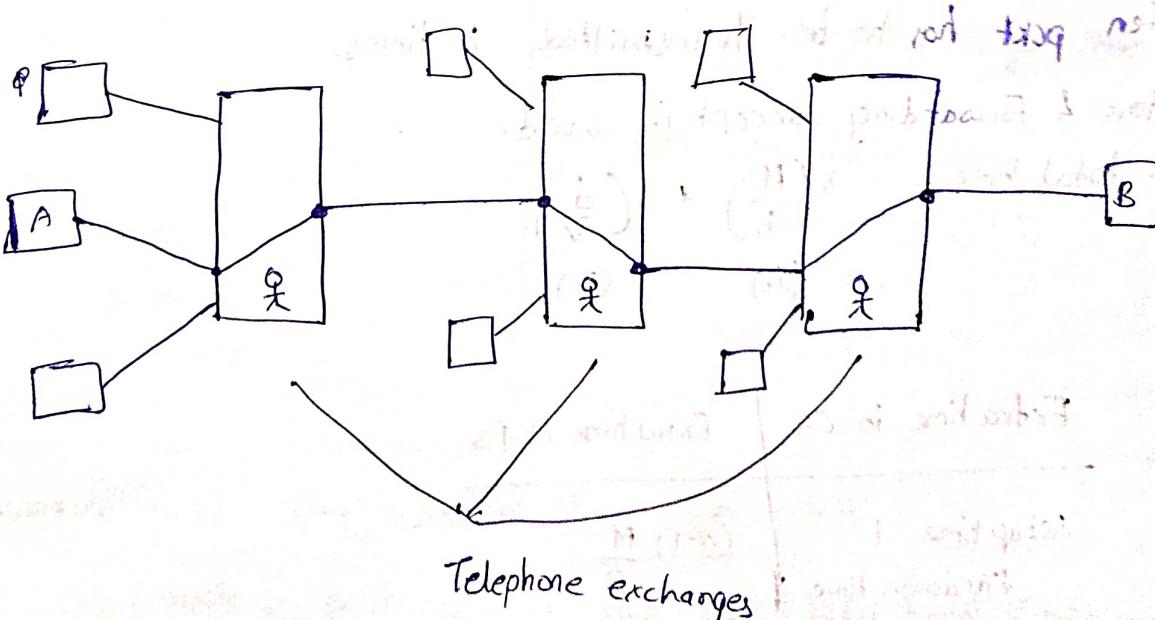
Switching:

Switching is process of connecting different networks.



Circuit Switching:

Physical Connection and disconnection is made physically.



$$\therefore \text{Total time req} = \text{setup time} + \frac{M}{B} + x \left(\frac{d}{v} \right) + \text{teardown time.}$$

$$(T_t) \quad (T_p)$$

$M \rightarrow$ size of message

$B \rightarrow$ Bandwidth

$x \rightarrow$ no of hops

$d \rightarrow$ distance b/w two hops

$v \rightarrow$ velocity of signal.

setup time \rightarrow time to setup connection

teardown time \rightarrow time to remove connection

\rightarrow Human intervention is required.

\rightarrow Here no header ~~are~~ are required as path is already been setup.

\rightarrow Data will always be sent in order.

\therefore No need of sequence numbers.

\rightarrow Circuit switching is applied at physical layer.

Packet Switching:

\rightarrow Here at each switch we maintain a software which forwards packet to appropriate switch.

\rightarrow Here pkts has to be transmitted at each switch.

\rightarrow If there are x no of hops \rightarrow b/w source & Destination
then pkt has to be transmitted x times.

\rightarrow Store & Forwarding concept is used.

$$\therefore \text{total time} = x\left(\frac{M}{B}\right) + x\left(\frac{d}{v}\right)$$

$$(T_E) \quad (T_P)$$

Extra time in CS | Extra time in PS

$$\text{setup time} + \frac{(x-1) M}{B}$$

↓

teardown time

Thus for large data, CS is ~~better~~ better

for small data PS is better

Total

→ H

→ S

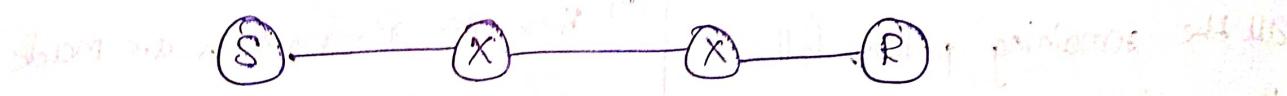
Packetization in packet switching (pipelining):

- Here data to be transmitted is divided into packets so that pipelining can be obtained.
- By this two switches will be transmitting parallelly.

Eg: Consider data = 1000 B; BW = 10^6 MBPS; Header = 100 Bytes.

Assume $T_p = 0$ for easier comparison

without packets Consider we have 3 hops b/w sender & receiver



without packetization

$$T_t = \frac{1000}{10^6} = 100 \mu\text{s}$$

Data to be transmitted

$$= 1000 + 100 = 1100 \text{ B}$$

$$\therefore T_t = \frac{1100}{10^6} = 1.1 \text{ ms}$$

Total time

$$= 3 \times T_t$$

$$= 3 \times 1.1$$

$$= 3.3 \text{ ms}$$

with packetization

Consider we ~~transmit~~ divide data into S pkts.

$$\Rightarrow \text{Size of each pkt} = \frac{1000}{S} = 200$$

$$\therefore \text{Size with header} = 200 + 100 = 300$$

$$\therefore T_t \text{ of a pkt} = \frac{300}{10^6} = 0.3 \text{ ms}$$

Total transmission time

$$= [3 + (S-1)] T_t$$

$$= 7 \times T_t$$

$$= 2.1 \text{ ms}$$

→ However pkt size cannot be too small.

For example if pkt size is 50 B then total time = 3.3 ms

and this equal to original value.

These are 2 types of pckt switching

- (i) Virtual circuits (Eg: ATM)
- (ii) Datagram (Eg: IP)

In both the types data is divided into pkts.

Virtual Circuit

(i) Here all the pkts follow same path

(ii) As the 1st pkt is transmitted all the remaining pkts follow the path taken by 1st pkt. So as the 1st pkt moves along the path it makes reservations at each switch for CPU, BW & buffers.

(iii) Connection Oriented

(iv) 1st pkt has global header and remaining pkts have local header.

global header help in finding path from source to destination whereas local header just helps the pkt follow the 1st pkt

(v) Same path is followed and hence pkts arrive in order

(vi) Reliable
with costly

Datagram

(i) Here each pkt follows different path and is transferred independently.

(ii) Here no reservations are made and path taken by each pkt may be different.

(iii) Connectionless

(iv) All the pkts must have header

(v) Different path are followed and hence pkts may arrive out of order.

(vi) Not reliable
(vii) Not costly

Interne

IPH Header

ver

Min

Header

Max

But

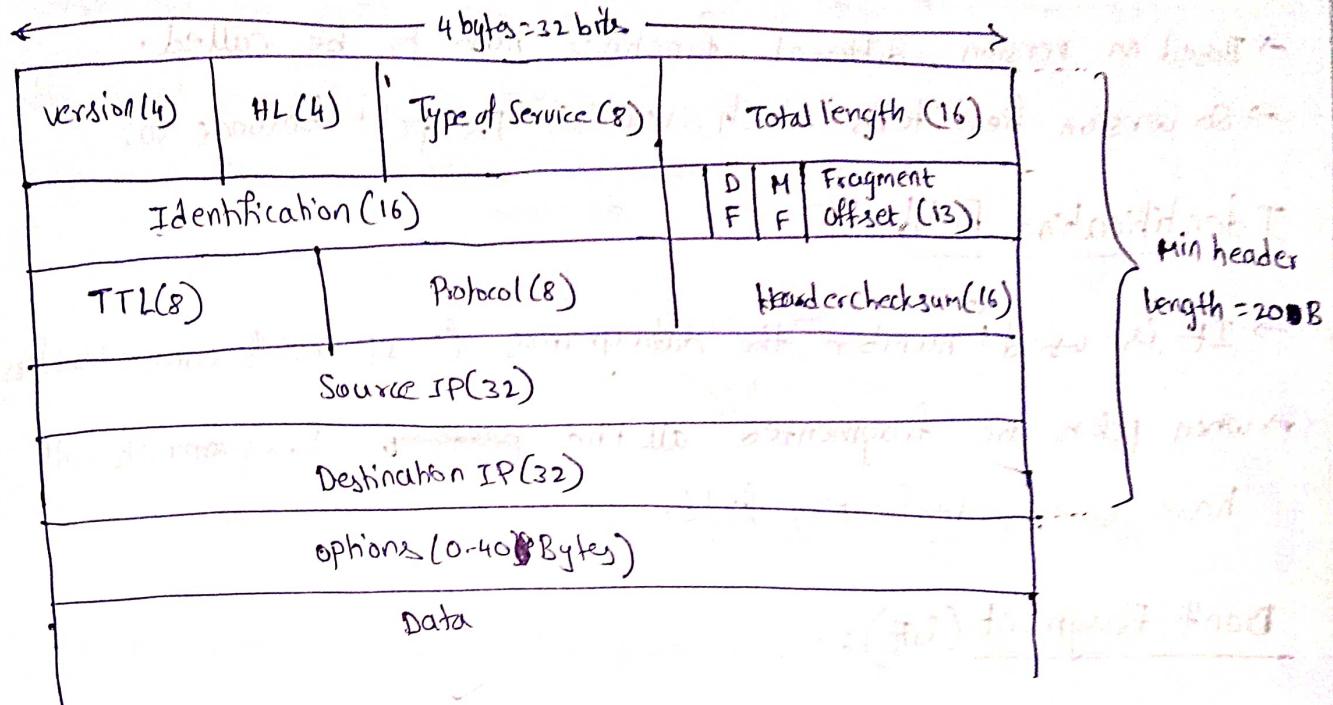
Note:

IP

So in

Internet Protocol:

IPv4 Header:



Min header = 20 Bytes; Max header = 20 + 40 = 60 Bytes

Header length (HL):

Max value possible in HL is $2^4 - 1 = 15$

But max header length is 60

so to obtain header length, we multiply header length field with 4

$$\text{Header length} = 4 * \text{Header length field}$$

Range of header length : 20 - 60

Range of header length field : 5 - 15

In exam,
Based on value given,
we understand which is
given

Note:

If header length is 30 then Header length field = $\frac{30}{4} = 7.5$

So in this case we put 8 in HL field and pad 2 bytes to the header.

→ In worst case we may need to pad 3 bytes.

Version:

- IP has various versions (IPV4 to IPV6)
 - Based on version, different functions have to be called.
 - So version field tells which version the packet belongs to.

Identification Field

- It is used to number the datagrams (\therefore IP uses connectionless)
 - when pkts are fragmented all the ~~packets~~ fragments will have same identification field.

Don't Fragment (DF):

- If DF = 1, then routers are not allowed to fragment the pkts.
 - In this case if router cannot forward it, the router will discard the pkt and will send an ICMP error message to sender.

More Fragment (MF)

- If $MF=1$, it means that more fragments are following.

Fragment offset.

- when packet is fragmented, ~~each~~ ^{at fixed interval} the fragment offset say no of ~~fragments~~ data bytes which are ahead of this fragment.

~~ier 1st segment gets fragment block~~

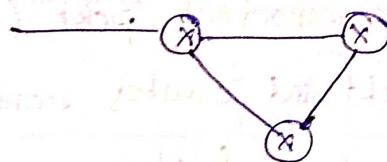
Time to live (TTL)

91

→ when a pckt is being routed, if a router don't know which now the pckt has to be forwarded to, it forwards the pckt to default router.

~~Sometime there may be a cycle~~

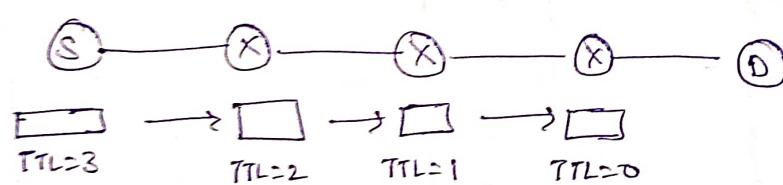
Eg:



Here the pckt will loop b/w the 3 hops infinitely.

So to ~~solve~~ overcome this problem we used TTL field.

→ The value in TTL say max no of hops the pckt can take.
After every hop the value in TTL is decremented.



∴ Discard the pckt.

→ Destination also decrements TTL

→ The main purpose of TTL is to ~~not~~ discard pckts that go into infinite loop.

Protocol

TL	TCP, UDP
NL	ICMP, IGMP IP

The datagram may follow ICMP or IGMP or TCP or UDP.

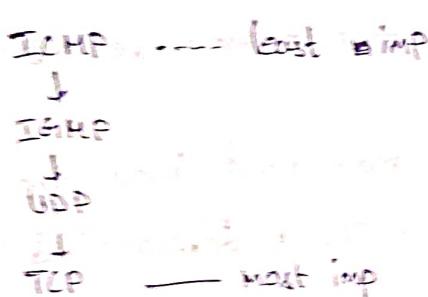
To indicate which protocol is being followed we used the protocol field.

Actually identifying the protocol that is being followed is possible by looking into pkts.

But routers can't do that because we don't have transport layer in routers.

→ Also when router buffer is full it discards new incoming packets. However if incoming pkt is more important (say TCP) and router's buffer has less important pkt (say ICMP) then router will discard ICMP pkt and creates room for TCP pkt. So to do this we need protocol field.

→ The order in which packets are discarded is



Header Checksum:

→ Header checksum is checksum of header part alone. (but not of data)

→ Finding checksum:

 i) Divide header part into smaller parts each of size 2 bytes.

 ii) Treat each part as a number and add them.

 iii) Now negate the result and store it in the checksum field.

while computing the header checksum, we take value corresponding

to header checksum is 0.

→ At every router checksum has to be recomputed as value of TR~~is changed~~ (fragment offset, MF, total length), (options, header length) may change.

Hence checksum is computed only header. If it were to be computed ~~log~~ on data, every router has to do lot of computation.

→ However the protocol of the pkt will compute the checksum on the data part (on sender side)

Source IP & Destination IP

IP add		Description	Used as	
NID	HID		Source IP	Dest IP
✓	✓	Valid IP add	✓	✓
✓	0's	Network id	X	X
✓	1's	Directed Broadcast	X	✓
1's	1's	Limited Broadcast	X	✓
1's	0's	Network Mask/Subnet mask	X	X
0's	✓	Host within same NW	✓	✓
0's	0's	I don't have IP (A host requesting for IP)	✓	X
127	✓	Loop back address	X	✓

Options

→ This field is optional and it provide some options

i) Record Route:

This is used to find the path followed by the pkt.

→ Every router adds its IP address if the pkt passes through it.

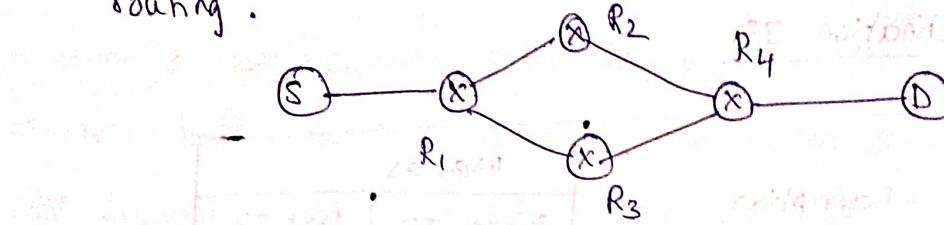
Size of IP add is 4B and option field is 40B.

So max no of router address possible is 10, but we also need to mention what kind of option we are storing.

So in practice max no of router addresses possible to be stored e.g.

(ii) Source Routing:

Using this option sender can select the path that has to be followed by the pkts. This is known as source routing.



Source routing is ~~2 types~~ said to be strict source routing if entire path to be followed is given. Otherwise it is called loose source routing.

strict source routing: E.g. [R₄R₂R₁]

loose source routing: E.g.: [R₁]

→ This option is used by N/w administrator to test if the path is working or not.

→ Also both options record route, source routing ~~is~~ not allowed to be used by user. They are generally used by N/w administrator.

(iii) Padding:

If ^{size of} options added is not a multiple of 4, the extra bits are padded.

(iv) Time Stamp: Using ~~this~~ this option, we record in-time and out-time of a pkts at each router in the path. Using this we can know the delay at each router.

Total length:

It specifies length of ~~packet~~ ^{datagram} including header.

This is 16-bit field.

$$\therefore \text{Max possible length} = 2^{16} - 1 = 65,535 \text{ (Data + Header)}$$

Min header size = 20B

$$\therefore \text{Max possible size of data} = 65,515$$

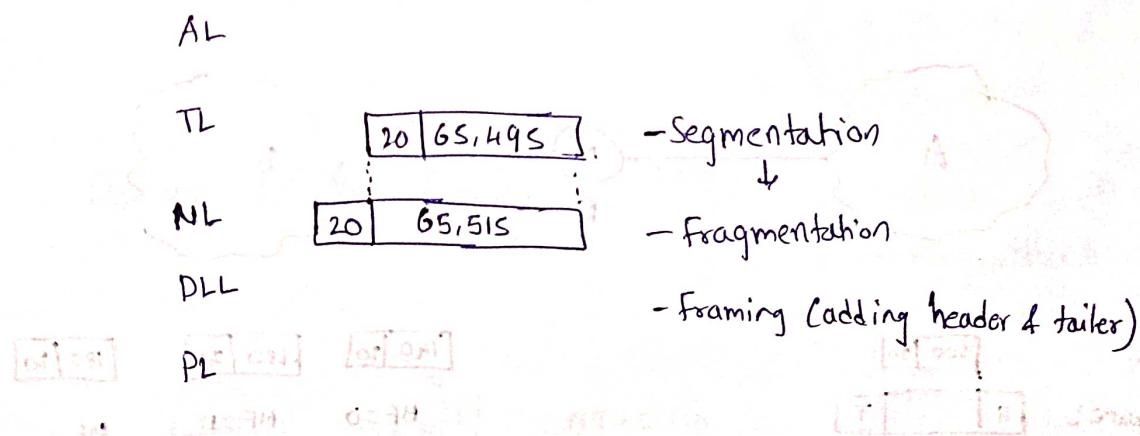
\therefore Maximum segment size at transport layer = 65,515

Also TL has 20B header.

$$\text{So max payload at TL} = 65,495.$$

But App. layer may provide data of any size to TL. So

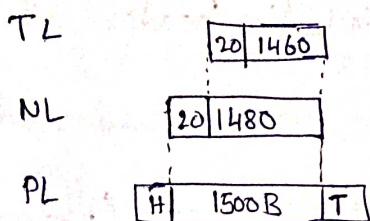
TL does the job of segmentation



→ However if the DLL is ethernet then max frame length is possible is 1500B. So for this reason NL does the job of fragmentation.

- In some cases DLL will be able to transmit frame of any size. If not, performing segmentation followed by fragmentation is waste of time. So it is better the TL finds least frame size / least datagram size and do segmentation accordingly

In this case segmentation done by TL is as shown below



Here only segmentation is done
but not fragmentation

• Fragmentation is not done at sender side. Segmentation itself is done such that there will be no need of fragmentation.

→ The maximum allowed size of frame is called ~~Max~~ in a NW.

Maximum Transmittable Unit (MTU).

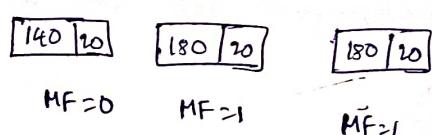
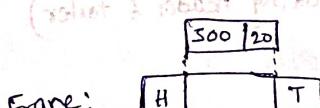
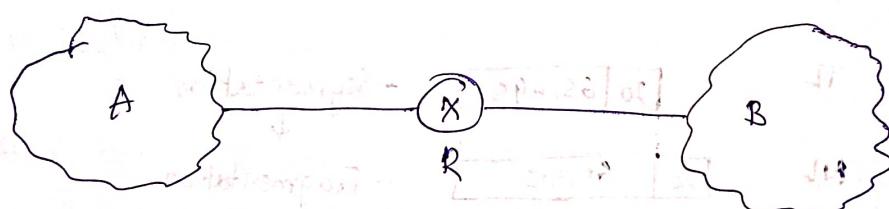
09/12/20

Fragmentation:

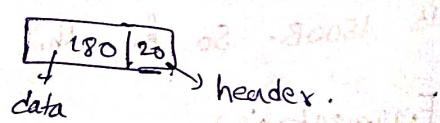
→ Fragmentation is done at routers but not sender.

• MTU may be different from network to network.

Consider 2 networks A & B with $MTU_A = 520$ & $MTU_B = 200$



In NW B MTU is 200



→ So the router R has to perform fragmentation before it sends it to B.

→ After fragmentation, the fragments are routed independently since it is datagram service. So the fragment may be received out of order.

- Identification number of all the fragments by 7 bits.
- The fragment offset field is used to know the order of fragments.
- For the last fragment MF is set to 1, which means that for remaining fragments MF is set to 0.

Assigning fragment offset to fragments:

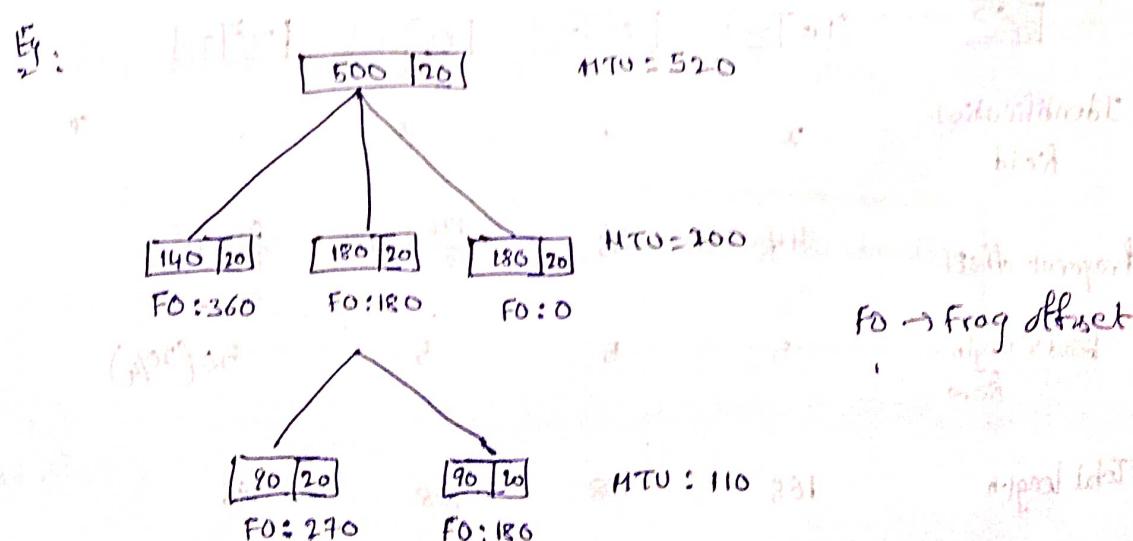
Assigning:

Assigning fragment offset sequentially from 0 won't be scalable.

For example, consider a case where packet P is fragmented into 3 packets P₁, P₂, P₃ and their fragment offsets are 0, 1, 2 respectively.

If we need to fragment P₂ again then we can't assign fragment offset to it. So we number fragments in a different way.

- Fragment offset is given such that, fragment offset is equal to number of data bytes ahead of the fragment.



∴ Final fragment offsets are (0, 180, 270, 360)

- But here we have a problem.

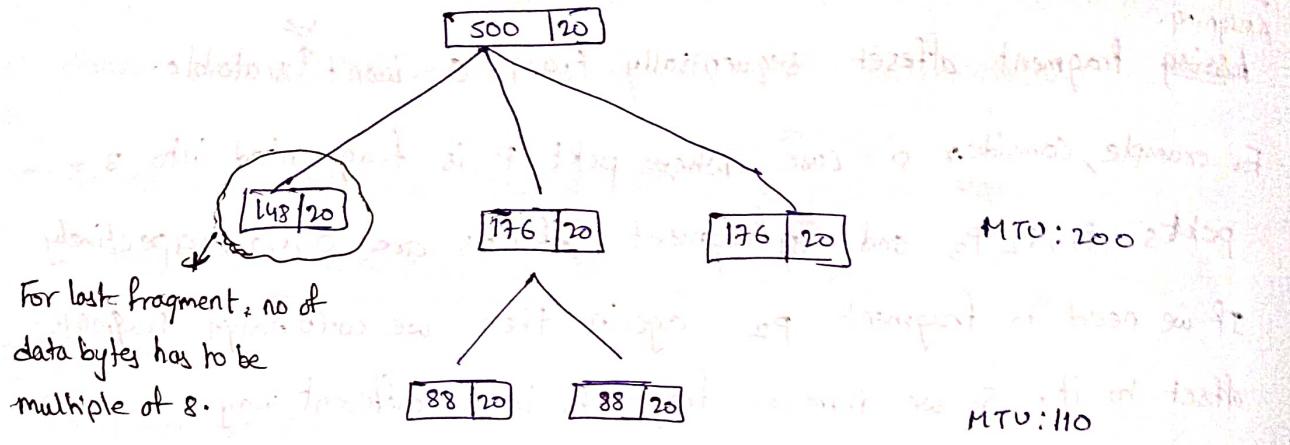
Fragment offset field has 12 bits and total length of ~~this~~ ~~the~~ possible has 16 bits. In worst case if last fragment has 1 byte, then fragment offset has to be $2^{16-20-1} = 65,514$.

To meet this problem, we ~~can~~ divide fragment offset with 8 ($\because \frac{2^{16}}{2^{13}} = 8$) and store it in the fragment offset field. 48

∴ 60

with 8 ($\because \frac{2^{16}}{2^{13}} = 8$) and store it in the fragment offset field.

Now this brings a new restriction, that original fragment offset has to be a multiple of 8. It means data bytes in each pkt should be a multiple of 8 (except last fragment). Now we perform fragmentation as shown below



Now the fragments are forwarded to the switch in the transport layer.

Identification field	x	x	x	x	x
fragment offset	44	33	$\frac{176}{8} = 22$	$\frac{0}{8} = 0$	
Header length field	5	5	5	5	$5 \in (20/4)$
Total length	168	108	196		
MF	0	1	1	1	

→ Here we are totally transmitting 4 fragments.
So we transfer 4 headers.

$$\text{Actual data to transfer} = 500 + 20 = 520$$

$$\text{Actual data that is transferred} = 500 + 4 \times 20 = 580$$

Note:

→ Fragment

→ Reassembly

Reassembly

May there

∴ 60 bytes is overhead from network layer.

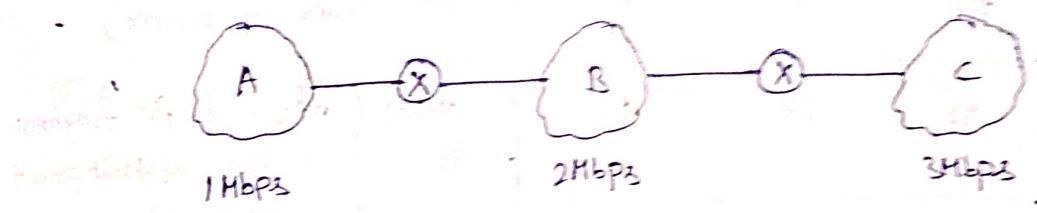
99

$$\text{Network layer overhead} = (\text{no of fragment} - 1) \times (\text{header length})$$

$$\therefore \text{Efficiency} = \frac{\text{useful data sent}}{\text{total data sent}} = \frac{500}{500}$$

$$\Rightarrow \text{Bandwidth utilization} = \eta \times R \\ (\text{effec. Bandwidth})$$

- Consider 3 networks



To find bandwidth utilization we take least bandwidth (bottleneck bandwidth) among the networks.

$$\text{Bandwidth utilization} = \eta * \text{bottleneck Bandwidth}$$

(η)
Effec. BW

Note:

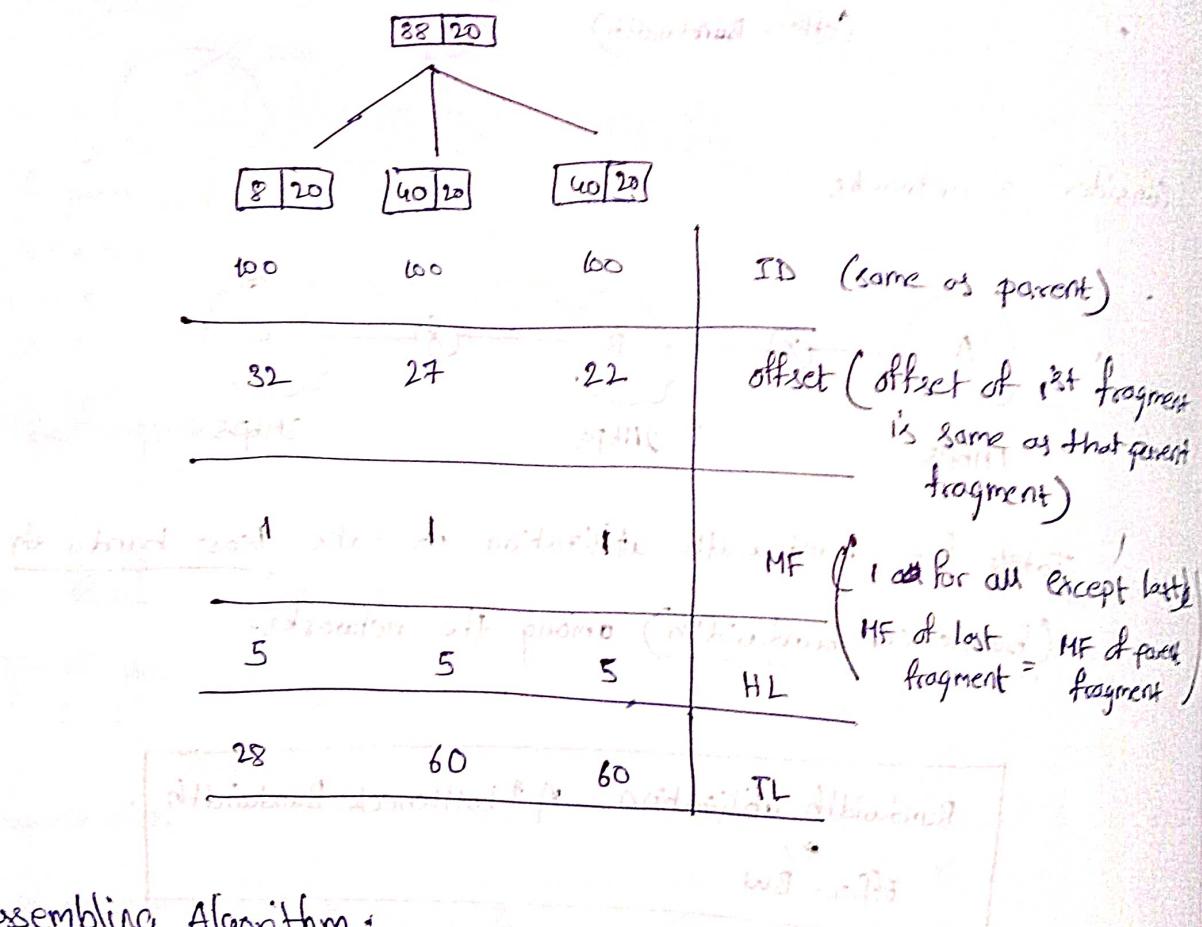
- Fragmentation is done only at routers, but not at source.
- Rearranging is done only at destination, but not at router.
Rearranging is ^{not} done at routers because all fragments may not meet at the same router. Also even if they meet, there may be a need for fragmentation again.

→ For fragmenting or refragmenting, the given fragment and its header information is sufficient. No other extra information is needed.

Consider a fragment $\boxed{28|20}$ with

frag offset = 22; MF = 1; header length = 5; total length = 108; ID = 105;

Fragment the pkct if its passed through N/w with MTU = 60.



Reassembling Algorithm:

MF offset

0 → 1st fragment

not 0 → intermediate fragment

0 → last fragment

0 → No fragmentation is done. } No fragmentation!

} Fragmentation done

Algorithm:

- Destination should identify that fragmentation is done (using MF, off)
- Destination should identify all fragments belonging to the same datagram using identification field
- Identify 1st fragment ($MF=1 \& offset=0$)
- Identify subsequent fragments ~~until MF=0~~ using (~~TL, HL, offset~~)
i.e. if f_i is a fragment, then we need to identify f_{i+1}

$$\text{offset of } f_{i+1} = \left(\text{offset of } f_i + \frac{TL_i - HL * 4}{8} \right)$$
- If $MF_i = 0$ then we don't find f_{i+1}
- Repeat (iv) until $MF=0$

Eg: Consider fragment

	1	2	3	4
offset	0	22	33	44
MF	1	1	1	0
HLF	5	5	5	5
TL	196	108	108	168
Id	600	600	600	600

- Identify 1st fragment ($MF=1 \& off=0$)

$$\text{offset of 2nd fragment} = 0 + \frac{196 - 5 \times 4}{8} = 22$$

and 2nd fragment with offset 22 is present

offset of 3rd fragment

$$\Rightarrow 22 + \frac{108 - 5 \times 4}{8} = 33$$

offset of 4th fragment $\Rightarrow 22 + 11 = 33$ (exists)

offset of 4th fragment

$$\Rightarrow 22 + \frac{108 - 5 \times 4}{8} = 33$$

MF of 4th fragment = 0

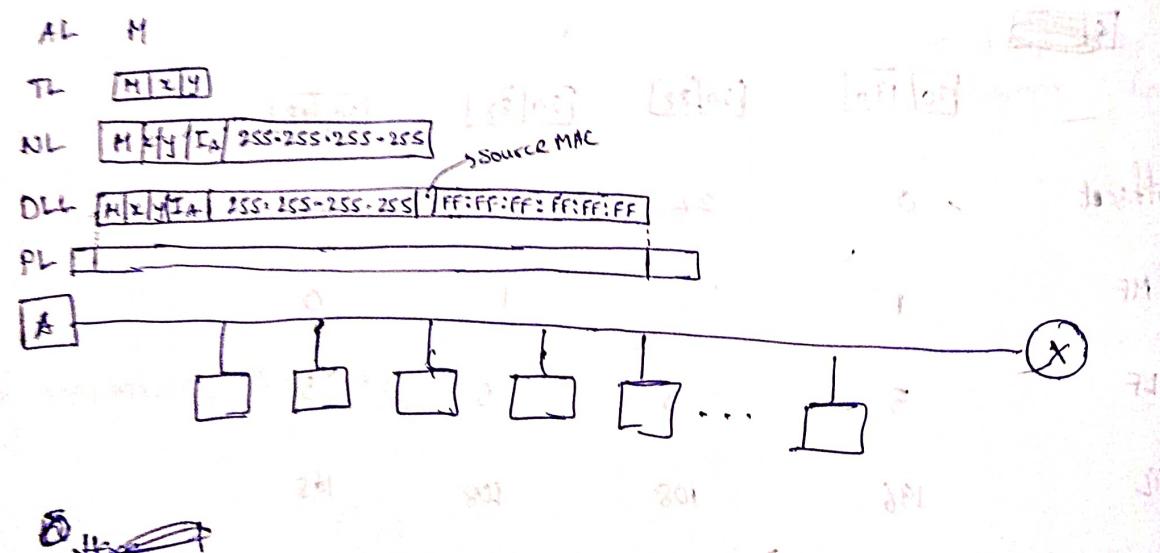
∴ Reassembling is finished

frame ready for delivery

Protocols & Concepts at Network Layer:

Implementation of broadcasting:

Limited Broadcasting:



For limited broadcasting destination IP = all 1's

destination MAC = all 1's

→ Every NIC is configured such that it accepts ~~pkts~~ pkts if destination MAC is all 1's

→ when the pkts finally reaches Router R, (where the Net ends) the router R will accept and discard the pkts.

Directed Broadcasting:

Destination IP = ~~all 1s~~ 1111...1111 (all 1s in thid)

Destination MAC = MAC of ~~the~~ gateway router

→ Now this packet is routed until it meets the router which is directly connected to the destination N/w.

→ At this router forwards packet into the destination N/w.

Before forwarding it sets destination IP to all 1s
destination MAC to all 1s

→ Like in the limited broadcasting this pkt will also ~~not~~ be discarded by the router of the network.

Address Resolution Protocol

→ ARP is used to find MAC address of a destination given its IP add.

~~Receiver in same N/w~~

~~- Receiver in same N/w~~:

→ On sender side, ARP can find whether receiver is on the same N/w or not using subnet mask of sender and destination IP.

→ If it is found to be on same N/w, sender broadcasts an ARP request packet which contains destination IP and MAC of sender. whichever destination IP matched will send its mac address to A.

the IP matched will send its mac address to A.

IP	MA	FF:FF:FF:FF:FF:FF
----	----	-------------------

Destination IP

+ source MAC

Broadcasting MAC

Receiver on different Nw

with dest IP = IP of router
default gateway

- In this case sender ~~sends~~ broadcasts ARP packets to know the MAC of the router, which is then used to forward the original data packet to the destination.
- After this original data packet is sent to router from which is sent to the destination.

Note: Host can't do this by itself because address didn't change.

- ARP request is broadcast message
- ARP reply is unicast message
- ARP is app used in below cases

(i) Host needs MAC of Host

(ii) Host → via Router → destination

(iii) Router → " " Host

(iv) Router → " " Router

- If source and destinations are in diff networks then source can never know the MAC of destination and destination can never know the MAC of source.

Special address 127:

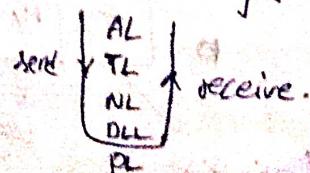
→ It starts from 127 to form 127.0.0.1

→ The CIDR 127 of class A is reserved for special purpose.

It is used for testing self connectivity.

→ If a system needs to check if its NIC is working properly or not then it can set destination add. as 127.0.0.1

→ This pkts will not go through physical layer but goes through remaining layers.



- 105
- However the addresses $127.0.0.0$ & $129.255.255.255$ should not be used.
 - Usage: ping 127.0.0.1
- If we are not working then we get timeout message.
- Since pkt comes back to same system this address is called loop back address.
 - One more application is that

If we are implementing server on a host then to test if the server is working or not we also put a client processes on the same host and we can start communication b/w them.

Here the processes are identified using port numbers

Message	source port	dest port	source IP	loop back add
---------	-------------	-----------	-----------	---------------

Q: Let IP add is IA.

port num of server is 80
port num of client is 90

M(80 90 IA 127.0.0.1)

RARP (Reverse ARP)

- We need to know IP add of a host given MAC add of the host.
- When a system is turned off or newly brought into Net, it doesn't know its IP address.
- RARP server contains MAC add & corresponding IP addresses of the systems. The below shown table is maintained at RARP server

MAC	IP
M1	IP1
M2	IP2

- Now ~~the host~~ sends RARP request pkt with
 - source destination IP = 0.0.0.0 and dest IP = 255.255.255.255
 - source MAC = MAC of the host
 - destination MAC = All 1's (i.e. Broadcast)
- Looking at IP = 0.0.0.0 remaining hosts in N/w discards this, but RARP server replies the host with ~~it~~ the host's IP.
- Disadvantages:
 - But for this protocol to work, there must be RARP server at every network (\because we are broadcasting the RARP req pkt)
 - Due to this the data is distributed across different RARP servers and if there is same data, then updating has to be done at every server.
 - The IP addresses assigned are static. Even if some hosts are not active, the IP addresses are kept assigned.
 - \therefore no of IP add \geq no of hosts

However ~~this~~ RARP is not currently under use.

10/12/20

~~BootP~~ Bootstrap Protocol

- This also used to know IP address of a host given its MAC.
- ~~like~~ Here also we have a BootP server. This server will have a static table of MAC & IP addresses.
- Here also pkt is sent in the same way, requesting for IP address.
- The main difference b/w RARP & BOOTP is that RARP operates at network layer (and may not contain other layer) whereas BOOTP operates at application layer.

→ Relay agent is used to obtain IP addresses if BootP server is not present in the network.

If a station needs to find its IP add, it broadcasts the request pkt. Since BootP server is not in the N/w relay agent accepts this message and unicasts this to bootP server on the other N/w. Then BootP server replies to relay agent and relay agent replies to the station.

Adv:

→ We can have only one bootP server and hence data is centralized.

Disadv: Data is static i.e., mapping MAC & IP is static

i.e., Mapping b/w MAC & IP is static

DHCP (Dynamic Host Configuration Protocol)

→ Its working is same as BootP. Here we have 2 DHCP servers. Here also we have relay agents.

→ The main difference is that ~~the~~ mapping b/w MAC & IP is dynamic ~~both~~ has 2 parts

is Static is Dynamic



Given to stations which
runs forever



Given to station that
runs only for some time.

MAC	IP
:	:

Static

MAC	IP	leaseTime
M ₁	I ₁	10 min

Dynamic

→ For dynamically assigning, a pool of IP addresses is maintained. whenever a host requests, an IP address is assigned and an entry is created in the table. After the lease time finishes, the station has to renew request if it needs IP for some more time. otherwise, the IP address will be kept back in the pool and entry is deleted.

Adv: (i) only one DHCP Server is required.

(ii) dynamic table

∴ no of IP add = no of hosts online

(iii) DHCP is also compatible with BootRequest. so for this purpose port numbers of DHCP & BootP are same

ICMP (Internet Control Message Protocol)

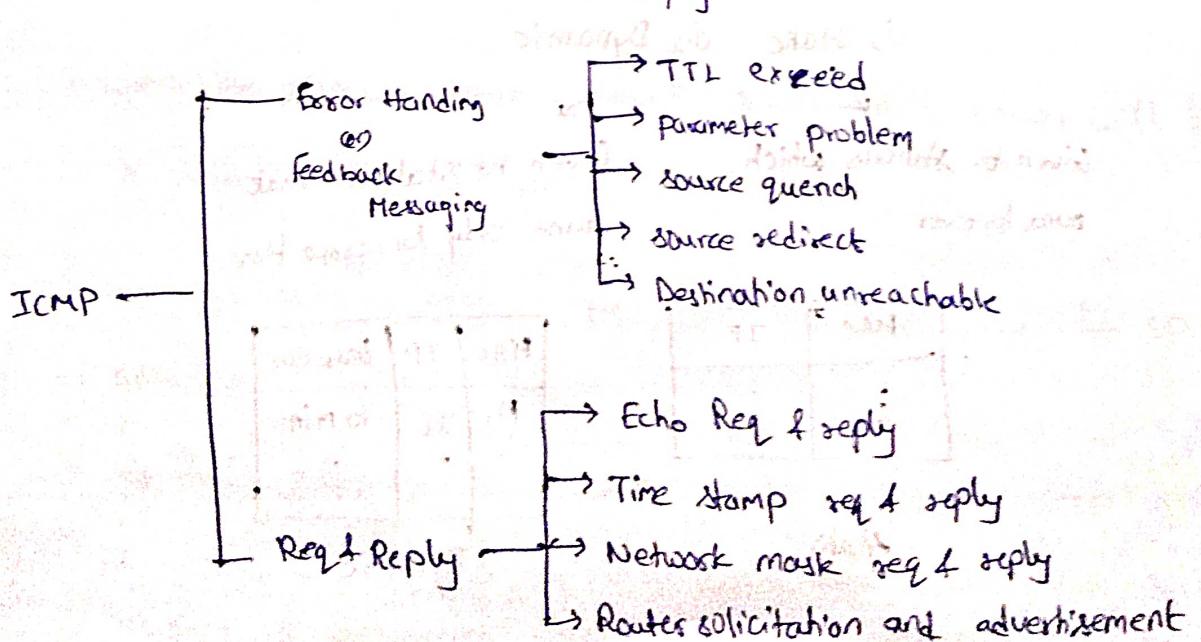
→ ICMP packet is sent to sender if the sent

packet is discarded at router or destination.

→ ICMP packet is also used for req. & reply.

Here ICMP packet is sent as request and

an ICMP packet is obtained as reply



→ when IP pkts is discarded, an ICMP pkt is sent.

~~But~~ But if this generated ICMP packet is discarded again, then no ICMP packet will be generated again. and in this case sender will not know that its pkt is discarded.

∴ (IP + ICMP) is still "not" reliable. However TL (TCP) is reliable.

ICMP Error Messages:

TTL exceed:

when TTL exceeds router discards the pkts and sends ICMP packet.

Here ICMP type = TTL exceed

Source Quench:

When a source (system or by router) sends lot of pkts to a single router and if the receiving router can't handle all the data, then source quench ICMP pkt will be sent to source.

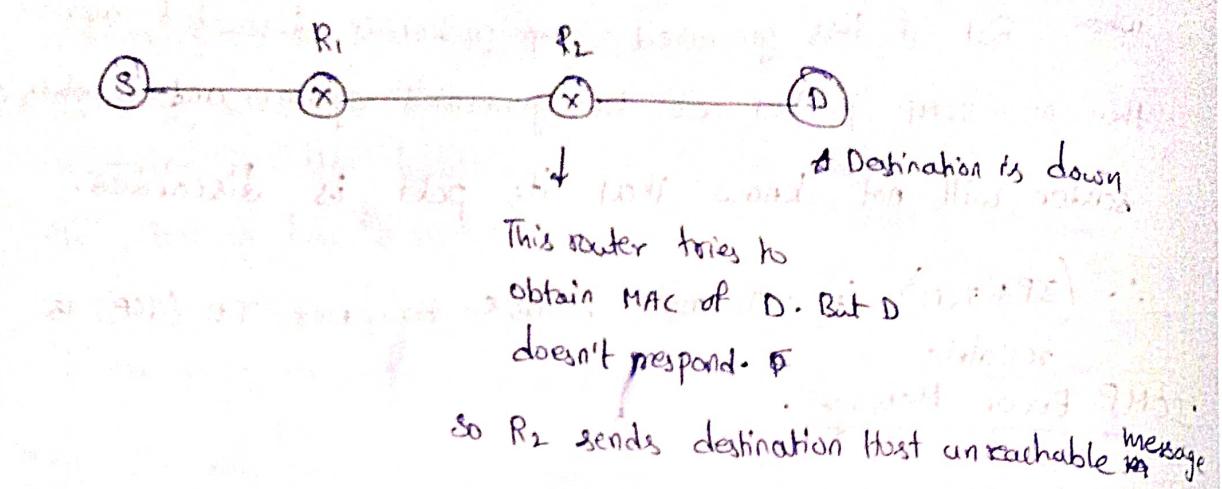
Seeing this packet, source will reduce the transmission.

Parameter Problem

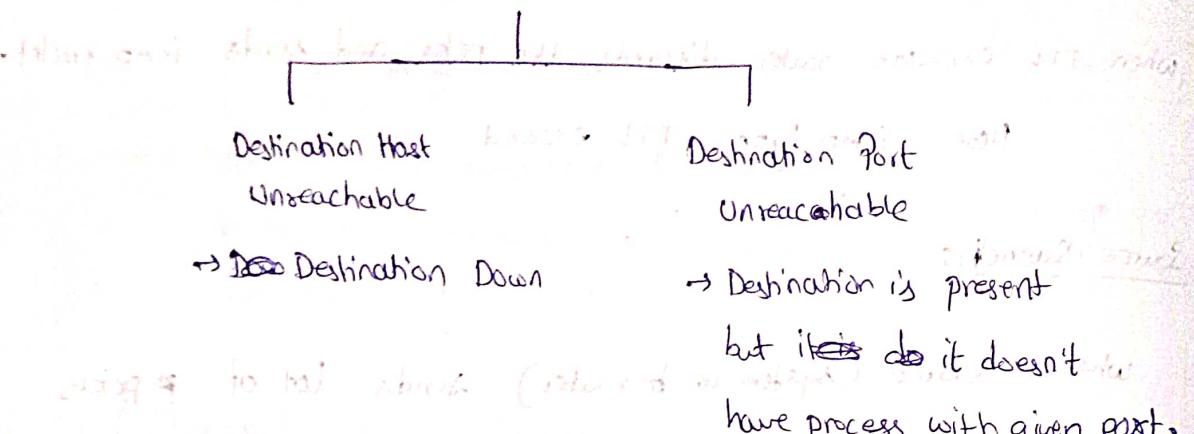
Assume we are using strict source routing, if a router given in the path is found to be down at the time of routing then ~~the~~ this ICMP message will sent.

Also this message is sent if the packet sent is corrupted.

Destination Unreachable



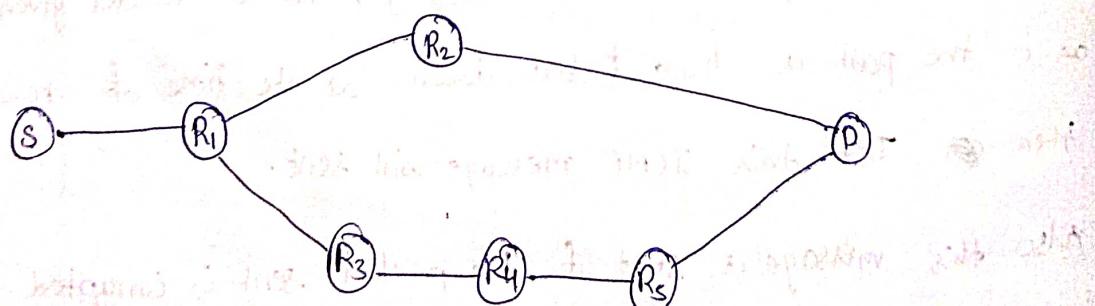
Destination Unreachable



- Also destination unreachable occurs when DF=1 and MTU of the network.
- In this ICMP message, it is specified that fragmentation is required and DF=1 and MTU of the network.

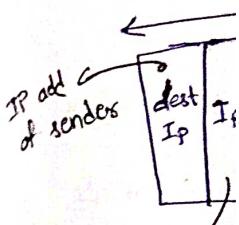
Source Redirect:

Consider



Here the best path is S-R₁-R₂-D

ICMP mes



IP add of
(who discar
pckt)

→ ICMP is

TCP or

→ If the

discar

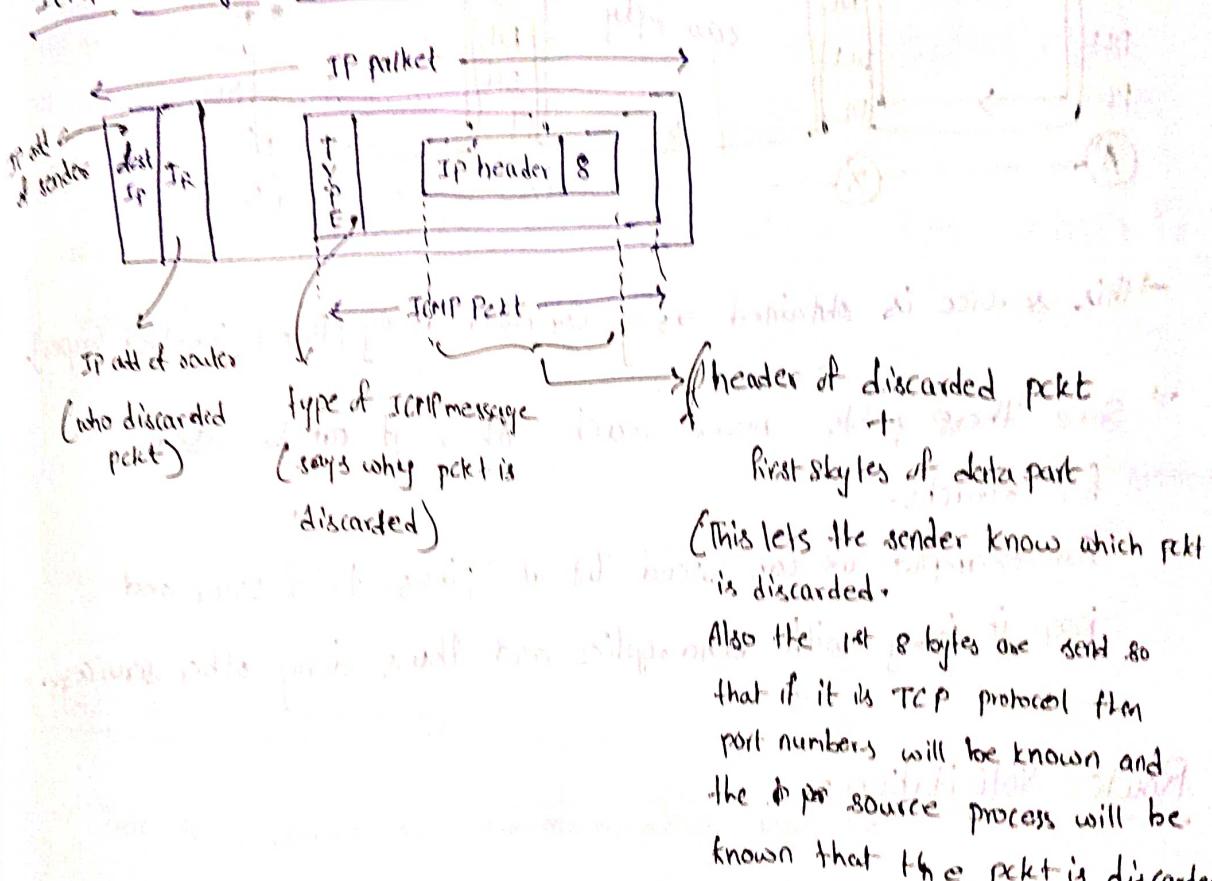
Even if

is discar

But consider R₁ has forwarded packet to R₃.
 Now if R₃ knows that there is better path, then R₃ sends ICMP msg pkts to R₁ (warning message) saying that there is a better path.

However, R₃ doesn't discard the packet and from next time onwards R₁ forwards to R₂.

ICMP message format:



→ ICMP is generated only if the discarded packet is either

TCP or UDP.

→ If the datagram has several fragments, then ~~the 1st~~ and if it is discarded then ICMP is generated only for 1st packet fragment.

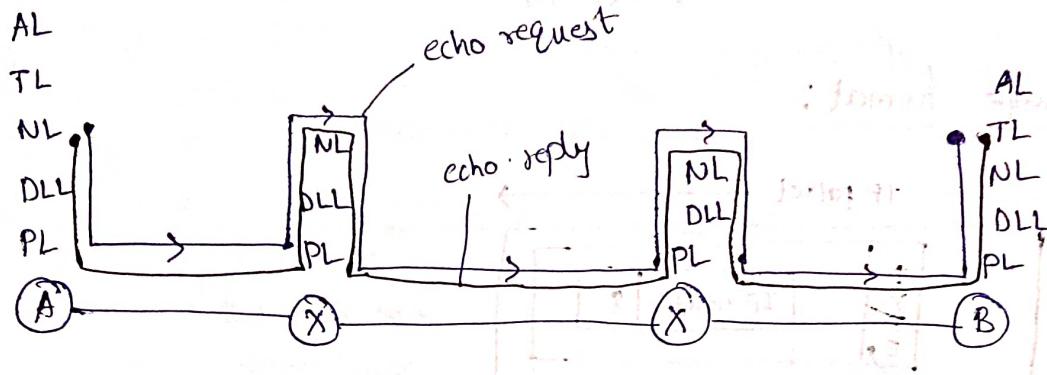
Even if 1st fragment is transmitted correctly and if 2nd fragment is discarded, then also ICMP will not be generated.

ICMP request and reply:

Echo request & reply:

It is used (to see if the destination is present (online) or not.)

The ICMP pkt never meets Application layer & transport layer.



→ This service is obtained using command ping (Packet InterNet Groper).

→ Since these pkts never reach AL, it can be used for attacks.

For example we can send lot of pings to a server and keep it busy with echo replies and thus deny other services..

Router Solicitation:

→ This ICMP pkt is sent by a station to know who is the default router.

→ So this ICMP pkt (router.solicitation ICMP pkt) is broadcasted and all the routers in the NW will reply.

→ One among these is used as default.

Router advertisement:

Whenever a new router is brought into network, this router ~~sends~~ broadcasts router advertisement ICMP pkts saying that you can use me as default router.

Network Mask Req & Reply:

→ This ICMP msg is sent by a host to know its Net mask or subnet mask.

Note:

Before any communication starts the below 3 steps ~~are~~ has to be done:

(i) Getting IP add (RARP, BootP, DHCP)

(ii) known default router (Router solicitation / Router advertisement)

(iii) knowing Network mask (Network mask Req & reply)

Time stamp req & reply

→ Two stations which are communicating may be in different time zones. So we need to synchronize the communication.

→ Time stamp req & reply is used to know the time at destination and destination uses this to know the time at sender.

→ However this is highly unreliable. and there are better ways to synchronize time.

Trace Route:

114

- This is an application of ICMP.
- This is used to know the path (routers involved in the path) of communication.
- The option record route is used to know the path ~~for~~ for destination.

But trace route is used to know the path for sender.

Implementation:

- Here we send a pkt with TTL = 1.

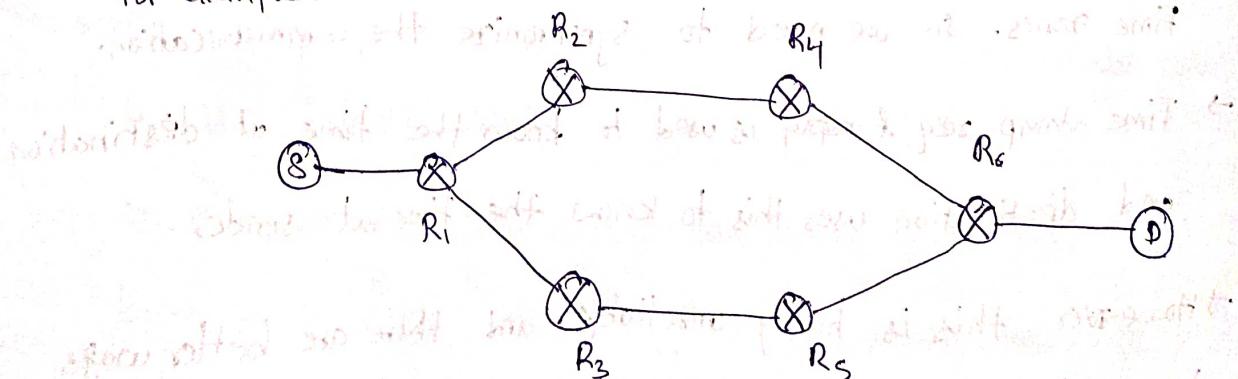
At ~~router~~ first router TTL becomes 0 and we can know the first router. (\because router sends TTL exceeded ICMP)

Similarly we do ~~the~~ same with TTL = 2 and know the 2nd router in the path.

We continue this process until we get ICMP pkt from the destination. (To get ICMP pkt from destination we intentionally put unreachable port number in the pkt)

- However in some case route obtained may be wrong.

For example,



$$\text{TTL} = 1 \Rightarrow R_1$$

$$\text{TTL} = 2 \Rightarrow R_2$$

$$\text{TTL} = 3 \Rightarrow R_5$$

$$\text{TTL} = 4 \Rightarrow R_6$$

$$\text{TTL} = 5 \Rightarrow D$$

$\therefore R_1 - R_2 - R_5 - R_6 - D$ is not a valid path

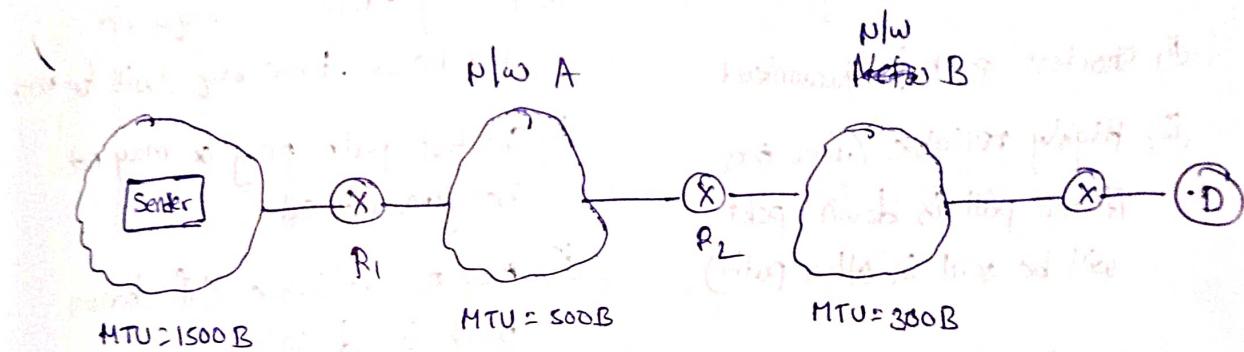
(But this is generally least probable case
cuz routes don't change that fast)

PMTUD (Path MTU Discovery):

- It is another application of ICMP.
- It is used to know ~~the~~ MTU along the path.
- Thus, using this, we can know least MTU in the path and we can send datagram of that size and thus avoid overhead of fragmentation.

Implementation:

Consider below scenario



- Initially we send pckt with maximum allowed size in sender network with DF = 1.
- Once it reaches N/w A, destination unreachable ICMP pack is sent by Router R₁ to sender. After this sender reduces the size and sends again.

This process is continued until pckt reach destination.

(To know that pckt has reached destination we put invalid port num in destination port and thus destination sends ~~invalid~~ destination port unreachable message to the sender.)

- However here also there is a chance that path followed for each packet may be different.

Routing

- Routing is process of preparing routing table ~~for~~ and this routing table is used for switching.
- ~~If there is no routing then we flooding.~~
- Flooding is process of sending obtained pkts to all the links except to the link the packet has arrived on.

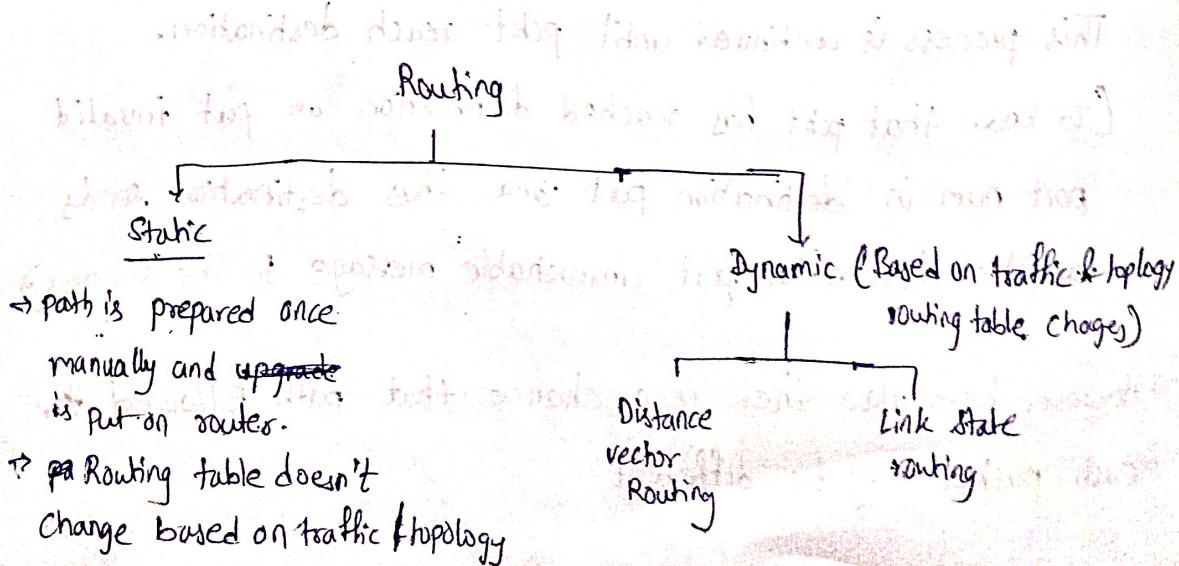
Advantages of flooding:

- No need of routing
- Shortest path is guaranteed
- Highly reliable (since even if one path is down, pkt will be sent in other path)

Disadv of flooding:

- Many duplicate pkts are received at destination
- More traffic

Types of Routing Algorithms:



Distance
Every router
If a
Based one
as shown

Des	Dis
A	1
B	7
C	11
D	0

Disadvantages of Routing

- Routing table are required and hence time req will be more.
- Shortest path may or may not be guaranteed.
- May not reliable (if chosen path is down).

Advantage of Routing

- No duplicate pkts will be received.
- Low traffic.

Des	Di
A	
B	
C	
D	

Step 2:
Every
At A: DV
At B: DV
At C: DV dec
At D: DV dec
At E: DV dec

Distance Vector Routing (DVR)

Every router has a routing table with below structure

Destination	Distance	Next-hop
:	:	:

Distance vector

If a router is not directly connected then distance is ∞ .

Step 1:

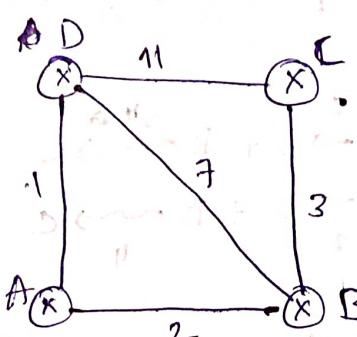
Based on the local information every router builds routing table as shown below.

Des	Dis	NH
A	1	A
B	7	B
C	11	C
D	0	D

Des	Dis	NH
A	∞	-
B	3	B
C	0	C
D	11	D

Des	Dis	NH
A	0	A
B	2	B
C	∞	-
D	1	D

Des	Dis	NH
A	2	A
B	0	B
C	3	C
D	7	D



Step 2: Every router sends its distance vector to adjacent routers.

(DV)

At A: DV received from B,D

At B: DV received from A,C,D

At C: DV received from B,D

At D: DV received from A,B,C

All this sharing will happen

at once in parallel (\because the channels are full duplex)

Using these distance vectors of neighbouring routers, each router builds new routing table.

At A:

From B

2
0
3
7

From D

1
7
11
0

New routing table (while building)

A	0	A
B	2	B
C	5	B
D	1	D

New routing table
old table shouldn't be used

$$A \rightsquigarrow B = \min \left\{ \begin{array}{l} A \xrightarrow{2} B + B \rightsquigarrow B \\ A \xrightarrow{1} D + D \rightsquigarrow B \end{array} \right.$$

→ : edge
~~: path

$$A \rightsquigarrow C = \min \left\{ \begin{array}{l} A \xrightarrow{2} B + B \rightsquigarrow C \\ A \xrightarrow{1} D + D \rightsquigarrow C \end{array} \right.$$

$$A \rightsquigarrow D = \min \left\{ \begin{array}{l} A \xrightarrow{1} D + D \rightsquigarrow D \\ A \xrightarrow{2} B + B \rightsquigarrow D \end{array} \right.$$

Shortcut

	A	B	C	D
B:	2	0	3	7
↓ AB (i.e., 2)	4	2	5	9

	A	B	C	D
D:	1	7	11	0
↓ AD (i.e., 1)	2	8	12	1

Pick Minimum

A	0	A
B	2	B
C	5	B
D	1	D

(since it's A to A, it is 0, but not min of above two)

At B:From A

0
2
0
1

$$\xrightarrow{+AB} (2)$$

2
4
0
3

From C

3
0
4
14

5
3
14
0

From D

1
7
11
0

8
14
18
70

A	2	A
B	0	B
C	3	C
D	3	A

A	2	A
B	0	B
C	3	C
D	3	A

At C:from D

1
7
11
0

from B

2
0
3
7

5
3
6
10

A	5	B
B	3	B
C	0	C
D	10	B

At D:ABC+1+7+11

1
3
0
2

9
2
10
14

0
14
11
22

A	1	A
B	3	B
C	10	C
D	0	D

At the end of 2nd round we get shortest paths containing at most 2 edges.

Since we have 4 routers, we need to go for 3rd round also for the final routing tables.

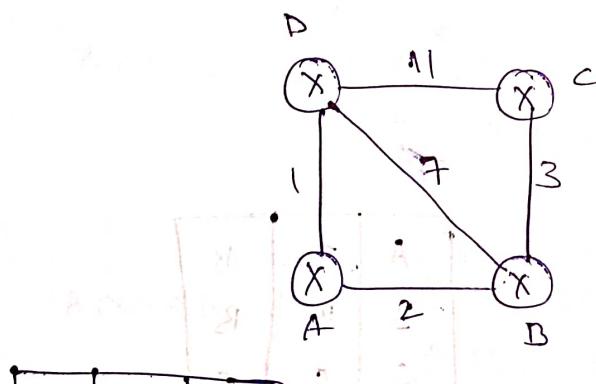
(But in exam we find out shortest path directly from the graph)

So the final routing tables are

A	1	A
B	3	A
C	6	A
D	0	D

A	5	B
B	3	B
C	0	C
D	6	B

Even after getting final routing table, ~~exchanging~~ DVs will continue forever periodically.



A	0	A
B	2	B
C	5	B
D	1	D

A	2	A
B	0	B
C	3	C
D	2	A

→ In routing table of D, C is not present as next hop for any entry. Similarly, in routing table of C, D is not present as next hop for any entry.

This means the link DC is not at all used. Similarly, the link BD is also not used.

Thus

shortcut for finding unused links:

A link PQ will be unused if there is ~~not~~ better other path from P to Q than the direct path b/w P & Q.

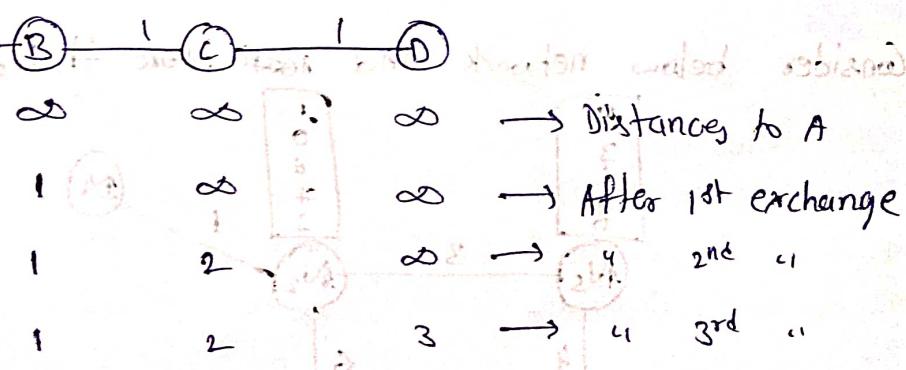
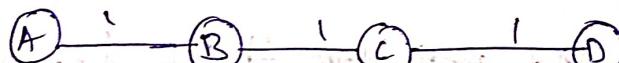
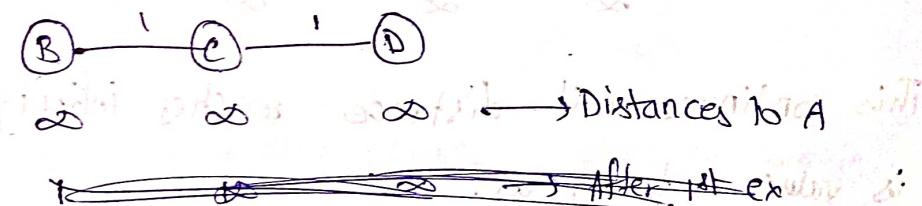
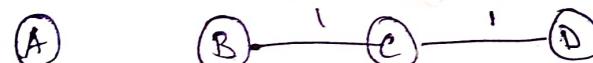
(Try this method for previous example)

Counts ~~to~~ infinity problem

- This is a disadvantage of DVR
- The problem is bad news ~~spear~~ spreads slow
good news ~~spear~~ spreads fast.

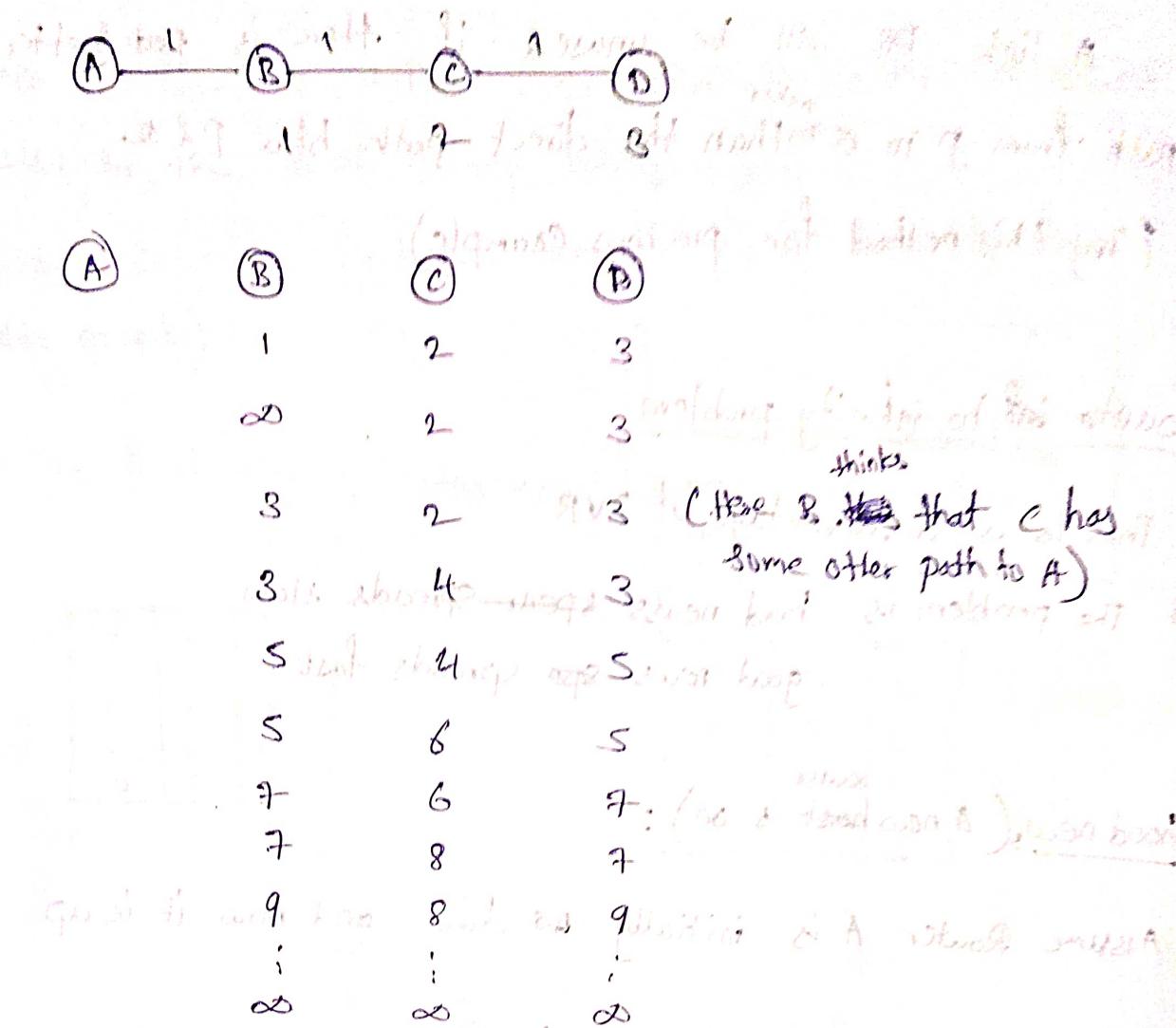
hood news (A new ~~host~~ is on):

Assume Router A is initially ~~is~~ down and now it is up



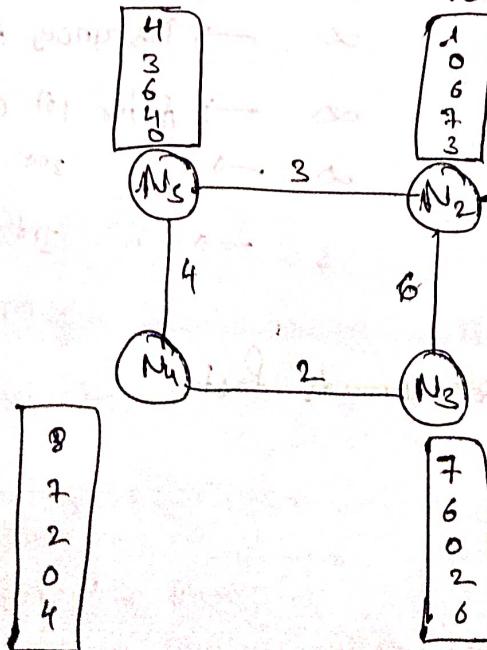
Thus good news ~~spear~~ spreads fast.

Bad news: (when a router is down)



This continues until distance reaches infinity and here infinity is value set by us.

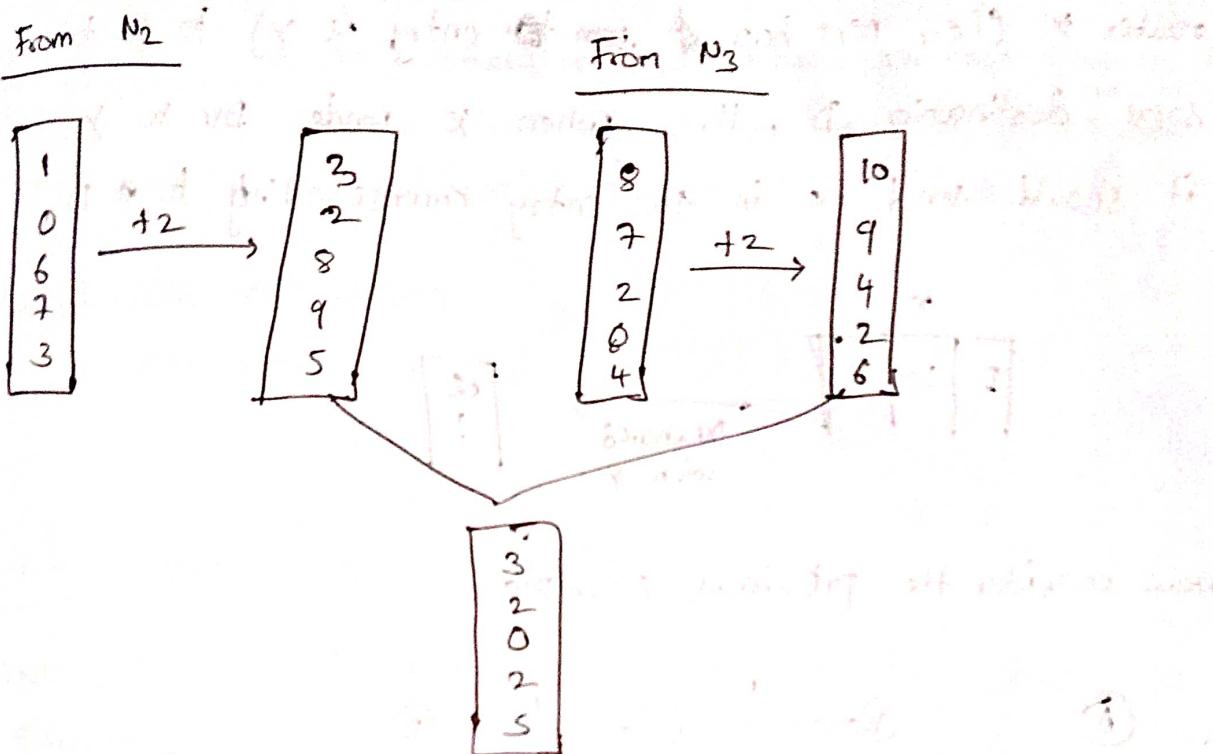
Q) Consider below network and write their distance matrix



If somehow distance b/w N_2 & N_3 fall to 2, then

what will the new routing table at N_3 after one exchange.

e.g:



Split Horizon

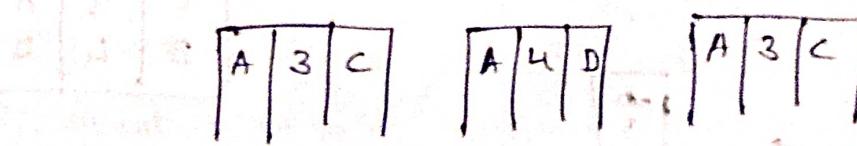
→ Count to infinity problem • also causes a packet to loop in the network. For example consider

(A) — (B) — (C) — (D)

1 2 3

∞ 2 3

3 4 3



Now consider B wants to send a packet to A.

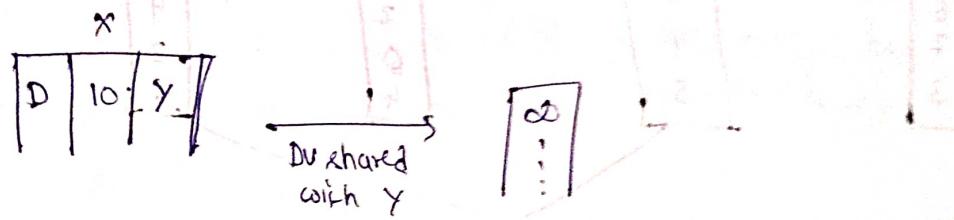
* B sends it to C; C sends to D; D sends to C;

C sends to D; D sends to C; ...

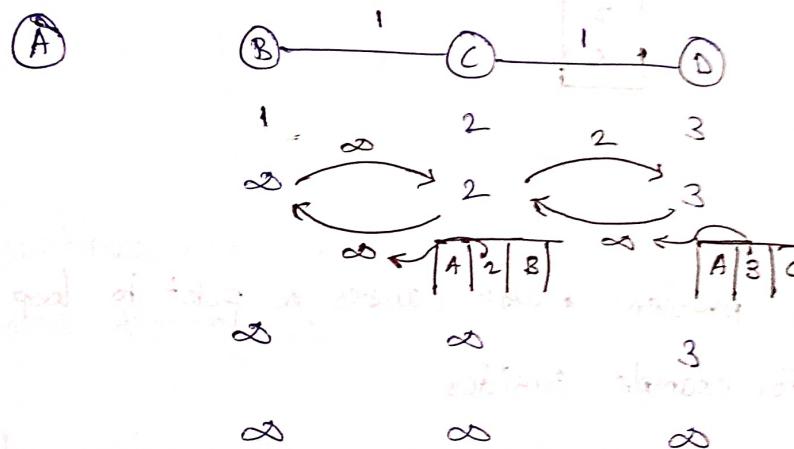


Split Horizon:

- This is solution for count to infinity problem.
- Split Horizon says if a router ~~x~~ ^{says} x is depending on router y (i.e., next hop ~~at~~ entry is y) to go to some destination D , then when x sends DV to y , it should send ∞ in the entry corresponding to D .



Now consider the previous example



And thus the problem is solved

Eg: Let routing table of router B be

A	1	A
B	0	B
C	3	A
D	4	D

Then

DV sent to A

∞
0
∞
4

DV sent to D

1
0
3
∞

Disadv without split horizon:

i) Count to infinity

ii) loops

iii) convergence is slow

i.e., new updates or modifications are not known quickly to routers that are far away from ^{the point} where the modification is done

Disadv with split Horizon

→ convergence is slow.

Link State Routing (LSR):

→ DTR is based on local knowledge whereas LSR is based on global knowledge.

→ Here every station prepares link state pkts.

A link state pkt ~~send~~ of a router contains distances of its neighbouring routers.

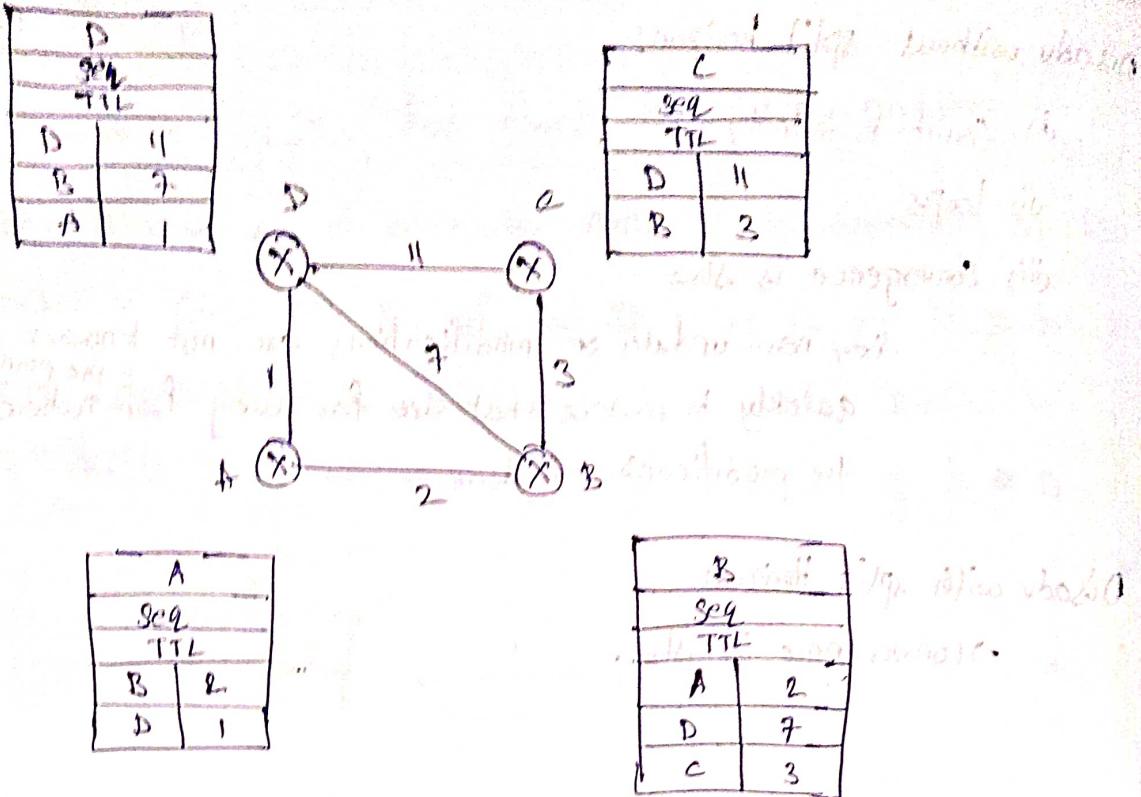
The neighbours and the corresponding distances are found using 'hello' packets.

→ After creating the link state pkt, the pkt is flooded to every other router.

So every router receives link state pkt of every router i.e., global knowledge

For this information, a router can build entire graph of the network.

1	A
0	B
3	A
4	D



- Now each router applies Dijkstra's algorithm and knows shortest path abg to each router.
- ∴ convergence is faster.
- Need of seq num:
 - Since flooding is used there is a chance that a router receives same pkkt twice. Assume by the time 2nd copy arrives, the router has already obtained latest packet (from same router). But still this accepts & uses old pkkt. To over come this we use sequence number field.

Every router maintains latest seq num of each router's link state pkkt

Router	latest seq num	life time
R1	10	6 min
R2	9	10 min
:	:	:

Thus using seq num old packets are discarded and are no more flooded and thus seq num field even helps to decrease flooding.

→ Need of TTL field:

since flooding is being used, there is a chance the packet may loop infinitely in the network. To avoid this TTL is used.

→ Need of lifetime field:

Assume a case where seq num is 156.

Assume a case where a link state packet with seq num 15 (00001111) is being transmitted and say due to a single bit error it is changed to 143 (10001111).

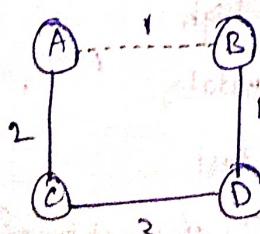
Now all the link state packets upto seq num 143 will be discarded and ≥ 144 will be accepted.

To overcome this problem every entry has lifetime/validity after which we accept any link state pkt that arrives.

Problems in LSR:

The problems in LSR are not transient (non-persistent)

1) Black hole problem:



If the link AB is down then A & B will know after this for some duration of time. If link AB will come up so any packets sent by A to B will not reach B during this time. After some time A will get to know that the link is down and will send pkts in different path.

(ii) Looping:

Here looping is present but for only some duration.

→ Consider the previous case where the link AB is down.

After A knows the link is down, A chooses the path to A - C - D - B. And so the loop was formed.

But C knows that the link is down after some duration so for C the path to B is still C - A - B.

→ So if C needs to send a pkt to B then it sends to A and A sends it to C, it will send it back to A again.

and this process continues until C discovers that the link AB is down.

DVR	LSR
1) used in 1980's	1) used in 1990's
2) BW req is less	2) BW req is more
3) Based on local knowledge	3) Based on global knowledge
4) Similar to Bellman Ford	4) Dijkstra's algo
5) Traffic is less	5) Traffic is high
6) Periodic updates	6) Periodic updates
7) Convergence is slow	7) Convergence fast
8) Count to inf is present	8) No problem of count to inf
9) persistent looping	9) Transient looping
10) Implementation: RIP	10) Implementation: OSPF

Routing Information Protocol (RIP):

- This is implementation of DVR.
- Here metric is hop count.
- ∴ Every edge has weight 1.

→ Here ∞ is equivalent to 16.

→ Computation is simple (compared to OSPF).

Open Shortest Path First (OSPF):

→ Computation is complex.

→ ~~Avoiding traffic of flooding:~~

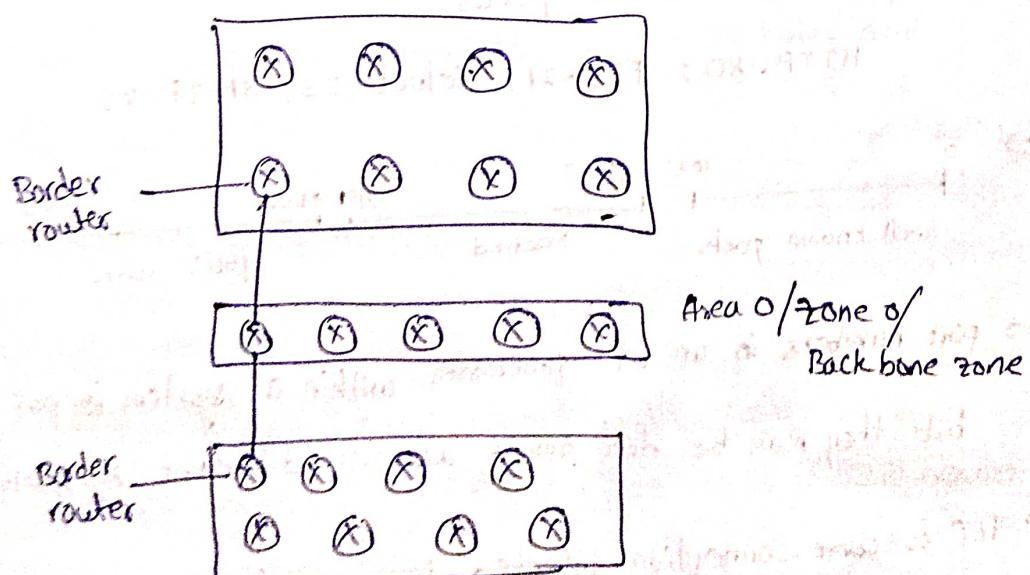
→ Here routers are divided into several regions and each region has a border router.

→ All the border routers are connected to area 0/zone 0/back bone.

→ A border router summarizes the flooded information and sends it to area 0 and from there this pkts will be forwarded to other regions.

Thus using this process traffic due to flooding is reduced.

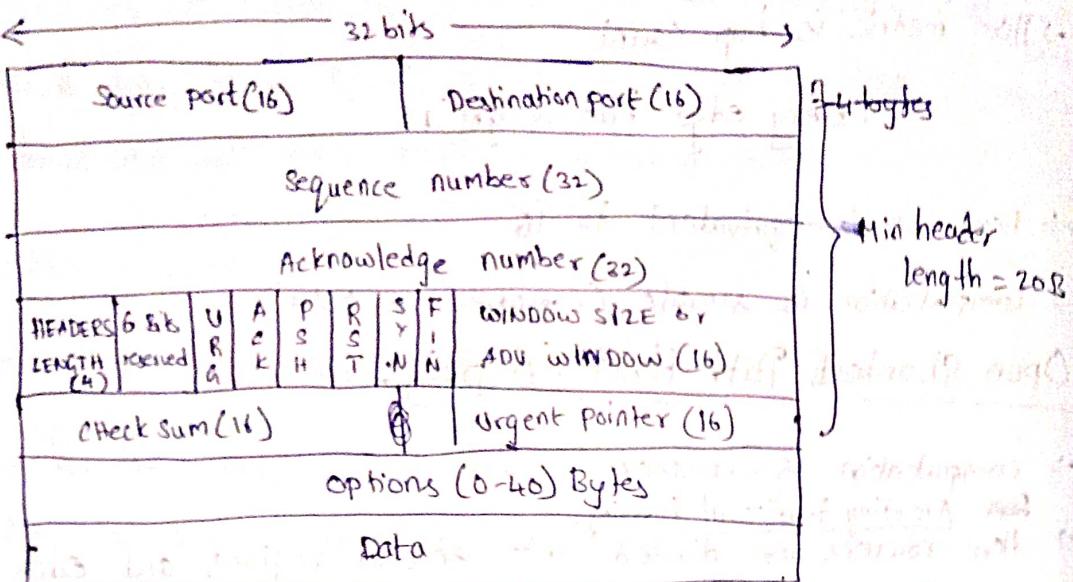
→ Metric is weight set by network administrator.



12/12/20

TCP

TCP Header:



$$\text{Max header length} = 20 + 40 = 60 \text{B}$$

→ TCP is a transport layer ~~part~~ protocol that ensures end-to-end communication.

→ So, TCP is an end-to-end protocol.

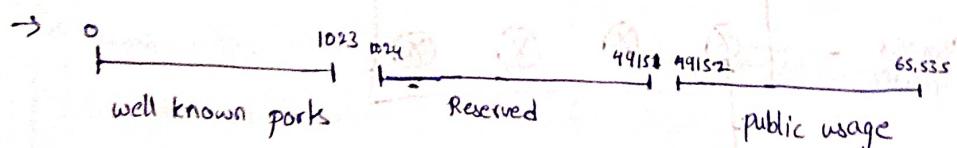
Source port & Destination port

→ Using port numbers, TCP does the job of multiplexing & demultiplexing.

→ port is a 16-bit number.

→ Example of well-known ports

HTTP - 80; FTP - 21; Telnet - 23; SMTP - 25



→ port numbers assigned to processes within a system are unique but they may be diff same across different systems.

→ TCP is ~~some~~ connection-oriented service. For any connection-oriented service, resources (BW, buffers etc.) are reserved.

"Identify port or only so we use to identify :)"

Sequence

→ TCP is a i.e. r E

It imp

→ The seq

in the

Acknowledge

→ It is

This num

For e

leg n

Then

→ How to calcu

TCP
calculation

b TL IP
in segmen
art

→ Identifying a connection uniquely is not possible using only port or only IP. is ~~not~~ possible.

→ So we use combination of IP add & port num called Socket to identify a connection uniquely.

∴ Socket = IP + port

∴ Socket is 48 bits

Sequence Number:

→ TCP is a byte stream protocol

i.e., Every Byte is counted

IP is packet stream protocol

It means logically we assume every byte is numbered.

→ The seq. num field contains number given to the first byte in the segment

Acknowledge Number:

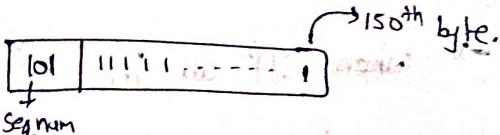
→ It is a byte number of next expected byte.

This number is sent as ack. by receiver.

For example consider a segment with 50 bytes and

seq. num is 101.

Then ack number is ~~151~~ 151

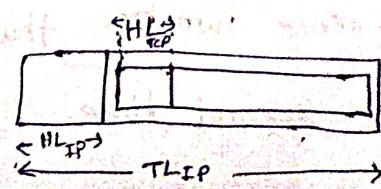


→ How to calculate ack number?

TCP header doesn't have total length field. So we calculate ack num using total length field of IP header

• $TL_{IP} - HL_{IP} - HI_{TCP}$ gives no of bytes

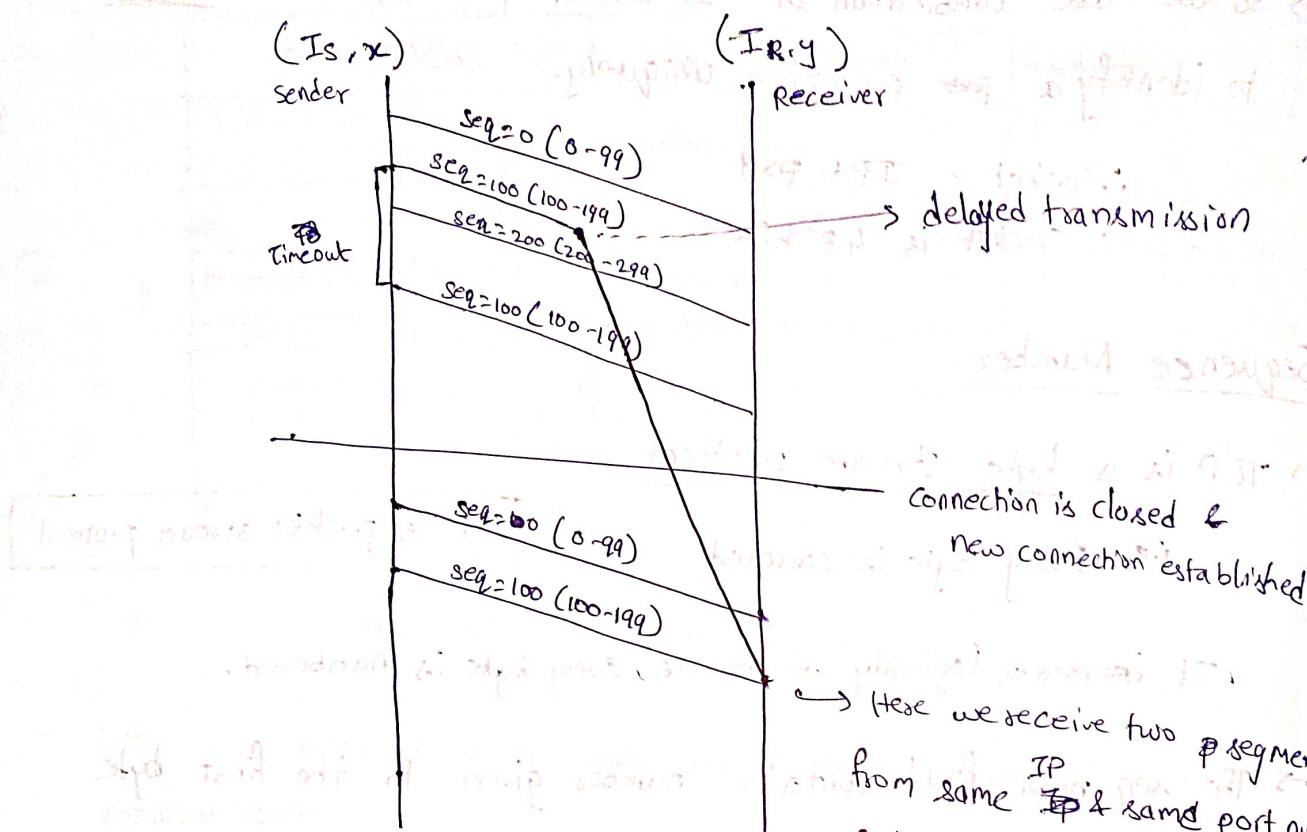
in segment and from that we calculate ack number.



Note:

Starting a seq num always from 0 has a problem.

Consider the below case.



→ Here we receive two segments from same IP & same port num and same seq num. So this is a problem.

→ To overcome this, TCP uses randomly generated numbers as starting sequence numbers.

However here also there is a chance of conflict but it is very negligible.

→ If all the 2^{32} seq numbers are assigned then we again start numbering from 0. This is known as wrapping around.

Wrap around time: The amount of time taken to use the sequence number that we started with (need not to be 0) for second time.

wrap around time depends on bandwidth.

$$\text{Ex: If } BW = 1 \text{ MBPS}$$

then

$$10^6 \text{ seq num} \xrightarrow{\text{are assumed}} 1 \text{ sec}$$

$$\text{All sequences } 2^{32} \text{ seq num} \xrightarrow{\frac{2^{32}}{10^6} \text{ sec}} \text{ i.e., } \approx 4296 \text{ sec}$$

Since wrap around time is this big, conflicts won't occur.

Life time

Maximum amount of time for which a packet can be in network.

Generally it is 2 min.

For no conflicts to occur,

$$\text{wrap around time} > \text{lifetime}$$

For example consider, let $BW = 1 \text{ Gbps}$

$$\Rightarrow \text{wrap around time} = \frac{2^{32}}{10^9} \approx 4.29 \text{ sec}$$

Here wrap around time < lifetime

The problem for this soln is to increase seq bits in seq num field.

\therefore given life time = 180 sec & $BW = 1 \text{ Gbps}$ we find no of seq num seq.

$$18 \rightarrow 1 \text{ G seqnum}$$

$$180 \rightarrow 180 \text{ G seqnum}$$

\Rightarrow no of seq num $\log_2 = 180 \times 10^9$

$$\Rightarrow \text{no of bits in seq num field} = \lceil \log_2 180 \times 10^9 \rceil = 38 \text{ bits}$$

However we have only 32 bits in the seq num field.

→ So the extra bits seq are kept in the options field.

These bits in the options field are called timestamp.

Header Length:

→ It is a 4-bit field.

But max header length possible is 60.

$$\therefore \text{Content in 4-bit field} = \frac{\text{original header length}}{4}$$

- Q) Consider IP datagram with TL = 1000 & HL = 5 and consider segment within IP datagram with HL = 5 and seq = 100. Find acknowledgement number.

Sol:

~~No size of data bytes in segment~~

$$\begin{aligned} \text{Data size of IP datagram} &= 1000 - 5 \times 4 \\ &= 980 \end{aligned}$$

$$\begin{aligned} \text{Data size of TCP segment} &= 980 - 5 \times 4 \\ &= 960 \end{aligned}$$

~~seq num~~ seq num of last byte in segment

$$= 100 + 960 - 1 = 1059$$

$$\Rightarrow \text{ack num} = 100 + 960 = 1060$$

Flags:
→ SYN - synchronization
→ ACK - Acknowledgment

- TCP is connection oriented
(i) Connection
(ii) Data transfer
(iii) Connection termination

SYN: This flag

other host

The packet

SYN packet

→ SYN is

Acknowledgement

In TCP, Acknowledged

For both

ack number is 0 then

ack number

any acknowledgement

when sending

When

When

Plz

Flags:

- SYN - Synchronization Flag
- ACK - Acknowledgement flag

→ TCP is connection oriented and it has 3 phases

(i) Connection Establishment

(ii) Data transfer

(iii) Connection Termination

SYN: This flag is ~~set~~ set, when we need to tell the other host that our starting sequence number is 20 and so. The packet with this information is called ~~a~~ SYN packet.

SYN packet consumes a seq num.

→ SYN is used to synchronize the sequence numbers.

Acknowledgement Flag:

In TCP, Acknowledgements are sent in two ways

(i) piggy backing

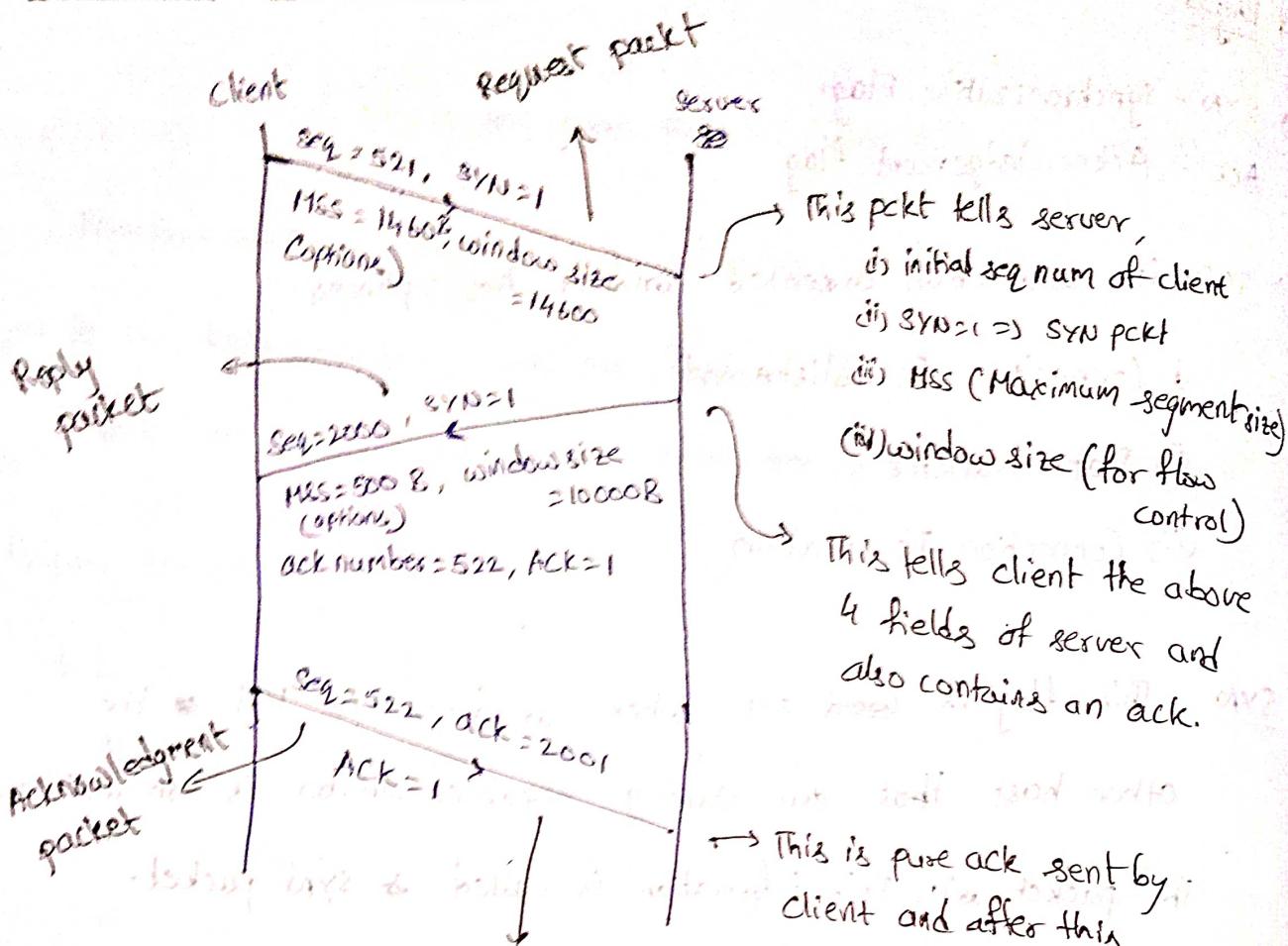
(ii) pure acknowledgement

For both acknowledgements the ack number is ~~set~~ put in ack number field and ACK flag is set to 1. If ACK flag is 0 then no matter what value is present in ack number field, the pckt is considered not to have any acknowledgement.

When sending pure acknowledgement only header is sent.

→ When pure ack is sent, no seq num is consumed.

Connection Establishment:



Since this is pure ACK, the seq num 522 is not consumed.

MSS: Maximum Segment Size is maximum DATA size in segment (i.e., excluding headers).

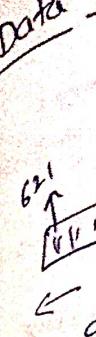
→ The seq num 521 from client & seq num 2000 from server are generated randomly.

→ Both server and client know MSS of each other and thus sends datagrams of the size which is minimum of the 2 MSS.

→ This connection establishment is called 3-way handshake.

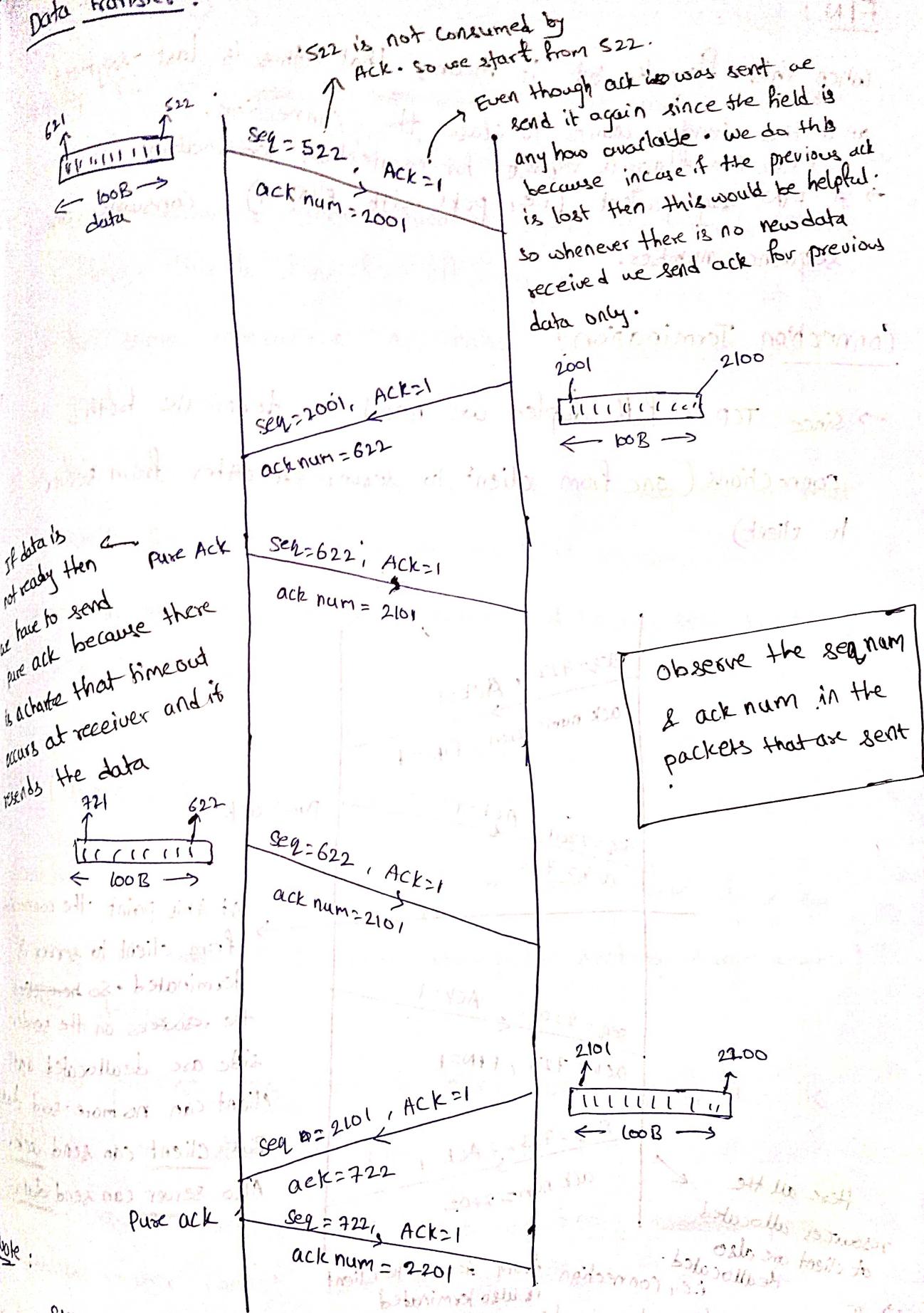
Note: As soon as connection is established resources (BW, buffers etc.) are allocated & reserved at both client & server.

→ TCP is a full duplex connection.



If data is not ready then we have to send pure ACK b/c it occurs at receiver.

Data Transfer:



SYN ACK

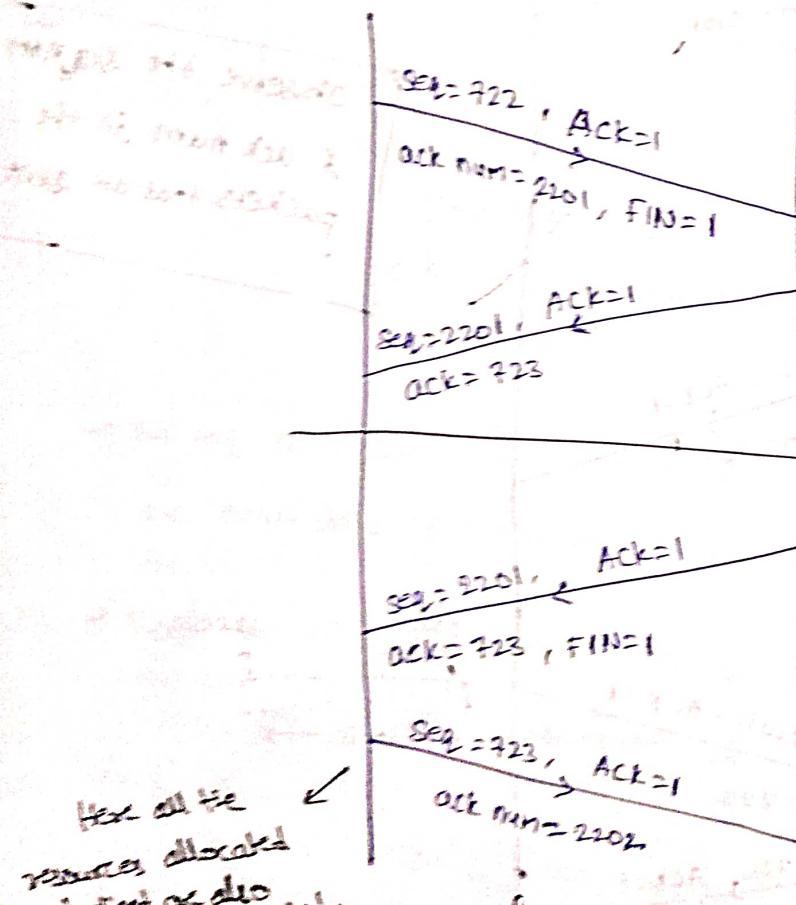
- 0. → 1st segment / Request packet
- 1. → 2nd segment / Reply packet
- 0. → Ack is present in header
- 0. → Invalid Combination

FIN

when this flag is set, it means that this is last segment and the sender wants to close the connection.
i.e., FIN flag is request for connection termination
→ A FIN ~~seq~~ packet (i.e., pkt with $\text{FIN}=1$) consumes one sequence number.

Connection Termination

- since TCP is full duplex we need to terminate both connections (one from client to server & other from server to client)



→ At this point the connection from client to server is terminated. So ~~here~~ then the resources on the sender side are deallocated and client can no more send data. But client can send ack. Also server can send data.

→ & client also deallocated.
 i.e. connection from server to client is also terminated

→ Thus, connection termination requires 4 packets in the worst case.
 However by setting FIN in the 2nd packet itself, we can terminate the connection in with 2 packets.

PSH (push flag):

Generally TL waits until it receives large enough data (i.e. MSS) and sends all the data at once. Until all the data is obtained from application layer, the data is stored in buffers. This is done for efficiency. But for some interactive application this technique won't be good.

So in this case AL sends PSH=1 to TL saying that transmit the data as soon as it is received.

Also on receive side when PSH=1, data is sent to AL as soon as it is received.

URG (Urgent) Flag and Urgent pointer:

Sometimes it may be needed that some data has to be sent first (than other pkts sent by same sender) to the destination.

For this purpose URG flag is given. When URG=1 then the pkt is treated as urgent pkt.

However router doesn't have TL so hence ~~it can't~~ router cannot look into URG flag.

So when URG=1 we set priority=7 in Type of Service field.

Whether URG has to be set or not is decided by AL.

When urgent pkt is received, this is sent to AL first.

Urgent pointer:

- This is used when only some part of segment is urgent, but not all.
- Urgent pointer indicates till what part the data is urgent.

Urgent pointer ~~tells~~ tells offset of bytes in urgent data.

$$\text{I.e., last byte of urgent data} = \text{Seq. num} + \text{Urgent pointer}$$

$$\therefore \text{no of bytes in urgent data} = \text{Urgent pointer} + 1$$

Note:

- When PSH=1, data is sent to AL as soon as it is received without buffering.

So the order in which pkts are received will not be changed.

- When URG=1, data is sent to AL even before the pkts that are received earlier.

So the order in which pkts are received changes.

RST Flag (Reset):

- This is used when there is anything wrong in connection and thus is used to make other host cautious. This is used whenever something out of normal occurs.
- One situation where this is used is if a receiver is expecting some seq. num pkt and if a third party sends some other pkt, ~~the~~ receiver assumes something is wrong and sends a pkt to sender with RST=1.

Window size
→ Used for flow control
→ At the connection field is used

As the sender the

If the sequence

then the

If window

sender side

is over

saying its

If it is

window is

Otherwise

sends ex

as non-ack

Wait again

This per byte

(saying)

and

15/12/20

Window size (or) Advertisement window

- Used for flow control.
- At the connection establishment stage, this window size field is used by receiver to tell its window size to sender.

→ As the communication proceeds, this field is used to tell the sender the available amount of space in receiving window.

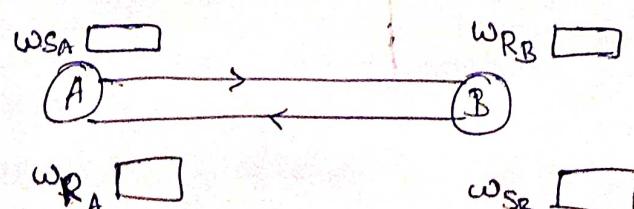
→ If the sender sends data more than the available space, then the data will be discarded.

→ If window size field is found to be 0 by sender then sender starts a persistent timer and waits until the timer is over or until it receives a message from receiver saying its window is free.

→ If it receives message saying that receiver window is free, then sender sends data.

Otherwise, it waits under persistent timer finisher and sends exactly 1 byte data. If sender gets reply with adv as non-zero then it continues with communication, else it will wait again.

→ This persistent timer will be useful in the cases where the message (saying window size is available) sent by receiver is lost.

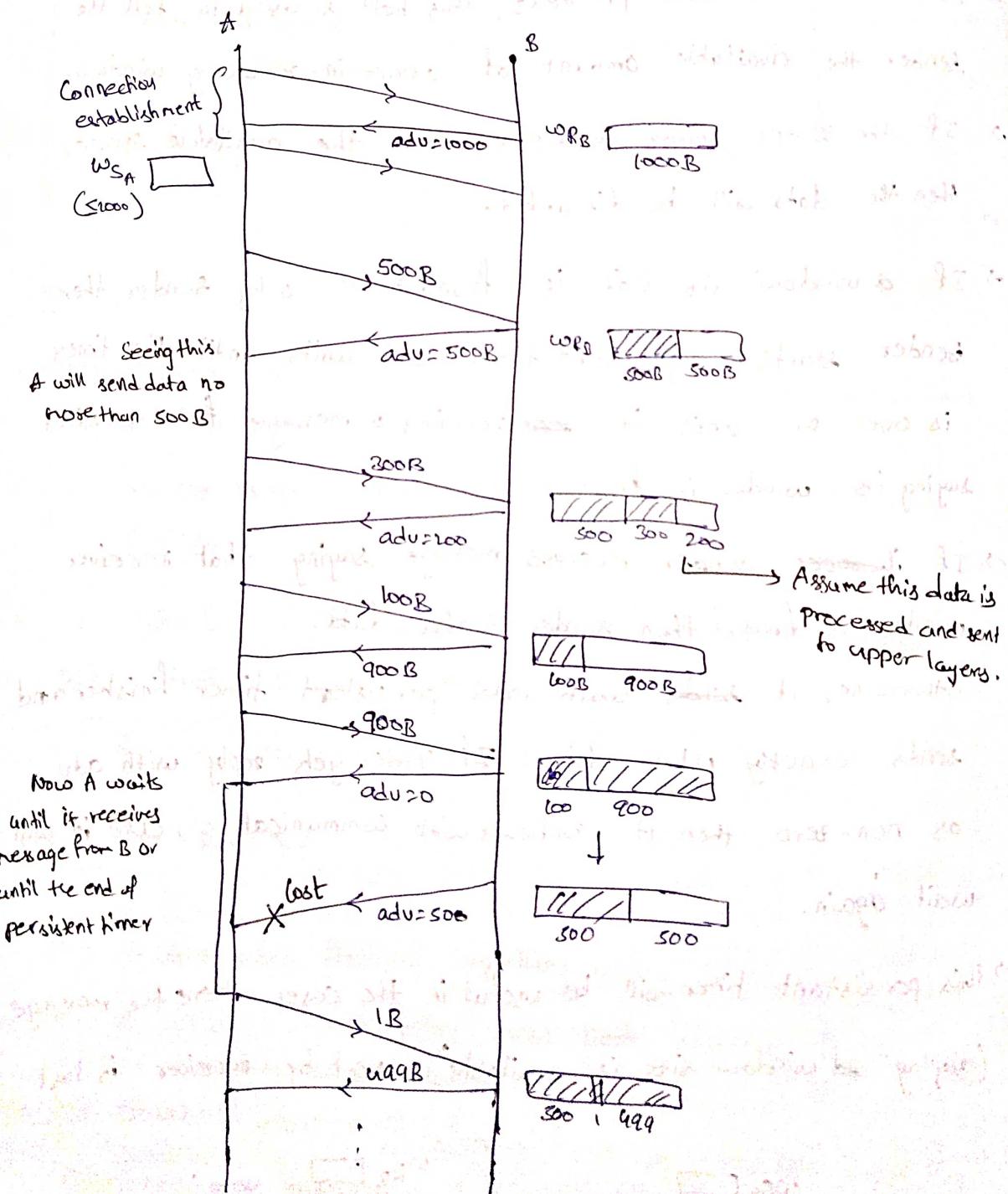


w_{RA} is known to B and w_{RB} is known to A at the time of connection establishment.

$$\omega_{S_A} \leq \cancel{\omega_R} \omega_{R_B} \quad \& \quad \omega_{S_B} \leq \cancel{\omega_R} \omega_{R_A}$$

Based on the values of w_{RA} and w_{RB} , the station B & A, respectively, sets their window sizes.

Consider below case where A sends data to B



pkt:
with 16 bit window size field,
the max size that can be advertised is $2^{16}-1$ Bytes

To overcome this we use extra bits from options

For example if 14B window is available

i.e., 2^{10}

then we take 14 bits from options field.

Checksum:

The checksum in IP header is computed only on IP header.

The checksum in TCP header is computed on IP header, TCP header, data-(~~segment~~).

But, we calculate ~~this~~ the checksum only on some fields of IP ~~and~~ header as many fields ~~are~~ change while the pkt is being transmitted.

In checksum computation we include only those fields of IP header that doesn't change.

~~Checksum~~ in checksum is calculated on below or below fields of IP header (i.e., pseudo IP header)

	Source IP	
	Dest IP	
cccccccc (8)	Protocol (8)	TCP segment length (16)

Total length
- header length

Options :

i) Timestamp: used to increase sequence number bits

It is used when wrap around time < life time

The MSB bits of seq. num are stored in the timestamp

ii) Window size extension: used when size of window is higher than 2^{k-1}

iii) Parameter Negotiation: at the time of connection establishment,

there may be a need to exchange some information. In that case we use this option.

Eg: MSS

iv) Padding: Used to make header size a multiple of 4.

Retransmissions in TCP

→ TCP uses some characteristics of SR and some of GBN

↳ If receiver gets duplicate ACK → congestion control
 $\rightarrow w_S = w_R$

↳ If receiver gets out of order packets are accepted

↳ Cumulative ACK are used

∴ ($7S-1$. SR + $2S-1$. GBN)

→ In TCP, if ack number received is 'x' then sender assumes that all the data before seq. num 'x' is received correctly.

IP uses
This is also
a means to
detect dupli-
cation after
for example
frame size of

create
stop
acks {

→ Retransmission after
also lost and hence
Congestion is also

Congestion Control:

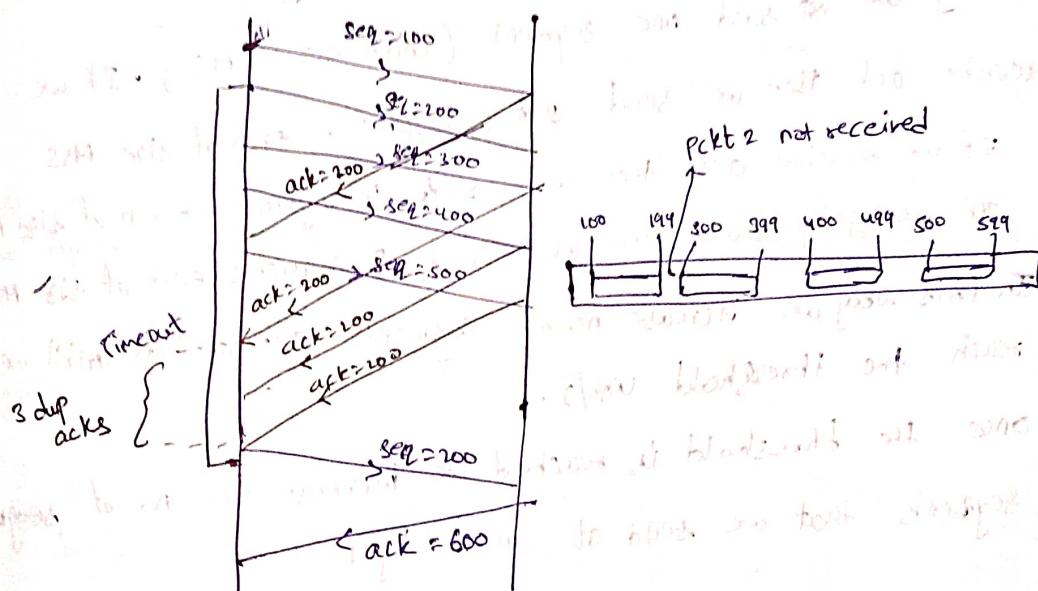
If sender sees
If network then
capacity of

→ TCP uses retransmission after 3 duplicate acknowledgments.

This is also known as Early retransmission.

It means that retransmission done after receiving either 3 duplicate acknowledgments or ~~within the timeout~~ after time out retransmission is done after time out.

For example consider transmitting 5 pkts and 2nd packet is lost. Assume size of each segment is 100.



→ Retransmission after time out indicates that, the duplicate acks are also lost and hence congestion is higher.

Congestion is also indicated by source quench ICMP message.

Congestion Control:

→ If a sender sends packets ~~more~~ than the capacity of network then congestion occurs.

→ If capacity of Ntw is W_c and receiver window size is W_R then

$$\text{Sender window size, } W_S \leq \min(W_c, W_R)$$

bits
≤ lifetime
used in the
process

window is higher

establishment

information. In

packet trades

between stations

multiple of 4.

of GBN

cumulative ack are used

sender assuming
received correctly.

However there is no chance that we can know the value of w_c .

→ So to meet this problem we use 'congestion control protocols'

working:

→ let receiver window size be w_R

so we define ~~threshold~~ threshold = $\frac{w_R}{2}$

→ Initially we send one segment (with size = MSS). If we receive ack then we send 2 segments each of size MSS.

If we receive ack then we send 4 segments each of size MSS.

If we receive ack then we send 8 segments each of size MSS.

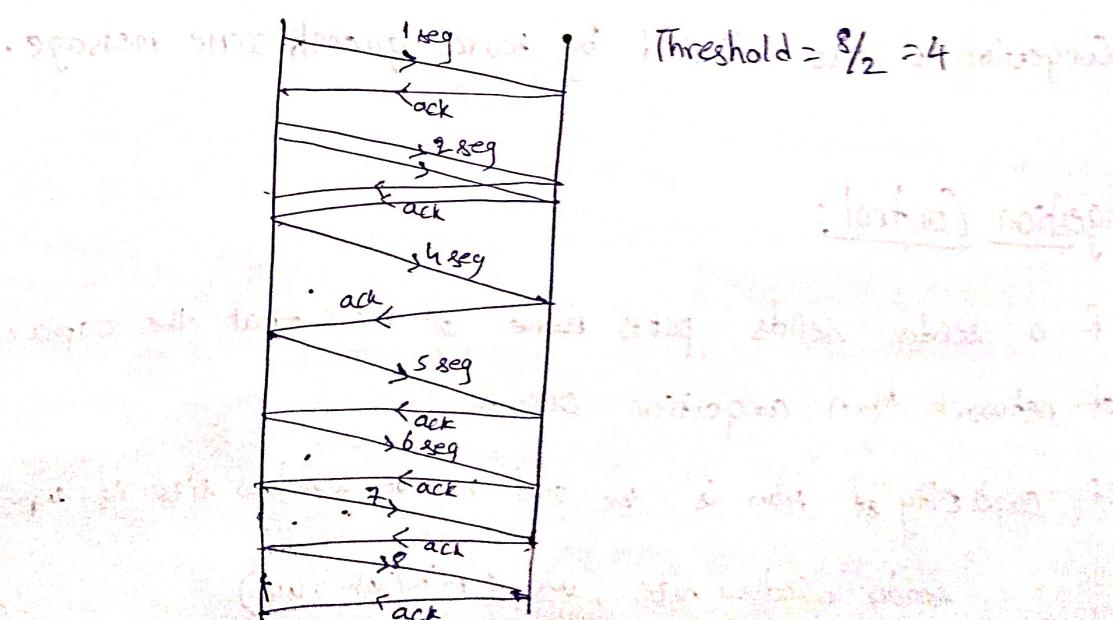
In this way we increase no of segments exponentially until we reach the threshold $w_R/2$.

→ Once the threshold is reached we increase the no of segments that we send at once by 1.

Eg:

Assume $w_R = 8$ segments and $w_c = 8$ segments

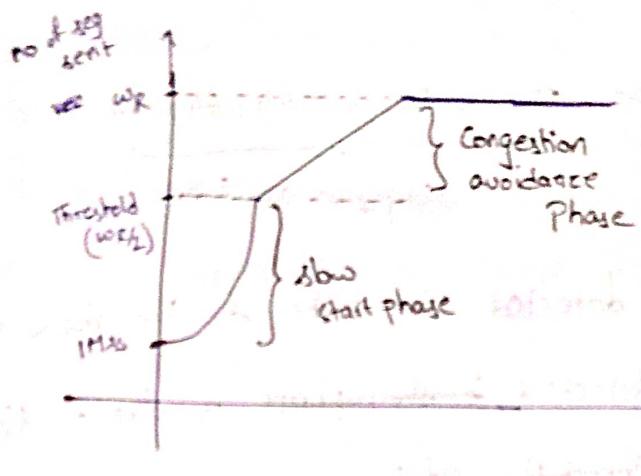
So initially we assume $w_c = 1$ and then increase it if needed.



is the value
of the
control protocols

- Q: After how many round trip times did we reach maximum capacity of ~~rate~~ received

Ans: 6



Note: wr = window size = number of segments sent in one round trip time.
Some implementations we may start with initial $wr = 2$ MSS.

Congestion Control Algorithm involves 3 steps:

- Slow start
- Congestion avoidance phase
- Congestion Detection

Q: Congestion may be detected in slow start phase or congestion avoidance phase.

It is detected in 3 ways

- Timeout (congestion is severe)
- 3 duplicate acks (mild congestion)
- ICMP source quench

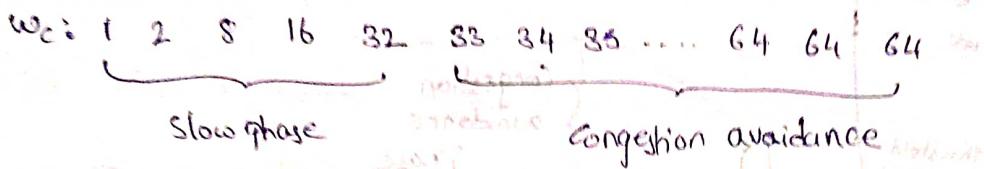
But this source quench talks only about one host but not about entire network. So this may not be considered a congestion.

Eg: $WR = 64 \text{ KB}$ (fixed window growth after loss of first 32 KB)

$MSS = 1 \text{ KB}$

$WR = 64 \text{ MSS}$

RTT Threshold = 32 MSS



Congestion may be detected in either of the above phases.

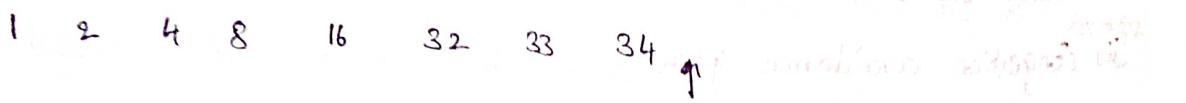
If congestion is detected in slow phase is due to timeout
then ~~threshold = add 1~~

then threshold = $Wc/2$ and enter slow start phase.

If congestion is detected is due to 3 dup ack then

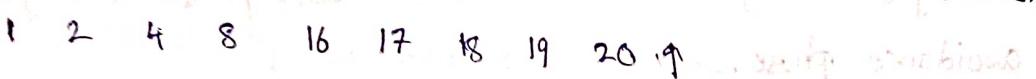
threshold = $Wc/2$ and enter congestion avoidance phase

Eg:



congestion detected by timeout

$\therefore \text{new threshold} = 34/2 = 17$ & enter slow start phase



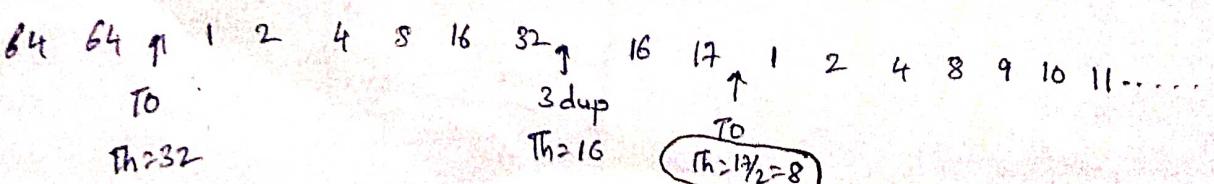
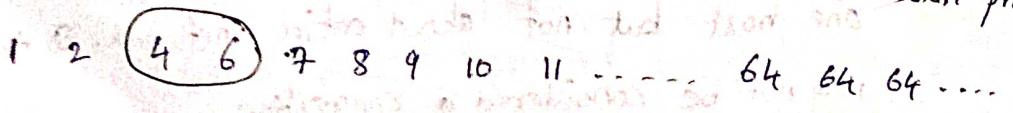
congestion detected by 3-dup pckts

$\therefore \text{new threshold} = 20/2 = 10$ & enter congestion avoidance phase



congestion detected by timeout

$\therefore \text{Threshold} = 12/2 = 6$ & enter slow start phase



TCP timer management:

There are 5 types of timers

i) Time-wait timer:

→ closing connection immediately may assign the same port number to a new process. Due to this the delayed packets of previous communication may be delivered to this new process.

So rather than closing connection immediately, we wait for time-wait timer.

$$\text{time wait timer} = 2 \times \text{life-time.}$$

→ So time wait timer handles delayed packets.

ii) keep alive timer:

This used to close idle connections.

→ These are connections that are opened but no communication is done.

Everytime a packet is received, the system restarts keep alive timer. If no packet is received until keep alive timer ends, then the station sends 10 probe messages. If no reply is received then we close the connection.

iii) Persistent timer:

This is used when a station receives window size of 0 from other host. After this timer we send 1 byte to test if the window is free.

above phases.

to timeout

slow start phase,
then

on avoidance phase

ed by timeout

slow start phase

by 3 dup pkts

avoidance phase

start phase

8 9 10 11 ...

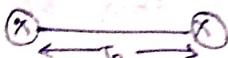
(iv) Acknowledgment timer:

This used to send cumulative acknowledgments.

Also using this we can wait until data is ready and thus implement piggybacking.

(v) Timeout timer

At DLL



$$\text{Round trip time} = 2 \times T_p$$

$$\text{timeout} = 2 \times \text{RTT}$$

But at transport layer deriving and finding $2 \times \text{RTT}$ in this way won't be easy. The path, traffic, may change dynamically. Hence the timeout timer at TCP also has to be dynamic.

→ Having a static TO timer may lead to congestion.

Consider a case where traffic is high but there is congestion.

If TO is small and fixed (static) then we will keep on retransmitting again and again causing congestion.

→ When TO is too high we wait a lot of time for retransmission and it is a waste of time.

(i) Basic algorithm

(ii) Jacobson's algorithm

Basic Algorithm:

Initially we predict RTT (say IRTT) and set $\text{TO} = 2 \times \text{IRTT}$.

Let Assume we received ack after some time and we say this is actual RTT (say ARTT). Using IRTT and ARTT we predict NRTT (next RTT).

$$NRTT = \alpha (IRTT) + (1-\alpha) ARTT$$

$\alpha = 1 \Rightarrow$ algorithm is static

$\alpha < 0 \Rightarrow$ algorithm is fully dynamic

α is called smoothing factor.

Eg: let $IRTT = 10 \text{ ms}$ & $\alpha = 0.5$

$$\Rightarrow TO = 2 \times IRTT$$

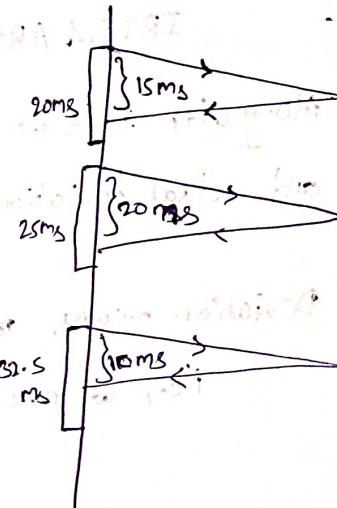
$$= 20 \text{ ms}$$

let $ARTT = 15 \text{ ms}$

$$\Rightarrow NRTT = 0.5(10) + 0.5(15)$$

$$= 25/2$$

$$\Rightarrow \text{new TO time} = 25 \text{ ms}$$



Now for 2nd transmission

$$IRTT = 25/2 \text{ ms}$$

$$ARTT = 20 \text{ ms}$$

$$\Rightarrow NRTT = 0.5(25/2) + 0.5(20)$$

$$= 6.25 + 10 = 16.25$$

$$\Rightarrow TO = 32.5 \text{ ms}$$

Now for 3rd transmission

$$IRTT = 16.25 \text{ ms}; ARTT = 10 \text{ ms};$$

$$NRTT = 13.125 \text{ ms}$$

$$\Rightarrow TO = 26.25 \text{ ms}$$

Note:

→ Here NRTT depends on entire history of RTTs but previous RTTs have lesser weights.

Disadv: The disadvantage with this algorithm is that $TO = 2 \times RTT$ i.e., we always multiply RTT with a fixed number(2)

Jacobson's Algorithm:

→ This algorithm overcomes the disadvantage of basic algorithm

i.e., $T_0 = 2 \times RTT$

↳ packet retransmission after $2 \times RTT$

Using this has no logic & reason.

→ Here also we have

$IRTT, ARTT$

Along with this we define 2 new parameters, initial deviation (ID) and actual deviation (AD) and next deviation (ND).

Deviation means ~~the~~ error in our guess of RTT.

i.e., we are expecting ARTT to be in range $[IRTT - ID, IRTT + ID]$

$$[IRTT - ID, IRTT + ID]$$

AD is actual observed deviation

ND is predicted deviation for next transfer

$$ND = \alpha(ID) + (1-\alpha)AD \quad \& \quad AD = |IRTT - ARTT|$$

→ Here timeout is given by

$$TO = \underbrace{(4 \times D)}_{\downarrow \text{deviation}} + RTT$$

This is experimentally found to give better results.

Eg: Let $IRTT = 10 \text{ ms}$, $ID = 5 \text{ ms}$, $\alpha = 0.5$

for 1st transmission:

$$TO = 4 \times D + RTT = 30 \text{ ms}$$

let $\Rightarrow ARTT = 20 \text{ ms}$

$$AD = |IRTT - ARTT| \approx 10 \text{ ms}$$

$$\Rightarrow NRTT = 0.5(10) + 0.5(20)$$

$$= 15 \text{ ms}$$

$$ND = 0.5(5) + 0.5(10)$$

$$= 7.5$$

for 2nd transmission:

$$IRTT = 15 \text{ ms} ; \Delta ID = 7.5 \text{ ms}$$

$$\therefore T_0 = 4(7.5) + 15$$

$$= 45 \text{ ms}$$

$$\Rightarrow \text{let } AD = ARTT = 30 \text{ ms}$$

$$\Rightarrow AD = |15 - 30| = 15 \text{ ms}$$

$$\Rightarrow NRTT = 0.5(15) + 0.5(30)$$

$$= 22.5$$

$$ND = 0.5(7.5) + 0.5(15)$$

$$= 11.25$$

Note (Karn's Modification):

Neither of the two algorithms talk about the case where ack is not sent within the timer.

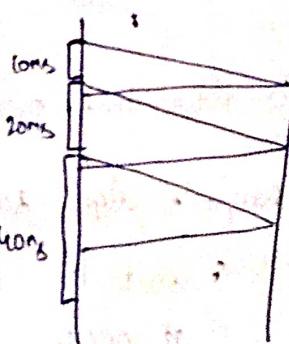
i.e., These algorithms don't tell what should be the initial prediction & timeout for retransmission.

Karn's modification to Jackson's algorithms says, whenever retransmission is required,

let T_0 to double of the previous T_0 .

whenever you receive ack in time, you can

continue with normal Jackson's algorithm.



18/11/20

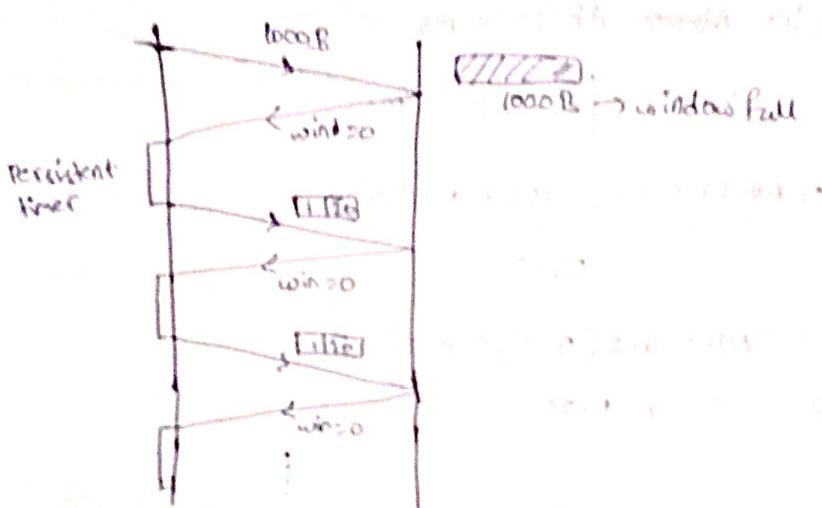
Silly Window Syndrome:

This

→ we say silly window syndrome problem occurs in 3 cases

- i) Receiver always advertises that its window size is 0 for certain amount of time.

But this problem is temporary and will be solved once window is available.



In this case, we use RST flag and restart the connection.

- ii) Second case where this problem occurs is when sender is slow and transmitting less amount of data (say 1 byte) at once. This means for 1 byte, ($1+20+20+20+20$) 81 bytes have to be transmitted. So wait for acknowledgement for each byte.

To solve this problem we use Nagle's algorithm.

Nagle's algo says if producing data at the rate of 1 MSS then wait for 1 RTT and send all the data produced in 1 RTT at once.

- iii) If data sufficient for 1 MSS is produced in less than 1 RTT then transmit it.

Traffic Shaping

- Another method before it enters
- Traffic shaping
- During connection a traffic pattern

Traffic shaping

- i) Le
- ii) Te

Leaky Bucket:

- here bucket
- Data may
- but it is taken
- of a constant

(iii) Another case in which this problem occurs is that when receiver accepts only one byte of data repeatedly (i.e., receiver advertises its window size as 1)

Clark's solution for this problem:

Receiver should not advertise until it gets freed

~~at least~~ half of the window or until it gets freed i MSS size of buffer

Traffic Shaping:

- Another method of congestion control is to shape the traffic before it enters network.
- Traffic shaping controls the rate at which the pkts are sent.
- During connection establishment, the sender and carrier negotiate a traffic pattern (shape)

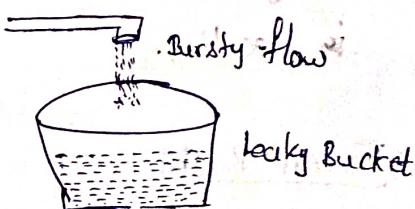
Traffic shaping is done using 2 algorithms:

i) Leaky bucket

ii) Token bucket

i) Leaky Bucket:

Here bucket corresponds to queue.



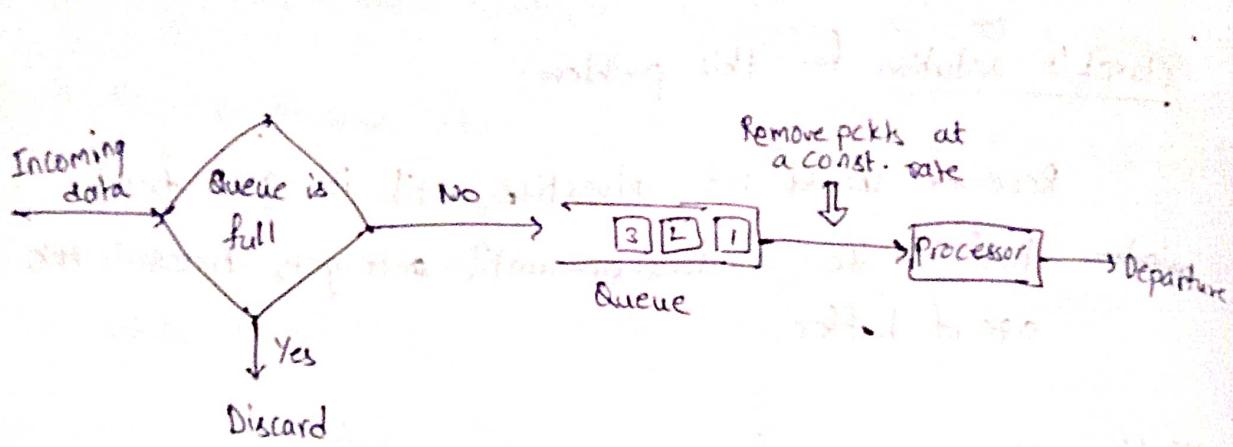
Data may come into queue at any rate

But it is taken out and put on the network at a constant rate.

However the queue (bucket)

However the bucket (queue) has some capacity.

If earlier bucket is full and new data comes in, then the new data is discarded.



- Here required outflow is x bytes per second and if pkt size is of variable length, then we use a counter to count no of bytes in queue and transmitting no of bytes.

Disadv:

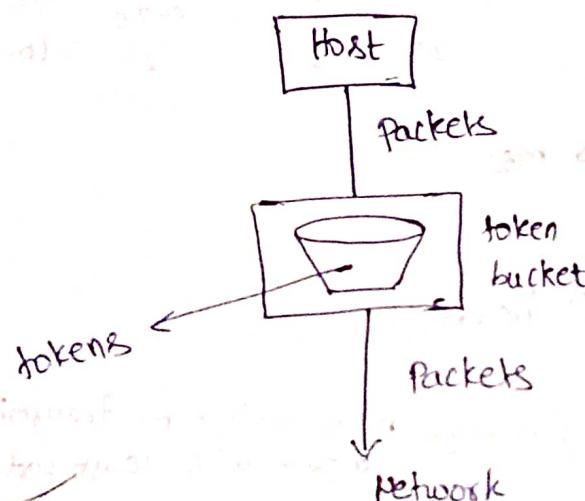
- If a station doesn't transmit data for long time and if it needs to transmit after sometime, the station will have some outflow rate and no extra rate will be given for the reason that the station didn't transmit earlier.

Token Bucket

- This overcomes the disadvantage of token leaky bucket.
- Each pkt needs a token to get on the network.
- Here we maintain a token bucket to which tokens are

added at a fixed rate.

- In this way even if a system doesn't transmit for long time and need to transmit later, it can use the tokens and transmit packets (one per each token)
- However, token bucket also has a capacity. Once token bucket is full, no more tokens will be added.



- If maximum capacity of token bucket is c tokens and token enter bucket at rate of g token/sec. then find max no of packets that can enter network during time interval t is _____

$$\text{Max no of packets} = c + g t$$

$$\Rightarrow \text{Max avg rate} = \frac{c + g t}{t} \text{ packets/sec}$$

- For large value of t , max avg rate = g
- Token bucket capacity = 250 kB; tokens arriving rate = 2MB/s
- If maximum output rate is 25MB/sec, then find burst time.

Sol:

War pigs were used in Roman warfare to scare the elephants. To enhance this, the pigs were set on fire.



since given token capacity = 250 kB

each byte is a token

Burst time is duration in which we send 25 MB

$\therefore C + rt$

$$\frac{C + rt}{t} = M$$

$$\Rightarrow C + rt = Mt \Rightarrow t = \frac{C}{M - r}$$

$$t = \frac{250}{25 - 2} = 0.25$$

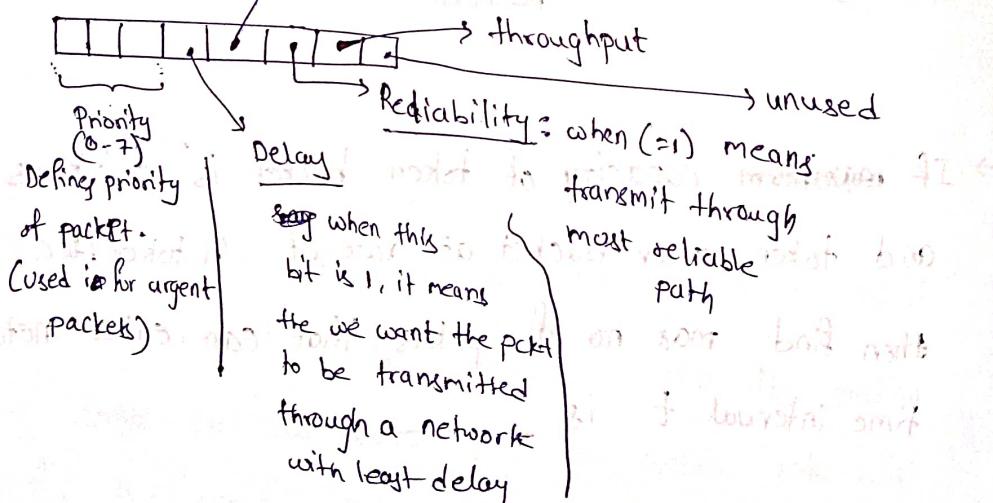
$$= \frac{0.25}{23} = 10.86 \text{ ms}$$

\therefore Burst time = 10.86 ms

Type of Service (field in IP header)

→ It is 8-bit field.

cost: when (=1) says pkts needs to be transmitted in a path with least cost



→ when throughput = 1, it means that packet has to be transmitted through a network where throughput is high

Portland was named after flipping a coin. Had the coin landed the other way, the city would be known as Boston.

The term astronaut is derived from two German words, which when put together mean 'star sailor'.

In medieval Germany, soldiers believed that kissing a donkey was the perfect remedy for a toothache.