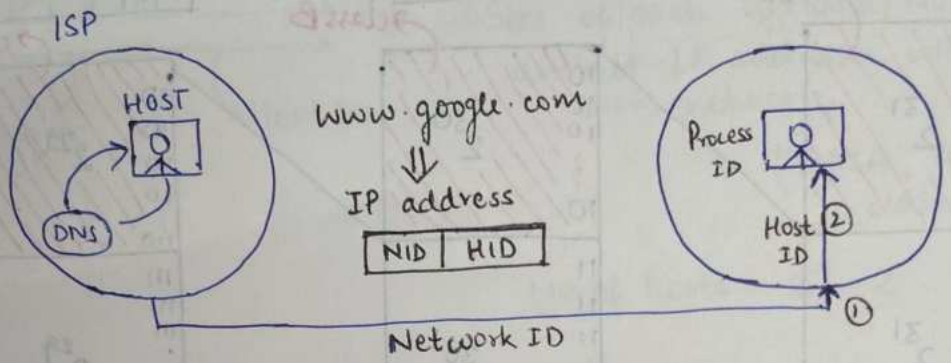
Introduction to Computer Networks and IP Address



- Service that is used to convert domain name into ~~A~~ into IP address is called domain name service.
- Port number is used to signify a particular process in the host. For well known services, port number is predefined & fixed.

  http :- 80 SMTP :- 25

  ftp :- 21

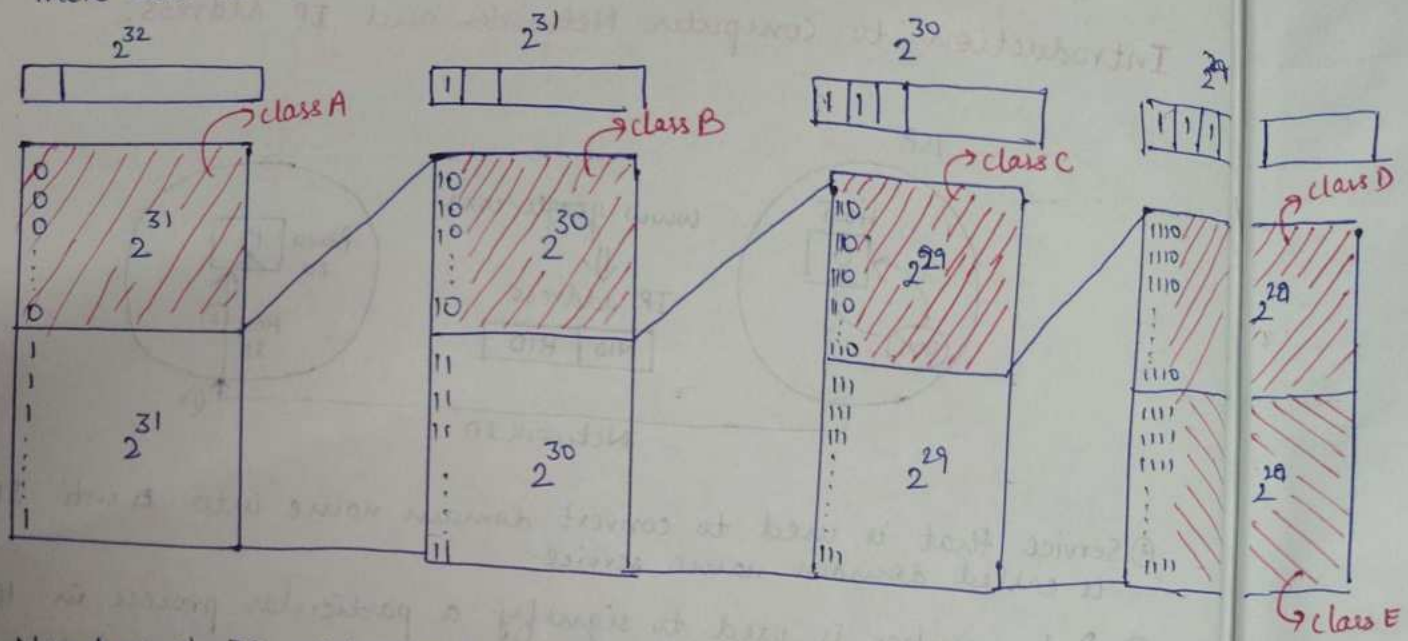
- Even though we need to reach google.com home page, we need to visit DNS which gives the IP Address of google.com home page. This is known as **DNS Overhead.**

  Rectification :- The IP address of google.com is stored ~~actually~~ locally in the computer system for some time. If again we want to visit that page, we can directly get IP address

  If IP address in the system expires, we have to go to DNS (no alternative)

Classful IP address classification

There are 32 bits in the IP address.



Number of IP addresses in a network of class A = 2^{31}

Class B = 2^{30}

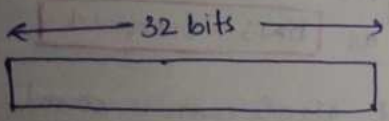
class C = 2^{29}

class D = 2^{28}

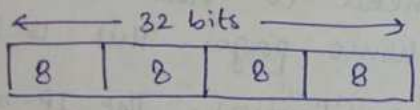
class E = 2^{28}

Popular representation of IP address —

→ Binary notation



→ Dotted Decimal Notation (4octets)



(192 · 168 · 173 · 9)
8 bits 8 bits 8bits 8bits.

Class A → starts with 0

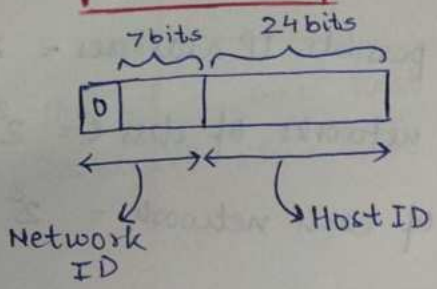
class B → starts with 10

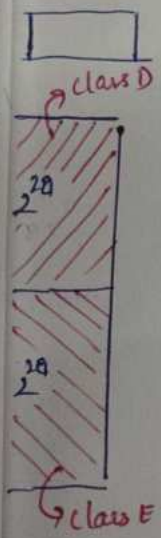
class C → starts with 110

class D → starts with 1110

class E → starts with 1111

CLASS A





Number of possible n/ws of class A = $2^7 = 128$

Size of each network / No. of distinct IP addresses within each network = $2^{24} = 16M$

(NASA, PENTAGON use class A network)

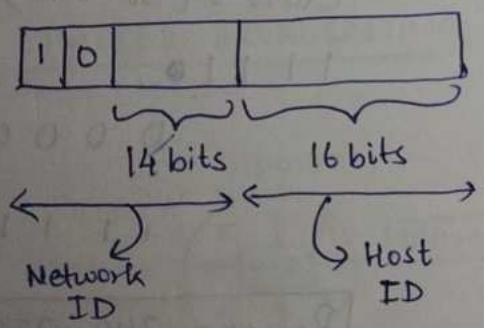
No. of hosts = $2^{24} - 2$

⊙ Starting address (All zeroes) and ending address (all ones) are not used

Number of networks in class A = 128 − 2 = 126

∴ Range = 0 − 127

CLASS B



No. of IP addresses possible = 2^{30}

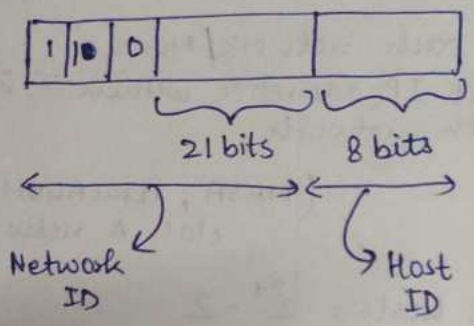
No. of networks of class B = 2^{14}

Size of each network = 2^{16}

No. of hosts = $2^{16} - 2$

Range = 128 − 191

CLASS C



No. of possible IP addresses = 2^{29}

No. of networks of class C = 2^{21}

Size of each network = 2^8

No. of hosts in each network = $2^8 - 1$

Range = 192 - 223

CLASS D And CLASS E

There is nothing like network IDs and host IDs.

Class D (1st octet)

1 1 1 0 _ _ _ _

0 0 0 0 = 224

⋮

1 1 1 1 = 239

Range = 224 - 239

- used for multicasting
- used for group emailing, group broadcasting.

Class E (1st octet)

1 1 1 1 0 _ _ _

0 0 0 0 = 240

⋮

1 1 1 1 = 255

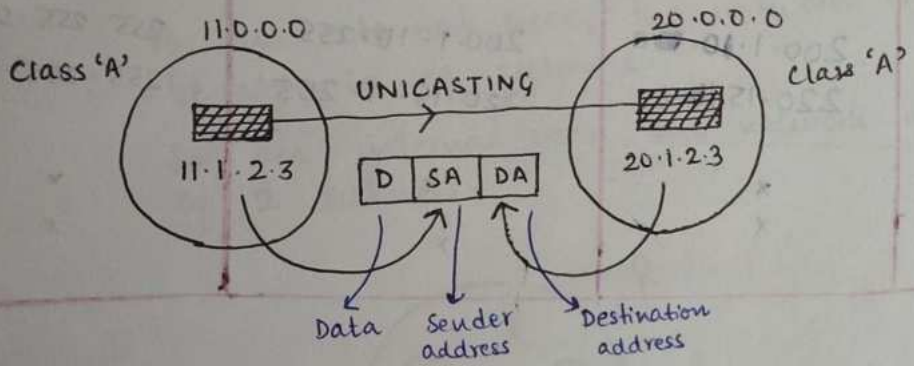
Range = 240 - 255

- used for military applications.

Types of casting

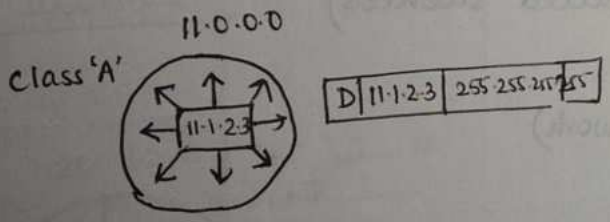
- Unicast
- Limited Broadcast
- Directed Broadcast

1. Unicast :- one host to one host
2. Broadcast :- one host to many hosts
   - Limited Broadcasting
   - Directed Broadcasting



When 'Host ID' has all 0's then the IP address represents the network. This is the reason first IP address in a network is never used as a valid IP address of the host.

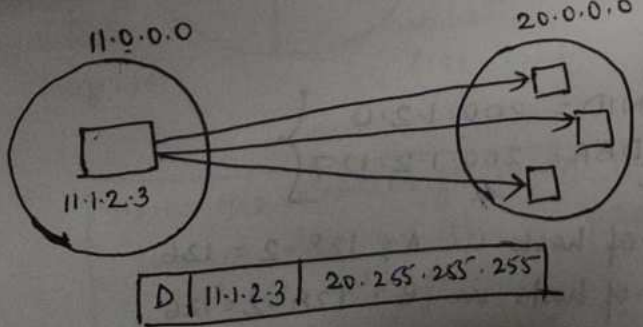
LIMITED BROADCASTING



If Destination address contains all 1's in IP address i.e. 255.255.255.255 then, the packet will be sent to all the hosts within that network.

LBA = 255.255.255.255

DIRECTED BROADCASTING



If Destination address contains all 1's in the Host ID part, i.e. X.255.255.255, then, the packet will be sent to all hosts in the network X000

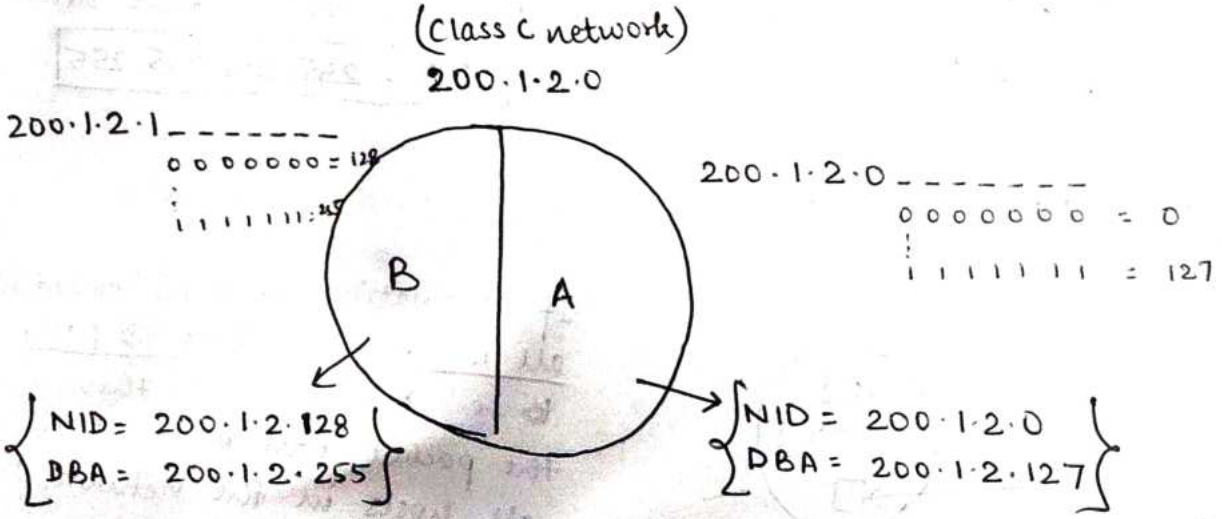
DBA : NID, HID = all 1's

	I/P address	Network ID	Directed Broadcast address	Limited Broadcast Address
Class A	1.2.3.4	1.0.0.0	1.255.255.255	255.255.255.255
	10.15.20.60	10.0.0.0	10.255.255.255	255.255.255.255
Class B	130.1.2.3	130.1.0.0	130.1.255.255	255.255.255.255
	150.0.150.150	150.0.0.0	150.0.255.255	255.255.255.255
Class C	200.1.10.100	200.1.10.0	200.1.10.255	255.255.255.255
	220.15.1.10	220.15.1.0	220.15.1.255	255.255.255.255
Class E	250.0.1.2	x	x	x
Invalid	300.1.2.3	x	x	x

SUBNETS, SUBNET MASK, ROUTING

- When the size of network is big, its maintainence will be difficult.
- Also there is a lack of ~~sec~~ security in a big network.

- **Subnetting :-** It is the process of dividing a network into smaller networks (called subnets)



No. of hosts in A = 128 - 2 = 126
No. of hosts in B = 128 - 2 = 126
Total hosts in 200.1.2.0 = 126 + 126 = 252
∴ Subnetting results in decrease in no. of valid hosts

"There will be loss of IP address due to SUBNETTING"

Ambiguity -

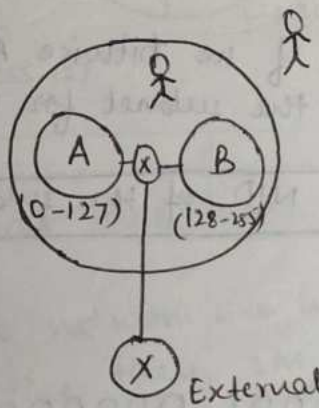
Q.1. 200.1.2.0 points to the whole network or only subnet A ?

Q.2. If 200.1.2.255 is the destination address of a packet, should it be sent to all hosts within the network or to only hosts in subnet B ?

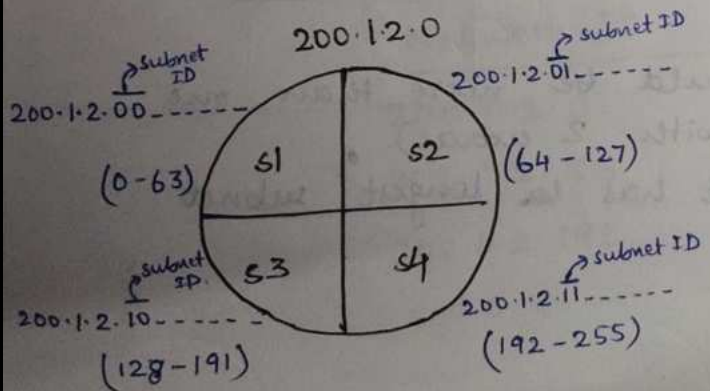
Answer :- It depends upon from where are we seeing the n/w

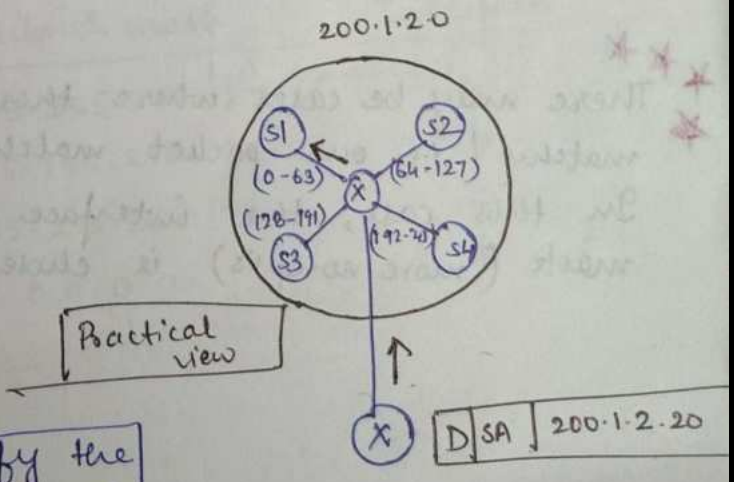
For an external user, the network would be 1 (he won't be able to see subnets.

For an internal user, the network would be made up of 2 subnets



Four subnets -





Here the challenge is to identify the subnet for which a particular IP address belongs.
This is done by using 'SUBNET MASK'

SUBNET MASK

Subnet Mask = 32 bit number

Network ID, subnet ID part — 1's Host ID part —— 0's

Subnet mask for the 4 subnet network shown above is

200 · 1 · 2 · _ _ HID
NID SID

⇒ 11111111 · 11111111 · 11111111 · 11000000

SM = 255 · 255 · 255 · 192

Given an IP address, if we bitwise AND with SM, we will get the NID of the subnet for which the IP belongs to.

SM & IP = NID of the subnet

IP : 11001000 · 00000001 · 00000010 · 10000010 = 200 · 1 · 2 · 130
SM : 11111111 · 11111111 · 11111111 · 11000000 = 255 · 255 · 255 · 192
NID : 11001000 · 00000001 · 00000010 · 10000000 = 200 · 1 · 2 · 128

∴ The IP address belongs to the subnet | 200 · 1 · 2 · 128 |

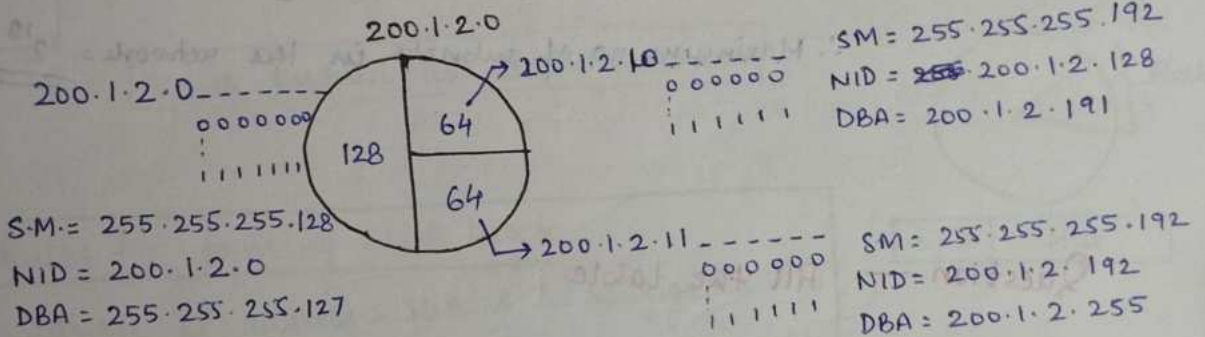
There may be cases where there could be more than one matches (i.e. our packet matches with 2 entries). In this case, the interface that has the longest subnet mask (more no. of 1's) is chosen.

VARIABLE LENGTH SUBNET MASKING

Dividing a network into subnets of unequal sizes is called variable length subnet masking (VLSM)

When subnet size is different, then each subnet will get a different subnet mask.

Three subnets



→ The networks that have same sizes have ~~equal~~ same subnet mask

→ When the network size is big, it means there are more hosts, then, SM will be small.

When the network size is small i.e. less no. of hosts, then, SM will be big.

Routing table

Network ID	Subnet mask	Interface
200.1.2.0	255.255.255.127	a
200.1.2.128	255.255.255.192	b
200.1.2.192	255.255.255.192	c
0.0.0.0	0.0.0.0	d

Finding number of subnets from the subnet mask & class of network

Given subnet mask = 255.255.255.192

= 11111111.11111111.11111111.11000000

= NID + SID = No. of 1's

NID + SID = 26

If the network belongs to class A, then,

NID = 8

∴ 8 + SID = 26 ⇒ SID = 18

∴ Maximum no. of subnets in the network = 2^{18}

Question — Fill the table.

Subnet mask	No. of hosts	Subnets in class A	Subnets in class B	Subnets in class C
255.0.0.0	$2^{24}-2$	1	—	—
255.128.0.0	$2^{23}-2$	2^1	—	—
255.192.0.0	$2^{22}-2$	2^2	—	—
255.240.0.0	$2^{20}-2$	2^4	—	—
255.255.0.0	$2^{16}-2$	2^8	1	—
255.255.254.0	2^9-2	2^{15}	2^7	—
255.255.255.0	2^8-2	2^{16}	2^8	1
255.255.255.224	2^5-2	2^{19}	2^{11}	2^3
255.255.255.240	2^4-2	2^{20}	2^{12}	2^4

00000000	0
10000000	128
11000000	192
11100000	224
11110000	240
11111000	248
11111100	252
11111110	254
11111111	255

Classless Inter Domain Routing (CIDR)

- No class A / class B / class C / class D / class E network is present.
- The number of bits in BID part is specified with the IP address in CIDR notation

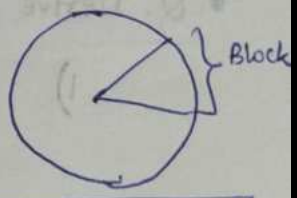
CIDR notation :- a.b.c.d/(n)

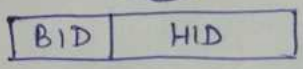
- a.b.c.d → 32 bit IP address
- (n) → No. of bits in BID

If n = 20, then,
BID = 20 bits
HID = 12 bits

Instead of NID, BID is used (Block ID)

n is known as slash number





Rules for forming CIDR block

- All IP addresses should be contiguous
- Block size should always be power of 2
  - ↳ No. of hosts in block
- First IP address in the block should be evenly divisible by the size of block.

Q:- Tell whether the following IP addresses form a CIDR block?

1) 100.1.2.32
   100.1.2.33
   ⋮
   100.1.2.47

Rule 1: ✓

Rule 2: No. of IP addresses = 16 = 2^4 ✓

Rule 3: First IP address —
100.1.2.00100000 ✓
↳ 0's ∴ divisible by 2^4

∴ The IP addresses form CIDR Block

2) 20.10.30.32
   20.10.30.33
   ⋮
   20.10.30.63

Rule 1: ✓

Rule 2: No. of IP addresses = 32 = 2^5 ✓

Rule 3: First IP address —
20.10.30.00100000 ✓
↳ 0's ∴ divisible by 2^5

∴ The IP addresses form CIDR block

3) 150.10.20.64
150.10.20.65
⋮
150.10.20.127

Rule 1: ✓
Rule 2: No. of IP addresses = 64 = 2^6 ✓
Rule 3: First IP address —
150.10.20.01000000 ✓
↳ 0's ∴ divisible by 2^6

∴ IP addresses form valid CIDR block

Q. Derive the range of CIDR block —

1) 20.10.30.35/27

BID = 27 bits HID = 5 bits

20.10.30.00100011
BID | HID

00000 (32)
00001 (33)
⋮
11111 (63)

∴ Range is
20.10.30.32
20.10.30.33
⋮
20.10.30.63

2) 100.1.2.35/20

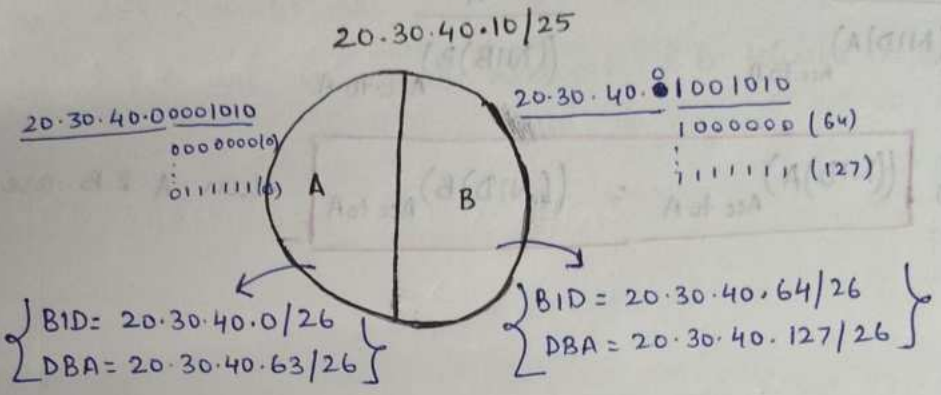
BID = 20 bits HID = 12 bits

100.1.00000010.00010100
BID | HID

0000.00000000 (0.0)
0000.00000001 (0.1)
⋮
1111.11111111 (15.255)

∴ Range is
100.1.0.0
100.1.0.1
⋮
100.1.15.255

Subnetting in CIDR



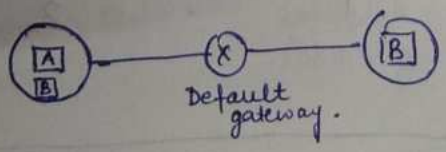
VLSM in CIDR blocks —

[Same as classful]

IMPORTANT QUESTIONS ON SUBNET MASK

- Subnet mask is also known as 'Network mask'
- 'ipconfig' command gives the following information —
  - → IPv4 address :- provided by ISP
  - → Default gateway :- default router connected to the network
  - → Subnet Mask :- Subnet mask that should be used.
  - → DNS :- conversion to IP address.

Let I_A = IP address of A
S_A = subnet mask of A
I_B = IP address of B
S_B = subnet mask of B.



On sending a packet from A to B, 2 cases are possible —

- If A & B are in same network, then, the packet can be transferred directly.
- If A & B are in different network, then, packet is sent from A to DGW and then from DGW to B.

A: I_A I_B

S_A S_A

Bitwise AND :- $((NID)A)_{Acc\ to\ A}$ $((NIB)B)_{Acc\ to\ A}$

If $\boxed{((NID)A)_{Acc\ to\ A} = ((NID)B)_{Acc\ to\ A}}$ then A & B are in same n/w

Example 1:-

$I_A = 200.1.2.10$ $I_B = 200.1.2.130$

$S_A = 255.255.255.128$

I_A : 11001000 . 00000001 . 00000010 . 00001010

S_A : 11111111 . 11111111 . 11111111 . 10000000

11001000 . 00000001 . 00000010 . 00000000

= 200 . 1 . 2 . 0

$((NID)A)_{Acc\ to\ A}$ = 200.1.2.0

I_B = 11001000 . 00000001 . 00000010 . 10000010

S_A = 11111111 . 11111111 . 11111111 . 10000000

11001000 . 00000001 . 00000010 . 10000000

200 . 1 . 2 . 128

$\therefore \boxed{((NID)A)_{Acc\ to\ A} \neq ((NID)B)_{Acc\ to\ A}}$

$\therefore$ A assumes A & B are in different network.

Example 2: (Practice)

I_A = 200.1.2.10 I_B = 200.1.2.69

S_A = 255.255.255.128 S_B = 255.255.255.192

Ans:- [A thinks A & B are in same network & B thinks A is in different n/w]

Supernetting or Aggregation

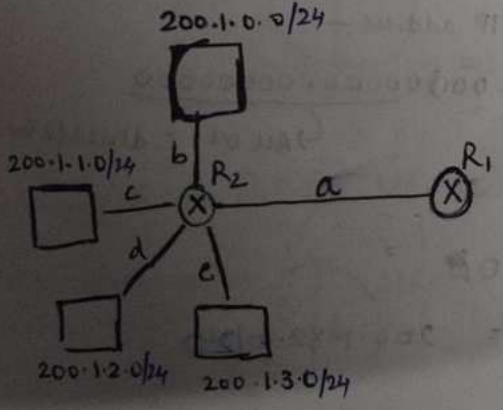
- Routing table contains single entry for each and every network.
- If no. of networks is large, then, the size of routing table will grow exponentially and the router takes a lot of time to process it.
- Hence we need to aggregate / combine the networks.
- This process is known as supernetting / aggregation.

Rules for aggregation

↳ All the network IDs should be contiguous.

↳ Size of each network should be same and they should be power of 2.

↳ The 1st IP address should be evenly divisible by size of the block.



Routing table for R_1 —

NID	Subnet mask	Interface
200.1.0.0	255.255.255.0	a
200.1.1.0	255.255.255.0	a
200.1.2.0	255.255.255.0	a
200.1.3.0	255.255.255.0	a

Redundancy

Q.1. Aggregate the following network IDs

1) 200.1.0.0/24
2) 200.1.1.0/24
3) 200.1.2.0/24
4) 200.1.3.0/24

Rule 1: ✓

Rule 2: Size of each n/w = 2^8 ✓

Total size of network = $4 \times 2^8 = 2^{10}$

Rule 3: First IP address —

200.1.00000000.00000000

↳ All 0's ∴ divisible by 2^{10}

∴ The networks can be aggregated.

SUPERNET MASK

32 bits

No. of 1's = Fixed part

No. of 0's = Variable part / no. of hosts

∴ Supernet mask = 255.255.252.0

N/w ID of the aggregated network is 200.1.0.0/24

Q.2. Aggregate the following network IDs

a) 200.1.32.0/24
200.1.33.0/24
⋮
200.1.47.0/24

Rule 1: ✓

Rule 2: Size of each n/w = 2^8

Total size of aggregated network = $2^4 \times 2^8 = 2^{12}$ ✓

Rule 3: First IP address —

200.1.00100000.00000000

↳ All 0's ∴ divisible by 2^{12}

∴ The networks can be aggregated.

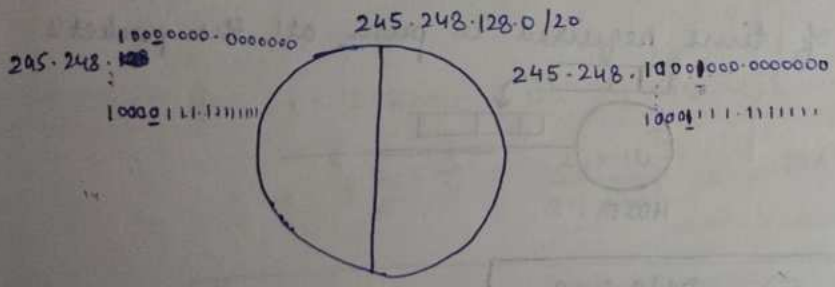
SUPERNET MASK = 255.255.240.0

N/w ID for aggregated network = 200.1.32.0/20

Gate 2012

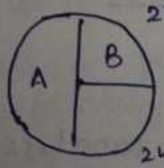
ISP has the following chunk of CIDR based IP addresses available with it : 245.248.128.0/20. The ISP wants to give half of this chunk of IP addresses to organization A and a quarter to organization B while retaining the remaining with itself. Which of the following is a valid allocation of addresses to A & B.

a) 245.248.136.0/21 and 245.248.128.0/22
b) 245.248.128.0/21 and 245.248.128.0/22
c) 245.248.132.0/22 and 245.248.132.0/21
d) 245.248.136.0/24 and 245.248.132.0/21



A can be 245.248.128.0/21 or 245.248.136.0/21

If A is 245.248.128.0/21 —



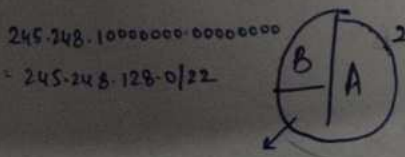
245.248.1000 1000.0000 0000 = 245.248.136.0/22

OR

245.248.1000 1100.0000 0000 = 245.248.240.0/22

} Not present in option.

If A is 245.248.136.0/21 —



245.248.1000 1000.0000 0000
245.248.136.0

∴ A = 245.248.136.0/21
B = 245.248.128.0/22

FLOW CONTROL METHODS

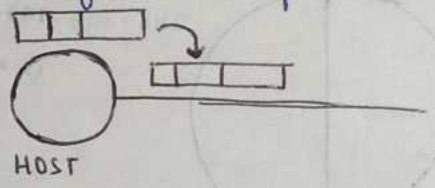
Delays in Computer Network:-

1) Transmission (T_t) Delay –

The time taken to transmit the packet from the host to the outgoing line is called transmission delay.

Also known as store-and-forward delay & packetization delay.

It is the amount of time required to push all the packet's bits into the wire.



$$\text{Transmission delay} = \frac{\text{Data size}}{\text{Bandwidth}}$$

Q:- Bandwidth = 1 bps [1 bit per second]
Data = 10 bits
∴ Transmission delay = $\frac{10}{1} = 10s$

Q:- Bandwidth = 1 kbps , data size = 1000 bits
Transmission delay = $\frac{1000}{1000} s = 1s$

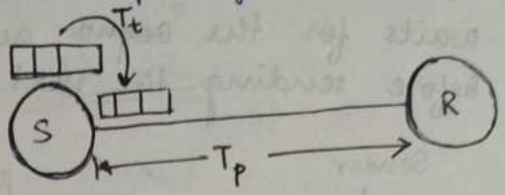
Q:- Bandwidth = 1 Kbps , data size = 1kB
Transmission delay = $\frac{1024}{1000} = 1.024\ s$

For data size, (bcoz data size is measured in binary)
- K = 1024
- M = 1024 × 1024
- G = 1024 × 1024 × 1024

For Bandwidth (bcoz bandwidth is measured in decimal)
- K = 1000
- M = 10^6
- G = 10^9

2) Propagation delay (T_p) –

The time taken by a bit to reach from one end of the link to the other end of the link is called propagation delay.



Propagation delay depends upon —

1) Distance b/w sender & receiver
2) Velocity of the signal in the transmission medium

$$T_p = \frac{d}{v}$$

For optical fibres,

v = 70% speed of light = $\frac{70}{100} \times 3 \times 10^8$ = 2.1×10^8 m/s

Q:- $d = 21$km $\quad v = 2.1 \times 10^8$ m/s $\quad T_p = ?$

$$T_p = \frac{d}{v} = \frac{21 \times 10^3}{2.1 \times 10^8} = 10 \times 10^{-5} \text{ s} = \underline{\underline{10^{-4} \text{ s}}}$$

Total time taken to send packet from sender to receiver $= T_t + T_p$

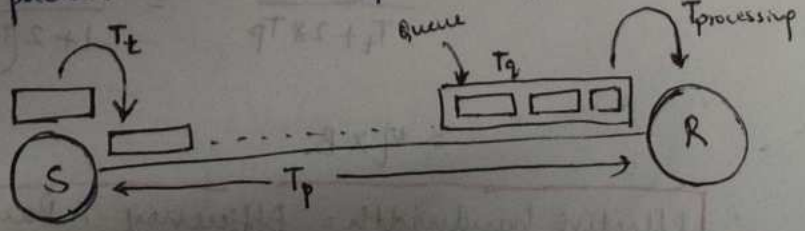
3) Queuing (T_q) delay –

When a packet reaches the destination, it is not processed immediately. All the packets are ~~processed~~ stored in the queue before being processed.

The amount of time a packet sits in the queue before it gets processed is called queuing delay (T_q)

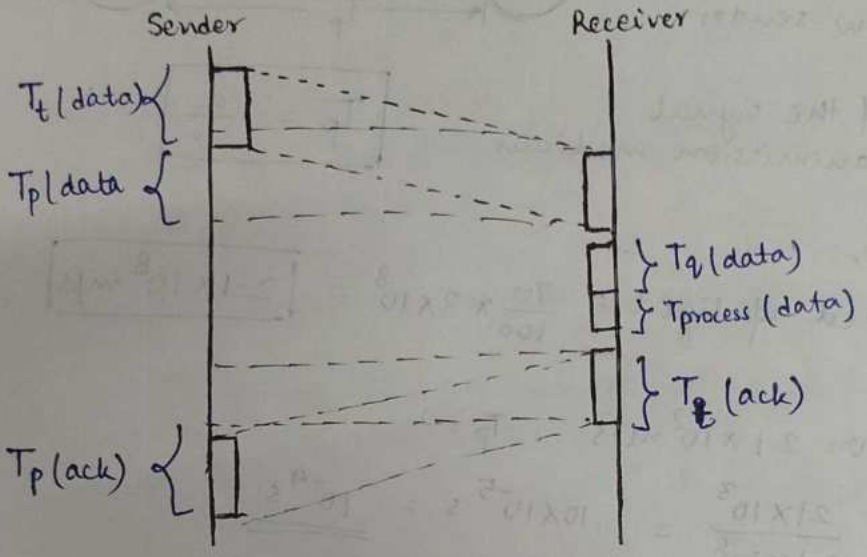
4) Processing delay

The time taken by the receiver to process all the bits of a packet is called processing delay.



Flow control ~~methods~~ stop and wait

Stop and wait is the simplest flow control mechanism in which the sender sends a packet and then stops and waits for the ~~acqua~~ acknowledgement from the receiver before sending the next packet.



Total time –

$T_t(data) + T_p(data) + \cancel{T_q(data)}^{0} + \cancel{T_{process}(data)}^{0} + T_t(ack) + T_p(ack)$

$= T_t(data) + 2T_p + \cancel{T_t(ack)}^{0}$

[bcz propagation time for every packet is same]

$= T_t + 2 * T_p$

$\therefore$ **Total time = $T_t + 2 * T_p$** for 1 packet

Efficiency, $\eta = \frac{\text{useful time}}{\text{Total time}} = \frac{T_t}{T_t + 2 * T_p} = \frac{1}{1 + 2(T_p/T_t)} = \frac{1}{1+2a}$

(where $a = T_p/T_t$)

Throughput = No. of bits that can be sent in 1 second using this protocol.

Throughput / Effective bandwidth / Bandwidth utilization / ~~Link utilization~~

$= \frac{L}{T_t + 2 * T_p}$ (L → size of packet) $= \frac{L \times B \times (1/B)}{T_t + 2 * T_p}$

$= \frac{T_t \times B}{T_t + 2 * T_p} = \frac{B}{1 + 2(T_p/T_t)} = \frac{1}{1+2a} \times B$

$= \eta \times B$

$\therefore$ **Effective bandwidth = Efficiency × Bandwidth**

Q:- Calculate efficiency—

1) T_t = 1msec T_p = 1msec

Efficiency, $\eta = \frac{1}{1+2a}$ where $a = \frac{T_p}{T_t} = \frac{1}{1} = 1$

$= \frac{1}{1+2} = \frac{1}{3} = \underline{\underline{33.33\%}}$

2) T_t = 2msec T_p = 1msec

Efficiency = $\frac{1}{1+2a}$ where $a = \frac{T_p}{T_t} = \frac{1}{2}$.

$= \frac{1}{1+2\times 1/2} = \frac{1}{2} = \underline{\underline{50\%}}$

Q. If η = 50%, what is the relation b/w T_p and T_t?

Given:- η = 50% = 0.5

$\therefore \frac{1}{1+2a} = 0.5 \Rightarrow 1+2a = \frac{1}{0.5} = 2 \Rightarrow 2a = 1 \Rightarrow a = 1/2$

$\therefore \frac{T_p}{T_t} = \frac{1}{2} \Rightarrow \boxed{T_t = 2T_p}$

Q. If BW = 4Mbps, T_p = 1msec L = ? so that η = atleast 50%.

BW = 4Mbps = 4×10^6 bits/sec $T_p = 1\times10^{-3}$ sec.

$\eta \geq 50\% \Rightarrow \eta \geq 0.5 \Rightarrow \frac{1}{1+2a} \geq 0.5$

$\Rightarrow 1+2a \leq 2 \Rightarrow 2a \leq 1 \Rightarrow a \leq 0.5 \Rightarrow \frac{T_p}{T_t} \leq 0.5$

$\Rightarrow 2T_p \leq T_t \Rightarrow \frac{L}{BW} \geq 2\times10^{-3}$ sec

$\Rightarrow L \geq 2\times10^{-3} \times BW$.

$\Rightarrow L \geq 2\times10^{-3} \times 4\times10^6$

$\Rightarrow \boxed{L \geq 8\times10^3 \text{ bits}}$

Factors which affect efficiency —

$$\eta = \frac{1}{1+2a} = \frac{1}{1+2\frac{T_p}{T_t}} = \frac{1}{1+2*\frac{d}{v}*\frac{BW}{L}}$$

Variables ——

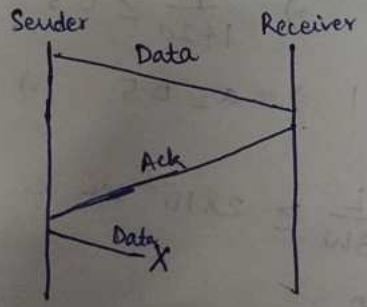
- d — inversely proportional
- v — (fixed)
- BW — (fixed)
- L — ~~data~~

If distance increases, η decreases
∴ Stop and wait is best for LANs

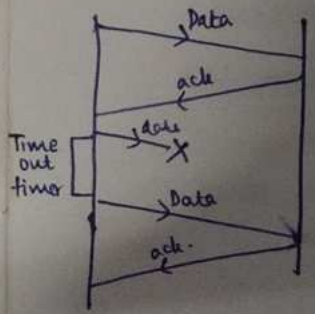
If data size increases, η ~~decreases~~ increases
∴ Stop and wait is good for big packets

Problems and their rectification in stop and wait protocol —

1) Data lost problem



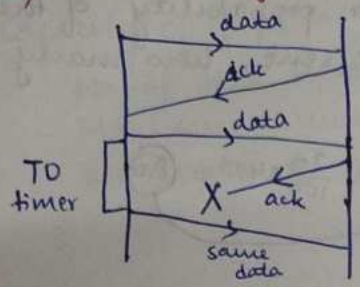
If data packet is lost —
Sender will wait for acknowledgement signal before transmitting next packet.
Receiver will never send acknowledgement signal bcoz it did not receive packet.
∴ DEADLOCK.



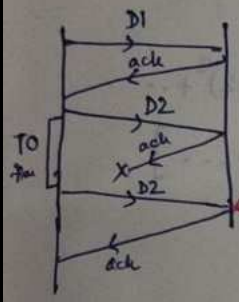
Rectification:- Timeout timer is used.
↳ If acknowledgement signal is not received within this time, same packet is sent again.

STOP and WAIT + TIMEOUT TIMER = Stop and wait ARQ
(Automatic ~~Repeat~~ Repeat Request)

2) Acknowledgement lost problem



- Acc. to sender, maybe data packet was lost due to which it did not get acknowledgement signal.
- It sends same data packet again
- Acc. to receiver, different data packet is received and it will be processed again.

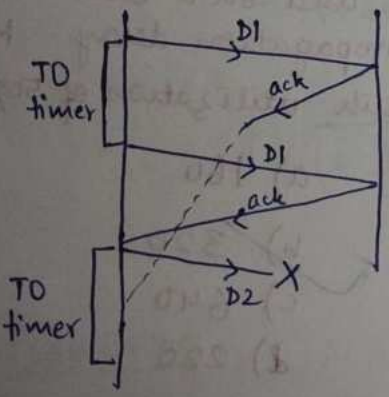


not processed.

Duplicate packet problem.

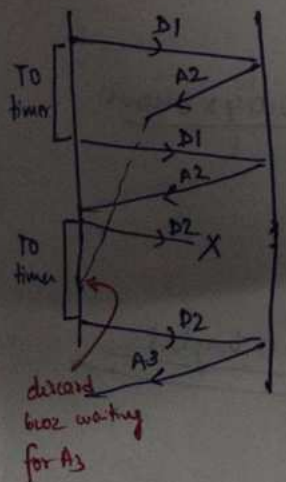
Rectification :- Have sequence number on each data packet / numbering of packets

3) Delayed acknowledgement problem.



- Acknowledgement signal is delayed.
- After timeout, sender again sends the same packet and receives acknowledgement
- It sends 2nd packet but it is lost and the delayed acknowledgement signal is received.

Acc. to sender — 2 packets are sent.
Acc. to receiver — 1 packet is received.



Solution :- Provide sequence number to acknowledgement signal

Q:- There is some problem in the communication channel due to which some of the bits are lost. Let error probability of the channel be 0.2 = 20%. If 400 packets are sent, how many packets are transmitted totally?

No. of packets which get lost out of 400 = $\frac{20}{100} \times 400 = 80$

Again from these 80, 20% will be lost and so on.

$\therefore$ No. of packets transmitted = $400 + 400(0.2) + 400(0.2)^2 + \dots$

$= 400[1 + 0.2 + (0.2)^2 + (0.2)^3 + \dots]$

$= 400 \cdot \frac{1}{1-0.2} = \frac{400}{0.8}$

$= \frac{400 \times 10}{8} =$ **500 packets**

Gate 2015

Suppose that stop and wait protocol is used on a link with a bit rate of 64 kb/sec and 20 msec propagation delay. Minimum frame size in bytes to achieve a link utilization of 50%.

a) 160
b) 320
c) 640
d) 220

Given:- $T_p = 20 \times 10^{-3}$ sec

Bandwidth, BW = 64000 bits/sec

Link utilization = $\eta \geq 50\%$.

$\Rightarrow \frac{1}{1 + 2\frac{T_p}{T_t}} \geq 0.5$

$\Rightarrow \frac{1}{1 + 2 \times 20 \times 10^{-3} \times \frac{BW}{L}} \geq 0.5 \Rightarrow 2 \geq 1 + \frac{0.04 \times 64000}{L}$

$\Rightarrow L \geq 40 \times 64 \Rightarrow L \geq 2560$ bits

$\therefore$ Minimum frame size = 2560 bits = **320 bytes**

CAPACITY OF PIPE AND PIPELINING

Capacity of pipe –

~~No of bits that ca~~

Maximum number of bits that can be present in a pipe at an instant is called capacity of pipe.

Capacity of pipe depends upon "bandwidth" & "propagation delay"

Capacity of pipe = BW * T_p for half duplex

Capacity of pipe = 2 * BW * T_p for full duplex

If capacity is high, it is called thick pipe
If capacity of low, it is called thin pipe.

For stop and wait,

$$\eta = \frac{1}{1+2a} = \frac{1}{1+2*\frac{T_p}{T_t}} = \frac{1}{1+\frac{2*T_p*BW}{L}}$$ (capacity)

$$= \frac{1}{1+\frac{C}{L}} \quad \therefore \eta \propto \frac{1}{C}$$

If capacity is more, efficiency of stop & wait is less

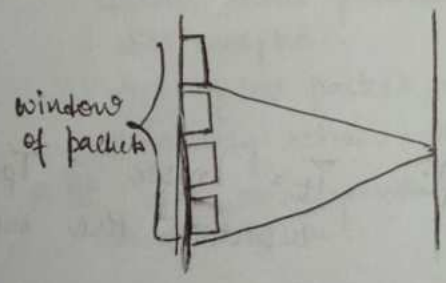
Pipelining

Time taken to transmit 1 packet in stop & wait protocol = $T_t + 2*T_p$

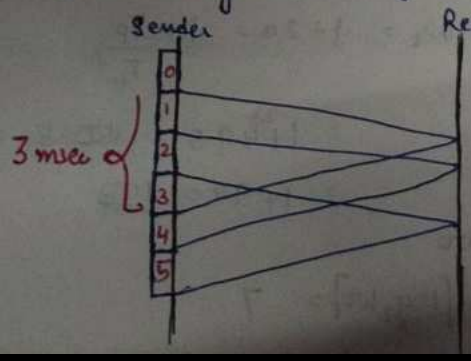
Given:- T_t = 1msec T_p = 1.5msec

$$\eta = \frac{1}{1+2a} = \frac{1}{1+2\frac{T_p}{T_t}} = \frac{1}{1+2*1.5}$$

$$= 1/4 = 25\%$$

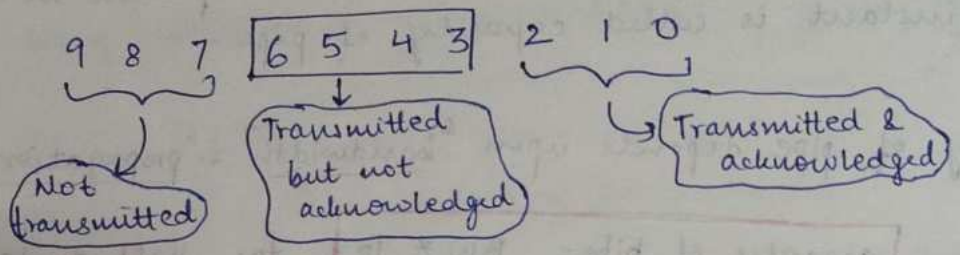


Increasing efficiency of stop & wait –



Round trip time = 2 * T_p = 3 msec.

Buffer/Queue is maintained at the sender side so that if a packet is lost, it can be retransmitted.
When acknowledgement signal for a packet is received, it is removed from the buffer.



Also known as sliding window protocol

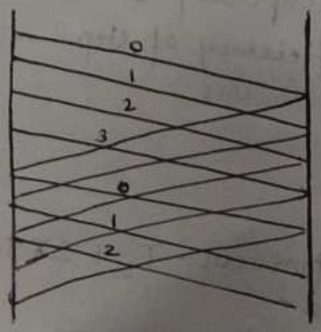
The sender window size in sliding window $W_s = 1+2a$

⊙ The sequence nos of the packets have to be stored in the header field of the packet.

Min. no of sequence nos = $1+2a$
Min. no. of bits in sequence no. fields = n —

$2^n = 1+2a \Rightarrow n \log 2 = \log(1+2a)$

$\Rightarrow \boxed{n = \lceil \log_2 (1+2a) \rceil}$



[Packet no. of packet transmitted & acknowledged can be used again.]

Q:- T_t = 1msec T_p = 49.5 msec
What is the sender window size for maximum efficiency.

For maximum efficiency,

Sender window size, $W_s = 1+2a = 1+\frac{2T_p}{T_t}$

$= 1+\frac{2\times}{1}49.5 =$ [illegible] $=$ [illegible]

$= 1+99 = \underline{100}$

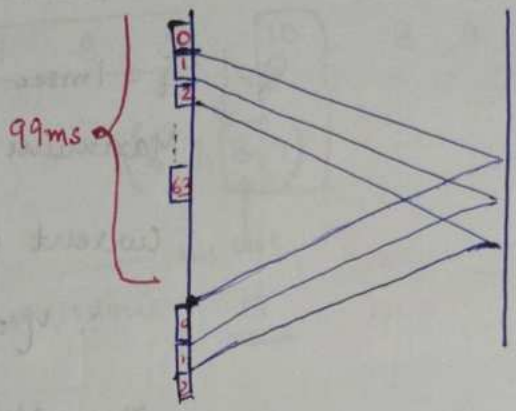
Sequence nos → 0 - 100
Minimum no. of bits = $\lceil \log_2 100 \rceil = 7$

⇒ If in the above question, the number of sequence field bits is given to be 6 then, 64 sequence numbers are possible ranging from (0-63)

$\therefore \eta = \frac{64}{100} = 0.64$

∴ Window size in stop & wait is limited by the number of bits in the sequence number field.

$W_S = \text{Math.min}(1+2a, 2^n)$



⊙ Sliding window protocol is a theoretical concept.

It is practically implemented by —

Sliding window protocol

Go Back N (GBN)

- ⊙ Sender window size is N (should always be greater than 1)
- ⊙ Receiver window size is always 1
  [Receiver will never accept out of order packets. Whole window is re-transmitted in case of a packet loss]
- ⊙ It uses cumulative acknowledgement.

No. of retransmissions is high

Selective Repeat (SR)

- ⊙ Sender window size, $W_S > 1$
  [For stop & wait, $W_S = 1$]
- ⊙ Receiver window size, $W_R = W_S$
  [Even if one of the packets is lost, other packets are still accepted. Only 1 lost packet is selectively retransmitted]
- ⊙ It uses independent acknowledgement.

No. of retransmissions is low

GO BACK N

⊙ Sete Sender size in Go Back N is N

Q:- T_t = 1msec T_p = 49.5msec ~~GBN~~ GO BACK 10 η = ? BW = 40 mbps

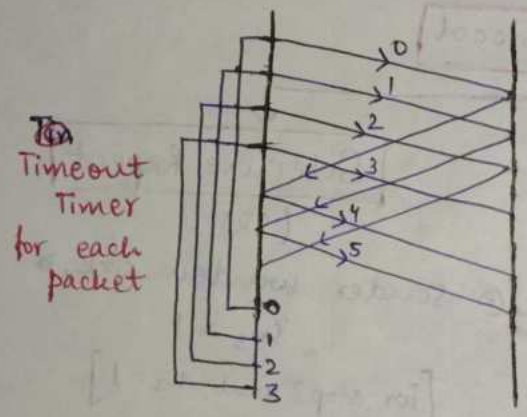
Maximum window size = $1+2a = 1+2\frac{T_p}{T_t} = 1+99 = 100$

Current window size = 10

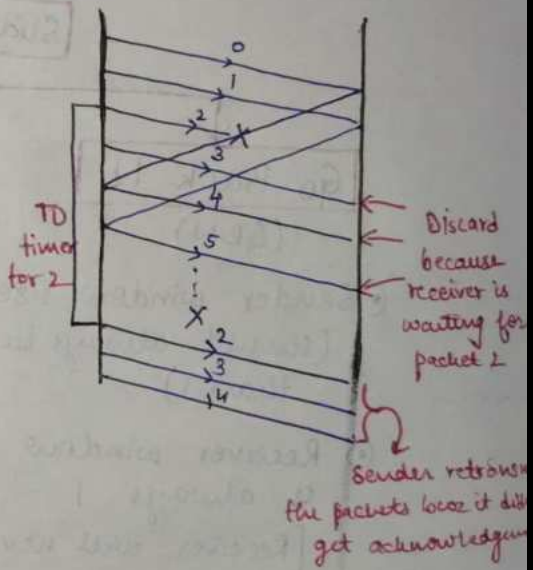
$\therefore \eta = \frac{10}{100} = 10\%$

Throughput = $\eta \times BW = \frac{10}{100} \times 40$ mbps = 4 mbps

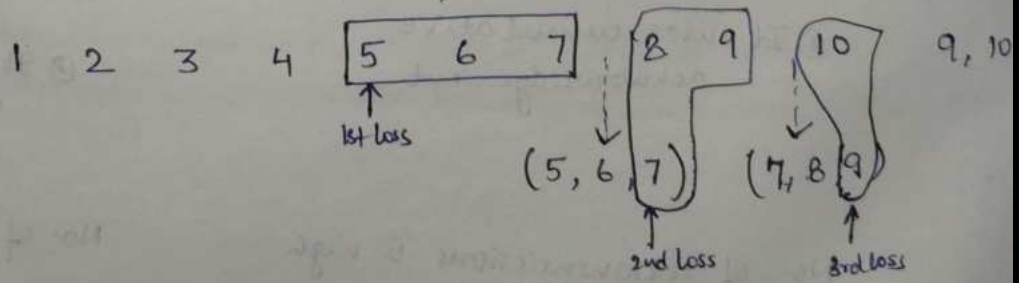
⊙ Receiver window size is 1



Suppose data packet 2 is lost.

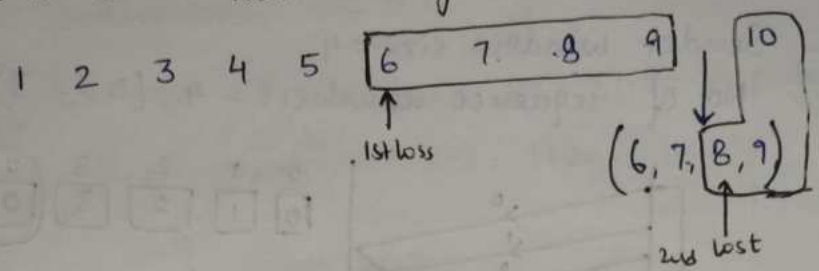


Q:- In GoBack3, if every 5th packet that is being transmitted is lost and 10 packets are to be sent, how many transmissions are required?



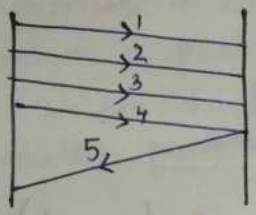
$\therefore$ Total no. of transmissions = 18

Q:- In GoBack-4, every 6th packet is lost. 10 packets are to be sent. How many transmissions?



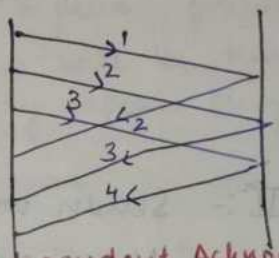
∴ Total number of transmissions = 17

⊙ Acknowledgements:- There are 2 kinds of acknowledgements



Cumulative Acknowledgement

Adv:- Less traffic
Disadv:- Less reliable



Independent Acknowledgement

Adv: High reliability.
Disadv: More traffic

GoBack-N uses cumulative acknowledgements.

At the receiver side, it starts an acknowledgement timer whenever receiver receives any packet. When it expires, it sends cumulative acknowledgement for the number of packets received in that interval.

If 'N' packets are received, N+1 acknowledgement no will be sent

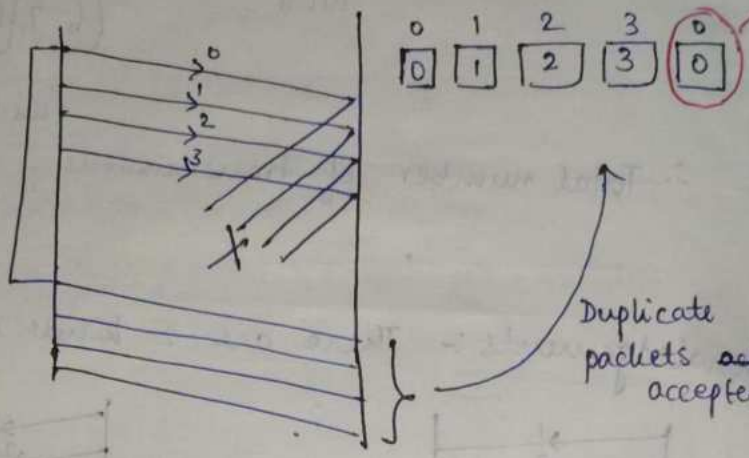
!! Imp:- Acknowledgement timer will not start until the receiver has received a packet

!! Timeout timer at sender side should be greater than acknowledgement timer at receiver side.

Timeout timer > Acknowledgement timer

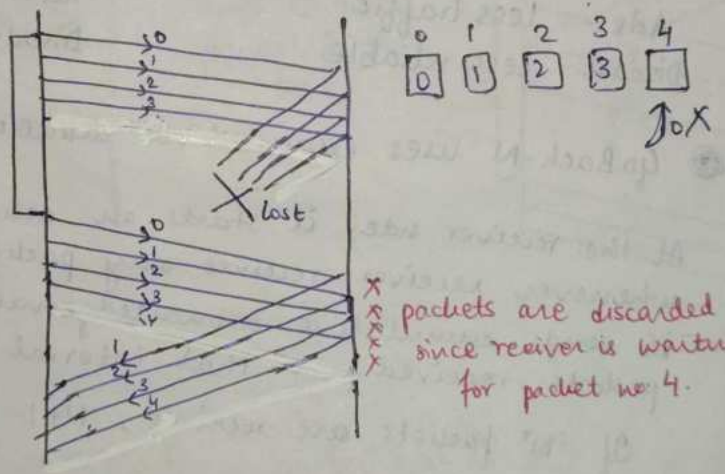
Relationship b/w window sizes and sequence numbers

Case I :- Sender window size = 4
No. of sequence numbers = 4 (0 ... 3)



Does not work!!

Case II :- Sender window size = 4
No. of sequence numbers = 5 (0 ... 4)



Works !

⇒ If sender window size = 'N' & receiver window size = '1', then to detect duplicate packets, 'N+1' sequence nos should be used.

For any sliding window protocol, if it has to work without any problem, the condition is $W_S + W_R \leq ASN$ (Available Sequence Numbers)

SELECTIVE REPEAT

⊙ Sender window size, $W_S > 1$

Q:- $T_t = 1ms$ $T_p = 49.5ms$ $W_S = 50$ SR protocol $\eta = ?$

Maximum sender window size $= 1+2a = 1+2\frac{T_p}{T_t} = 1+99 = 100$

$\therefore \eta = \frac{50}{100} = 50\%$.

⊙ Receiver window size is equal to sender window size

$$\boxed{W_R = W_S}$$

This protocol is called 'selectively repeat' protocol because whenever a packet is lost, there will be timeout at the sender side & sender selectively repeats only that particular packet.

Q:- $W_S = 3$ 10 packets are sent. Every 5th packet is lost. How many transmissions in SR protocol?

1 2 3 4 5 6 7 [5] 8 9 10 [9]

5 ↑ 1st loss

9 ↑ 2nd loss

Total no. of transmissions = 12

⊙ The Acknowledgements are independent.

!! In case of packet loss, GBN & SR behave in the same way.

!! In case of currupted data packet, GBN doesn't send acknowledgement signal while SR sends ~~NAKO~~ NACK (Negative acknowledgement)

The maximum window size should be half the max. seq. number.

Comparison of various flow control protocols

	Stop and wait (SAW)	Go Back N (GBN)	Selective Repeat (SR)
Efficiency	$\frac{1}{1+2a}$	$\frac{N}{1+2a}$	$\frac{N}{1+2a}$ (N → sender window size)
Buffers (sender buffers + receiver buffers)	1 + 1	N + 1	N + N
Sequence Numbers	2	N+1	2N
Retransmissions (if 1 packet is lost)	1	N	1
Bandwidth Required	Low	High (because of high transmissions)	Moderate
CPU requirement [complexity]	Low	Moderate (No sorting at receiver, No searching at sender)	High (out of order packets are also accepted) sorting is then done at receiver, searching of lost packet at sender.
Implementation	Low	Moderate	Complex
Acknowledgements	Independent	Cumulative	Independent

Gate 2016

Consider 128×10^3 b/s satellite communication one way propagation delay of 150 ms. Selective Retransmission protocol is used on the link to send data with frame size of 1kB. Neglect transmission time of acknowledgement. The minimum number of bits required for sequence number fields to achieve 100% utilization is

a) 2
b) 4
c) 6
d) 8

$\eta = \frac{W}{1+2a}$ where W = sender window size

$W \leq 1+2a$

$\Rightarrow W \leq 1 + 2\frac{T_p}{T_t}$

$T_t = \frac{L}{BW} = \frac{1024 \times 8}{128 \times 10^3}$

$= \frac{2^{13-7}}{10^3} = \frac{2^6}{10^3} = 0.064$

$\Rightarrow W \leq 1 + \frac{2 \times 0.15}{0.064}$

$\leq 5.68 \Rightarrow 5$

For SR protocol,

$W_S = W_R$

∴ No. of sequence nos required = $W_S + W_R$

$= 5 + 5 = 10$

No. of bits required = $\lceil \log_2 10 \rceil = 4$

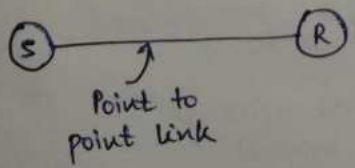
ACCESS CONTROL METHODS

Stations can communicate with one another using 2 types of links —

Types of links

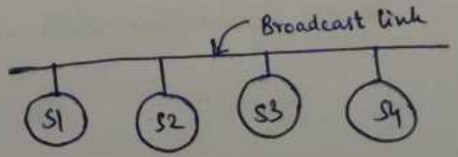
Point to point links

- dedicated link b/w 2 stations
- entire capacity of the link is used for communication b/w 2 stations only.



Broadcast links

- It is the common link to which multiple stations are connected.
- The capacity of the link is shared among the connected stations for communication



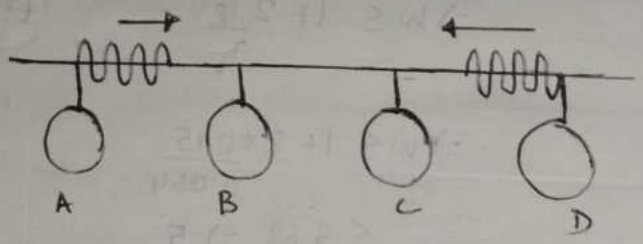
Access Control

Access Control is a mechanism that controls the access of stations to the transmission link.

Need :-

To prevent the ~~collision~~ occurance of collision or if the collision occurs, deal with it.

Collision of data packets results in data curroption.



A & D start transmitting simultaneously.

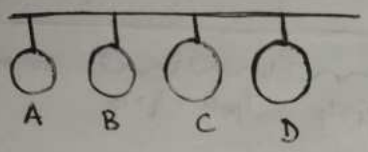
- Access control methods are the methods used for providing the access ~~code~~ control.
- ensure smooth flow of traffic on the network.
- implemented at the Data ~~bist~~ link layer of the OSI model.

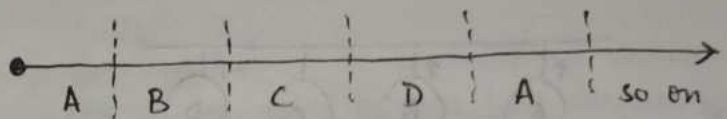
Access control methods

- → Time Division Multiplexing
- → Polling
- → CSMA/CD
- → Token Passing
- → Aloha

1) Time Division Multiplexing

- The link time is divided into slots of equal size and each slot is assigned/allocated to the stations in Round Robin manner.
- Each station transmits its data during the time slot allocated to it.
- If station does not have any data to transmit, the time slot goes waste.





Size of time slot

Size of each time slot is kept such that each station gets sufficient time to

↳ ~~trans~~ put the packet on the transmission link (T_t)

↳ last bit of the packet is able to get out of the link (T_p)

$\therefore$ **Size of time slot = $T_p + T_t$**

$$\text{Efficiency, } \eta = \frac{\text{useful time}}{\text{cycle time}} = \frac{T_t}{T_t + T_p} = \frac{1}{1+a}$$

Effective Bandwidth / Bandwidth utilization / Throughput = $\eta \times BW$

Disadvantage

If any station does not have data to transmit, then, the time slot is wasted. This leads to decrease in efficiency.

Q:- If $T_p = 1ms$, $T_t = 1ms$, $BW = 4Mbps$. find effective bandwidth, efficiency

$$\eta = \frac{1}{1+a} = \frac{1}{1+T_p/T_t} = \frac{1}{1+1} = 50\%.$$

Effective Bandwidth = $\eta \times (BW) = 0.5 \times 4Mbps = 2Mbps$

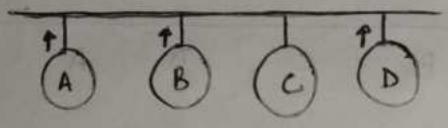
If each station requires 2kbps BW, find maximum no. of stations

$N \times 2kbps$ = Max available BW

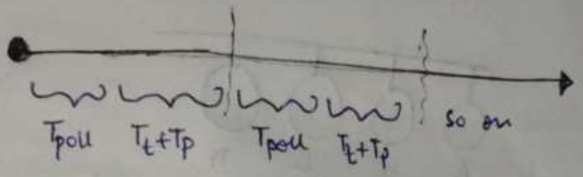
$\Rightarrow N \times 2kbps = 2mbps \Rightarrow N = 1000$

2) Polling

- One station is selected among a group of stations willing to transmit data by using a polling algorithm.
- The selected station then sends the data to the destination.
- Then, the cycle repeats.



arrow indicates willing



$$\text{Efficiency} = \frac{\text{useful time}}{\text{cycle time}} = \frac{T_t}{T_{poll}+T_t+T_p}$$

Advantages

- no time slot is wasted (unlike TDM)
- maximum efficiency & bandwidth utilization

Disadvantages

- Time is wasted during polling
- Each station has equal probability of being selected in each round. ∴ some stations may starve.

$$\text{Maximum Available Effective Bandwidth} = \text{Total no. of stations} * \text{Bandwidth Requirement of 1 station}$$

Throughput

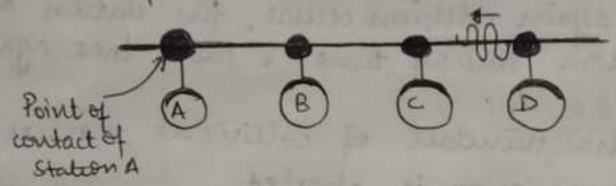
3) CSMA/CD

used in Ethernet

Carrier Sense Multiple Access / Collision Detection

Step 1:- Sensing the carrier

- Any station willing to transmit the data senses the carrier (checks if carrier is free or not).
- If it finds the carrier free, it starts transmitting the packet otherwise not.



If A senses the carrier, it will find carrier free while it is not bcoz D is transmitting data.

Step 2:- Detecting the collision

- Detection of collision is done by the transmitting station.
- For detecting collision, CSMA/CD implements the following condition –

$$T_t >= 2 * T_p \qquad L \geq 2 * T_p * BW$$

→ Minimum size of the packet.

If size of packet is less than the minimum size, extra bits are added to it.

2 cases are possible

- No collided signal comes back during transmission → indicates data packet is transmitted successfully
- Collided signal comes back during transmission → indicates collision occured

- Collision Detection is not possible if $L < 2 * Tp \times BW$

Step 3:- Releasing JAM signal

- JAM is a 48 bit signal
- released by transmitting station as soon as it detects collision
- It alerts other stations not to transmit their data immediately after collision (otherwise there is a possibility of collision with same data packet again)
- Frequency of JAM signal is different from that of data signals. This ensures that JAM signal does not collide with data the data signals undergone collision.

Step4: Waiting for Back Off Time

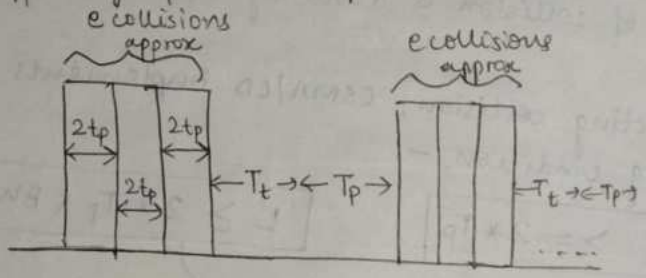
After the collision occurs, the transmitting station waits for a random amount of time called back-off time (determined by backoff algorithm).

After backoff time, it tries transmitting the data packet again.

If again collision occurs, the station again waits for some random backoff time & then tries again

If the number of collisions reaches its limit, the transmission is aborted.

Efficiency of CSMA/CD / Ethernet



$$\eta = \frac{\text{Useful time}}{\text{Cycle time}}$$

$$= \frac{T_t}{e \cdot 2t_p + T_t + T_p}$$

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

Maximum amount of data that can be sent through ethernet = 1500 bytes

---

Back Off Algorithm

n = No. of collisions of a data packet.

Transmitting stations choose a no. from $(0 \text{ to } 2^n - 1)$

Waiting time = $T_{slot} \times (K)$

→ Applicable for only 2 stations

→ If A won after 1st collision its probability for 2nd collision is higher.

→ Collision probability decreases exponentially

If distance decreases, η increases

∴ suitable for LANs
not suitable for WANs

If size of data packet increases, η increases

∴ suitable for bigger data packets.

Gate 2015

Consider a CSMA/CD network that transmits data at a rate of 100Mbps (10^8 b/s) over 1km cable with no repeaters. If the minimum frame size required for this network is 1250 bytes, what is the signal speed in the cable in km/sec

A) 8000
B) 10000
C) 16000
D) 20000

BW = 100Mbps = 10^8 bps
Length of cable = 1km
Minimum size of data packet = 1250 × 8 bits

$T_t = \frac{1250 \times 8}{10^8}$ s $= \frac{10000}{10^8} = 10^{-4}$ s

$T_p = \frac{d}{v} = \frac{1}{v}$

For CSMA/CD network,

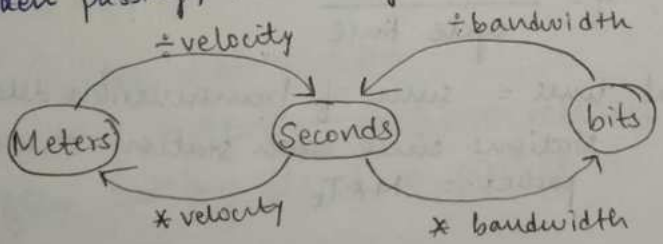
$T_t = 2T_p$ [∵ packet size is minimum]

$\Rightarrow 10^{-4} = 2 \times \frac{1}{v} \Rightarrow v = 2 \times 10^4 = 20000$ km/s

4) Token passing

Ring topology

In token passing, time may be expressed as meters/seconds/bits.



Terminology

1. Token :- ⊙small message composed of special bit pattern.
⊙represents the permission to send a data packet.
⊙A station is allowed to send data packet if & only if it possesses the token otherwise not.

2. Ring latency :-
Time taken by a bit to complete one revolution of the ring is called ring latency.

$$\text{Ring latency} = \frac{d}{v} + N * b$$

Length of the ring = d
Speed of the bit = v
No. of stations = N
Bit delay at each station = b

$\frac{d}{v}$ → Time taken by the bit to traverse the ring (T_p)

$N * b$ → Time taken by the stations to hold the bit.

3) Token Holding time

Time for which a station holds the token before transmitting to the other side.

4) Cycle time

The time taken by the token to complete one revolution of the ring is called ~~token time~~ cycle time.

$$\text{Cycle time} = \frac{d}{v} + N * THT = T_p + N \times THT$$

Efficiency –

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

Useful time = sum of transmission delay of N stations since each station sends 1 data packet. = $N * T_t$

$$\therefore \eta = \frac{N \times T_t}{T_p + N \times THT}$$

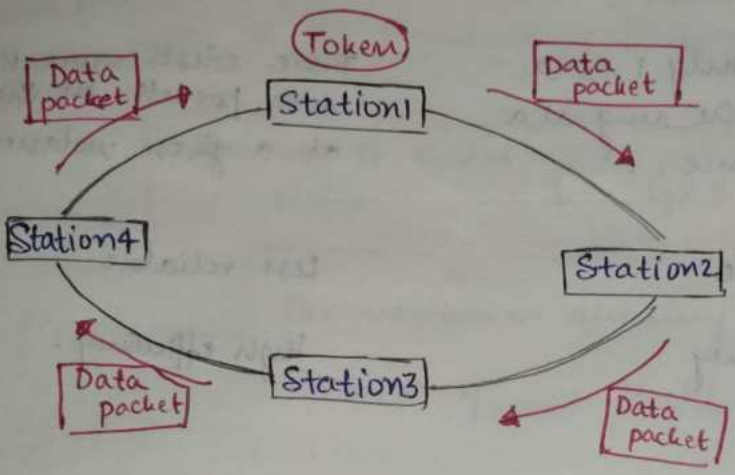
!! * 'Token Holding Time' depends upon the strategy implemented —

Token Passing strategies

- Delayed Token Reinsertion
- Early Token Reinsertion

Delayed Token Reinsertion

In this strategy, the station keeps holding the token ~~but~~ until the last bit of data packet transmitted by it takes the complete revolution of the ring & comes back to it



Token Holding Time (THT) = Transmission delay + Ring Latency.

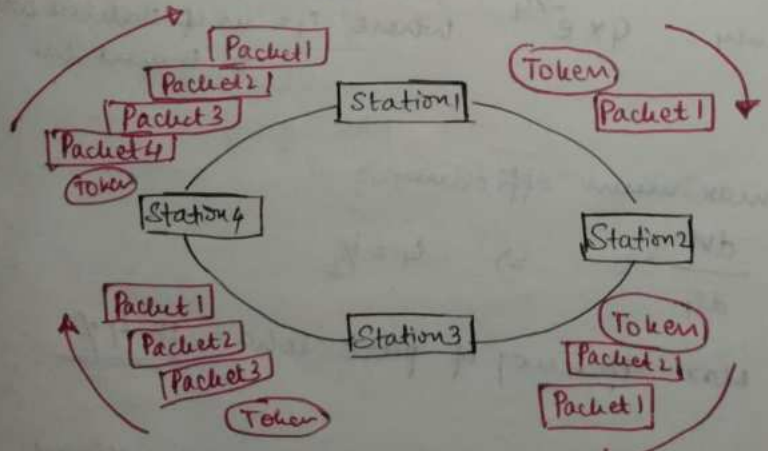
Assumption
- bit delay is 0 in most cases

$$\therefore \eta = \frac{N \times T_t}{T_p + N \times THT} = \frac{N \times T_t}{T_p + N(T_t + T_p)}$$

$$= \frac{1}{\frac{a}{N} + 1 + a} = \frac{1}{1 + \left(1 + \frac{1}{N}\right)a} = \frac{1}{1 + \left(\frac{N+1}{N}\right)a}$$

Early Token Reinsertion

In this strategy, station releases the token immediately after putting its data packet to be transmitted on the ring.



Token Holding Time (THT) = Transmission delay (T_t)

$$\therefore \eta = \frac{N \times T_t}{T_p + N \times T_t} = \frac{1}{1 + \frac{a}{N}}$$

Delayed Token Reinsertion (DTR)	v/s	Early Token Reinsertion (ETR)
Each station holds the ~~packet~~ token until its data packet reaches back to it.		Each station releases the token immediately after putting its data packet on the ring.
There exists only 1 data packet on the ring at a given instance.		There exists more than one data packets on the ring at a given instance.
More reliable		Less reliable.
Low efficiency		High efficiency.

5. ALOHA

→ Pure Aloha

- Stations can transmit data at any time whenever they want.
- After transmitting packet, station waits for some time.

If acknowledgement signal is received, then, transmission is successful.

If not received, then station waits for random time 'back off time' & then transmits again.

After backoff limit is reached, it aborts the transmission.

Efficiency = $G \times e^{-2G}$ where G = no of stations willing to transmit data

For maximum efficiency,

$$\frac{d\eta}{dG} = 0 \Rightarrow G = 1/2$$

∴ Max efficiency of pure aloha = 18.4%.

η_{max} is very less bcoz of large no of collisions.

→ Slotted Aloha

Time is divided into slots.
Every station can transmit only at the beginning of time slot.

[Collision only occurs when 2 stations try to transmit data at the beginning of same time slot]

Efficiency of slotted aloha, $\eta = G \times e^{-G}$

G = No. of stations willing to transmit.

For maximum efficiency, $G = 1$

$\therefore \eta_{max} = 36.8\%$

Pure Aloha	v/s	Slotted Aloha
Any stations can transmit at any time.		Any station can transmit at the beginning of time slot.
The time is continuous & not globally synchronized		The time is discrete and globally synchronized.
Vulnerable time = $2 \times T_t$		Vulnerable time = T_t
$\eta = G \times e^{-2G}$		$\eta = G \times e^{-G}$
$\eta_{max} = 18.4\%$ ($G = 1/2$)		$\eta_{max} = 36.8\%$ ($G = 1$)
Main adv — simplicity in implementation		Main adv — reduced collisions double efficiency.

Summary

Flow control methods :-

Make sure that the receiver is not overburdened with the packets.

- Stop and wait protocol — $\eta = \frac{1}{1+2a}$
- Sliding window protocol
  - Go Back N — $\frac{N}{1+2a}$
  - Selective Repeat — $\frac{N}{1+2a}$

Access control methods

Methods that control the access of stations to the communication link.

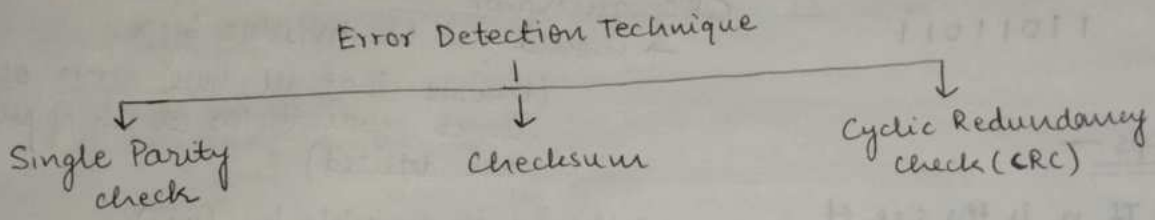
- Time Division Multiplexing — $\eta = \frac{1}{1+a}$
- Polling — $\eta = \frac{T_t}{T_{poll} + T_t + T_p}$
- CSMA/CD — $\eta = \frac{1}{1+6.44a}$
- Token Passing
  - Delayed Token Reinsertion — $\eta = \frac{1}{1+(N+1)a/N}$
  - Early Token Reinsertion — $\eta = \frac{1}{1+a/N}$
- Aloha
  - Pure Aloha — $\eta = 4 * e^{-2g}$, $\eta_{max} = 18.4\%$
  - Slotted Aloha — $\eta = 4 * e^{-4}$, $\eta_{max} = 36.8\%$

ERROR CONTROL METHODS

⊙ Process of detecting and correcting data frames that have been corrupted or lost during transmission.

→ Lost packets are handled by Flow control methods.

⊙ Error detection is the technique used to check if some error occured in the data during transmission.



Single Parity Check

In this technique, one extra bit called parity bit is sent along with original bits

Steps — [Even parity]

→ At sender side, total no. of 1's in data is counted. Parity bit is set to 0/1 to make the total no. of 1's even

→ Data + Parity bit is transmitted.

→ Receiver receives the packet & counts no. of 1's. If even, data is correct else it requests for retransmission.

Example —

① Data to be transferred = 1001001
even parity is used.

Parity bit = 1

Code word received by receiver = 10010011

Receiver counts no. of 1's & determines if it is even or not

② Data transferred = 1100100
odd parity is used.

Parity bit = 0

Code word generated = 1100100

If no curroption takes place, receiver receives 1100100.

Advantages

⊙ guarantees detection of odd number of bit errors.

Disadvantages

⊙ cannot detect even no. of bit errors.

Cyclic Redundancy Check

based on binary division.

CRC Generator :- algebraic polynomial represented as bit pattern. Bit pattern is obtained by the following rule—

Example—

CRC Generator is
$x^7+x^6+x^4+x^3+x+1$

→ Power gives the position of the bit.
→ Coefficient gives value of bit.

Bit Pattern is
11011011

CRC generator
→ should not be divisible by x
(ensures that all burst errors of length equal to the length of polynomial are detected)

→ should be divisible by (x+1)
(ensures that all the burst errors of odd no. of bits are detected)

Steps —

① If n is the size of CRC generator, n-1 bits are appended to the end of data.

② Divided by CRC generator.

③ Remainder obtained is called CRC.
CRC consists of (n-1) bits.

④ CRC is appended to the original data to be transmitted.

⑤ At Receiver side, code is again divided by CRC generator.
On division if remainder is 0, then, data is not currupted else it is currupted.

Example

Data to be transfered = 110101101
CRC generator → x^4+x+1
= 10011

```
10011 | 110101101
        10011
        ------
        01000111011
          10011
          -----
          00000101 1
```

∴ CRC = 1011

कुछ गलत है। देख लेना

Data transmitted → 110101101111011

At receiver's end—

```
10011 | 110101101111011
        10011
        ------
        01001110111011
          10011
          -----
          00000101111011
              10011
              -----
              00100D11
                10011
                -----
                000
```

Checksum

Sender's side:
- If m bit checksum is used, data unit is divided into segments of m bit.
- All the segments are added.
- 1's complement on the sum.
- The value obtained is called checksum

Receiver's side:
- Received data unit is divided into m bit segments.
- All the segments are added along with checksum value.
- Value obtained is complemented.

If result = 0, no corruption
result ≠ 0 corrupted data.

Example –

Data unit to be transmitted = 10011001 11100010 00100100 10000100

8 bit checksum is used

10011001 + 11100010 + 00100100 + 10000100

= 100100011 = 00100101 (8 bits)
(wrap up)

1's complement = 11011010

At receiver's end → sum of all segments + checksum value

= 00100101 + 11011010 = 11111111

1's complement = 00000000

∴ data is not corrupted

ISO OSI STACK

Functions needed to be implemented for communication

Mandatory functions

- Error control
- Flow control (receiver should not be overflooded)
- Access control
- Multiplexing / Demultiplexing
- Addressing

Optional functions

- Encryption / Decryption
- Checkpointing
- Routing

70 functions both optional & mandatory

To implement the above functionalities, various reference models are used —

- ISO-OSI
- TCP/IP
- ATM
- X.25
- IEEE (deals with LANs)

(in syllabus)

} Reference models

ISO-OSI model

International Standard Organization - Open System Interconnection

Layers in the OSI model —

User interactiveness
1) Application layer
2) Presentation layer
3) Session layer

Thick layer ← 4) Transport layer

Complex ← 5) Network layer

Both h/w & s/w ← 6) Data link layer

H/w ← 7) Physical layer.

Advantages of layering

- Divide and conquer
- Encapsulation
- Abstraction
- Easy testing.

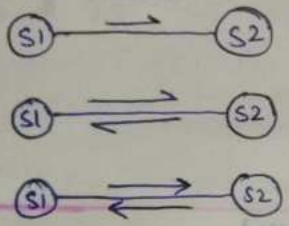
Newspaper Holder

1) Physical layer

EMFP

⊙ Physical layer deals with Electrical, Mechanical, Functional & Procedural characteristics

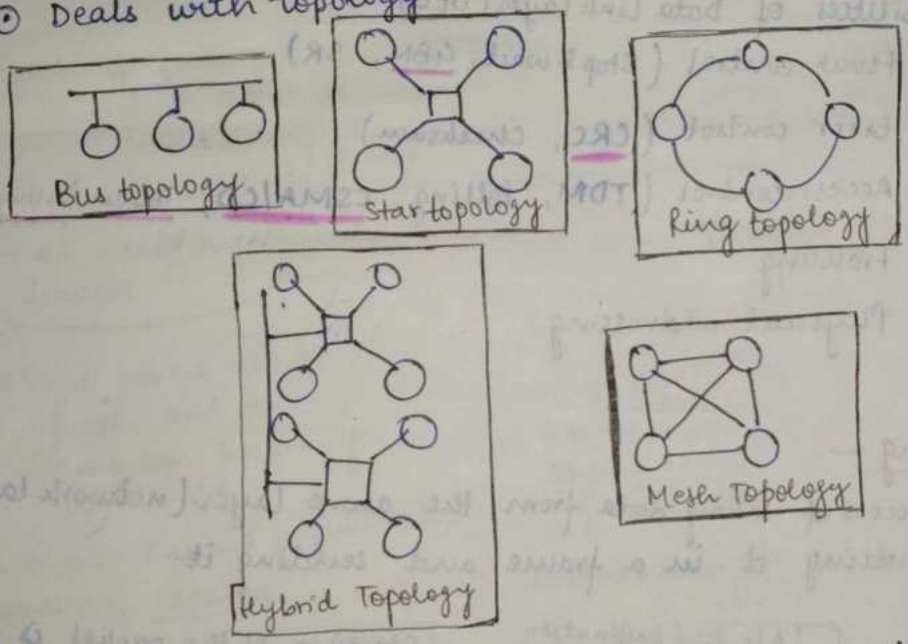
⊙ Transmission modes:



Walkie Talkie

2) DATA LINK LAYER

⊙ Deals with topology —



⊙ Deals with encoding (bits → signals/waves)

- Manchester encoding
  - 0 Represented by ⊓
  - 1 Represented by ⊔
- Differential Manchester Encoding
  - 0 Represented by ⊓/⊔
  - 1 Represented by ⊓/⊔

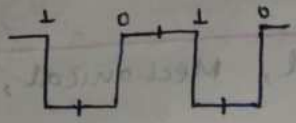
In both the encodings,

Baud Rate = 2 * Bit Rate

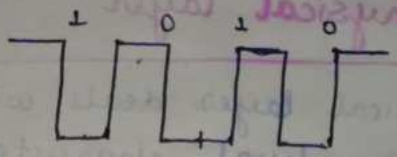
- Baud Rate: No. of voltages being sent per second
- Bit Rate: No. of bits sent per second

Example of encoding –

10 10



Manchester encoding



Differential Manchester encoding

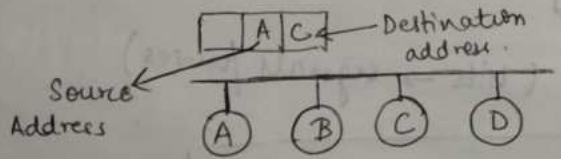
2) DATA LINK LAYER (limited, to be used within a network)

Responsibilities of Data Link Layer (DLL)

- Flow control (Stop & wait, GBN, SR)
- Error control (CRC, Checksum)
- Access control (TDM, Polling, CSMA/CD, Token Passing, Aloha)
- Framing
- Physical addressing.

Framing –

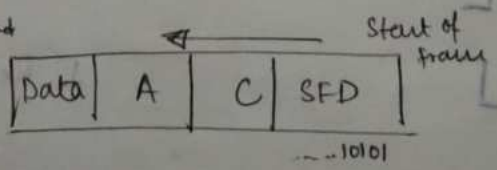
- process of taking data from the above layer (network layer), putting it in a frame and sending it.



Starting of the packet is seen by B & it will discard it since DA is C.

Every ~~frame~~ station should ~~no~~ know beginning of the frame.
∴ There is a special field in the beginning called

* **Starting Frame Delimiter (SFD)**



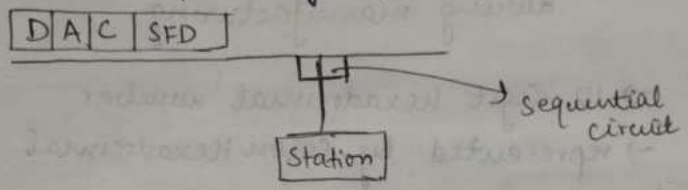
- SFD represents that a frame is coming and alerts all the stations.
- SFD → sequence of bits that is entirely different from data.
- Bits in SFD —

(10101010 .... 11) (0+1)*

SFD — Data

Regular expression → NFA → DFA → Sequential circuit (using flip flops)

Sequential circuit has the capability of recognising the bit pattern of SFD.



End of frame — [redacted]

Frames

Fixed size/length frames

↳ Since frame size is fixed, end of frame can be easily determined.

Problem:- Internal fragmentation

Solution:- Padding

↳ Adding Dummy bits.

Variable size/length frame.

Ethernet (CSMA/CD)

Having a field saying length of the frame

Problem:- Corruption in length field.

Token passing

Having an Ending Delimeter

Problem:- ED can match with the data available

ED Data A C SFD

Solution

costly & obsolete

Character stuffing

used when only characters are used in the data.

If data contains ED, a byte is stuffed into data to differentiate it.

ED:- $ escaped using 10$

If data contains 10$, 10|010$ is used.

Imp for gate

Bit stuffing

Sender stuffs a bit to break the pattern

Receiver receives frame.

If data contains 01110, it removes the 0 & reads the data

Data = 011100011110 ED = 0111

Data after bit stuffing —

0110100011011100

NIC - Network Interface Card

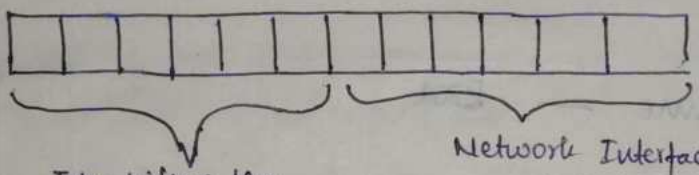
Physical Addressing

MAC (Media Access Control) is the physical address that works at the data link layer.

MAC Address = 48 bit no.
embedded into Network Interface Card (NIC) during manufacturing.

→ 12 digit hexadecimal number
→ represented by Colon Hexadecimal notation.

CC:46:D6 → Cisco
3C:5A:B4 → Google
3C:D9:2B - Hewlett Packard



Identifies the manufacturer. Called Organizational Unique Identifier (OUI)

Network Interface controller assigned by manufacturer (Data & Sl. No.)

Command to see MAC address → ipconfig/all

LAN Technologies like Token Ring, & Ethernet use MAC address as physical addres.

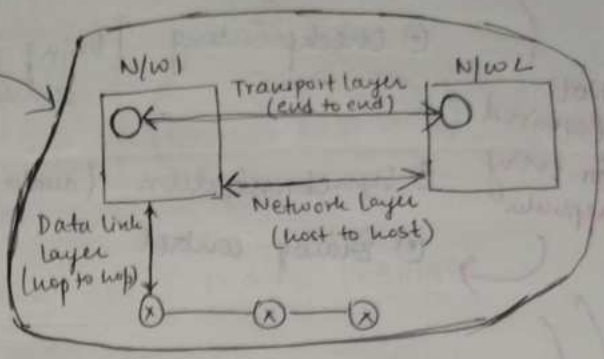
AppleTalk does not use MAC Address as physical addres.

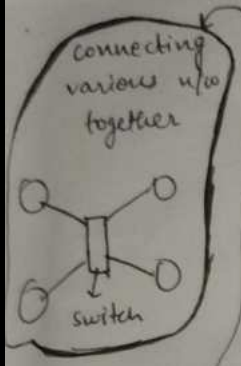
Data Link Layer
- → Logical Link control
  Error control, Flow control
- → Media Access Control
  handles Framing, physical addressing, accesscontrol, error control

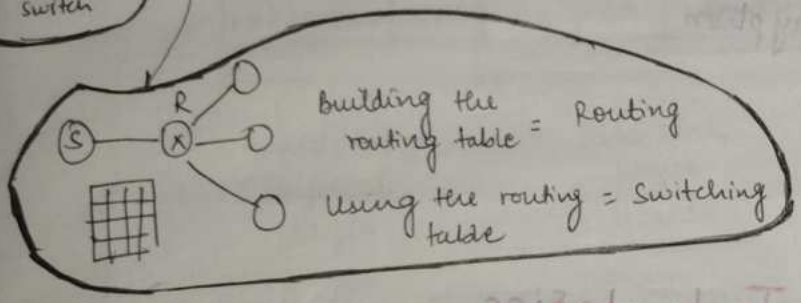
3) NETWORK LAYER

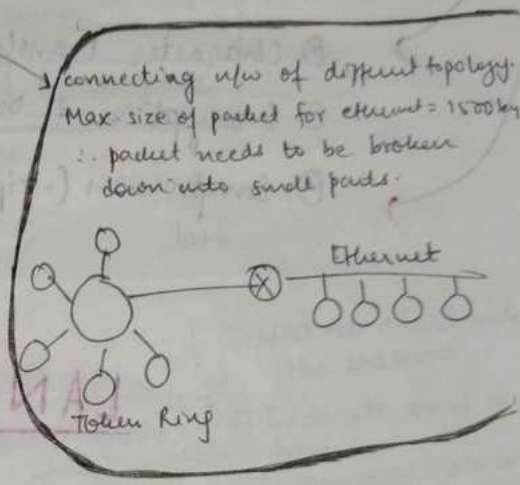
Main responsibilities —

- Host to Host Connectivity
- Logical Addressing (at IP)
- Switching
- Fragmentation
- Routing
- Congestion Control (at TCP)









4) TRANSPORT LAYER

तेनु मैं Love करदा, बेमतलब करया
बाहों में आजा सोनिए

Main responsibilities are

- End to end communication using port numbers
  (Port numbers = Service Point Addressing)
- Flow control (SR)
  SR → software layer (TL)
  GBN → hardware layer (DLL)
- Error control (checksum)
- Segmentation (Takes data from Application, Presentation, Session Layer) and divides into segments.
- Multiplexing & Demultiplexing
- Congestion Control

SESSION LAYER

- Authorization and authentication
- Checkpointing [losing interconnection, Torrent. movie downloads from the same point (not beginning) if internet is available]

Not Required for every application

- Synchronization (audio & video)
- Dialog control

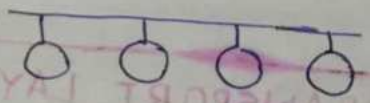
PRESENTATION LAYER

- Character translation
- Encryption & decryption
- Compression (.zip)

LAN Technologies

ETHERNET (IEEE 802.3)

- Topology :- Bus Topology

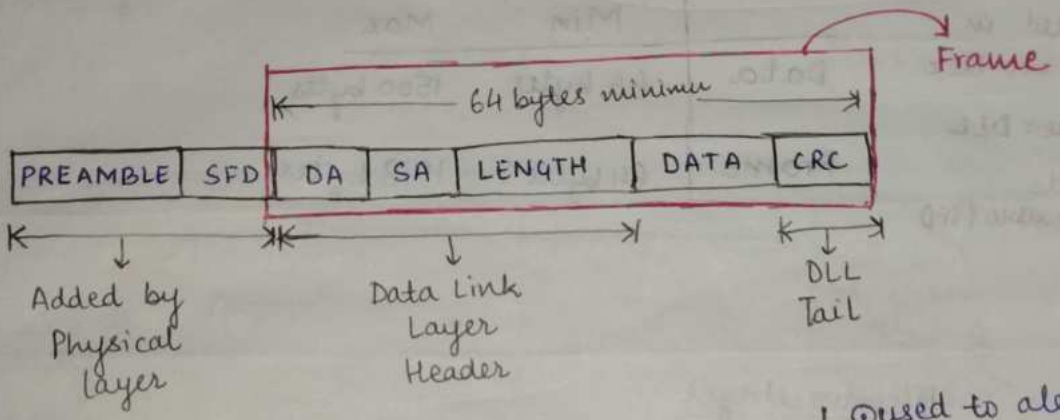


- Access Control Method :- CSMA/CD
- Encoding :- Manchester encoding
  - 0 represented by ⌊
  - 1 represented by ⌈
- No acknowledgement signal is sent.
- Data Rates (Bandwidth) 10 mbps — 100 mbps — 1 Gbps
  - 100 mbps → fast ethernet
  - 1 Gbps → Gigabit ethernet
- Ethernet operates at DLL
  (LAN Technologies are dealt in DLL)

Application layer (Message) → Transport layer (segment) → Network layer (Datagram) → Data Link layer (Frame) → Physical layer

IPDU (Single Protocol Data Unit)

FRAME FORMAT OF ETHERNET (IEEE 802.3 FRAME FORMAT)



→ PREAMBLE → 7 bytes 10101010 10
→ SFD : [Start Frame Delimiter] : 1 byte = 10101011
- ⊙ used to alert all the stations.
- ⊙ Indicate start of frame, synchronization

→ DA : Destination address
→ SA : Source address
} MAC addresses

→ CRC : Cyclic Redundancy check : 4 bytes (32 bit CRC generator is used)

Types of MAC addresses

UNICAST
LSB of 1st bytes is 0
1A:2B:3C:48:56:6F
SA is always unicast
Unicast DA means packet is sent to only 1 host.

MULTICAST
LSB of first byte is 1
11:2B:3C:48:56:6F
Multicast DA means that packet is sent to a group of hosts.
MAC address of group = Multicast MAC

BROADCAST
All the bits are 1's in MAC address.
FF:FF:FF:FF:FF:FF
The packet is sent to all the hosts in the network.

→ LENGTH : 2 bytes

- Ethernet follows variable length frames.
- In CSMA/CD,

$$L \geq 2 * BW * T_p$$

Substituting the standard values of ethernet,

$L \geq 64$ bytes (frame size)

- Max length of data = 1500 bytes

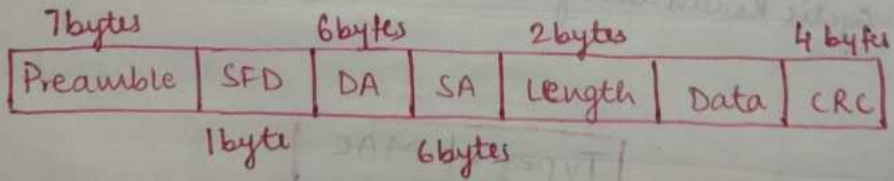
* Preamble & SFD are not included in frame size bcoz frame = DLL

DL adds preamble & SFD

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

Disadvantages

- Not applicable for real time applications
- Not applicable for interactive applications
- No priorities. So not suitable for client server applications



SWITCHING

Switching
- → Circuit switching (applied at PL, obsolete technology)
- → Packet switching
  - → Virtual circuits switching
  - → Datagram switching
- → Message switching

→ Switching is done at network layer.

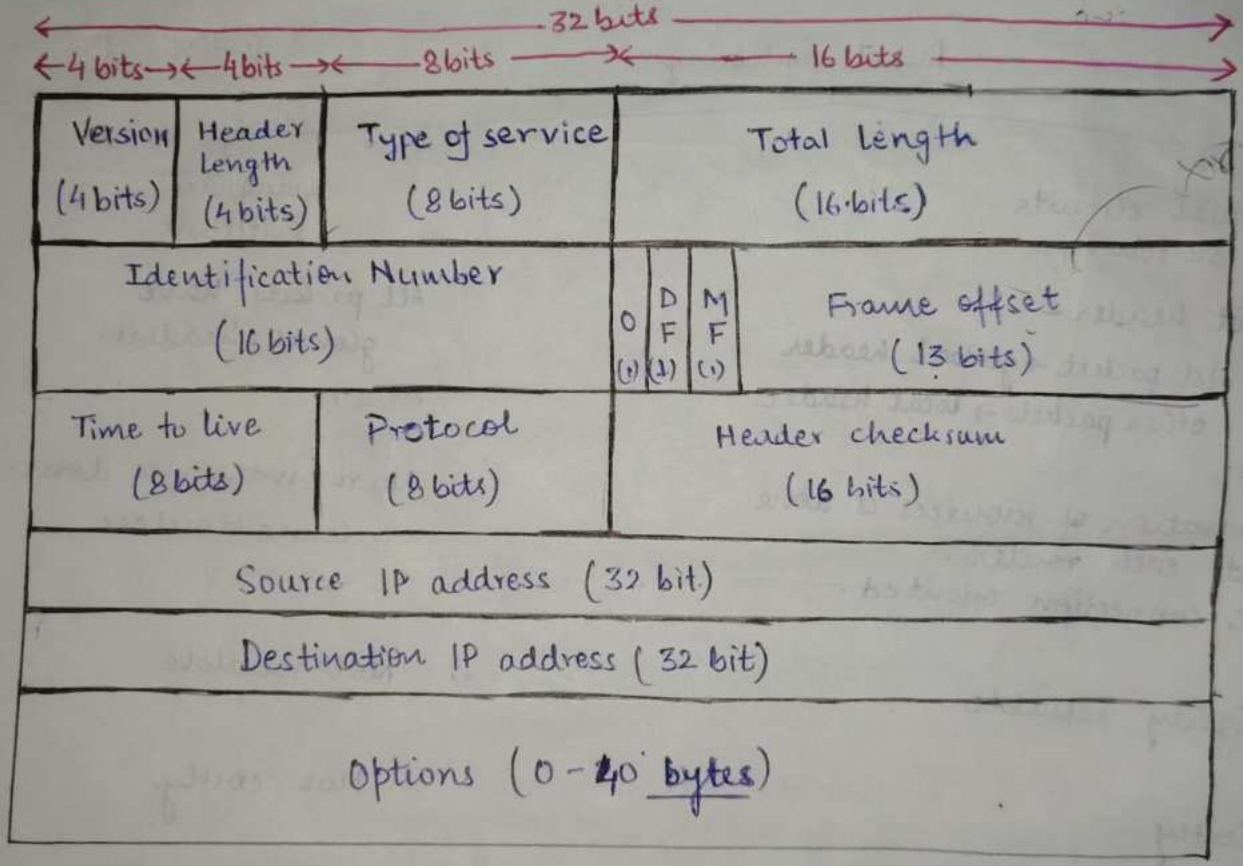
Packet switching

Virtual circuits switching	Datagram switching
Packet header – 1st packet → global header; other packets → local headers	All packets have global headers.
Reservation of resources is done at each router. ∴ connection oriented.	No reservation is done. ∴ Connectionless.
Highly reliable	Not reliable
Costly	Not costly
Packets are received in order.	Out of order packets may be received.

IPv4

- IPv4 is short form of Internet Protocol version 4 (fourth version of IP)
- Responsible for delivery of packets from source host to destination host.
- IPv4 is a connectionless protocol for use on packet switched network

IPv4 Header



1. Version

- 4 bit field that indicates version of IP being used.
- For IPv4, it is 0100
- Datagrams with different versions are parsed differently
  - → IPv4 datagrams are parsed by version 4 parsers.
  - → IPv6 datagrams are parsed by version 6 parsers.

2. Header length

- 4 bit word that tells the length of header in 4 byte words
- Example:- If length of header = 44 bytes,
  header length field contains $44/4 = 11$

20 bytes <= Header size <= 60 bytes

5 <= Header length field <= 15

Scaling factor = 4

3. Type of service

- 8 bit ~~field~~ field used for Quality of service.
- Special treatment is given to datagram for ~~type~~ a particular service.

4. Total length

- 16 bit ~~field~~ field.
- Total ~~length~~ length of datagram in bytes.

Total length = Header length + Payload (data)

Minimum length of datagram = 20 bytes
Maximum length of datagram = 2^{16} = 65536 bytes

5. Identification Number

- 16 bit ~~field~~ field.
- Identification of fragments in the original datagram.

When an IP datagram is fragmented,

- Each fragmented datagram is assigned the same identification number.
- The number is important for reassembly of ~~p~~ fragmented datagrams.

6. DF bit

Do Not Fragment bit.

0 → grants permission to intermediate devices to fragment datagram if needed.

1 → Datagram cannot be fragmented.

If n/w requires datagram to be fragmented to travel further but the settings do not allow, then, the datagram is discarded.

An error message is sent to the sender saying that the packet has been discarded due to its settings.

7. MF bit

More Fragments bit.

0 → indicates to the receiver that the datagram is either the last fragment in the set or it is the only fragment.

1 → indicates to the receiver that the current datagram is a fragment of some larger datagram.

More fragments are following.

MF bit is set to 1 on all fragments except the last one.

8. Fragment offset

- 13 bit field.
- Indicates the position of a fragmented datagram in the original unfragmented datagram.
- First fragmented datagram has a fragment offset 0.

> Fragment offset for a given fragmented datagram = No. of data bytes ahead of it in the original unfragmented datagram.

- Scaling factor = 8

9. Time to live

- TTL is a 8 bit field.
- indicates maximum number of hops a datagram can take to reach the destination.
- Main purpose of TTL is to prevent the IP datagrams from looping around forever in a routing loop.

Value of TTL is decremented by 1 when

!! - Datagram takes a hop to any intermediate device having network layer.

!! - Datagram takes a hop to the destination

Datagram is discarded if value of TTL is 0 before reaching destination.

10. Protocol

- 8 bit field.
- Tells the n/w layer at the routers / destination to which protocol the IP datagram belongs.
- Protocol no —

ICMP → 1 IGMP - 2

TCP → 6 UDP - 17

Priority of protocols — TCP > UDP > IGMP > ICMP

11. Header checksum

- 16 bit field.
- checksum value of entire header.

At each hop —

- Header ~~set~~ checksum is compared
- If mismatched, datagram is discarded.
- Router modifies the checksum field whenever it modifies IP header.

Computation of header checksum includes IP header only.

Errors in the data field are handled by the encapsulated protocol.

Fields that can be modified —

1. TTL
2. Options
3. Header length
4. Total length
5. Fragment offset.

12. Source IP Address

- 32 bit field
- logical address of sender of the datagram.

13. Destination IP Address

- 32 bit field
- logical address of destination of datagram.

14. Options

- 0 - 40 bytes

Purpose of options —

↳ **Record Route**

This option is used to record the IP addresses of the routers through which datagram passes.

Max. no. of IPv4 router addresses that can be recorded	= 9

↳ **Source Routing**

This option is used to specify the ~~root~~ route that the datagram must take to reach destination.

- **Loose routing** — only some routers are given in the path.
- **Strict Routing** — all the routers are given.

↳ **Padding**

To make header length multiple of 4.

FRAGMENTATION

- Process of dividing a datagram into fragments during its transmission.
- Done by intermediary devices like routers at the network layer.

Need –
Each n/w has its MTU (Maximum ~~Transmissible~~ Transmission Unit)
maximum size of datagram that can be sent to it.

~~Fragments~~
Datagrams of size greater than MTU cannot be transmitted through the network.

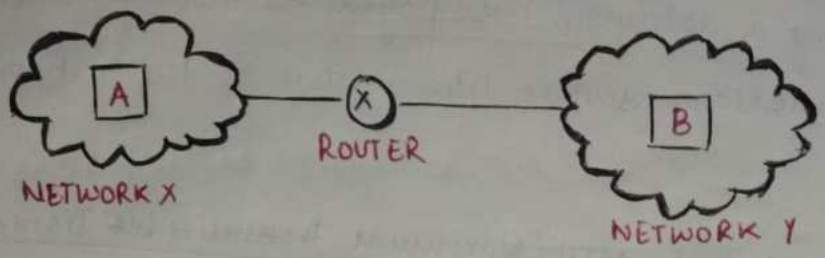
Datagram Fragmentation

When router receives a datagram to transmit further, it examines
- Size of datagram
- MTU of destination network.
- DF bit in IP header.

Cases	DF bit	Action
Size <= MTU	–	Transmit the datagram
Size > MTU	0	Datagram is fragmented
Size > MTU	1	Datagram is discarded

Changes made by the router during fragmentation
- Total length field is changed to the size of fragment.
- MF bit for every datagram other than last fragment is set to 1.
- It changes the fragment offset value
- Recalculates the header checksum.

IP Fragmentation example –



MTU of n/w X = 520 bytes
MTU of n/w Y = 200 bytes
Host A wants to send a message to host B.

Router receives a datagram from host A having
→ Header length = 20 bytes
→ Payload length = 500 bytes
→ Total length = 520 bytes
→ DF bit = 0

Length of datagram (520 bytes) < MTU of n/w Y.
∵ DF bit is set to 0, therefore, fragmentation is allowed.

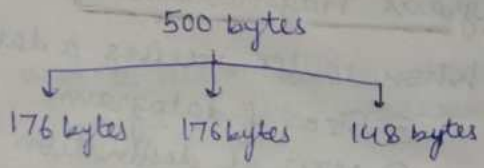
Size of fragments should be ~~or~~ multiple of 8 (because frame offset value is scaled to 8.

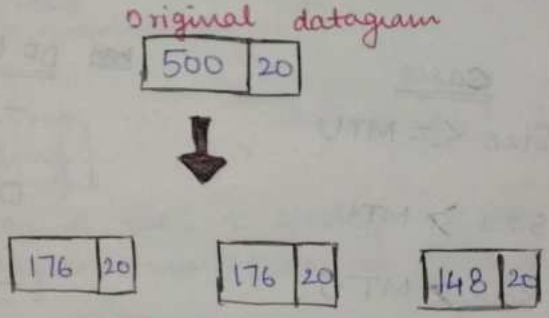
~~Maximum value~~
• Size of datagram header in each fragment = 20 bytes

∴ Payload length maximum that can be transmitted
= 200 − 20 = 180 bytes

Greatest no less than 180 which is the multiple of 8 = 176

∴ The datagram is divided as follows —





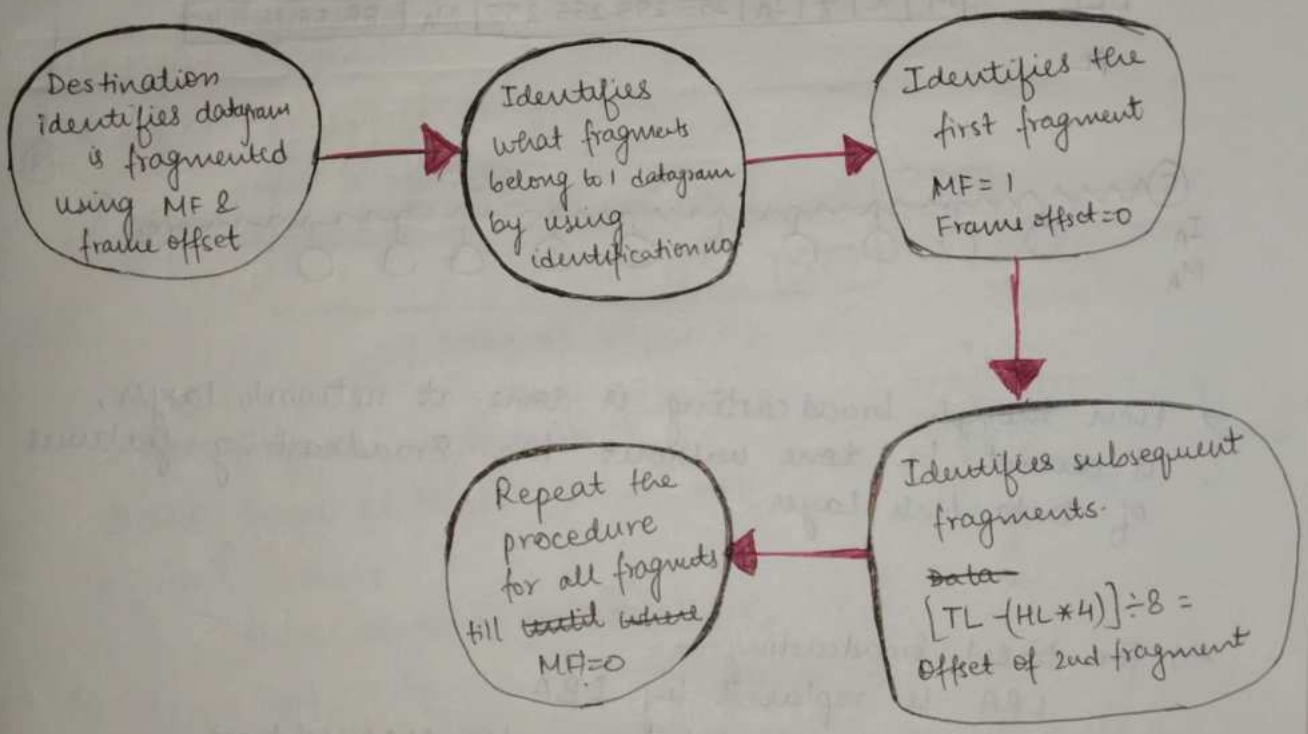
1st fragment –
① Header length field value = 5
② Total length field value = 196
③ MF = 1
④ Fragment offset = 0
⑤ Header checksum is recalculated
⑥ Identification no. is same as original

3rd fragment
Header length = 5
Total length = 168
MF = 0
Fragment offset = $\frac{176+176}{8} = 352/8 = 44$

Reassembly Algorithm

Depending upon MF and frame offset, receiver identifies that the datagram is fragmented.

MF	Frame offset	
1	0	1st fragment
1	!0	Intermediate frame
0	0	Only single datagram (No fragmentation)
0	!0	Last fragment.



Important points –

1. Different fragments from same datagram can take different routes to reach destination
2. Reassembly is always done at destination (never at routers)

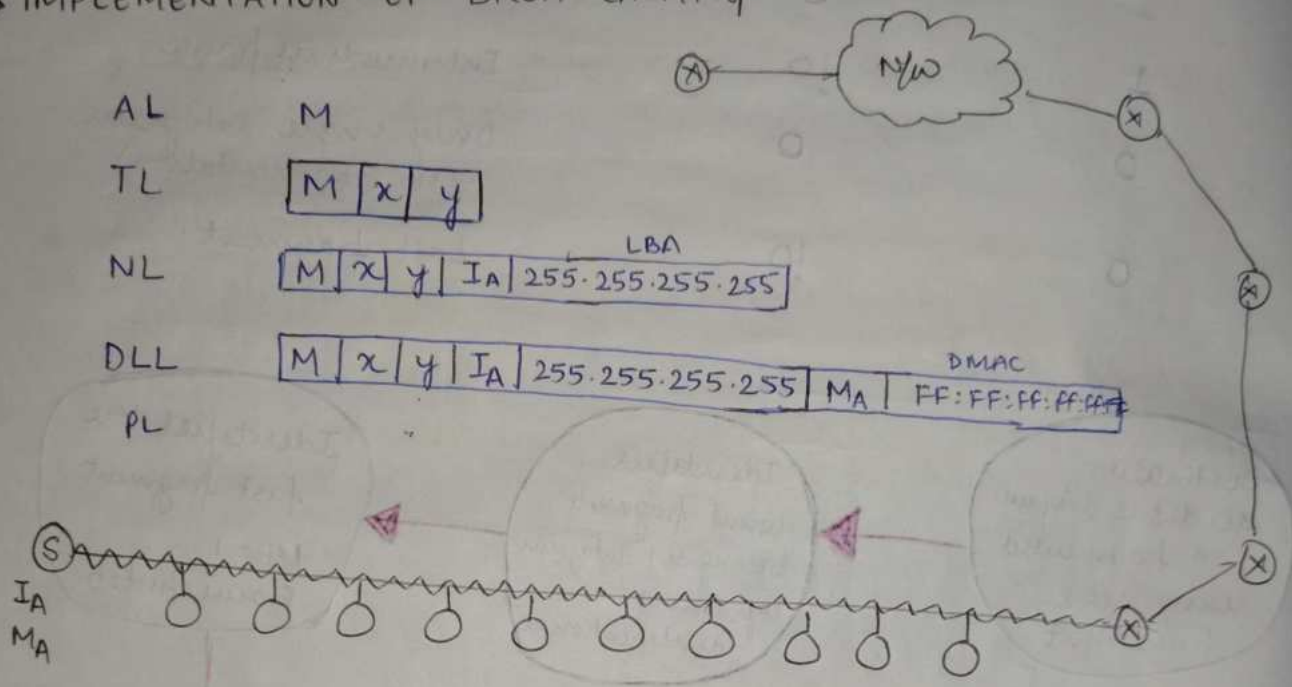
Total overhead = (No of fragments − 1) * Size of IP header.

Efficiency = $\frac{\text{Data without header}}{\text{Data with header}}$

Bandwidth utilization / Throughput = Efficiency * Bandwidth.

PROTOCOLS AND CONCEPTS AT NETWORK LAYER

→ IMPLEMENTATION OF BROADCASTING



→ Even though broadcasting is done at network layer, it cannot be done without the Broadcasting features of Data link layer.

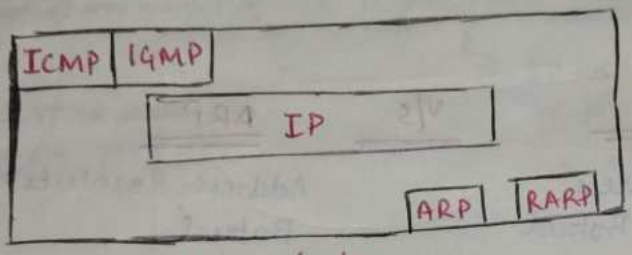
→ For Direct Broadcasting —
LBA is replaced by DBA
DMAC replaced by next router MAC address
Router routes the message as unicast message & when the router forwards to the n/w, then, DBA is changed back to LBA & DMAC will be FF:FF:FF:FF:FF:FF.

ARP (Address Resolution Protocol)

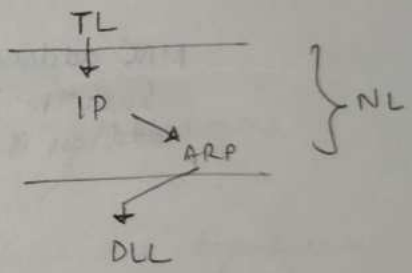
Need:- Most of the computer programs/applications use logical address (IP address) to send/receive messages.
However, the actual communication happens over physical address (MAC address)

So, destination MAC address is required to communicate with other ~~prot~~ devices.

Function of ARP ⇒ IP address → MAC address.

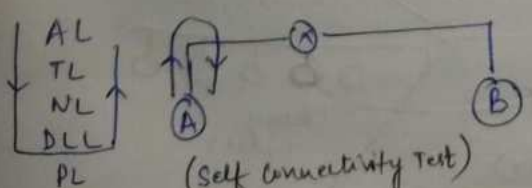


Network layer



① ARP Request is broadcast
② ARP ~~Reqst~~ Response is unicast
③ used when
Host needs to find MAC of host
Host → Router
Router → Host
Router → Router

~~Spec~~
Special Address 127 / Loop Back Address



(Self Connectivity Test)

If a packet is sent with destination address 127 then the packet is again going to come back to the same host.

The command to test is ping 127.1.2.3. We are supposed to see RTT as ≅ 15 ms.
If sth like Timeout appears, sth is wrong in our NIC and we should troubleshoot.

Not in gate

RARP (Reverse Address Resolution Protocol)

- IP of a system is not constant. Therefore it can't be stored in the ROM. While MAC address is stored in ROM.
- So, RARP is used to get IP address from MAC address.
- NFS → Network File server
  Contains a table of IP & MAC addresses.

Disadvantages

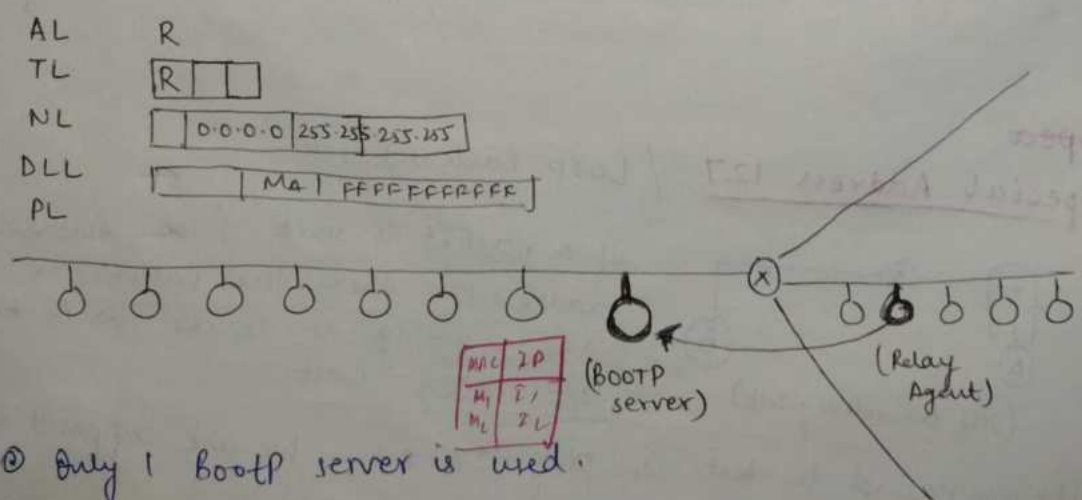
- Every n/w should have RARP server
- static mapping (no. of IP addresses > no. of hosts)

RARP v/s ARP

RARP	ARP
Reverse Address Resolution Protocol	Address Resolution Protocol.
MAC address is known IP address is requested	IP address is known & the MAC address is requested.

BOOTP (Bootstrap Protocol)

- Main difference b/w RARP & BOOTP — BOOTP works at AL, and RARP works at DLL.



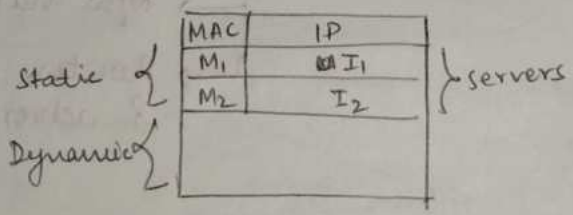
- Only 1 BootP server is used.
- Relay agents are present in those n/ws which do not have BOOTP server

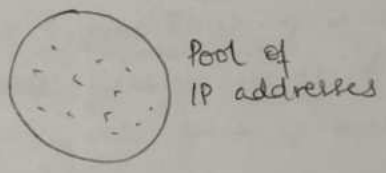
Relay agent can communicate with the BOOTP server because it knows the IP address of BOOTP server.

Adv → Only 1 Bootp server is used
Disadv → Mapping table is static

DHCP (Dynamic Host Configuration Protocol)

- Main difference b/w DHCP and BOOTP is that the table is not static in DHCP.
- DHCP has a table at DHCP server & the table has 2 parts — static part & dynamic part.
  - static part → servers are given permanent IP address
  - dynamic part → IP address is assigned when required. valid upto lease time





If host A, M_A wants its IP address then the dynamic part will look like.

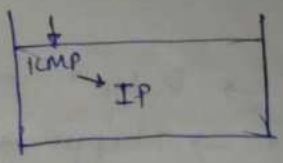
MAC	IP	Lease Time
M_A	I_A	10min
M_B	I_B	10min

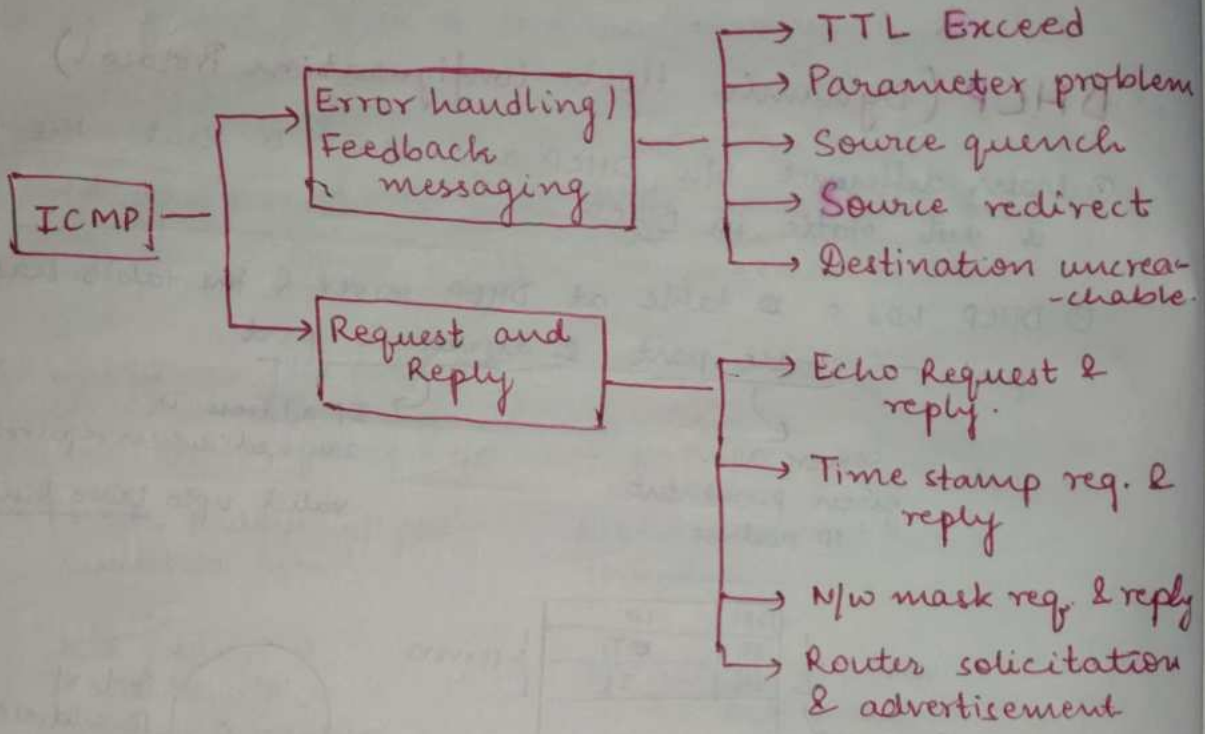
- Only 1 DHCP server is enough
- Table is dynamic. (No. of IP addresses used = No. of hosts online)
- To make DHCP compatible with BootP, both DHCP & BOOTP should have same port number.

→ DHCP is operated at application layer
∴ cannot be implemented on Router as Router has only 3 layers.

ICMP [Internet Control Message Protocol]

② works at Network layer





Error Handling / Feedback messaging

whenever there is an error / anything goes wrong, ICMP protocol is used.

IP is unreliable.

Packets are generally discarded at routers due to congestion

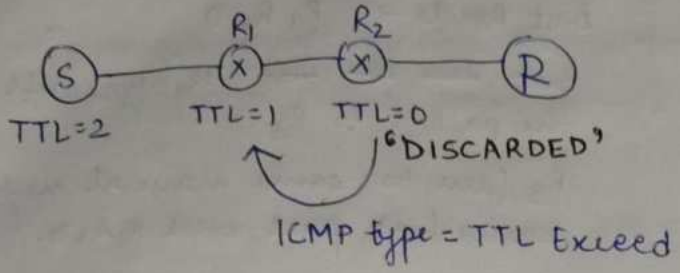
→ If IP packet is lost, ICMP packet is generated

→ If ICMP packet is lost, no ICMP packet should be generated

bcoz it may fall into infinite loop

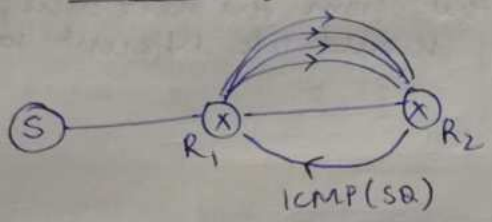
∴ (ICMP + IP) → unreliable

1. TTL Exceed



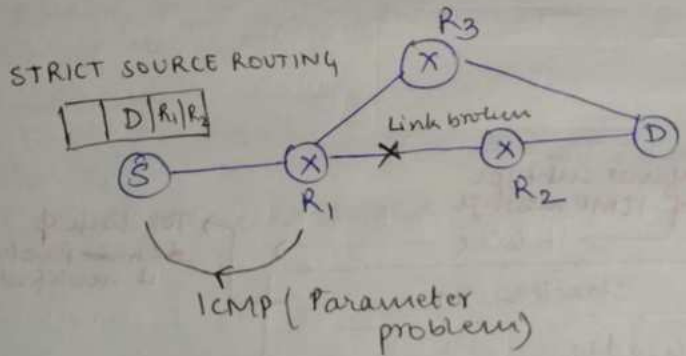
If TTL is 0 before the packet has reached its destination, then, router sends ICMP packet to the source.

2. Source quench



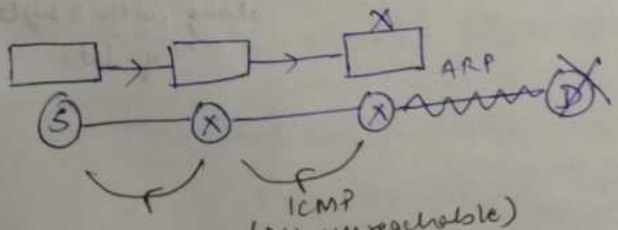
R_1 sends a lot of packets & R_2 cannot handle them. It sends ICMP packet of type 'source quench' asking R_1 to send packets via a different route.

3. Parameter problem



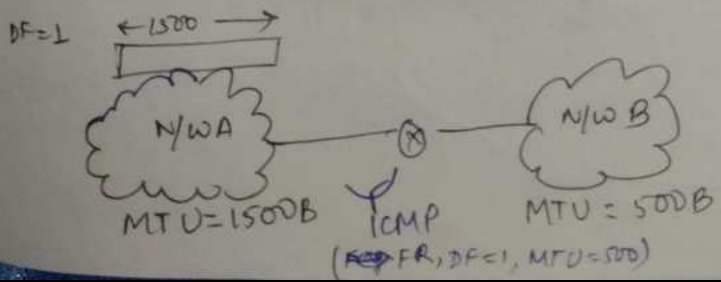
Strict Source Routing is used. Options field is set as D, R_1 R_2 but $R_1 \rightarrow R_2$ link is broken. Packet is discarded. ICMP (Parameter Problem) is sent to the sender.

4. Destination unreachable



→ It is of 2 types
- → Destination Host Unreachable
- → Destination Port unreachable

→ R_2 sends ARP request for getting MAC address of the host. If host is down, R_2 sends ICMP(DHU) error message.

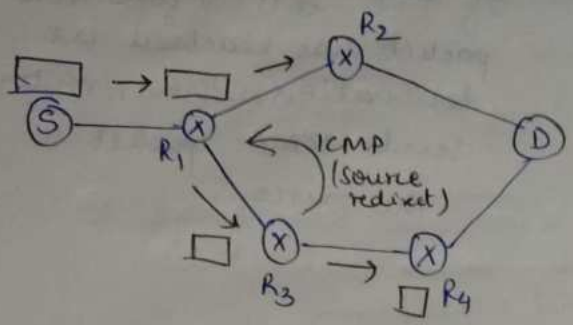


DF bit is set to 1 but MTU requires fragmentation.

FR = Fragmentation Required.

5. Source Redirect

① packet is not discarded.
② it is warning not error message



Best Route → $R_1 R_2 D$

But ~~due to sour~~ R_1 forwards the packet to R_3..

R_3 (due to some manual mechanism) knows best part was $R_1 R_2 D$.

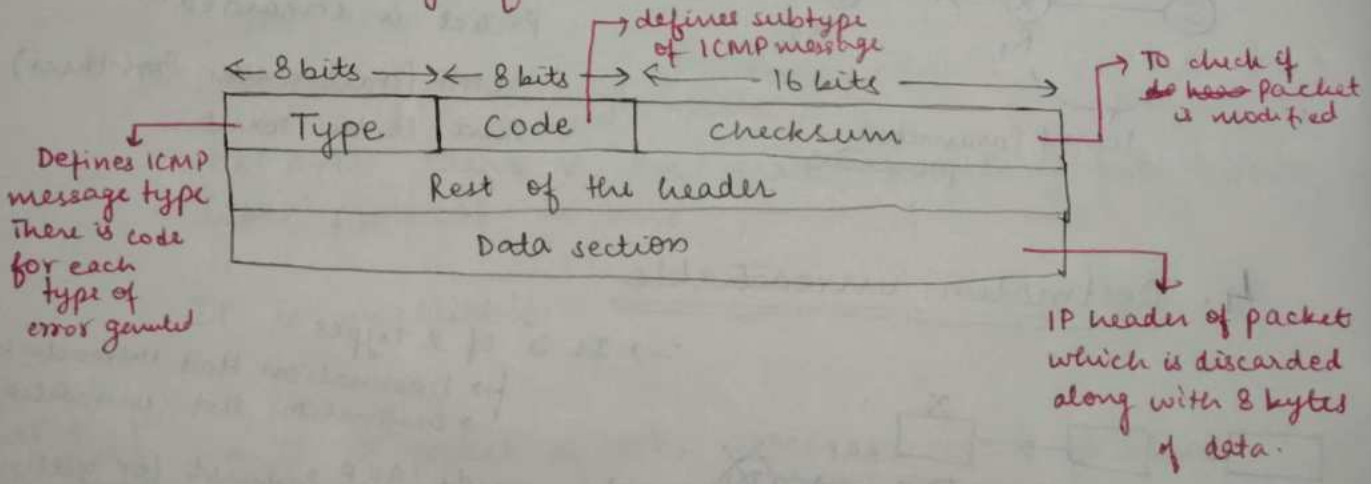
It will forward the packet to R_4 but at the same time, it will send ICMP packet to R_1 telling it that from the next time, it should use different route

* If a packet is fragmented into 100 parts and it is discarded, then only 1 ICMP packet is sent.

Bcoz anyway the sender will have to send the ~~entire~~ all ~~msg~~ 100 fragments again.

→ If 3rd packet is lost, no ICMP is generated.

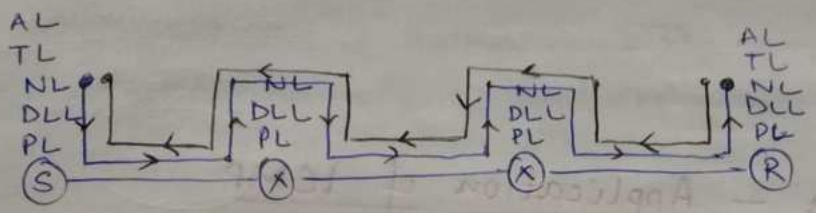
ICMP message format



ICMP Request and Reply messaging

① Echo Request and Reply

- used to test whether the NL of destination and the intermediate routers are working or not.
- PING command is used = Packet Internet Grofer.
  - works at NL not AL
  - ∴ not client-server



② Router solicitation and advertisement.

Station present in n/w sends broadcasting message to all the routers connected to it

Router which is available sends back the reply.

It is then made default router / gateway.

Newly added router to a network sends message to all the hosts that it is available to be used as default router.

③ Time stamp Request and Reply.

- N/w devices are available in different parts of the world.
- different time zones.

→ Problem of synchronization

→ As a result flow of messages

For better synchronization, special type of ICMP message called Time stamp request & reply is used.

Network Mask Request and Reply

- N/w mask is used in routers.
- N/w Administrator is responsible for setting up the router in such a way that the n/w mask is present in the router.
- Every host can send ICMP packet to the router to get the subnet mask

Traceroute – Application of ICMP

Traceroute provides a map of how data on the internet travels from the source to the destination.

Procedure –

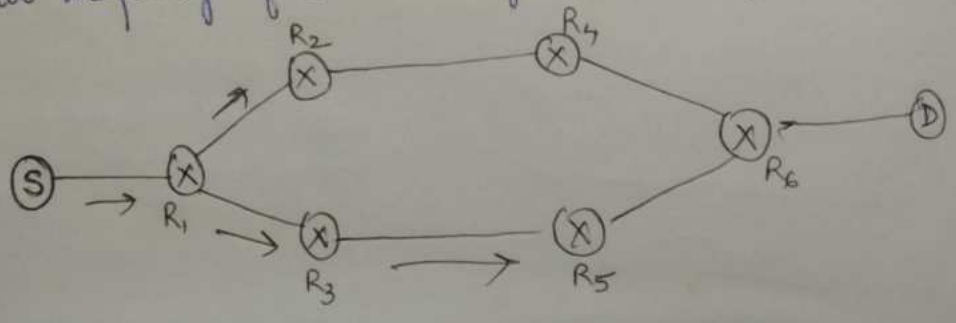
- Send a packet with TTL = 1. TTL = 0 at first router and it sends back IP of the router.
- Repeat the process with TTL = 2, 3, 4 ....
- Destination doesn't send ICMP packet when TTL = 0 bcz packet isn't discarded.

  ∴ A dummy port no. that doesn't exist at the destination is used.

  In this case, destination ~~prot~~ sends ICMP packet of type Destination Port Unreachable.

- The process is stopped when ICMP(DPU) ~~path~~ is received.

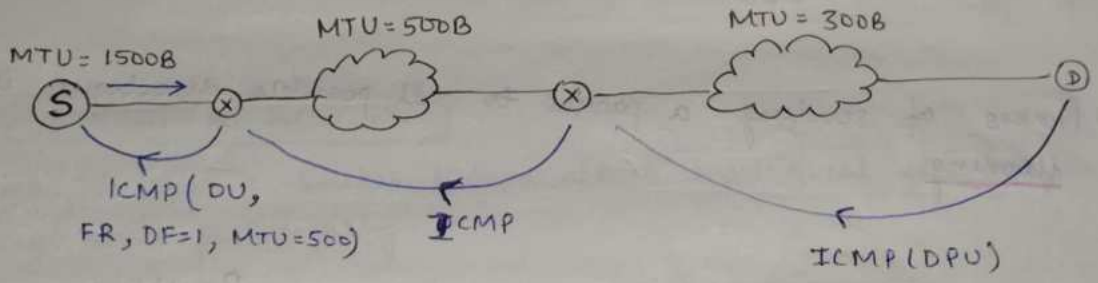
→ Traceroute might not ~~be~~ give the actual path. But in majority of cases we get actual path.



Traceroute = ~~R~~ R_1 R_2 R_5 R_6 = not valid path
bcoz packet can move in any direction independently.

PMTUD — Path MTU Discovery — Application of ICMP

Finding ~~minimum~~ maximum size of datagram that can be sent from source to destination such that no fragmentation takes place.



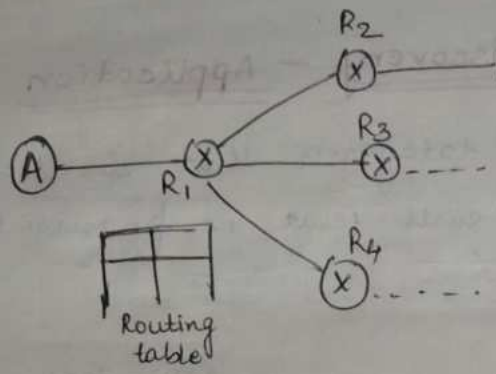
Sender sends a packet of size 1500B
It receives ICMP (DU, FR, DF=1, MTU = 500)

Sender sends a packet of size 500B
It receives ICMP (DU, FR, DF=1, MTU = 300)

Sender sends a packet of size 300B
It receives ICMP (DPU)
↳ because dummy port no. is used.

ROUTING

- Process of preparing a routing table (at every router) is called routing.



Routing table is used to decide to which router the packet should be sent next.

- Process of sending a packet to all possible directions is called flooding.

Flooding

- No routing tables are required
- Shortest path is guaranteed
- Highly reliable

Disadv

- Duplicate packets at destination
- More traffic

Routing

- Routing table is required
- Shortest path is not guaranteed
- Less reliable.

Adv

- No duplicate packet at destination
- Less traffic

Routing algorithms

Static

→ Manually prepare and upload routing table offline.

→ If topology changes (link broken or new link added), or traffic changes (high traffic at a router), then, manual intervention is required.

∴ Not used

Dynamic

Topology changes & traffic changes are handled automatically by the algorithm.

- Distance Vector Routing (DVR)
- Link State Routing (LSR)

DISTANCE VECTOR ROUTING

DVR requires that a router inform about topological changes to its neighbours periodically.

The routing table at a router contains 3 fields—

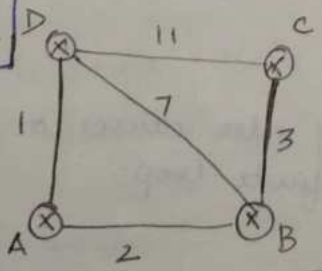
1) Destination 2) Distance 3) Next Hop.

Algorithm—

① Router transmits its distance vector to each of its neighbours.
② Each router receives & saves the most recently received distance vector from its neighbours.
③ Router then recalculates its distance vector.

1)

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

⊙ Each router sends the distance vector to neighbouring routers.

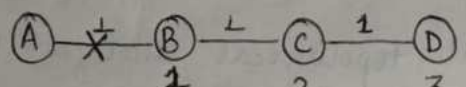
⊙ Routing table is again designed by using the latest data.

Distance = minimum of the values from neighbours

Bellman Ford Algorithm

Count to infinity – problem of DVR

If a link is down, then, it takes a lot of time for the routers to update their ~~to~~ routing tables.



Link b/w A & B is down suddenly

A	B	C	D	(distances to A)
1		2	3	
∞	2	3		
3	4	3		
5	4	5		
5	6	5		
7	6	7		
⋮	⋮	⋮		
∞	∞	∞		

The distances gradually tend to infinity.

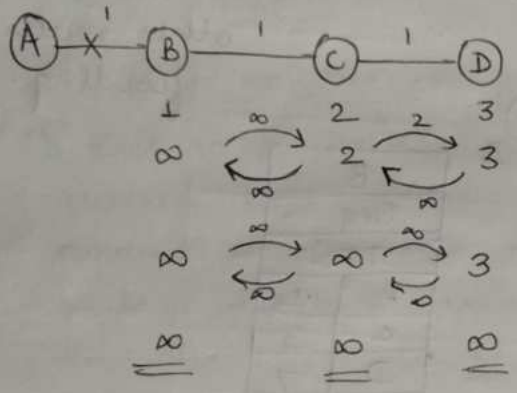
Reason for the above problem –
Only distance vector is ~~sent~~ exchanged b/w the routers. (not the next hop).

Count to infinity also causes a packet to fall into infinite loop.

Split Horizon – Solution to Count to Infinity Problem.

Solution to count to infinity is called 'Split Horizon'

While sending distance vector from A to B, distances in A which depend upon B (ie with next hop as B) are set to ∞.

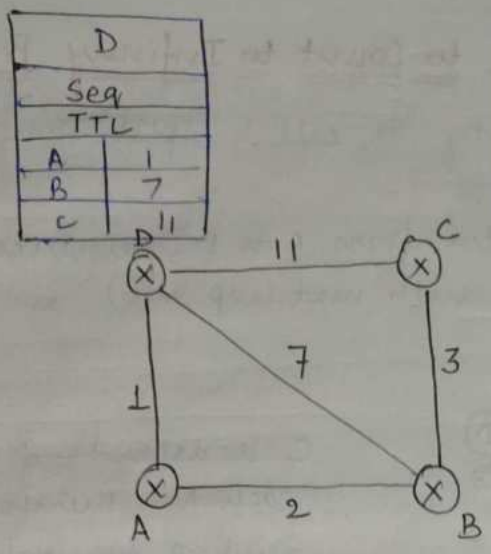


C is depending upon B to reach A. Therefore, instead of sending the value 2, it sends the value ∞.

- By using split horizon, convergence is fast, no loops are formed.
- In case of only DVR, convergence is slow and loops will occur.

LINK STATE ROUTING (LSR)

- Every router creates something called Link State Packet
- Link state packet contains –
  - Neighbouring nodes
  - Distance to the neighbouring nodes.
- Every router receives link state packet from every other router in the network.
- The link state packets are sent by flooding.
- By using the ~~packet~~ info in the packets received, each router creates its routing table using DIJEKSTRA ALGORITHM



The Link state packets are sent to every other router by flooding.

A

Seq.	
TTL	
B	2
D	1

B

Seq.	
TTL	
A	2
C	3
D	7

★ Every node has global database.

DVR – local database (depends only upon info of neighbour)
LSR – global database (depends upon info of all routers).

Single source Shortest Path algoritham is applied at each node ★

DIJEKSTRA ALGORITHM.

∴ Routing table at A

Des	Dis	NH
A	0	A
B	2	B
C	5	B
D	1	D

Routing table at B

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

* LSR converges faster than DVR

~~LRS~~

LSR – Problems

⊙ Heavy traffic due to flooding.

⇒ Sequence number is present on each link state ~~Rou~~ Packet.

D Each router stores the seq. no of the latest packet received from a particular host. If a packet with sequence no less than that already received reaches a router, (due to a different path which takes longer time), then, the packet is discarded.

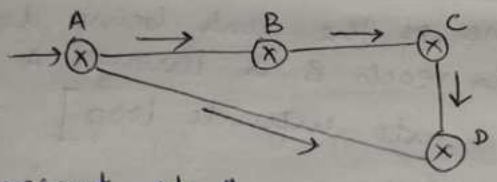


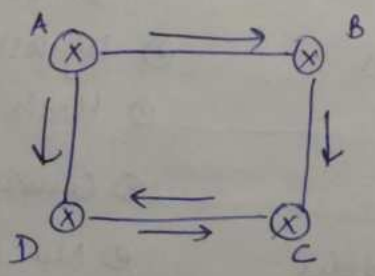
Table present at A –

Router	Latest
B	~~10~~ 15
C	20
D	30

(B, 8) – Discarded ×
(B, 15) – Accepted ✓
(B, 11) – Discarded ×

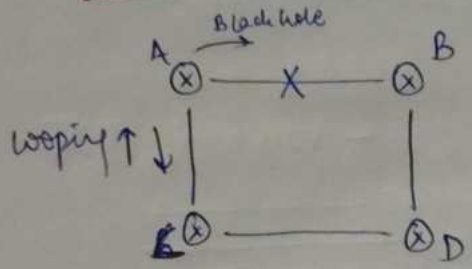
* Flooding can be controlled by using sequence numbers. If old packet is received at a router, it is not transmitted further.

⇒ Time to live field is used to prevent a packet from falling into infinite loop.



⇒ Sequence bits can get currupted. ~~Therefo~~
To solve this problem, validity / lifetime column is used. After a fixed amt of time, the entry is discarded & the currently travelling packet seq. no is accepted.

Other problems with LSR



LSR has 2 ~~persistent~~ pro' transient problems (for short time)

1) **Black hole** [If A–B link is down suddenly, it takes 'timeout' time to discover that link is down. Upto that time, packets sent from A to B will fall into black hole]

These problems are present only for a short amount of time and get rectified automatically

2) **Looping** [After timeout, A redesigns its routing table and finds that shortest path to reach B is via C→D→B. It sends the packet to C.

C is not aware of the link being down. Acc to C, shortest path to reach B is through A.

∴ Packet falls into infinite loop]

Differences b/w ~~ER~~ DVR & LSR –

Distance Vector Routing	Link State Routing
⊙ used in 1980s	⊙ used in 1990s
⊙ BW required is less (no flooding, vector is sent only to neighbours)	⊙ BW required is more (flooding, sent to every router)
⊙ Every router has local knowledge	⊙ Every router has global knowledge
⊙ Bellman Ford algorithm	⊙ Dijikstra algorithm
⊙ Less traffic	⊙ High traffic
⊙ Converges slowly	⊙ Converges faster
⊙ Count to infinity problem	⊙ No count to infinity proble
⊙ Persistent looping	⊙ Transient loop
⊙ Implemented by using RIP protocol	⊙ Implemented by using OSPF

RIP (Routing Information Protocol)

- RIP is the implementation of (DVR) → simple to implement
- Metric = Hopcount (weights)
- Infinity is represented by number 16 (i.e. if 16 hops are there, infinity is reached).

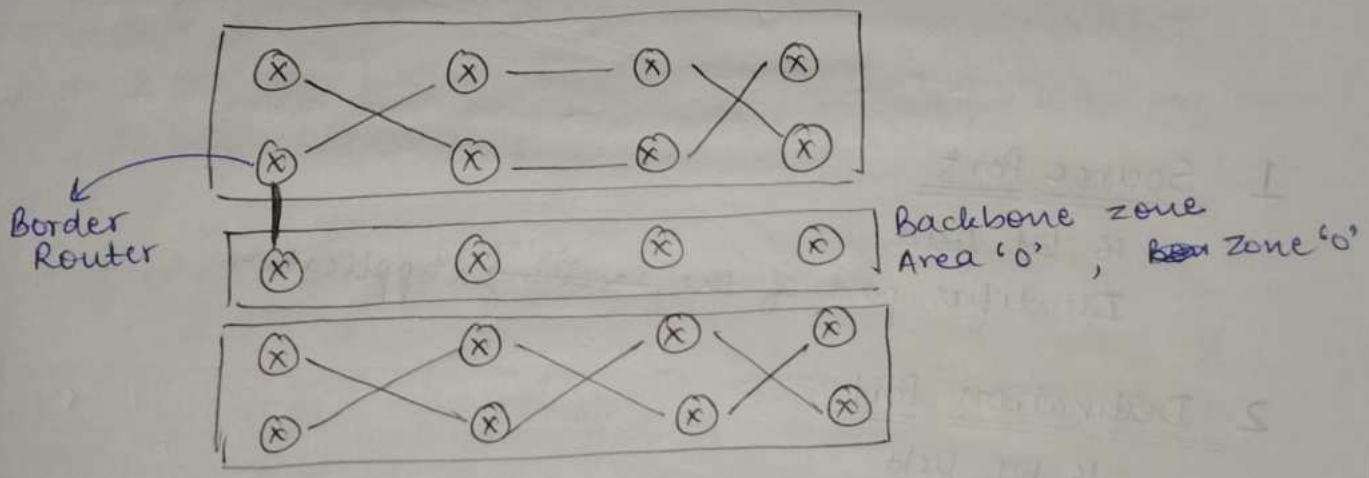
OSPF

Implementation of (LSR) → computation is complex.

OSPF divides all the routers into some regions and flooding is restricted in a particular region.

There is a router designated as 'Border Router' which receives all the flooded information, prepares summary.

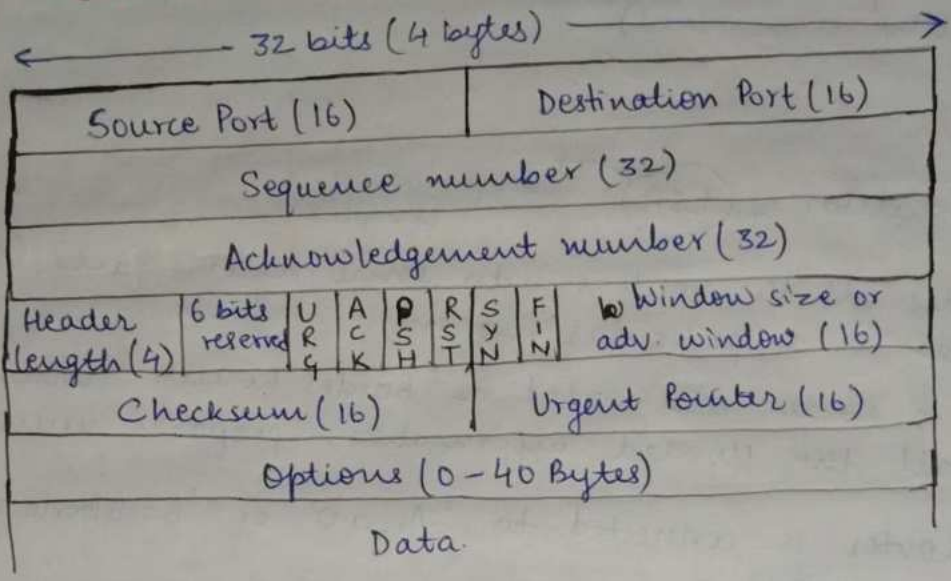
Border Router is connected to 'Area 0' or 'Backbone zone'



EIGRP protocol = RIP + OSPF

Transmission Control Protocol (TCP)

TCP header –



1. Source Port

16 bit field.
Identifies port of the ~~receiving~~ sending application

2. Destination Port

16 bit field
Identifies port of the receiving application

$0 - 2^{16} - 1$ port numbers can be represented
$(0 - 65535)$

http: 80
ftp: 21
Telnet: 23
SMTP: 25

0-1023 → well known services port no

1024 - 49151 - Reserved

49152 - 65535 - Private

Port no + IP address = identifies connection uniquely

TCP is connection oriented protocol (Resources are reserved)

Socket = Port no. + IP
(16bit) (32bit)
48 bit no which identifies process

3. Sequence Number

- 32 bit field (randomly chosen)
- TCP assigns a unique sequence number to each byte of data contained in the TCP segment.
- This field contains the sequence number of first data byte

4. Acknowledgement Number

- 32 bit field.
- Contains sequence number of the data byte that receiver expects to receive next from the sender.
- Acknowledgement number = Sequence number of last received data byte + 1.

5. Header length

- 4 bit field.
- scaling factor = 4.
- helps in knowing from where the actual data begins.

Length of TCP header always lies in the range 20 – 60 bytes

Header length field can contain values in the range 5 – 15

Wrap Around Time

The time is the time ~~required~~ after which a particular sequence number will be used again.

∵ there are 32 bits in sequence no. field
∴ No. of sequence numbers possible = 2^{32}

Wrap Around Time (WAT) depends upon bandwidth.

Example — Bandwidth = 1MBps

$\Rightarrow$ 1 sec $\rightarrow$ 1MB $\Rightarrow$ 10^6 bytes $\rightarrow$ 1 sec

$\Rightarrow$ 10^6 seq. no. $\rightarrow$ 1 sec $\Rightarrow$ 1 seq. no. $\rightarrow \frac{1}{10^6}$ sec

$\therefore$ 2^{32} seq. nos $= \frac{1}{10^6} \times 2^{32}$ sec $= 4296.967$ sec

In today's internet, there is a concept of lifetime

Lifetime → Maximum amount of time for which the packet is alive (or the time taken to reach the destination in worst case)

Lifetime = 3min = 180 sec

As long as WAT > LT, no problem.

If WAT $\ll$ LT, problem arises due to same sequence nos.

Solutions —

↳ Decreasing bandwidth (but not practical because we would DIE for bandwidth. Right?)

↳ Increasing the number of sequence numbers [This is done by utilizing options field]

Example- Bandwidth = 1GB

1 sec — 1GB ⇒ 1 sec → 1G sequence nos.

⇒ 180 sec → (180 × 1G) seq. nos. required.

No. of bits in sequence number field = $\lceil \log_2(180 \times 1G) \rceil \approx 42$

∴ No. of additional bits required for seq. no. = 42 − 32 = 10

These bits are used in options and are called time stamp.

* No. of bits in seq no. field required to prevent the problem of wrap around time = $\lceil \log_2(BW \times LT) \rceil$

Establishing TCP connection
Three Way Handshaking

Step 1 :- [SYN]

For establishing a connection,

- Client sends a request segment to the server.
- Request segment only consists of TCP header with an empty payload.
- Then, it waits for the reply segment from the server

Request segment consists of

→ Initial sequence number
→ SYN bit set to 1
→ Maximum segment size (MSS) of the network.
→ Receiving window size

Step-2:- [SYN + ACK]

After receiving the request segment,

- ⊙ Server responds to the client by sending the reply segment.
- ⊙ It informs the client of the parameters at the server side

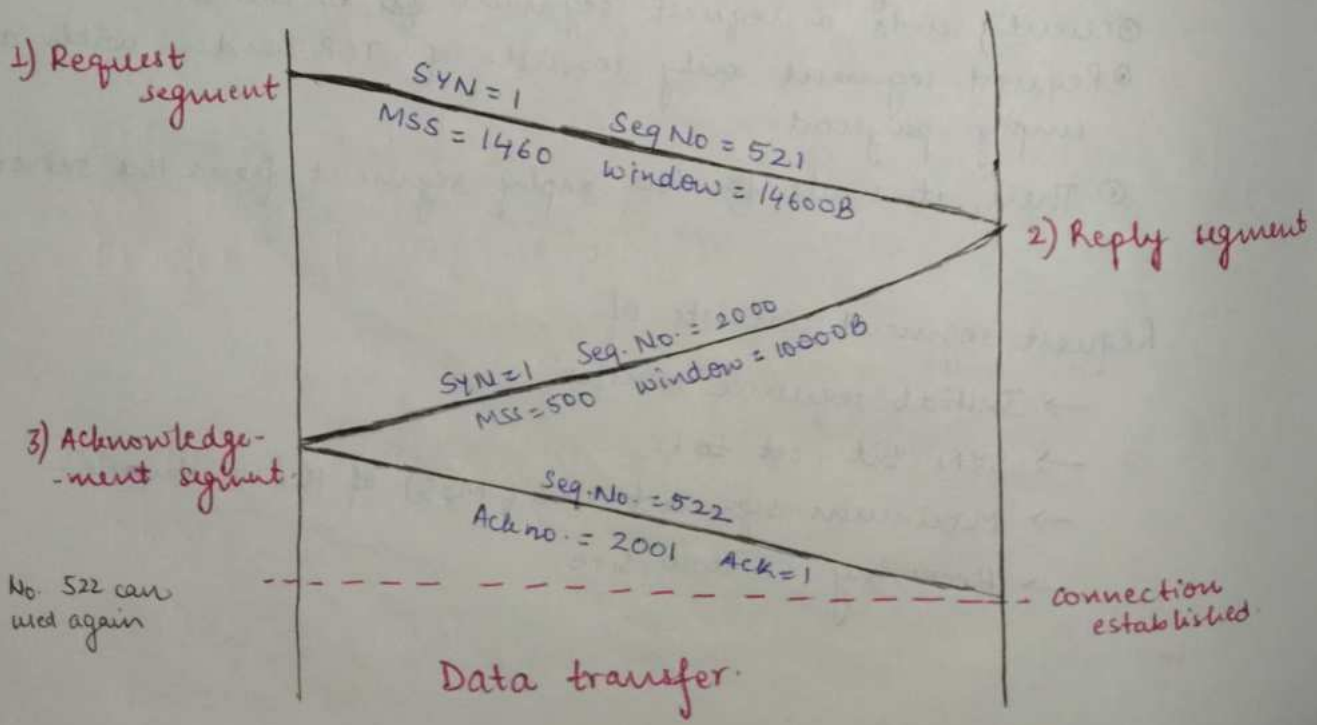
Reply segment consists of the following information -

- → Initial sequence number.
- → SYN bit set to 1
- → Maximum segment size (MSS) of the n/w.
- → Receiving window size
- → Acknowledgement number.
- → ACK bit set to 1.

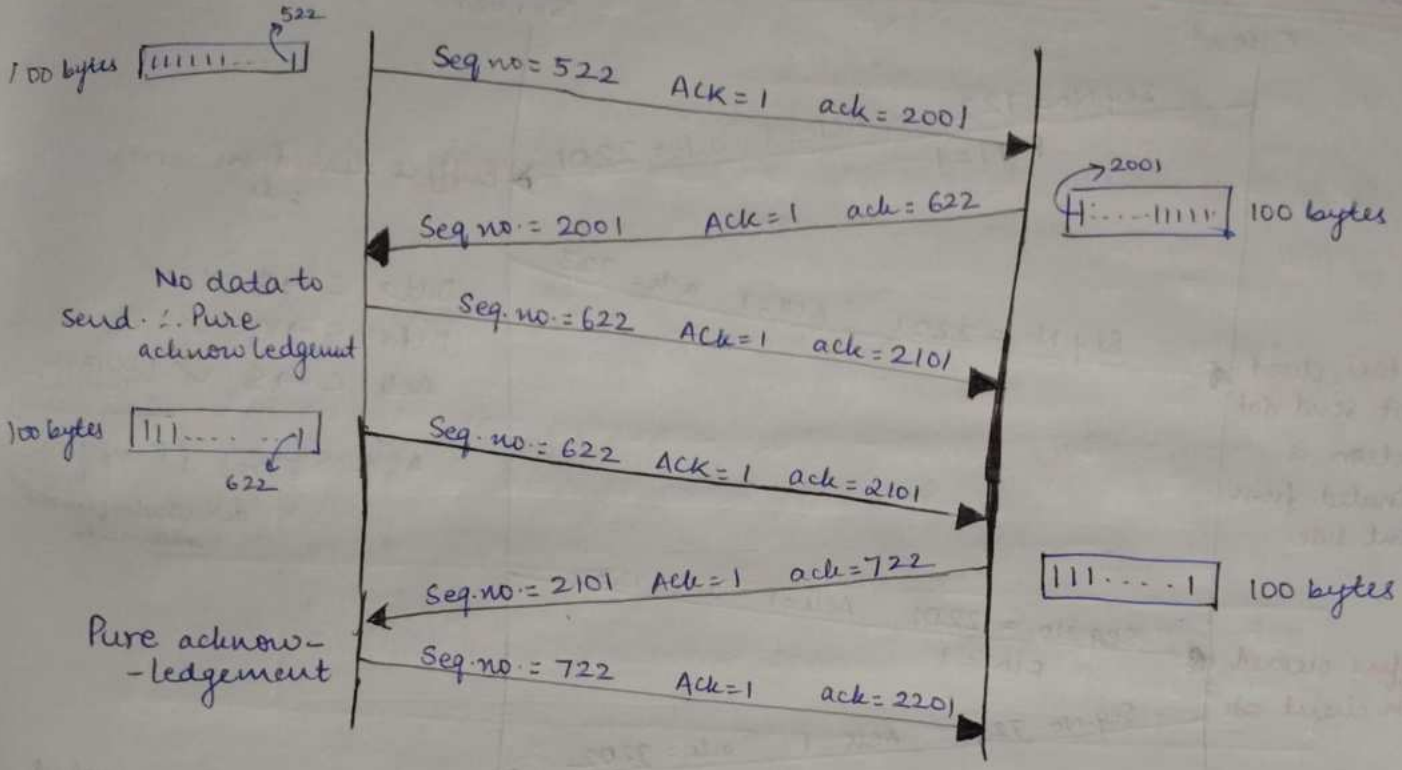
Step-3 :- [ACK]

After receiving the reply segment,

- ⊙ Client acknowledges the response of the server.
- ⊙ It acknowledges the server by sending a pure acknowledgement.



DATA TRANSFER AFTER CONNECTION ESTABLISHMENT

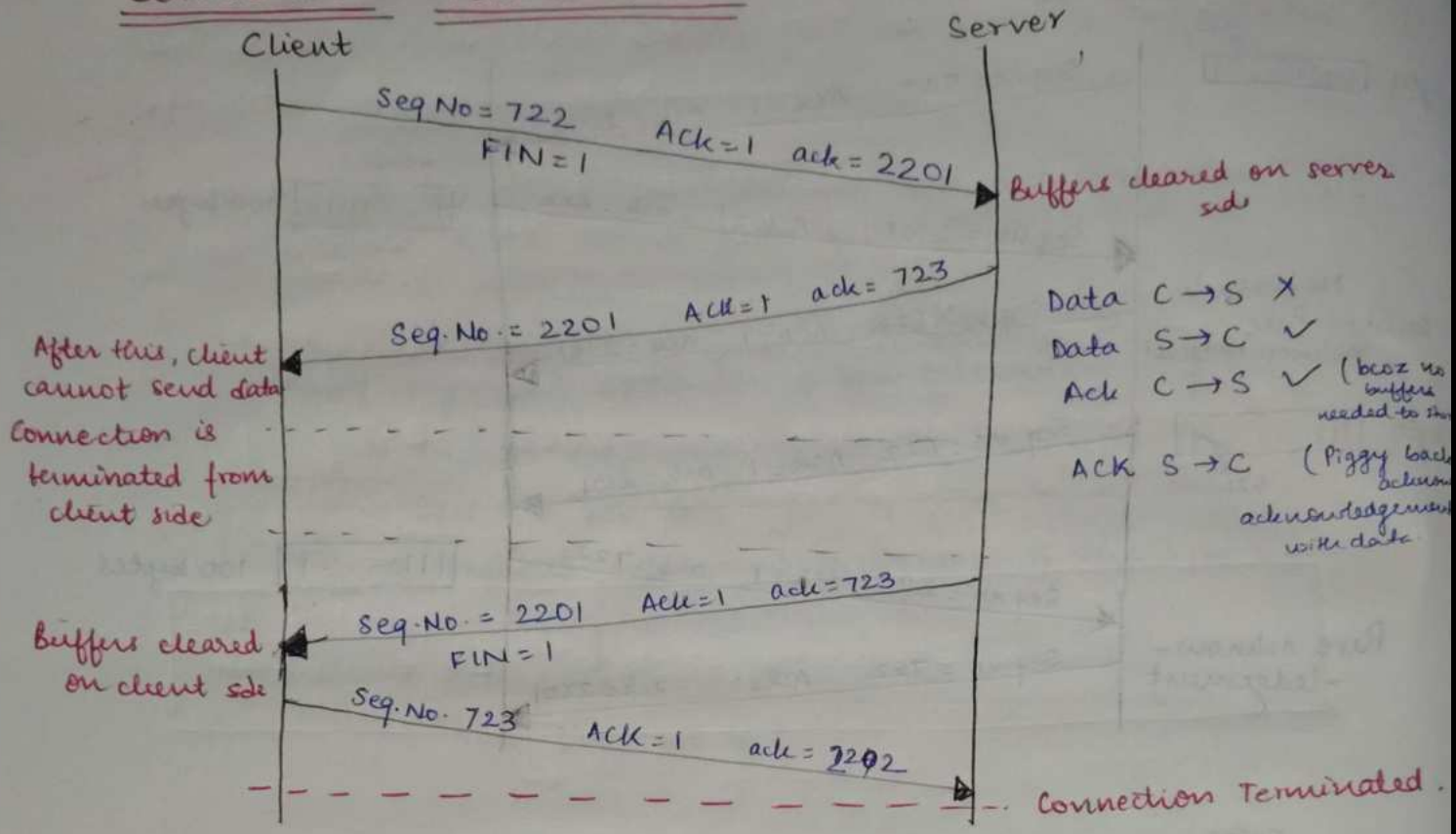


SYN	ACK	
1	0	1st segment (Request segment)
1	1	2nd segment (Reply segment)
0	1	Pure acknowledgement.
0	0	Not possible

⊙ ~~Sender~~ If there is no data to send immediately, then, instead of waiting, pure acknowledgement is sent.

No Sequence number is consumed for pure acknowledgement.

CONNECTION TERMINATION



- 3 segments send for ~~data~~ connection establishment.
- Data transfer
- 4 segments (max) or 2 segments (min) for connection termination.

Flags used

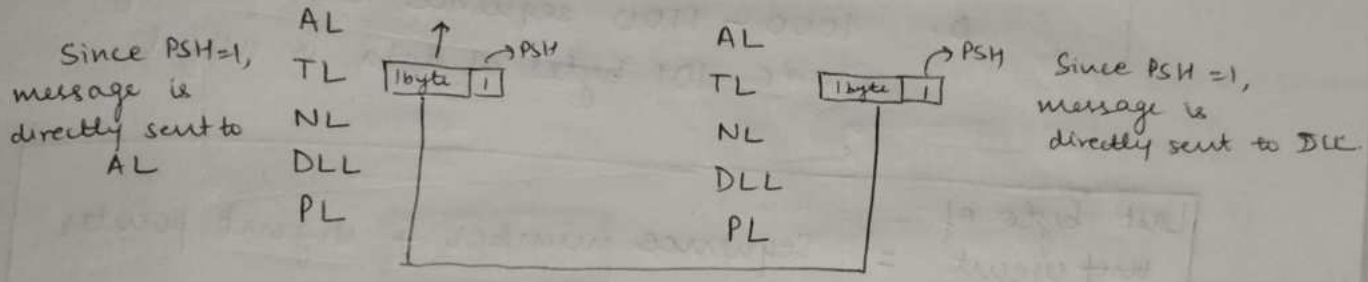
SYN flag :- (1 bit) (S.N.) Synchronizing sequence numbers [telling client/server which sequence no. is randomly chosen]
takes 1 in sequence number.

ACK flag :- (0 S.no.) Tells whether acknowledgement number field is valid or not.

FIN flag :- (1 s.no.) Request for connection termination

PSH (Push flag)

- Generally, TL takes data from AL ~~one~~ some bytes at a time and waits for more data until the size of data reaches MSS.
- PSH flag is used to tell TCP not to buffer the data and directly push the data (though of small size) to the DLL.
- Interactive applications [Chat, Telnet, SSH, ~~Puppy~~ Putty, Rlogin] etc. have PSH=1.



URG (Urgent Flag)

- used to tell that the given segment is urgent and should be executed first.
- On the receiver side, a segment with URG flag set to 1 is processed first.

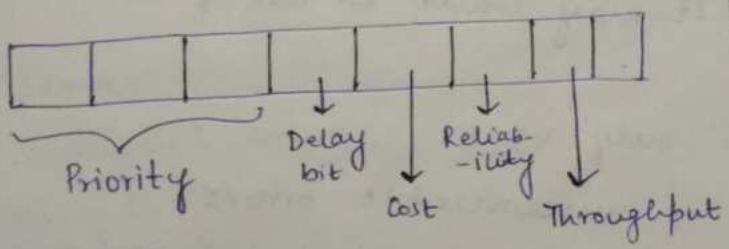
→ Thus, this flag helps in out of order processing of data.

Routers do not have TL. ∴ they cannot see the URG flag.

∴ Type of service field (IP datagram) is used.

↳ 8 bits.

For urgent ~~bit~~ flag, priority bits = 111 (7)



URGENT POINTER (16 bits)

In a TCP segment, if only some part of data is important, urgent pointer is used.

Urgent pointer tells the last ~~seq~~ sequence number upto which data is important.

If seq no = 1000, urgent pointer = 100. Then,

b. 1000 – 1100 sequence nos. are useful.
i.e. 101 bytes of data is useful.

Last byte of ~~data~~ urgent data	=	Sequence number + urgent pointer

RST flag (Reset)

used to reset the TCP connection.

When RST bit is set to 1,

- It indicates the receiver to terminate the connection immediately.
- It causes both the sides to release the connection & all its resources abnormally.
- The transfer of data ceases in both the directions.
- It may result in loss of data that is in transit.

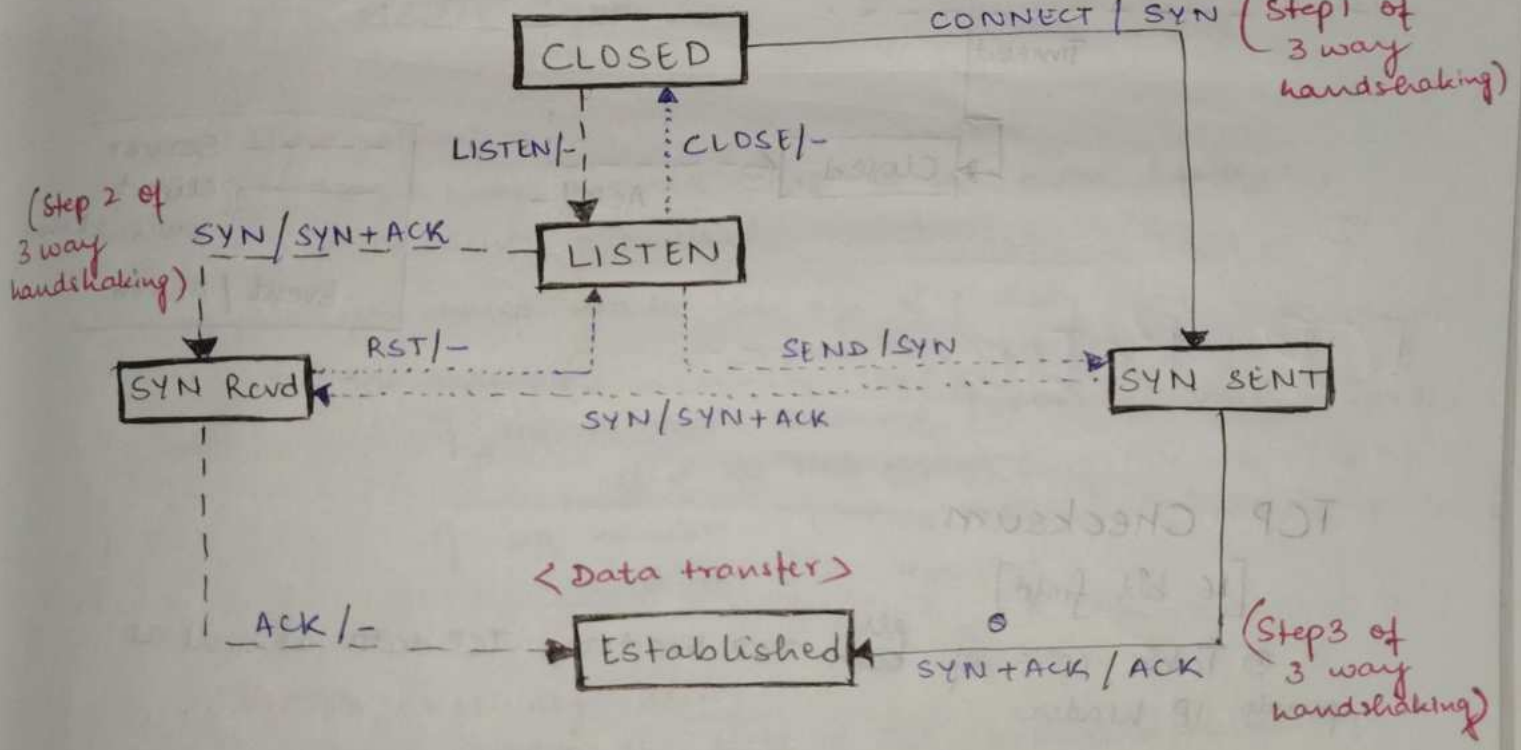
used only when

→ unrecoverable errors
→ no chance of terminating the TCP connection normally.

Window size

- 16 bit field.
- contains the size of the receiving window of the sender.
- advertises how much data (in bytes) the sender can receive without acknowledgement.
- Window size is used in flow control.
- Adv. window changes dynamically.
- If advertisement window size = 0, persistence timer is used on the sender's side after which it sends 1 byte of data to check if advertisement window is available or not.

TCP State Transition Diagram — Connection Establishment

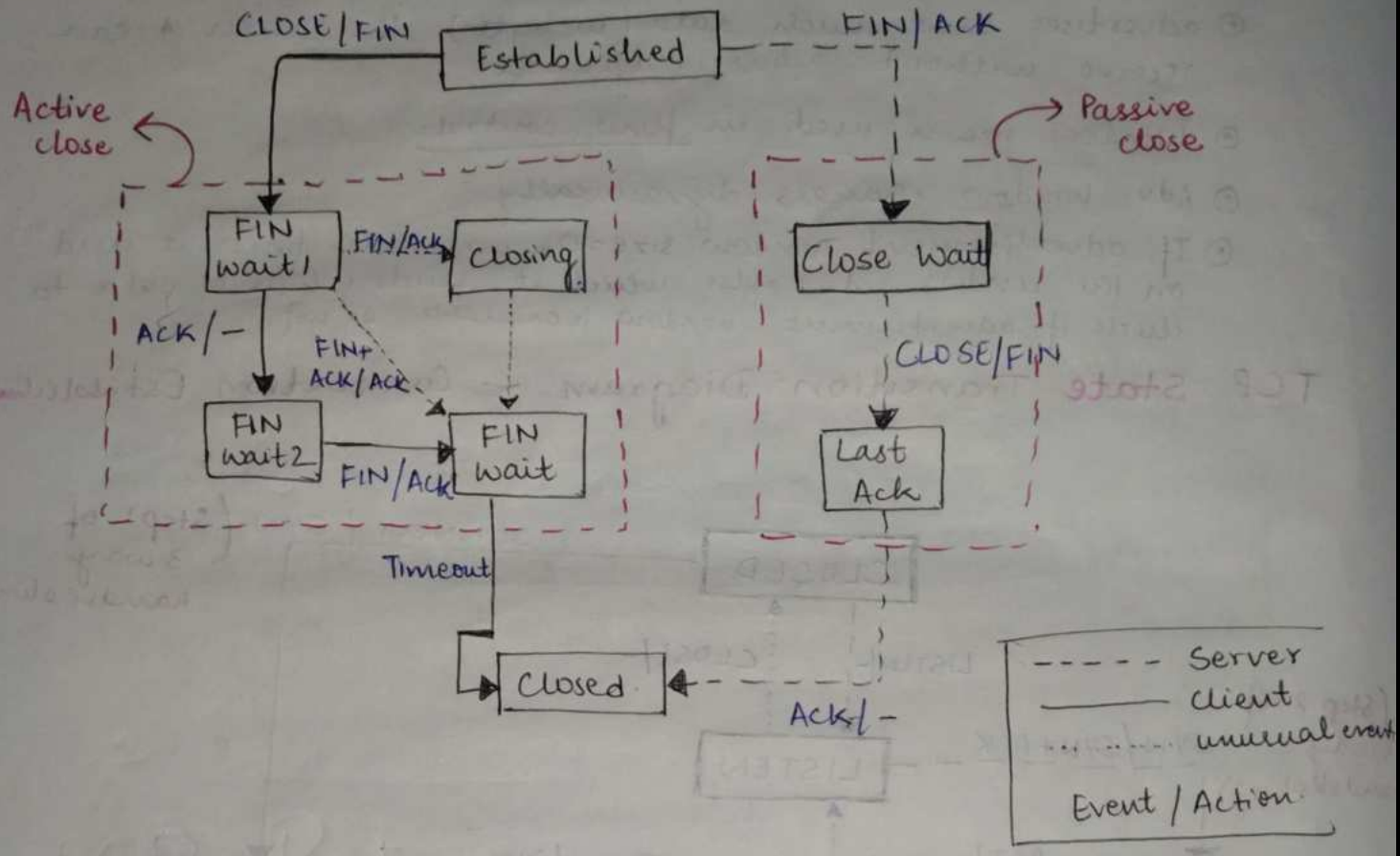


- – – – – server
- ..... unusual events
- _____ client.

(Event / Action pair)

Hamsamida

TCP State Transition Diagram – Connection Release.



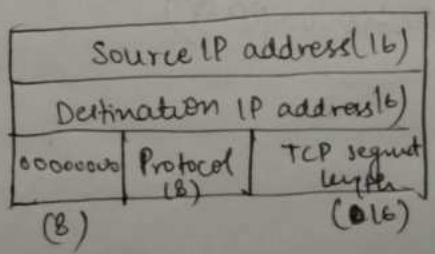
TCP Checksum

[16 bit field]

⊙ Takes care of ~~both~~ all TCP header, TCP data as well as pseudo IP header.

From IP header, checksum is calculated only on the fields in pseudo IP header. Because other fields in IPv4 ~~add~~ header change when packet is received by the receiver.

Pseudo IP Header.



Why IP header ~~check~~ is ~~a~~ included in ~~calculated~~ calculating TCP checksum?

In order to have double checking.

To make sure that the ~~so~~ message always reaches correct destination.

Options

0 – 40 bytes.

Options field is used for following purposes –

1. Time stamp
2. Window size extention
3. Parameter negotiation
4. Padding.

Time stamp –

When wrap around time is less than lifetime, timestamps are used.

Timestamp marks the age of packet.

For example,

if seq number = 000...0 (32 0's) & time stamp = 0, it is the first segment.

if seq number = 000...0 (32 0's) & time stamp = 1, it is 2^{32} th segment.

Window size negotiation

- Using window size field of TCP header, window size of only 16 bits can be represented.
- If the receiver wants to receive more data, it can advertise its greater window size using this field.
- The extra bits are appended in options field.

Parameter negotiation

During connection establishment, both sender and receiver have to specify their maximum segment size (MSS).

There is no such field for MSS.

∴ It is specified in options field.

Padding

Addition of dummy data to fill up unused space in the transmission unit and make it conform to the standard size is called padding.

Header length field should be multiple of 4

TCP Retransmission

⊙ Retransmitting of a segment that is lost before reaching the receiver is called retransmission.

⊙ When sender discovers that the segment sent by it is lost, it retransmits the same segment to the receiver.

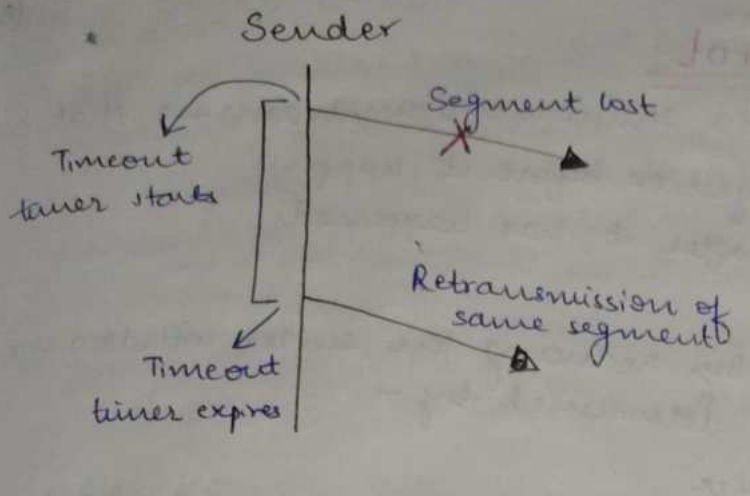
TCP segment is lost when —

- Either TimeOut Timer expires
- Or it receives 3 duplicate acknowledgements.

Retransmission after timeout expiry —

Each time sender transmits a TCP segment to the receiver, it starts TimeOut Timer.

If sender does not receive acknowledgement for the sent segment before the timer goes off, it retransmits the same segment to the receiver & resets the timer.



Retransmission after receiving 3 duplicate acknowledgements.

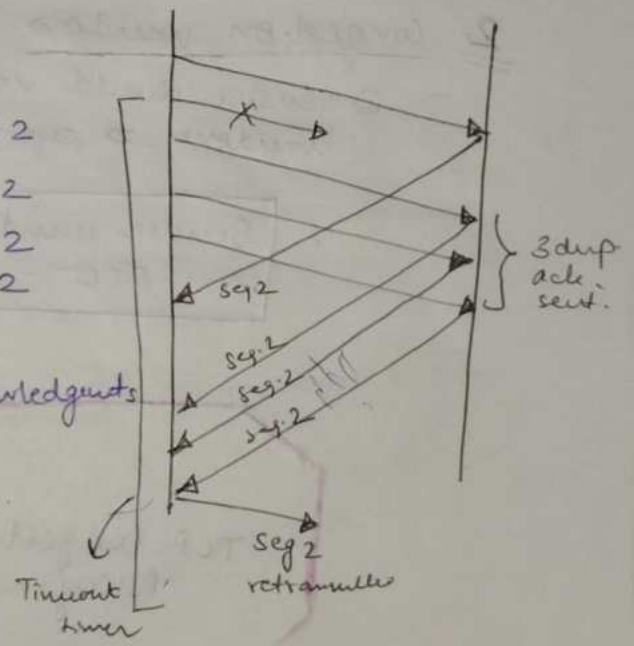
Sender assumes that a packet has been lost if it gets 3 duplicate acknowledgements for a TCP segment sent by it.

This is known as Early retransmission or Fast Retransmission.

Sender sends 5 TCP segments.
2nd segment gets lost.

Segment 1 sends ack for segment 2
Segment 3 sends ack for " 2
Segment 4 " " " " 2
Segment 5 " " " " 2

Sender receives 3 duplicate acknowledgments in total.
∴ it assumes segment 2 is lost.



TCP Congestion Control

Congestion control refers to the techniques & mechanisms that can
- Either prevent the congestion before it happens
- Or remove congestion after it has happened.

TCP reacts to congestion by reducing the sender window size.
Sender window size is determined by -
- → Receiver window size
- → Congestion window size

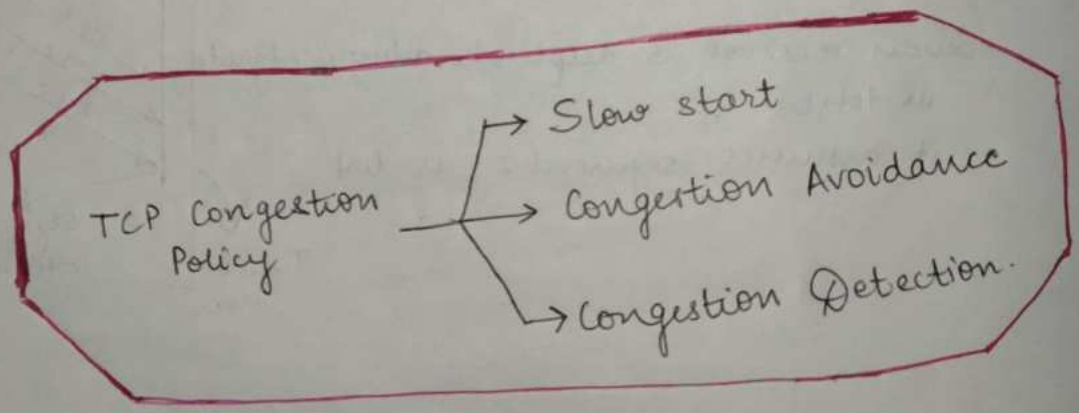
1. Receiver window size -

- Advertisement window.
- Sender should not send data greater than receiver window size.
- Receiver dictates its window size through TCP header.

2. Congestion window -

- Sender should not send data greater than congestion window size.

$$\therefore \text{Sender window size} = \text{Minimum}\left(\text{Receiver window size}, \text{Congestion window size}\right)$$



Slow start phase

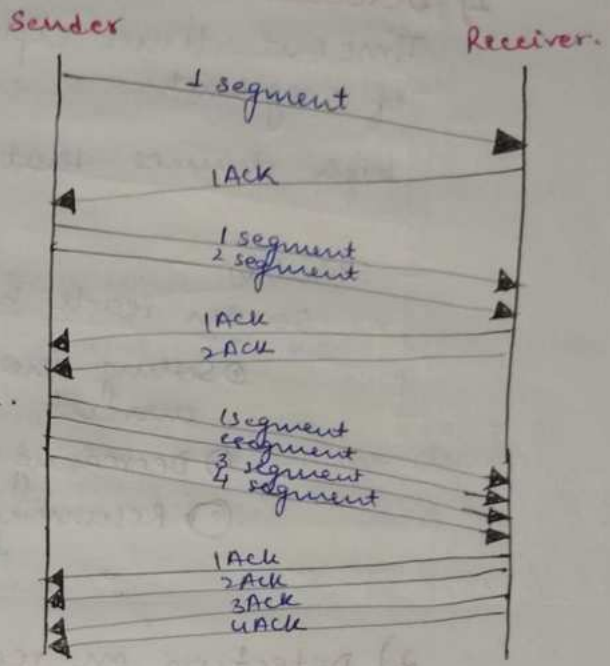
Initially, sender sets congestion window size = Maximum segment size (MSS).

The congestion window size is increased exponentially when acknowledgements are received.

After 1 round trip,
congestion window size = $(2)^1 = 2$ MSS

After 2 round trip,
congestion window size = $(2)^2 = 4$ MSS

After 3 round trips,
congestion window size = $(2)^3 = 8$ MSS ...



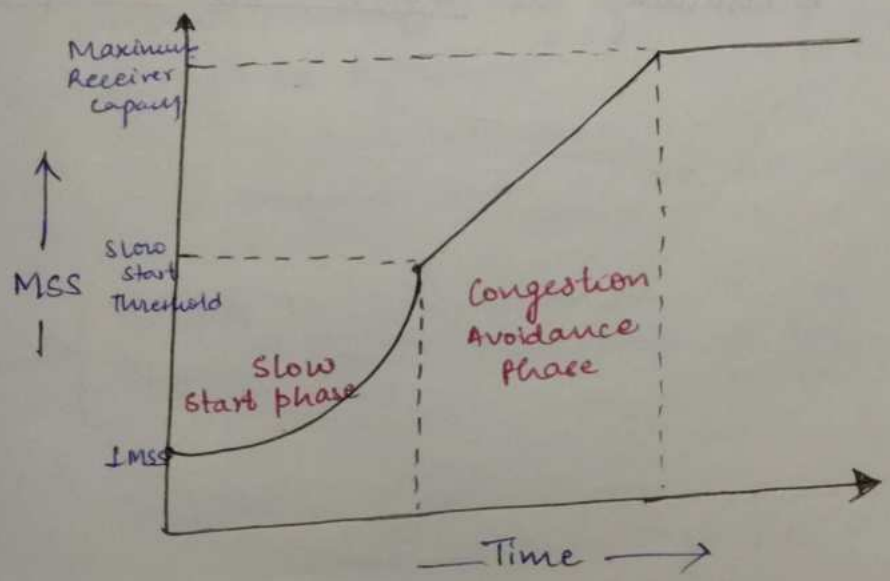
This phase continues until the congestion window size reaches slow start threshold.

Threshold = $\frac{\text{Receiver window size}}{\text{MSS} * 2}$

Congestion Avoidance Phase

After reaching threshold,

- Sender increases congestion window size linearly to avoid the congestion.
- On receiving acknowledgement, sender increments the congestion window size by 1.



Congestion Detection phase

When sender detects the loss of segments, it reacts in different ways depending upon how the loss is detected.

1) Detection on Timeout

Timeout timer expires before receiving the acknowledgement of a segment.

High chances that it was due to congestion in n/w.

Solution –

Sender reacts by –

- Setting the slow start threshold to half the current window size.
- Decreasing CWS to 1MSS.
- Resuming slow start phase.

2) Detection on receiving 3 Duplicate Acknowledgements

- Sender receives 3 duplicate acknowledgements for a segment.
- weak possibility that it was due to congestion in the n/w

Solution –

Sender reacts by –

- Setting the slow start threshold to half the current congestion window size.
- Decreasing CWS to slow start threshold.
- Resuming the congestion avoidance phase

TCP Timer Management

1) Time wait timer –

- Time wait timer is used for connection termination.
- Time wait timer = 2* lifetime.
- Started when sender sends the second FIN segment

2) Keep Alive Timer

- used to terminate long idle TCP connections.
- Each time server hears from client, keep alive timer is reset.
- After it expires, 10 probe segments are sent to the client.
- If no response is received, connection is terminated.

3) Acknowledgement timer

used for cumulative acknowledgement + piggy back acknowledgement.

4) Persistent timer

TCP uses persistent timer to deal with zero window size deadlock situation.

5) Timeout timer

Time for which sender should wait for acknowledgement without retransmitting the packet.

should be dynamic

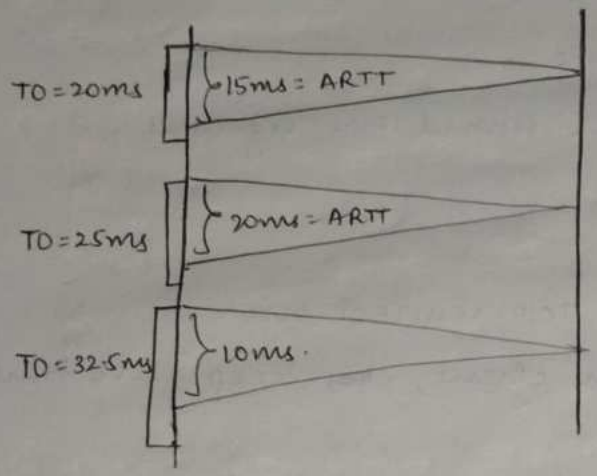
Timeout timer = 2* Round Trip Time.

Problem
- Timeout timer is too small → unnecessary retransmission
- Timeout timer is too large → large waiting time if packet is lost.

Solutions → Basic algorithm
↳ Jacobson's Algorithm

Basic Algorithm for Timeout Timer computation

⊙ Timeout timer is calculated dynamically.



IRTT = 10ms (guessing)
TO = 2 * IRTT = 20ms
ARTT = 15ms

NRTT = α IRTT + $(1-\alpha)$ ARTT

α = Smoothening factor
(will be given in ques)
$0 \le \alpha \le 1$ $\alpha = 0.5$ (Avg)

NRTT = 0.5×10 + 0.5×15
= 12.5 ms.

⇒ TO = 25ms (for next packet)

⇒ This algoritem is dynamic in nature.

⇒ Disadvantage — TO = (2) * RTT
↓
No logic why 2 is used.

- - - - - - - - - - - -

For next packet,
IRTT = 12.5ms
TO = 25ms
ARTT = 20ms
NRTT = 0.5 × 12.5 + 0.5 × 20
= 16.25 ms

IRTT =	Initial Round Trip Time
ARTT =	Actual Round Trip Time
NRTT =	Next Round Trip Time (expected)

Jacobson's Algorithm for Timeout Timer Computation

Similar to ~~same~~ basic.

ID = Initial Deviation
AD = Actual Deviation

TO = 4 * ID + IRTT

AD = | IRTT − ARTT |

NRTT = α IRTT + $(1-\alpha)$ ARTT

ND = α ID + $(1-\alpha)$ AD

Karn's modification says that if we don't receive the acknowledgement of the packet within timeout, keep doubling the timeout timer for the next packets.

Need For UDP [User Datagram Protocol]

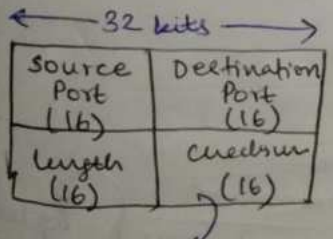
TCP is disadvantageous for some situations —

1) When the application needs just 1 request and 1 reply.
   Ex:- DNS, BOOTP, DHCP, NTP (Network Time Protocol), NNP (Network News Protocol), TFTP (Trivial File Transfer Protocol), RIP, OSPF

2) For broadcasting & multicasting, buffers are needed to be allocated in TCP for each host. This leads to inefficiency.
   ∴ UDP is used for broadcasting / multicasting.

3) When we require speed than reliability.
   Ex→ watching a youtube video in HD format → TCP
   [if 1 frame is lost, video pauses]
   but when video is not in HD, some frames may be lost but video will not be clear. not pause. Also, quality of video may drop due to loss of frames.

UDP Header



computed on UDP header, data, pseudo IP header.

8 bytes of header

- UDP should communicate the options to the IP Datagram. The various options are
  1) Trace Route
  2) Record Route
  3) Time stamp.

- Additional responsibility of UDP is to inform application layer about ICMP error packets.

Application Layer protocols

Application layer is responsible for all the services that internet provides.

Application layer protocols are —

DNS → DN→IP , HTTP → Web pages , FTP → Files , SMTP → Email , POP → Email Retreiving.

1) DNS (Domain Name Service)

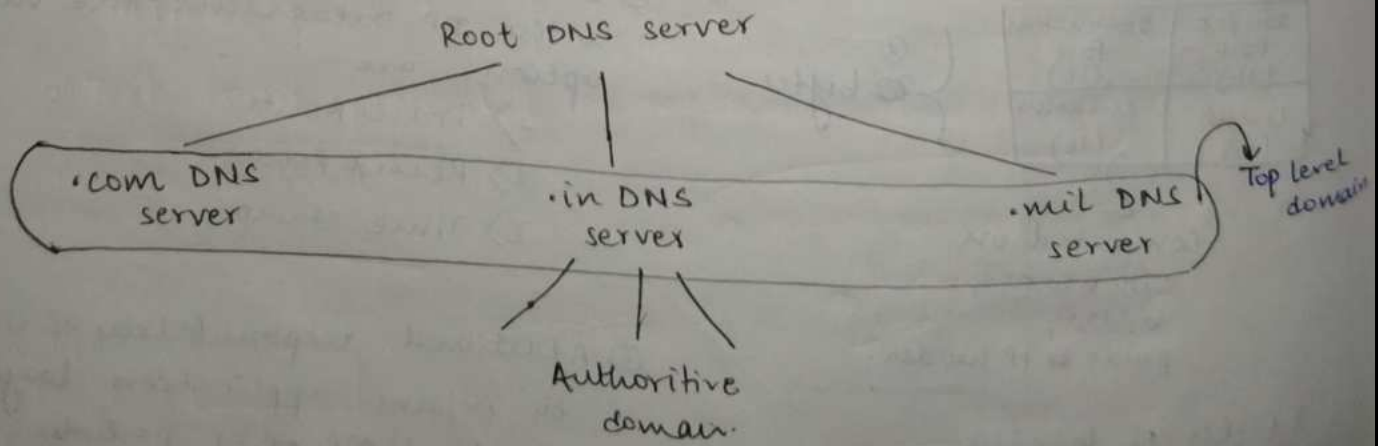
(Port number : 53)

- DNS uses UDP at transport layer
- used to convert Domain name to IP address.

- Types of domains —
  - Generic domain (.com, .edu, .mil, .org, .net)
  - Country domain (.in, .us, .uk)
  - Inverse domain (Given IP address → Find domain name)

- DNS is also used for load balancing

DNS database is organized in hierarchial manner.

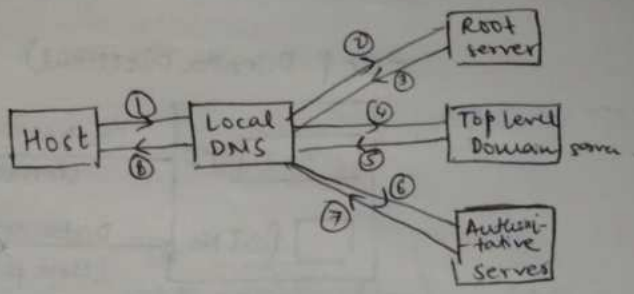


⊙ Internet Engines Task Force (IETF) manages 13 root servers across the world to deal with failures.

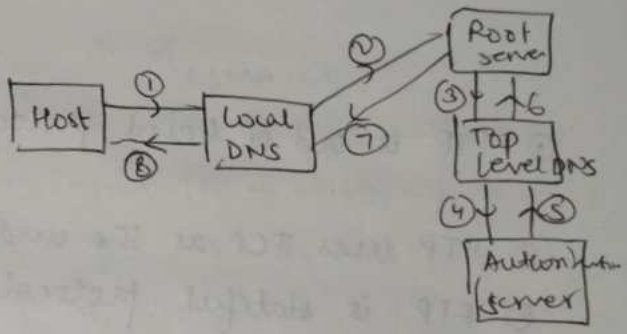
⊙ Local DNS is used to prevent DNS overhead.
It stores freq. used domain names & IP addresses for a time duration

Local DNS contacts Root server in 2 ways —

1) Iterative way →



2) Recursive way.



2) HTTP (Hyper Text Transfer Protocol)

⊙ HTTP always needs reliability (it does not implement it by itself)
⊙ HTTP uses TCP at transport layer.
⊙ HTTP is in band protocol (both commands & data go in 1 connection).
⊙ HTTP is stateless protocol (user activity is not tracked on server side)

2 versions — HTTP 1.0 (non persistant connection)
1 connection is used for each object.

HTTP 2.1 (persistent connection)
1 connection for all objects

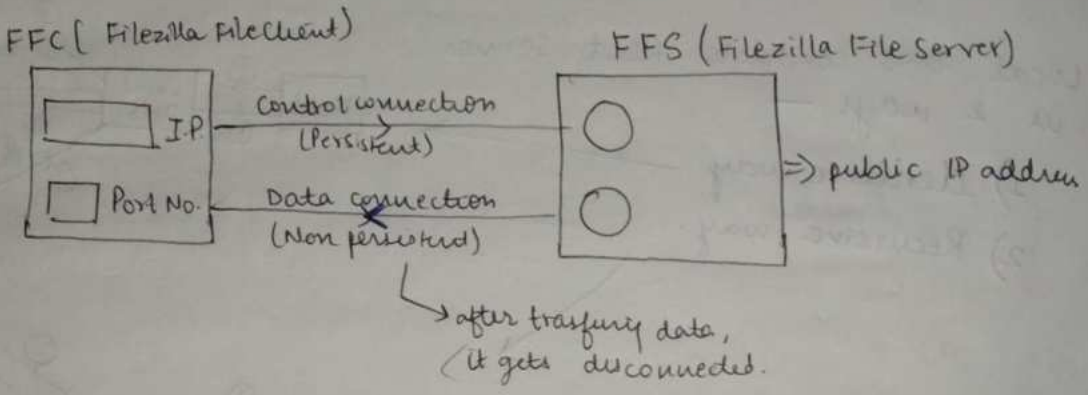
Methods supported by HTTP

1. Head → returns header (metadata) of a webpage
2. Get
3. Post } used with forms
4. Put → to upload an object.
5. Delete → to delete the object
6. Trace → Trace the servers which are sending data.
7. options
8. Connect → used by https (for authentication)

3) File ~~and~~ Transfer Protocol (FTP)

Port Number = 21

- used to transfer FTP.
- Tectia, Filezilla are popular FTPs



- FTP is out of band protocol (commands & data flow through different connections)
- FTP uses TCP as the underlying protocol.
- FTP is stateful protocol.

4) SMTP (Simple Mail Transfer Protocol)

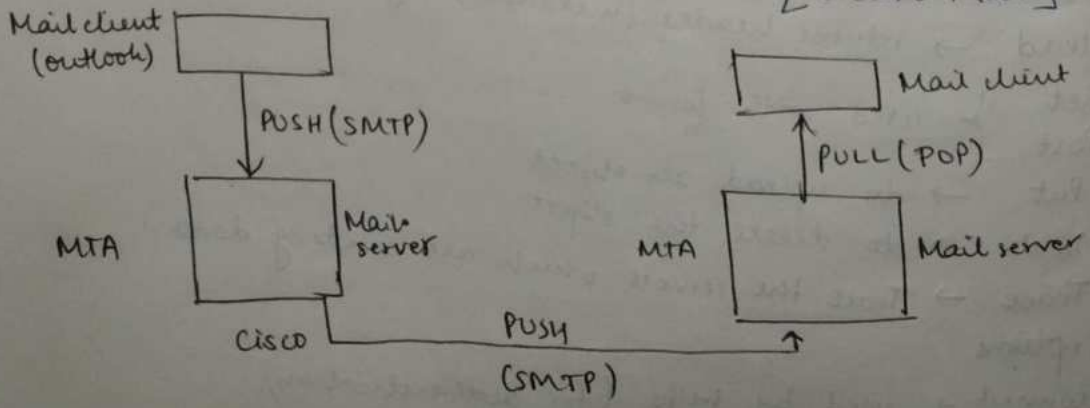
- Emails are transferred using SMTP.
- FTP cannot be used for emails because to transfer files using FTP, both server and client should be online.

TMTA = Mail Transfer Agent

- Text based protocol (For multimedia transfer, MIME software used)

SMTP & POP are in band protocol

↳ Multipurpose Internet Mail Extension
{Text → Non-text, Non text → text}



For any given situation, efficiency of Token Ring is always greater than CSMA/CD

$$\eta_{CSMA/CD} = \frac{1}{1+6.44A} \quad (N \to \infty)$$

$$\eta_{TR-E} = \frac{1}{1+a/N} \qquad \lim_{N\to\infty} \eta_{TR-E} = \lim_{N\to\infty} \frac{1}{1+a/N} = 100\%.$$

$$\eta_{TR-D} = \frac{1}{1+a(1+1/N)} \qquad \lim_{N\to\infty} \eta_{TR-D} = \lim_{N\to\infty} \frac{1}{1+a(1+1/N)} = \frac{1}{1+a}$$

No need to solve numerical if comparison is req!

$$\therefore \quad \underbrace{\eta_{TR-E}}_{\frac{1}{1+\left(\frac{N+1}{N}\right)a}} > \underbrace{\eta_{TR-D}}_{\frac{1}{1+\frac{a}{N}}} > \underbrace{\eta_{CSMA-CD}}_{\frac{1}{1+6.44a}}$$

Hardwares in computer network

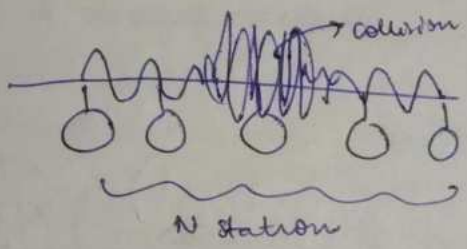
① Host has 5 layers — AL TL NL DLL PL.

② Cables can be

10 base T (10mbps, No multiplexing, 100m)
10 base 2 (" " , 200m)
10 base 5 (" " , 500m)

↓
length of LAN segment.

→ ~~Dpl~~ Operate at physical layer. (completely hardware)
→ Problem – attenuation (signal strength reduces)
→ Collisions are possible



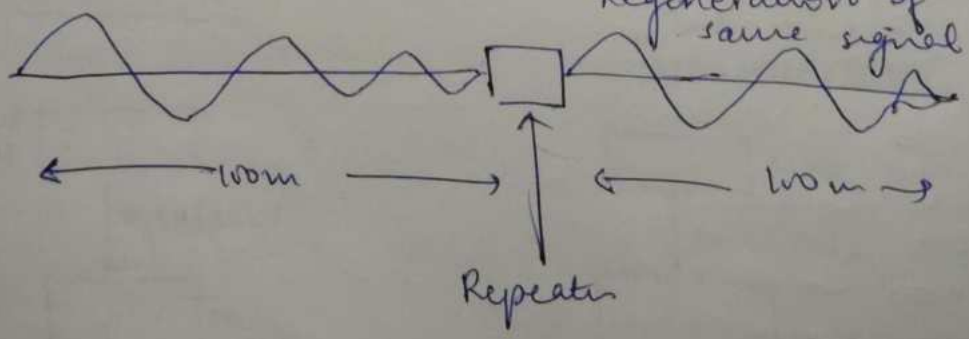
Collision Domain = N

Repeaters –

used to increase the length of LAN segment.

For example – 10 base T cables are used.
max. distance upto which signal is alive is 100m.
∴ Repeaters are used to transmit signal to greater distances.

LAN spans for 200m



→ used to connect 2 LAN segments of similar type.
→ Physical layer.
→ Collisions are possible.

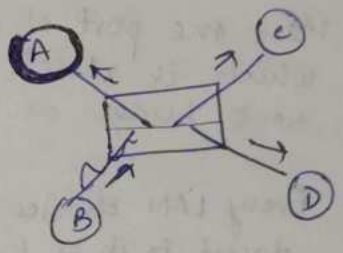
Collision Domain = n [Collision Domain is unaffected]

If we cut a wire and introduce repeater to connect them, no change in collision domain.

→ Range of LAN is increased.

Hub

Multiport repeater is called hub.

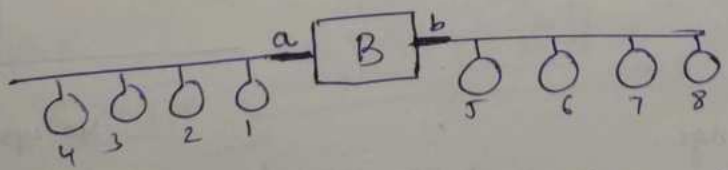


→ Traffic is very high
→ works purely at physical layer.
→ Collisions are possible.
Collision domain = N

Cost is less.

~~Brigo~~ Bridge

used to connect LAN segments of different types. (PL, DLL)



2 types

Static
manually filled.

MAC	Port
1	a
2	a
3	a
4	a
5	b
6	b
7	b
8	b

Dynamic/ Learning/ Transparent.
When a station sends message, its MAC & port are noted.
Time taking but dynamic.

a) Filtering
b) Forwarding
c) Flooding

able to store and forward
∴ no collision inside bridge.

Collision domain = n/2

Avoiding infinite loops in ethernet at I-P → Time to live field is used.

Avoiding infinite loops in n/w containing multiple bridges → Minimum spanning tree al

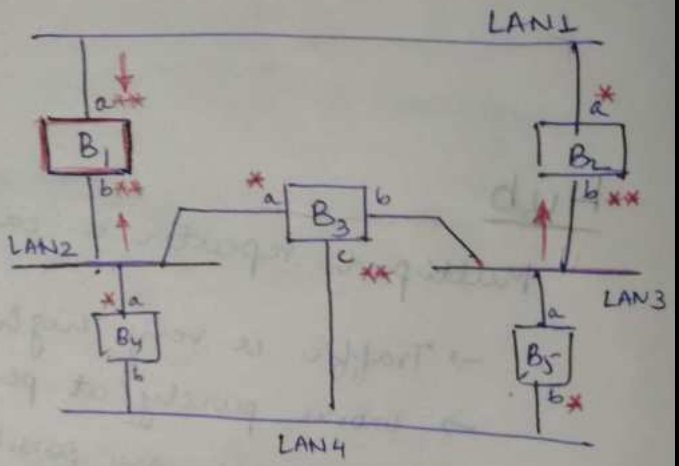
Spanning Tree Algorithm

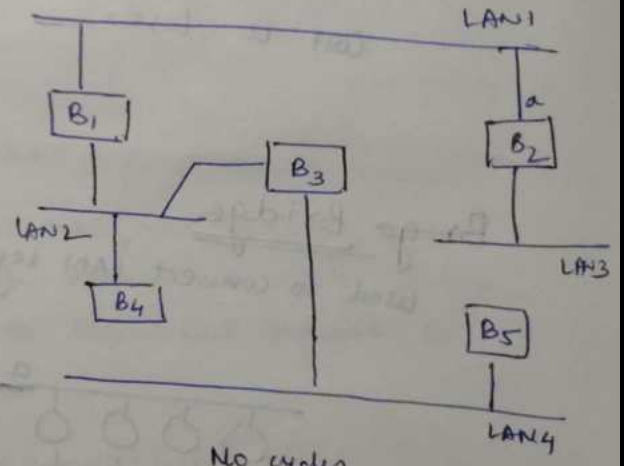
Every bridge has a built a ID.
The one with smallest ID is taken as root bridge.

Mark one port of each bridge which is closest to the root bridge as the root port(*)

Every LAN chooses the bridge closest to it as the designated bridge for that LAN. Make the corresponding port as designated port (**).

Make the root ports and designated ports as forwarding ports and block the remaining.

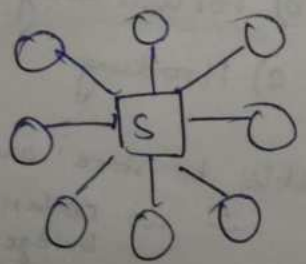




No cycles

Switch

Multiport bridge.
It connects hosts instead of LAN segment.
Collision domain is 0.



Traffic is less.

It is costly.

Every one can communicate at the same time.

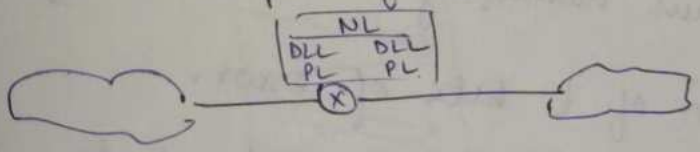
Device	Broadcast Domain	Collision Domain
Repeater	Same	Same
Hub	Same	Same
Bridge	Same	Reduces
Switch	Same	Reduces (0)
Routers	Reduces	Reduces
Gateways	Reduces	Reduces

Routers

Connecting 2 different networks

Every Router has NL, DLL and PL.

Each port of router has DLL & PL.

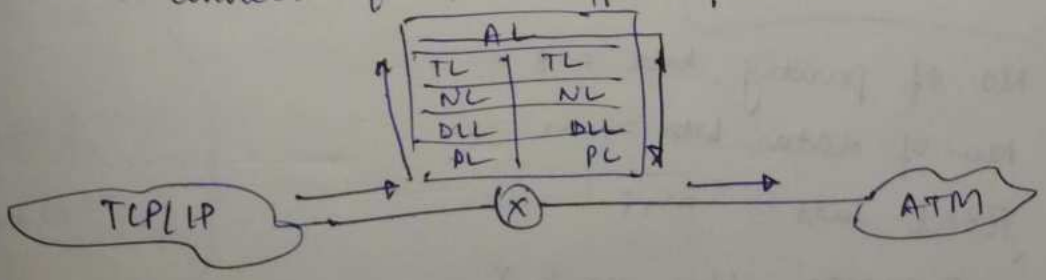


No collision inside routers

Gateways

protocol converter.

Used to connect n/w with diffent protocols.



Used for proxy

Application layer — Application & Gateway

Transport layer — Transport gateway

N/w layer — Router

DLL — Bridge, Switch

Physical — Repeater, hub

Hamming code

For detection of ~~t~~ bit error,

minimum hamming distance $d_{min} = t+1$

For correction of t bits of error,

minimum hamming distance $d_{min} = 2t+1$

No. of parity bits = r

No. of data bits = m

Total bits = m+r

Relation b/w m & r

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

Congestion ~~control~~ detection phase

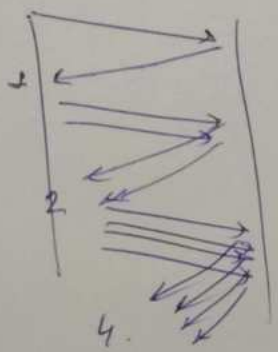
Slow start phase

- ⊙ cwnd increases by 1 MSS on every successful acknowledgement.
- ① cwnd doubles after every round trip (set of acknowledgements are reached).

Congestion avoidance phase

- ⊙ cwnd increases by 1 MSS ~~on every successful~~ after every RTT.
- ① cwnd increases by 1/cwnd after every successful acknowledgement.

If cwnd is window size, then cwnd acknowledgements will arrive in 1 RTT



Slow start Threshold = Receiver window size / 2x MSS

TCP leaky bucket & Token bucket

"Life was never meant to turn into thiss!"

COMPUTER NETWORKS

1. Forouzan Data Communications & Networking

2. Kurose Ross Computer Networking: A Top-Down Approach.

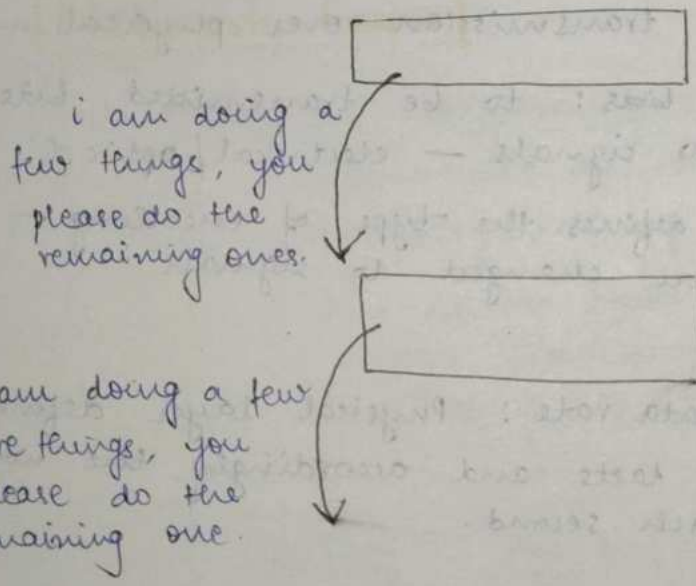
3. Tanenbaum Computer Networks.

Why Layering?

1. **Modularity** – one problem is decomposed into a number of smaller more manageable subproblems.
   ⇒ more flexibility in designing, modifying and evolving computer networks.

2. **Functional Reuse** – a common functionality of lower layer can be shared by many upper layers.

⊙ Lower layer provides services to upper layers.



This is how the layers would communicate if they could actually TALK!

Protocol

⊙ An agreed upon convention for communication
   ↳ both endpoints need to understand the protocol.

⊙ Protocols must be formally defined and unambiguous.

Layering and Headers

↳ Each layer needs to add some control information to the data in order to do its job.

Header [↳ This data is typically ~~prependend~~ prepended to the data before being given to the lower layer.

↳ Once the lower layers deliver the data and control information — the peer layer uses the control inform

Physical Layer

⊙ coordinates bit stream transmission over physical mediam

↳ representation of bits: to be transmitted, bits must be encoded into signals — electrical/optical.
Physical layer defines the type of encoding — how 0s and 1s are changed to signals.

↳ bit length — data rate: Physical layer defines how long a bit lasts and accordingly the no. of bits sent each second.

Data Link Layer

⊙ framing: The D.L.L divides the stream of bits received from the n/w layer into manageable data units called frames.

⊙ physical addressing : The D.L.L adds a header to the frame to specify the NIC address of appropriate receiver on the other side.

MAC address (48 bits hardcoded into NIC)

⊙ error control : The D.L.L adds reliability to the physical layer by adding a trailor of information necessary to detect / recover damaged / lost frames.

⊙ access control : when 2 or more devices are connected to the same link, the DLL determin which device has control over the link at any given time.

Network Layer

⊙ logical addressing : The physical addressing implemented by the data link layer handles the addressing / delivery problem locally.
If a packet passes through the n/w boundary another addressing system is needed to help distinguish b/w source & destination n/w.

⊙ routing : The N.L provides the mechanism for routing / switching packets to their final destinati

⊙ fragmentation and reassembly :
The N.L sends messages down to the D.L.L for transmission. Some D.L.L technologies have limits on the length of messages that can be sent. If the packet that the N.L wants to send is too large, the N.L must split the packet up, send each piece to D.L.L and then have pieces reassembled once they arrive o

Routing v/s Forwarding

Routing is the process of finding the routes

↳ Routing protocols can run even if no packets need to be sent. Keeps the table ready to forward a packet.

Forwarding is the process of using a route to send a packet towards the destination.

↳ Done only when a packet arrives at the node whose final destination is not this node.

↳ Done in real time.

Transport Layer

⊙ port addressing – Computers often run several processes at the same time. For this reason, process to process delivery means delivery not only from one computer to other but also from specific process to other.

∴ Transport layer address includes a type of address called port address.

⊙ segmentation and reassembly –

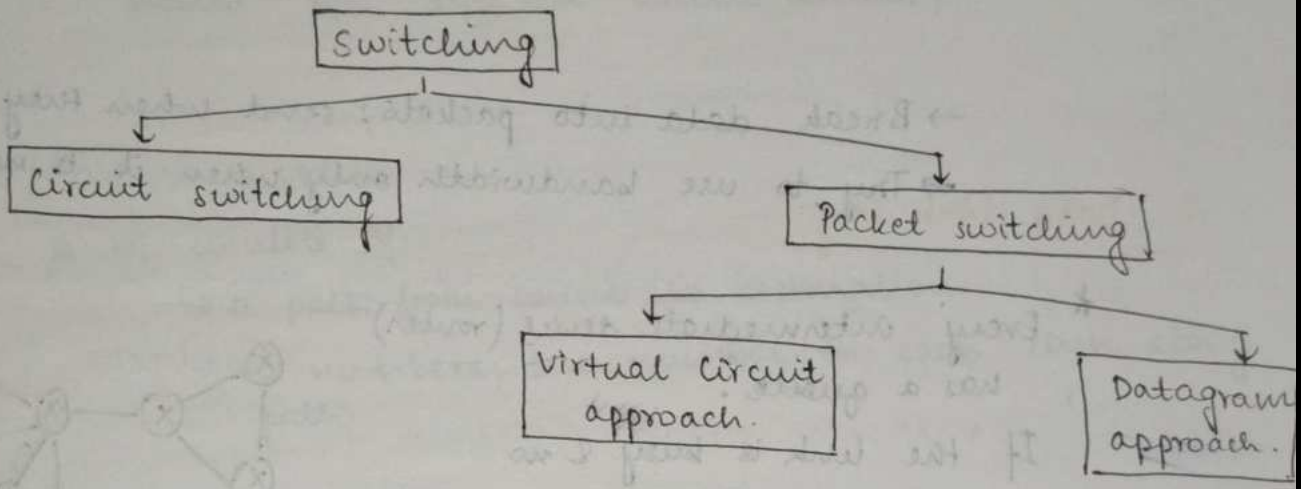
A message is divided into segments, each segment containing a sequence no. These nos enable the transport layer to reassemble msg correctly & to identify & replace packets that are lost

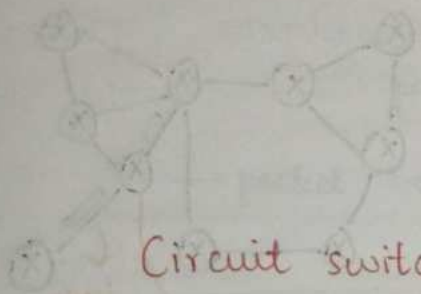
⊙ flow and error control –

Flow & error control in this layer are performed end to end rather [illegible]

Switching

Long distance transmission is typically done over a n/w of switched nodes





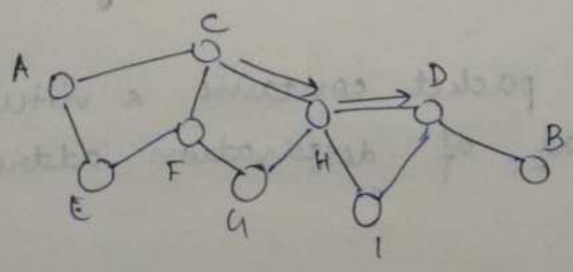
Circuit switching –

- ⊙ data sent continuously → data is not broken into parts.
- ⊙ creating a session reserves dedicated bandwidth in series of switches b/w caller and ~~receip~~ recipient
- ⊙ Guaranteed capacity (in both directions) so long as session up.

★ → Before sending, must reserve the capacity for session.

→ Success = bandwidth is guaranteed for life of session.

 → often the links may be busy.



A has to communicate to B.
C & D are already communicating
A has the dedicated path A → C → D → B which is blocked.
So, A cannot communicate even through other path A → E → F → G → H → I → D → B

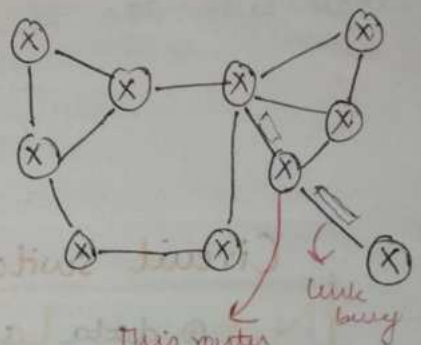
Packet switching

- ⊙ Link is shared
- ⊙ To each frame, a little bit of information (header) is added which tells the switch how to forward it

→ Break data into packets; send when they are ready
→ Try to use bandwidth only when it is needed.

* Every intermediate device (router) has a queue.

If the link is busy & no alternate link is there, packets are stored in queue and transmitted when link is available.



→ If queue itself is full, drop the packet.

*

Virtual circuit approach

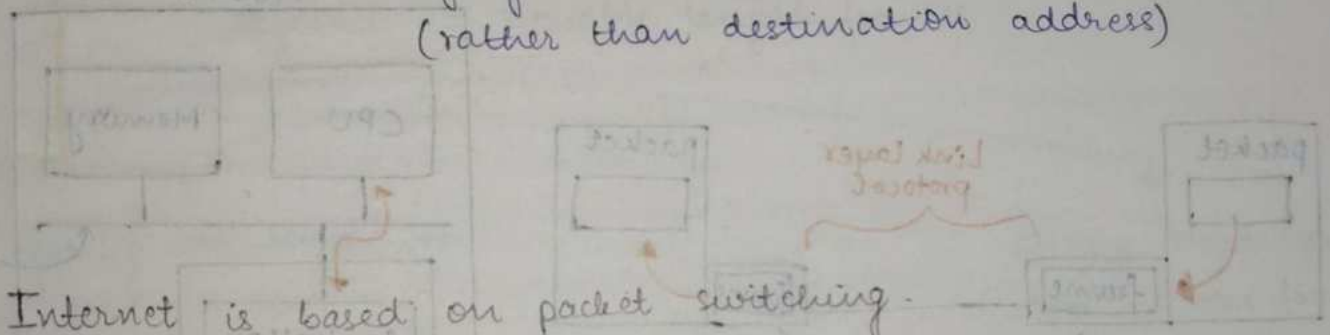
- ⊙ Preplanned route established before any packet is sent
- ⊙ Call request and call accept packets establish connection (handshaking)
- ⊙ Each packet contains a virtual circuit identifier instead of destination address

- ⊙ No routing decisions required for individual packets.
- ⊙ Clear request packet is used to drop circuit.
- ⊙ Not a dedicated path (links in a path may be shared b/w different virtual circuits).

VC implementation –

a VC consists of
- ↳ a path from source to destination
- ↳ VC numbers, one number for each link along path.

→ packet belonging to VC carries VC number (rather than destination address)



Internet is based on packet switching.

Internet applications are bursty
- ↳ email only consumes bandwidth when mail is sent or received.
- ↳ web browsing only consumes bandwidth when you visit site.

⊙ Packet switching (datagram) is much more efficient way to support bursty applications.

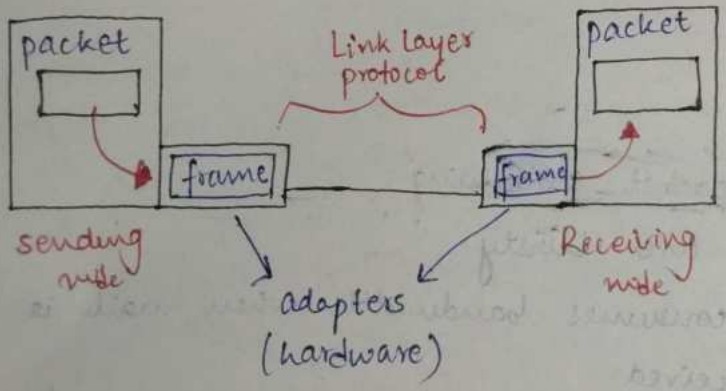
1. Physical Layer

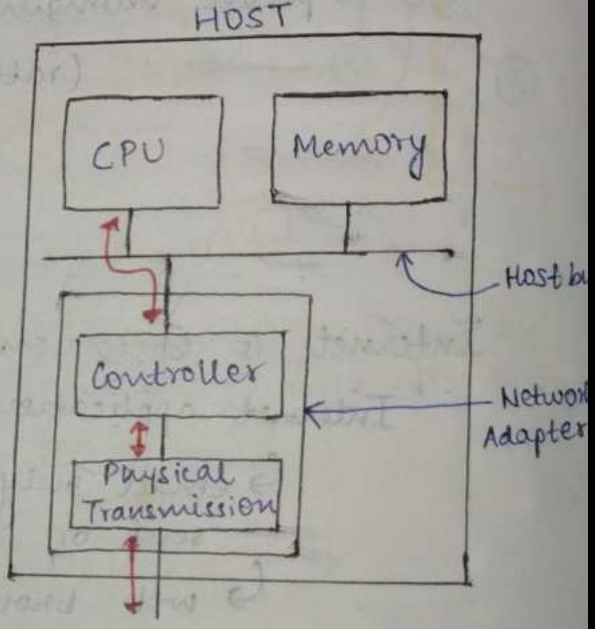
Scope → concerned with how signals are used to transfer message bits over a link.

2. Data Link Layer —

Link layer is implemented by using hardware

- Link layer is implemented in 'adapter' (NIC)
- Adapter receives datagram from nw layer and creates frames.





1. Framing

The data stream is broken up into discrete chunks.

Receiver needs to know when a frame starts and ends.
otherwise errors from misinterpretation of data stream.

Approach1: Time gaps.

(send next frame only after a certain time interval)

network may squeeze out the gaps during transmission.

Approach2: Fixed frame size

We might be restricting ourselves for smaller/larger frame.

what should be the optimal frame size?

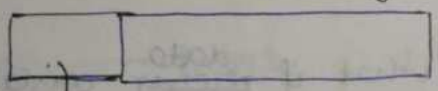
→ large frame size ⇒ wastage of resources if small amt of data is there

→ small frame size ⇒ large no of frames.

Approach 3: Variable length frames

specify the length of frame in the very beginning

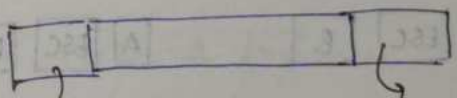
(Byte count)



length of frame.

count field may get currupted

specify start and end of the frame explicitly.

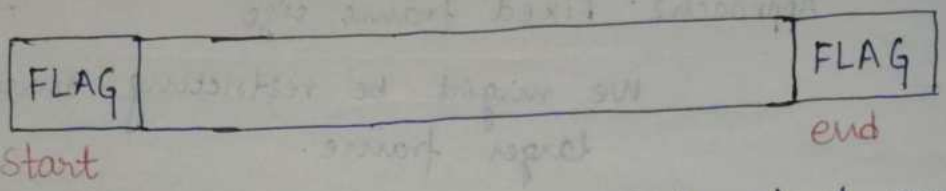


Byte stuffing

Bit stuffing

Byte stuffing

Add 1 byte of flag at the start and end of the frame.



Two consecutive flags indicate start and end of frame.

→ what if FLAG itself is there in the data (afterall, its just a sequence of bits. RIGHT?)

Solution →

use escape character if FLAG is present in data. #FLAG.

ORIGINAL BYTES	AFTER STUFFING
A FLAG B	A ESC FLAG B
A ESC B	A ESC ESC B
A ESC FLAG B	A ESC ESC ESC FLAG B
A ESC ESC B	A ESC ESC ESC ESC B

Rule at receiver end –
whenever # is present, ignore escape character and consider next byte as part of data.

→ what if ~~receiver~~ data contains escape character in its sequence?

Solution →

use one more escape character

FLAG __ ## FLAG ___ FLAG → treated as end FLAG.

__ #

~~## FLAG~~ –

FLAG __ ## ___ FLAG

→ __ # ___

ESC, FLAG → These take up 1 byte each time used.

Therefore byte stuffing is not optimal.

Bit stuffing

Simple algorithm —

01111110 used as flag

At transmitter → after every 5 consecutive ones, insert a zero.

At receiver → In a pattern of 111110 anywhere in the data, remove the trailing 0

TRANSMITTER		RECEIVER
01111110	→	011111010
1111100	→	11111000
11111	→	111110

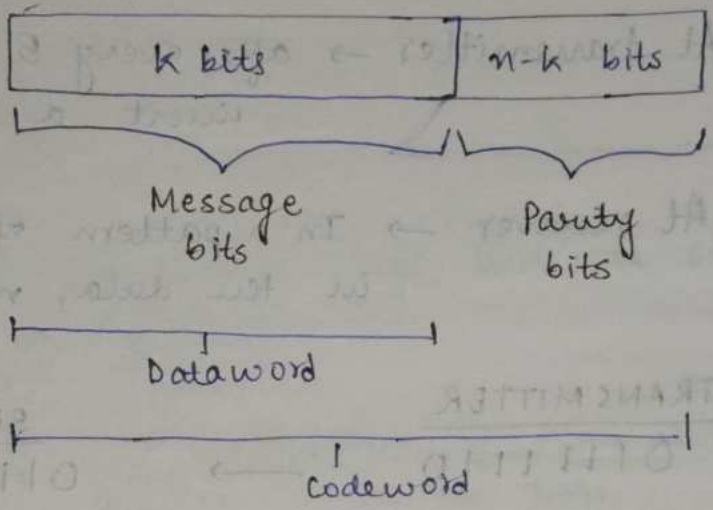
2. Error Handling

— Detect and/or correct errors in received frames.

Errors →
- Single bit error: only 1 bit gets corrupted. (0 0 0 0 [0] 0 1 0 → 0 0 0 0 [1] 0 1 0 corrupted.)
- Burst error: multiple bits get corrupted. ([0] 1 0 1 [1] [0] 0 → [1] 1 0 1 [0] [1] 0)

Block coding

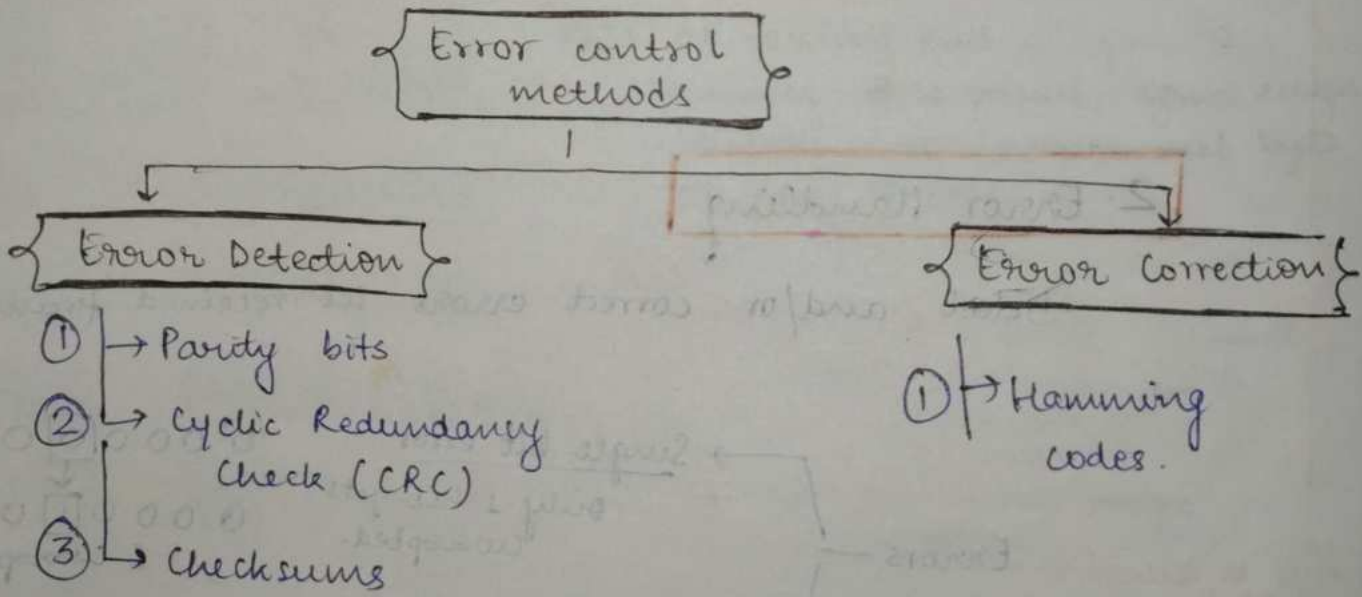
↳ a few bits are appended to the message



The bits can be appended anywhere

→ start ✓
→ middle ✓
→ end ✓

If there are k bits for dataword,
No. of datawords possible = 2^k.



Receiver always knows the list of all valid codewords.

Parity Checking

- simplest scheme
- add an extra bit to the data, such that the total number of 1 bits is even (even parity) or odd (odd parity)

→ Even parity :- make the no. of 1s in a bit string an even number.

commonly used

→ Odd parity :- make the no of 1s in a bit string an odd number

Check at the receiver.

Given a message of k data bits, $D_1, D_2, \ldots D_k$ append a parity bit P to make a codeword

→ P is exclusive OR of the data bits

$$P = D_1 \oplus D_2 \oplus D_3 \oplus \ldots\ldots \oplus D_k$$

→ Receiver can detect error only if the number of corrupted bits are odd.

Hamming distance

The hamming distance b/w 2 words (of same size) is the number of differences b/w the corresponding bits.

$d(000, 011) = 2$

$d(10101, 11110) = 3.$

$d(01101010, 11011011) = 4$

Minimum Hamming distance –

The minimum hamming distance is the smallest hamming distance b/w all possible pairs in a set of words.

$C = \{0000, 0011, 0101, 0110, 1001, 1010, 1100, 1111\}$

Minimum hamming distance = 2.

★ Minimum hamming distance in case of single parity codes → 2

$\boxed{d_{min} = 2}$

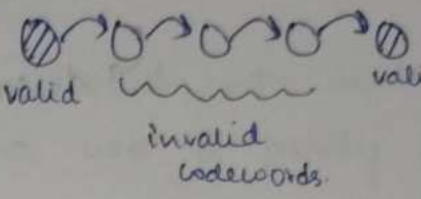
★ IMP

If the minimum Hamming distance b/w any 2 valid codewords is represented as 'd'. Maximum no. of bits of error that can be detected by the receiver = $d-1$.

To guarantee ~~detect~~ detection upto s errors, what should be the minimum hamming distance in the set of codewords?

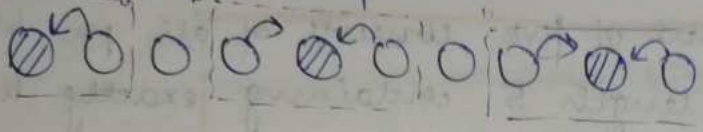
A. s
B. s-1
✓ C. s+1
D. None of these

Suppose s = 3.



This way, no set of s errors in a single bit could turn one valid codeword into some other valid codeword.

Only single bit error possible



if invalids are non overlapping, then, we can always correct errors.

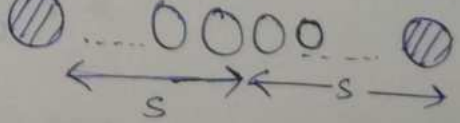
⊙ We can correct errors if we are really sure about the original codeword that caused it.

d_{min} = Minimum hamming distance

Maximum bits of error that the receiver can correct = $\frac{d_{min} - 1}{2}$

To guarantee CORRECTION upto s errors, what should be the minimum hamming distance in the set of codewords?

A. 2s
B. 2s-1
✓ C. 2s+1
D. None of these



$\therefore 2s+1 = d_{min}$

If minimum x invalid codewords in between,

↳ we can detect 'x' bits of error

↳ we can correct $\left\lfloor \frac{x}{2} \right\rfloor$ bits of error

↳ $d_{min} = x+1$.

Ques

The 'two out of five' consists of all possible binary words of length 5 containing exactly two 1s.

1. No. of codewords → _ _ _ _ _ $^5C_2$
2. Minimum hamming distance ⟶ 2
3. Errors that the code can detect → 1
4. Errors that the code can CORRECT → 0

Summary of Hamming distances —

↦ All errors of d or fewer bits in a codeword can be detected if and only if the minimum hamming distance of the code is $(d+1)$

↦ All errors of c or fewer bits in a codeword can be corrected if and only if the minimum hamming distance in the code is $(2c+1)$

↦ A code is a c error correcting AND d error detecting $(d \geq c)$ code if and only if its minimum hamming distance is $(d+c+1)$.

Single Error Correcting Code (SEC)

Designing code which can correct single bit error

Ex → Hamming code

Basic idea → Divide the dataword into multiple overlapping groups and use a parity for each group separately.

m → Data bits

r → parity bits.

It always satisfies the relation —

$$2^r \geq m + r + 1.$$

2^r → all possible combinations from r parity bits.

$m+r+1$ → no. of cases of error/no error
- ↦ error in 1 of m data bits
- ↦ error in 1 of r parity bits
- ↦ No error.

Let k be the minimum number of bits needed to correct all possible t bits error then, which of the following conditions must satisfy?

No. of parity bits = $\geq k$

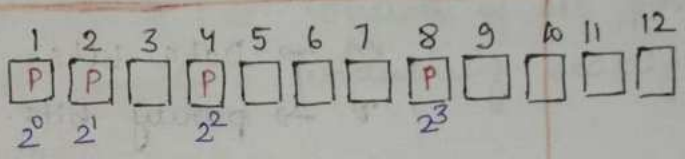
$$2^k \geq 1 + \binom{n}{1} + \binom{n}{2} + \binom{n}{3} + \ldots\ldots + \binom{n}{t}.$$

Hamming Code

→ Construction

① Number the bits starting from 1 : bit 1, 2, 3, 4, 5...

Step 1 :- Mark all bit positions that are powers of 2 as parity bits.
(positions 1, 2, 4, 8, 16, 32 ...)

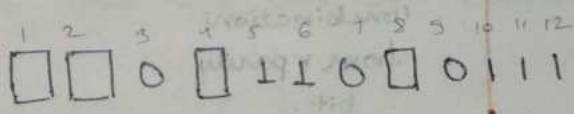


Hamming magic

⑥ Find the bad parity bits (parity bits that fail) and add their locations to find the location of corrupted data bit

Step 2 :- Insert data bits in remaining positions

Data to send = 01100111



Step 3 :- Calculate the Parity bits

Each parity bit calculates the parity of some of the bits in the codeword. The position of the parity bit determines the sequence of bits that it alternately checks & skips.

Position 1 :- 1, 3, 5, 7, 9 ....

Position 2 :- 2, 3, 6, 7, 10, 11, ...

Position 4 :- 4, 5, 6, 7, 12, 13, 14, 15 ....

∴ Codeword is

~~0 1 0 0 1~~

0 1 0 1 1 1 0 1 0 1 1 1

$P_1 = 3^{rd} \oplus 5^{th} \oplus 7^{th} + 9^{th} + 11^{th}$

$= 0 \oplus 1 \oplus 0 \oplus 0 \oplus 1 = 0$

$P_2 = 3^{rd} \oplus 6^{th} \oplus 7^{th} \oplus 10^{th} \oplus 11^{th}$

$= 0 \oplus 1 \oplus 0 \oplus 1 \oplus 1 = 1$

$P_4 = 5^{th} \oplus 6^{th} \oplus 7^{th} \oplus 12^{th}$

$= 1 \oplus 1 \oplus 0 \oplus 1 = 1$