

Problems associated with Computer Network:

- * communication
- * identification
- * connection.

1. communication: protocol is a language of computers needed for communication of the computers.

some of the protocol are:

- HTTP : web browser
- SMTP : Mail communication
- FTP : File communication
- NTP : Network time protocol

* Location of the protocol is NOS (Network operating system)

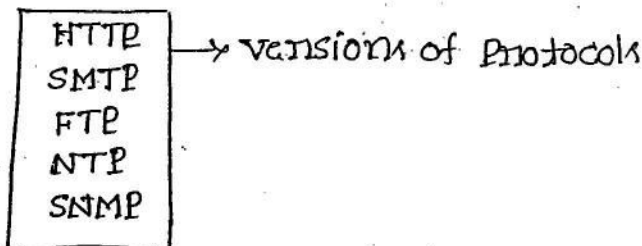
Network time protocol:

transactions are done by storing source time slot and converting to the destination time slot and mailing to the user.



Network Management protocol:

DOS + All protocols = window NT



HTTP: (Hypertext transfer protocol): Browser requesting for a web page

HTTP: 1.0, 1.1, 2.0 (Application protocols).

* RFC (Request for Comments) \Rightarrow standards for computer networks.

* concept of computer n/w have an RFC number

RFC₁

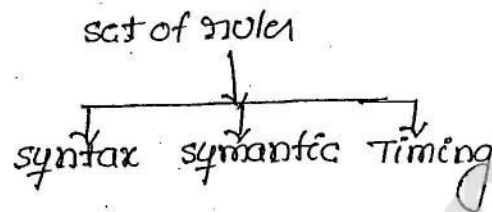
RFC₂

:

RFC 793 \rightarrow TCP/IP.

RFC 3700

Protocol: A set of rules and regulation or conventions

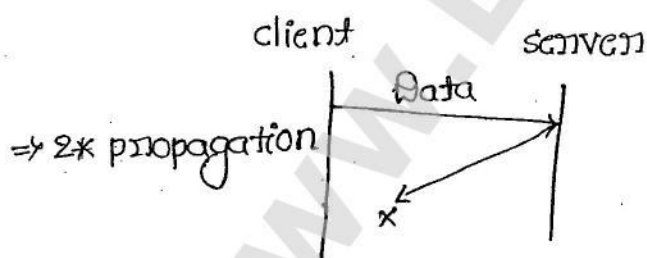


* Rules and regulations must be crystal clear i.e. there must not be any duplicates and any invalid syntax is not acceptable.

* Timing i.e. starting and ending a task must be mentioned.

* syntax \rightarrow send acknowledge

* semantic \rightarrow Receive acknowledge.



* RTT is measure of delay b/w 2 hosts

* Minimum acknowledge waiting time = 2 * propagation (RTT)

* Maximum ack. waiting time = 2 * (Min. ack. waiting time)

$$= 2 * (2 * \text{propagation})$$

$$= 2 * \text{RTT}$$

Round Trip Time \rightarrow RTT Turn over time \rightarrow Time out (ol)

* Protocol is an agreement between the communicating parties on communication is to proceed.

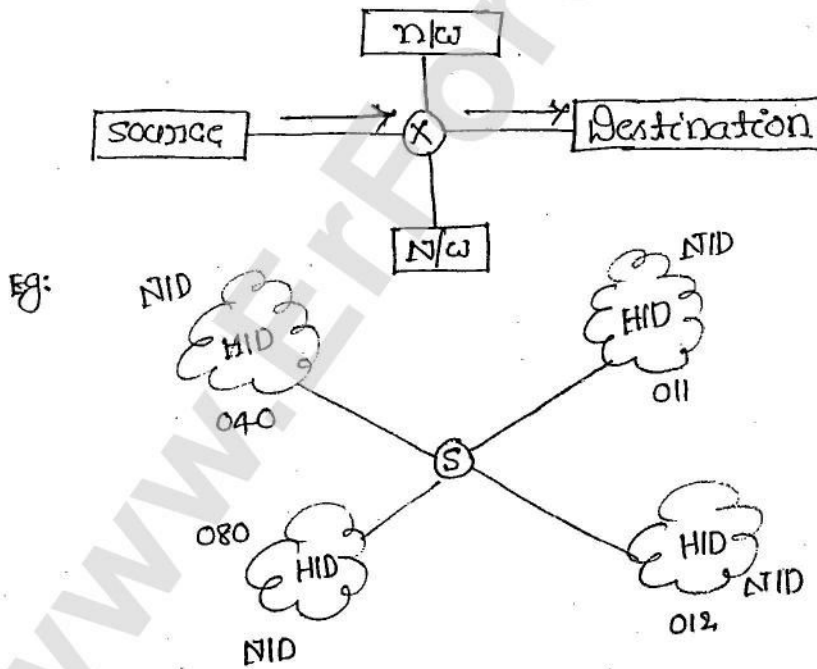
* IP address does not refer to a host actually, it really refers to interface, if a host is on two networks, it must have two IP.

* A system may have multiple IP addresses and multiple physical addresses.

2. Identification:

To send a packet from source to destination we have the identification steps:

- * identify the network → logical
- * identify the host within the network. i.e. among all physical ← the destination, one system is identified
- * identify the process within the host → service point



040 4752469
 3 7
 02145 39764
 5 5
 078914 0823
 6 4

- * Each num is 10 digits
- * Two paths.
- * Each num is unique
- * Hidden meaning.

1. 32-bit number
 8 8 8 8

2. Two paths → NID
 → HID.

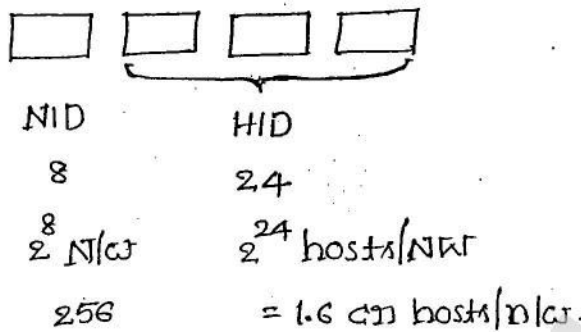
3. HID must be unique

4. Hidden meaning.

Classification of IP Addresses:

To covers the needs of diff types of organizations IP addresses are divided into Five classes.

class A:

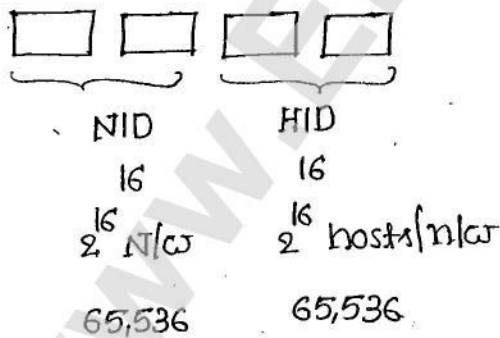


* Govt. org uses this network, eg: Defense n/w

eg: APSWAN Andhra pradesh state wide Area Network

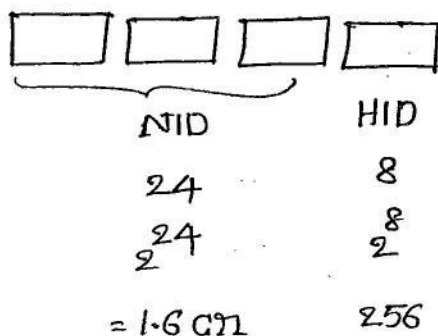
* IP address can be assigned to any electronic device.

class B:



Eg: Big organizations, MNC, Banks.

class C:



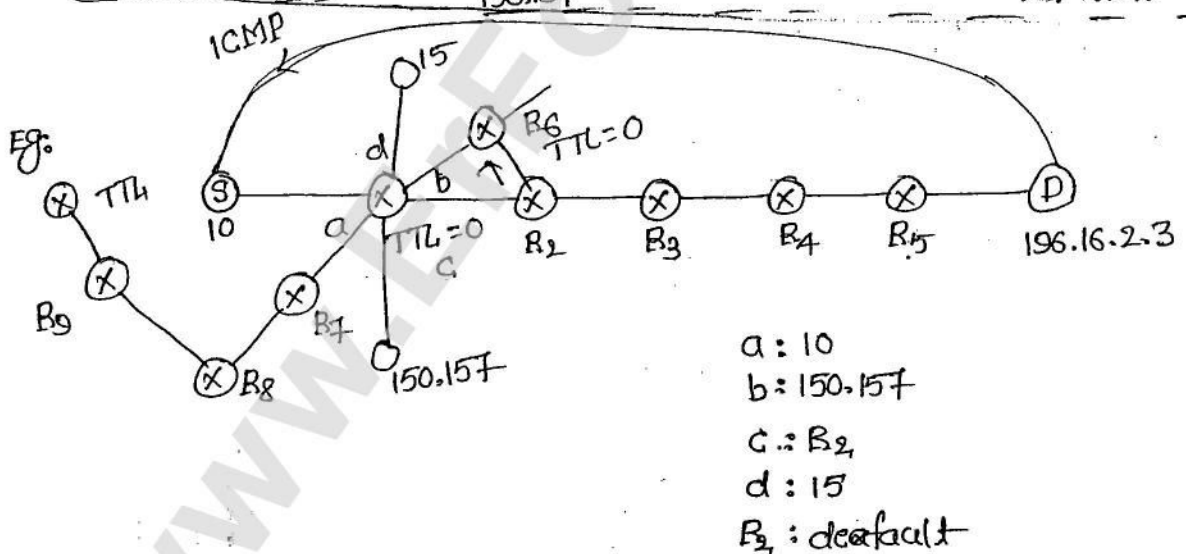
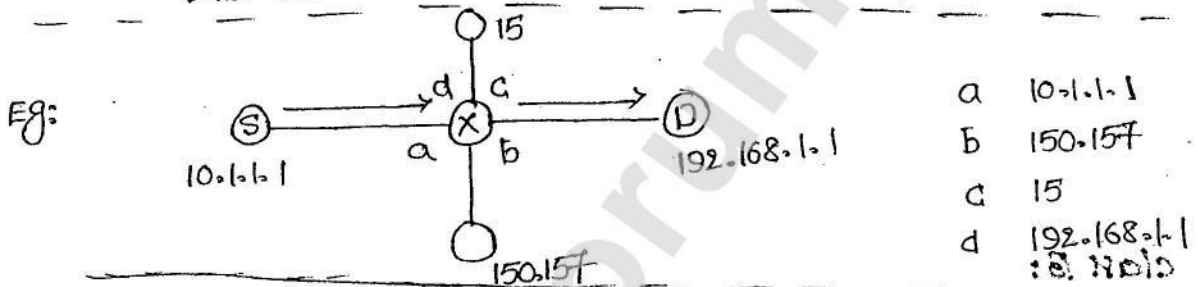
eg: Engineering colleges.
Medium orgs

- class A : 1-126
- class B : 128-191
- class C : 192-223
- class D : 224-239
- class E : 240-255

* 127 is a special IP address.

Eg: $\frac{150.167.1.1}{\text{NID}} \frac{1.1}{\text{HID}}$ class B
 $\frac{192.168.1.1}{\text{NID}} \frac{1.1}{\text{HID}}$ class C

} used in routing process

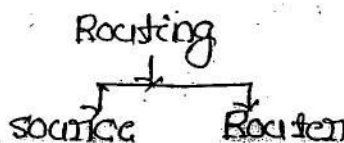


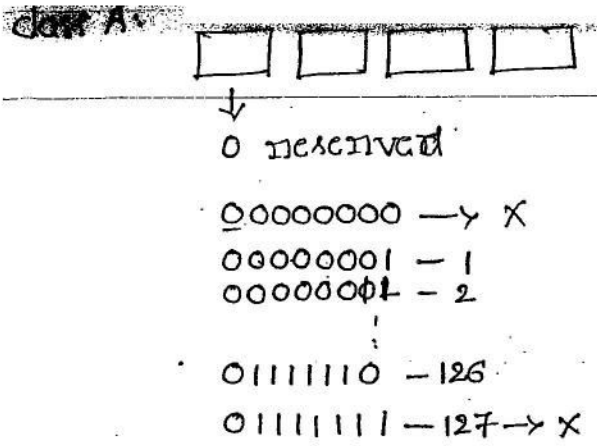
* consider default route (make dynamic route to avoid the wrong route through R₇ to B₉)

* Make TTL = 0

Eg: TTL = 3 min = 180 sec.

NOTE: TTL is used for to avoid the infinite loop. : 0 hold

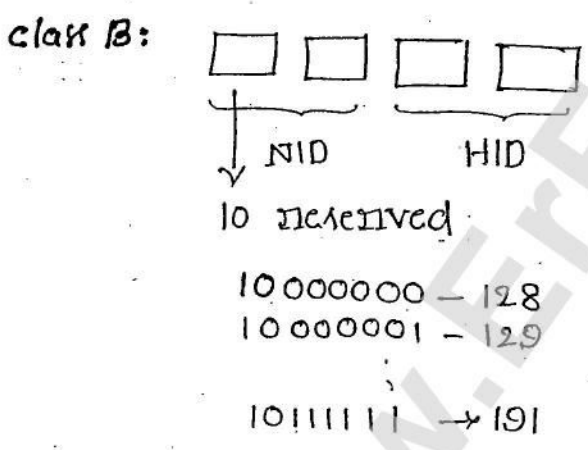




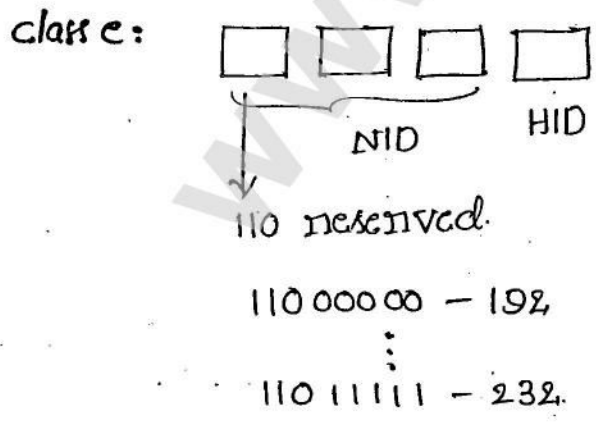
NOTE: But "0" and 127 are reserved for special purpose,

∴ 10.0.0.0
10.255.255.255 } Not used

2^8 N/w	2^{24} hosts/N/w
$2^7 - 2$ N/w	$2^{24} - 2$ hosts/N/w



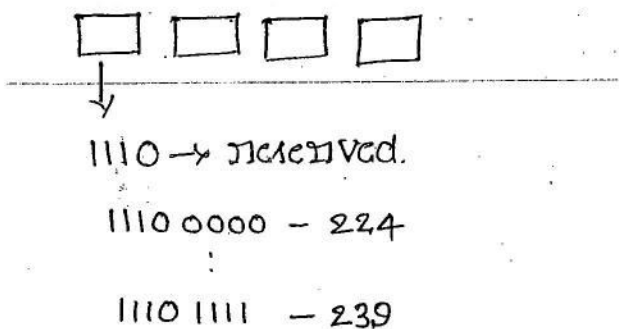
2^{16} N/w 2^{16} hosts/N/w
↓
 2^{14} N/w $2^{16} - 2$ hosts/N/w



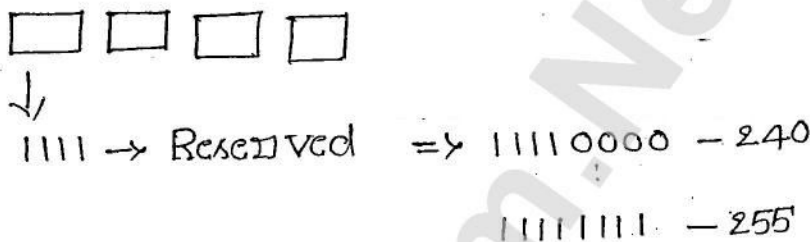
2^{24} N/w 2^8 hosts/N/w
↓
 2^{21} N/w $2^8 - 2$ hosts/N/w

class D: it is designed for multicasting there is no NID on HID.

* The whole address is used for multicasting.



class E: it is reserved for internet for special use.



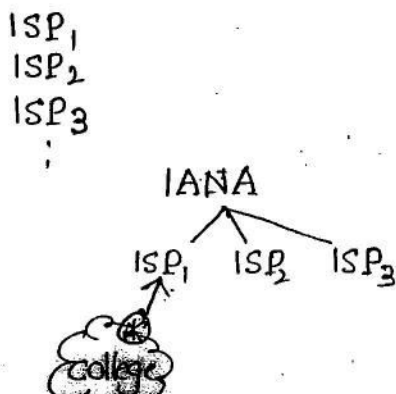
$$\begin{aligned}
 A &: (2^7 - 2) * (2^{24} - 2) \\
 &+ \\
 B &: (2^{14} - 2) * (2^{16} - 2) \\
 &+ \\
 C &: (2^{21} - 2) * 2^8 - 2 = 4 \text{ billion IP Address.}
 \end{aligned}$$

- * IP₄ \rightarrow 32-bit addressing system
- * IP₆ \rightarrow 128-bit " "
- * windows 7 supports IP₆
- * Linux 10.6 also supports IP₆.

IANA: Internet Assigned Number Authority

\rightarrow it is used to assign the unique IP address for sys.

ISP: Internet service provider.



* A user can ~~not~~ contact directly to IANA for address but it is time consuming so a mediator known as ISP handles the connection IANA and Provider IP address.

IANA (or) ICANN: Internet Corporation for Assigned Names and Numbers.

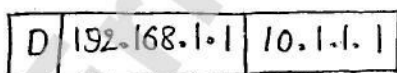
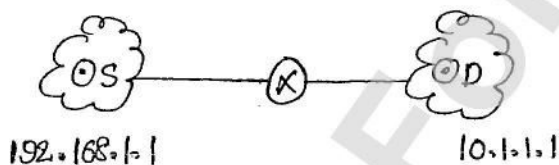
spoofing

spoofing: using IP address of others by unauthorized users.

Types of communication:

- * unicast → one to one
- * Multicast → one-to-many
- * Broadcast → one-to-all
- * Anycast → one to one and one to all.

unicast:



Eg: Mail application (browsing webpages).

Broadcast:

- Directed Broadcast
- Limited Broadcast

- * sending packets to all systems in same n/w → Directed Bc
- * sending the packets to other sys in our own n/w → Limited Bc

Multicast:

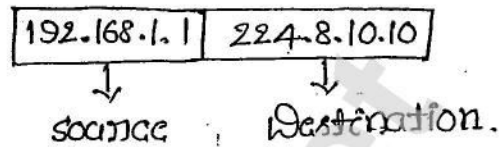
- * it is created from class D.
- * All the IP addresses are stored in a group.
- * For class D, i.e. for group communication IGMP is used instead of IP.

Eg: sending group mail

yahoo.mail

Group IP address

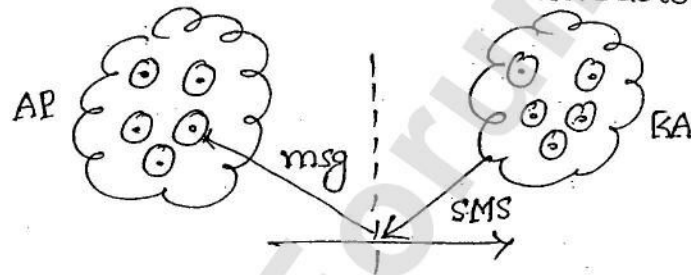
10.1.1.1
10.1.1.2
150.157.1.107
200.200.200.1



NOTE: * Majority of communication present in multicasting.

Anycast:

* it is used in mobile host communication.



* in multicasting 28 bits are available for identifying group.
so million groups can exist at the same time.

* Two types of group addresses are supported for multicasting
→ permanent
→ Temporary.

Eg: For any cast : Laptop.

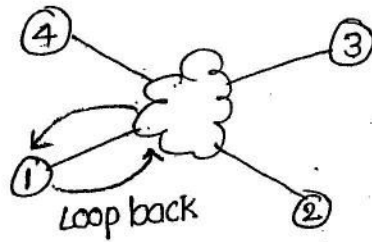
* The nearest access agent takes care of communication of particular mobile.

* so it has 1:1 and 1:all communication.

⇒ 127 Special IP Address:

* 127 IP address is used for connectivity purpose.

PING: Packet internet group.



* Address can be used source and destination are same.

* in command prompt type: c:\192.168.1.2 to ping system 2 to system 1.

* if a system sends the request i.e ping to other then if there from other then it is called "requested timeout".

→ positive message

→ Requested timed out

→ Destination unreachable.

Characteristics:

* it is called as loop back address because packet is delivered the source and again received by the source.

* its first octet should be 127 but no restriction on other

eg: 127.0.1.1
127.50.255.255

* it never fall under any classification

* it is also used for interprocess communication (IPC).

* "localhost" is a URL to 127.1.1.1 address. if source & dest have similar address → not valid.

Limitations of logical addressing sys: :

- * There is no flexibility.
- * There is no security
- * it is not permanent.

See solutions:

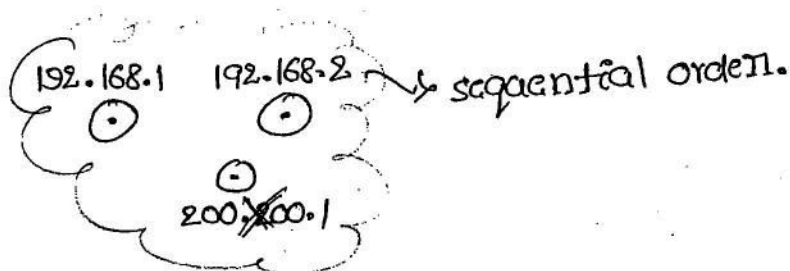
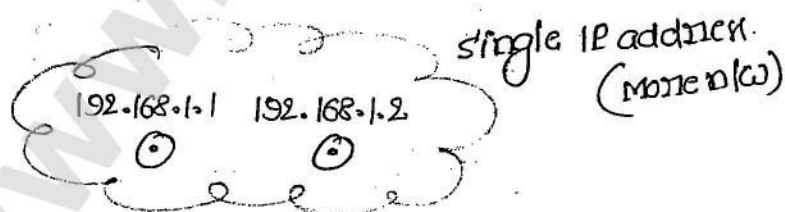
1. supernetting
2. subnetting
3. Physical Addressing system.

Supernet:

The process of aggregating two or more networks to generate the IP address for the group is known as "Supernet"

Limitations:

- * it is applicable for two or more networks.
- * All the networks in the supernet must be of same class.
- * Network ID's of the networks in the supernet must be in the order.



Advantages:

- * it improves flexibility of IP allotments.
- * it reduces no. of routing table entries.
- * it is used to improve security.

Subnet:

- * Network is partitioned into small subnets and are connected connectors called "Bridge".
- * process of dividing a single network into multiple subnets is subnetting.
- * Filtering and forwarding approach.

Advantages:

- * it improves security.
- * Maintenance and administration are simple.
- * Restructuring of the net is simple.
- * systems within the same subnet can communicate without any bridge.
- * if a system within one subnet needs to communicate with other subnet then it must pass through the "Bridge".

NOTE * The process of borrowing bits from host ID to generate subnet ID is known as "subnet".

- * no. of bits borrowed is depends on our requirements.

Eg: To have 3 subnets in class A net, we suppose borrow 2-bits.

$$\therefore \text{No. of subnets possible} = 2^2 = 4$$

$$\therefore \text{no. of systems per subnet} = 2^6 - 2$$

Eg: class B net. we have 100 subnets, we need 7-bits from
HID

$$\text{no. of subnets} = 2^7$$

$$\text{no. of hosts per subnet} = 2^9 - 2.$$

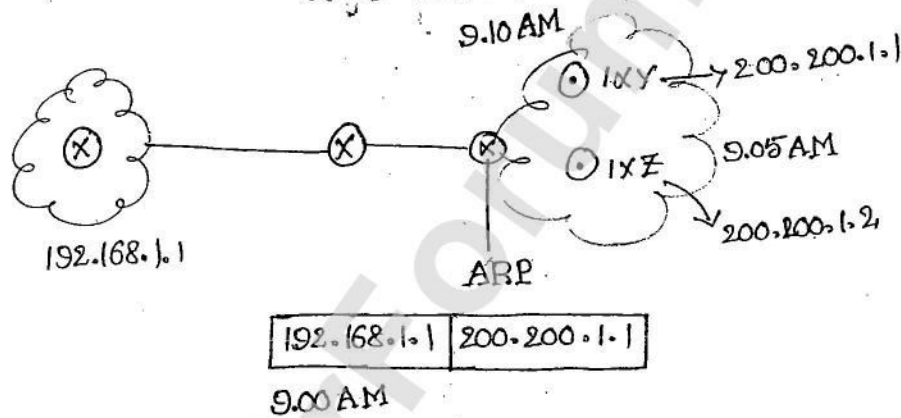
Limitations:

- * it complicates communication process. (4 step process)
- * we will loose IP address during this process.

Step procedure:

- > identify the network
- > identify the subnet network
- > identify the host id.
- > identify the process.

Physical Addressing system:

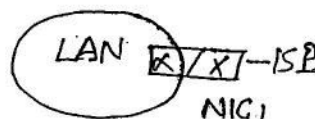
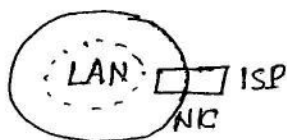


* if a system having IP = 192.168.1.1 send data to other systems 200.200.1.1 at 9.00 AM. Meanwhile if the destination system changed its IP, then data is sent other systems.

* In order not to have data misusage, a physical addressing system maintains the IP addresses of changed systems and provides the data to it.

200.200.1.1 => logical => different

IXY => physical => unique (IMEA)



* By using one IP address a hacker can't easily attack the sys
 so proxy IP address is maintained. such that if he has
 IP, through then the connection b/w IP₁ and IP₂ is disconnected
 so the system within the LAN are safe.

Logical: 32-bit - Network layer - IP - s/w → not permanent

Physical: 48-bit - Data link layer - ARP → H/w → permanent

service point: 16-bit → Transport → TCP/UDP - s/w → fixed.

Service point Addressing sys:

* it is 16-bit addressing system

so 2^{16}
 0
 ⋮
 1024 } well known port number
 (RFC)
 ⋮
 49,000 } Assigned port number (Nos)
 ⋮
 64,000 } Dynamic port number (Application)
 =

various types of objects in computer networking:

- * workstations and servers (7 layers)
- L₁ * Hub - 1 (Physical layer)
- L₂ * switch - 2 (Physical, Data link layer)
- L₂ * Bridge - 2 (PL, DLL)
- L₃ * Router - 3 (PL, DLL, NL)
- * Brouter - 3 (PL, DLL, NL)
- * Gateway - 7.

1. work station and servers:

- * A particular OS server acts as the domain key and all the client systems acts as workstations.
- * The server maintain some Access control list (ACL) which represent the accessibility of programs by the client.
- * The unaccessible programs are denied by server.
- * servers may have several applications.
eg: OS server, DB server.

2. HUB:

- * it is used to connect multiple workstations and servers.
- * it is a passive device, no sw associated with this.
- * it is a broadcasting device.

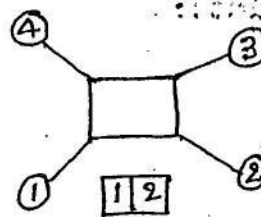
Advantages:

- * cost of the hub is low.
- * operation is simple.

Disadvantages:

- * Network traffic is high.

causing unnecessary disturbance at various systems.

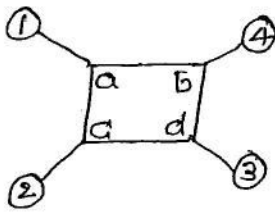


Switch:

- * combination of a hub and bridge
- * used to connect multiple workstations.
- * it maintains a look-up table to keep track all the systems.

Advantages:

- * Network traffic is less.
- * No unnecessary disturbances at various locations.
- * Because of above two reasons performance is good.



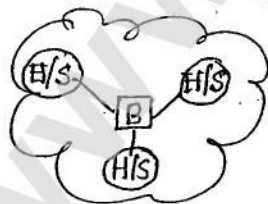
a	1
b	2
c	3
d	4

DisAdv:

cost of switches is 2 to 3 times of the hub.

Bridge: (PL, DLL)

- * A bridge can be used to connect multiple LAN's (similar) or multiple subnets.
- * its design criteria is filtering and forwarding.
- * its operation principal is based on physical addressing sys
- * it also maintain lookop table.



H/S \Rightarrow Hub/switch.

Router:

- * it is a sophisticated WAN device and its principal is based on addressing system
- * it is used to connect two or more different similar networks.

* it requires a lot of configuration. where as bridge and switch are

* All routing algorithms are running in a router. so the cost of a router is very high.

6. Brouters:

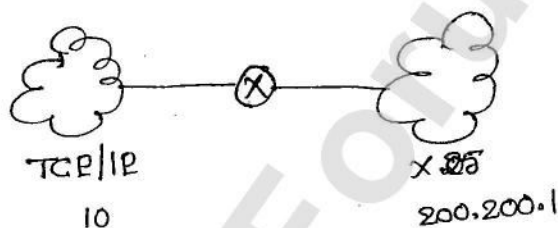
* Brouters are devices that combine the functions of both bridges and routers.

* They operate at both the data link and network layers.

* it is combination of Router and Bridge.

7. Gateway:

* it is used to connect 2 or more different but similar networks



* A gateway is a protocol converter.

* A gateway can be

-> stand alone computer with special software and several NIC.

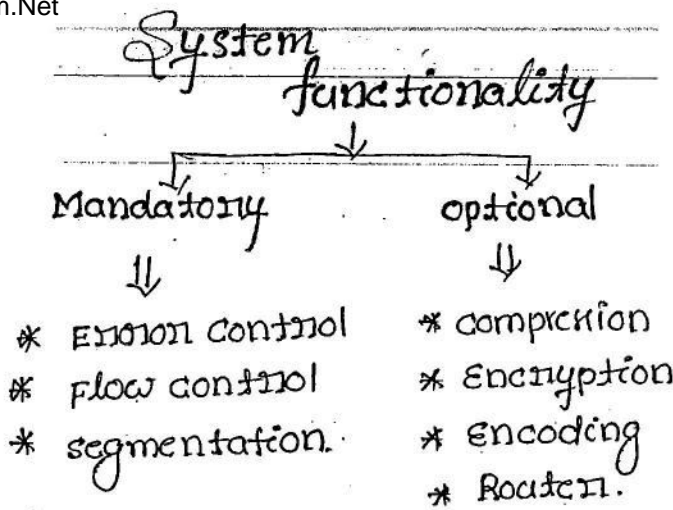
-> software installed in a router.

-> A front end processor (FEP) in a mainframe.

=

Appⁿ layer Gateway = Router

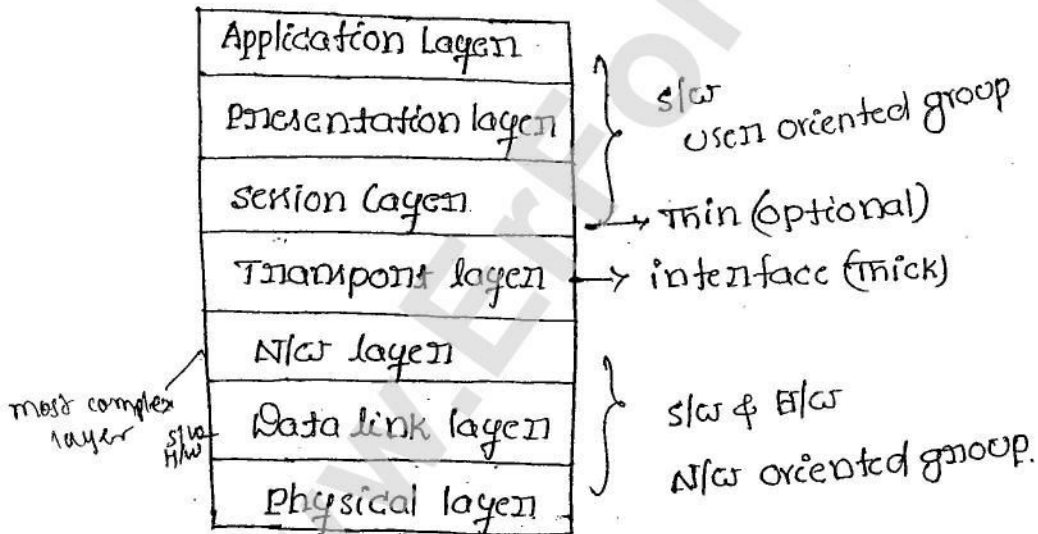
Network layer Gateway = Gateway.



Total => 70 functionalities
Reference model:-

- * OSI
- * TCP/IP
- * FR
- * ISDN
- * ATM - WAN/LAN
- * IEEE
- LAN - * X.25

TO access all these functionalities
reference model OSI



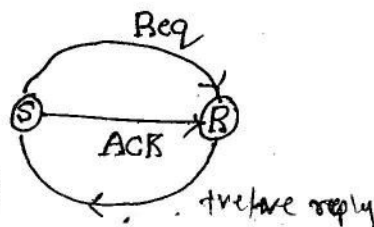
OSI reference model divide the 70 functionalities into 7 individual groups.

connection-oriented & connectionless communication:

connection-oriented:

Three-way-handshake

1. connection establish
2. Transfer
3. Terminate connection

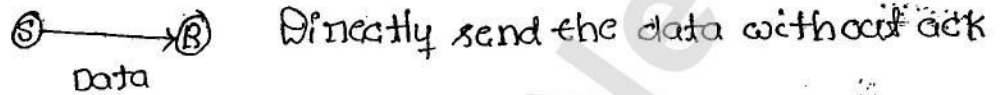


* Reliability is high

if accept => +ve reply
otherwise "-ve"

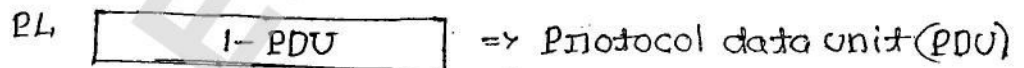
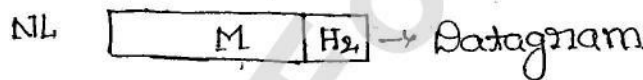
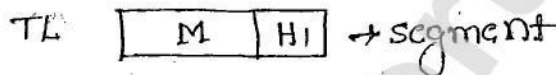
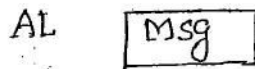
* Protocol is TCP (Transmission control protocol).

connection-less:



∴ UDP (User Datagram protocol)

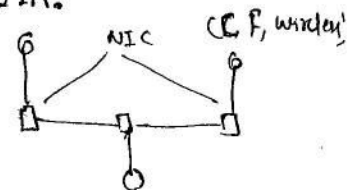
user



1. Physical Layer:

* it defines electrical, mechanical, functional and procedural specifications of interface and media are providing for sending a bit stream on a computer network.

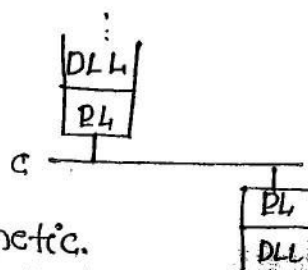
* interface is ^ NIC Network interface card.



Representation of bits:

c: Electrical (cooper wire)

F: Light signal



Physical layer converts the digital to electrical and vice versa.

1. it defines transmission mode.

→ simplex ⇒ keyboard cable

→ Halfduplex ⇒ one can talk at a time simultaneously
(walky-talky)

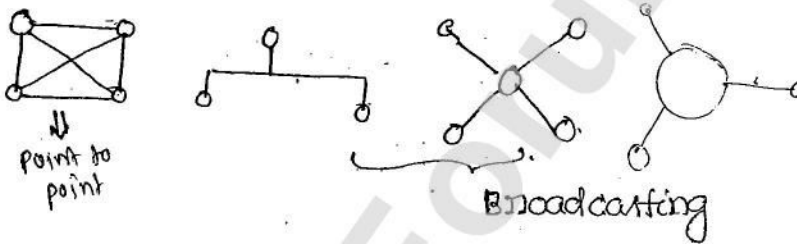
→ Full-duplex ⇒ Both can talk simultaneously.

NOTE: session layer decides either half duplex or full duplex connection
(Upper layer protocol) →

2) it defines link configuration

* point-to-point link (A dedicated channel for one source)

* Broadcasting link (A single channel for all sources)



* it defines topology configuration.

→ it maintains fixed rules.

2. Data link Layer :

* DLL transmits frames of data from computer to computer.

* Responsibility

→ Error control

→ Flow control

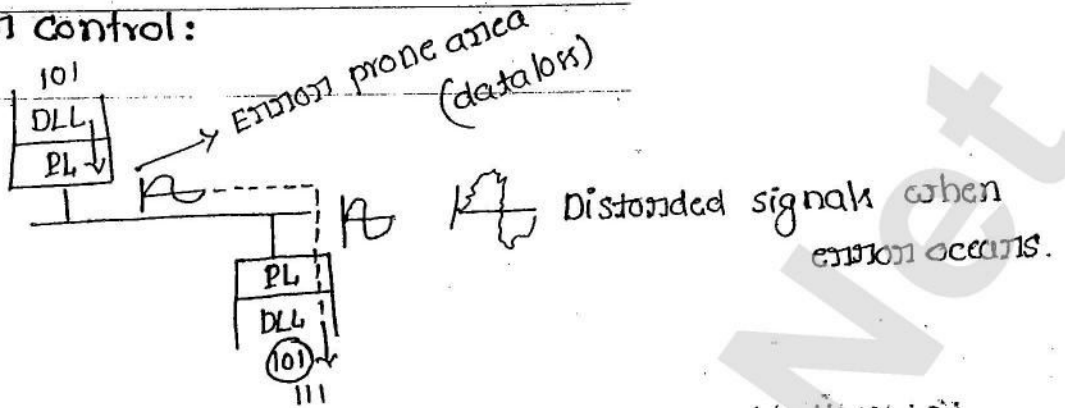
→ Access control

→ framing

→ physical addressing system 48 bit

MAC
Ethernet
LAN
NIC

1. Error Control:



* if there are errors in the transmission of bits or signals then a chance of distorted signals and bit representation is changed.

- * so the destination of the DLL must verify these functions:
 - Error Detection
 - Error correction
 - Re-transmission (sending -ve ack to sender then sending again).

2. Flow control: → A fast sender and slow receiver is leading to flow problems.
→ It is used only in connection oriented.



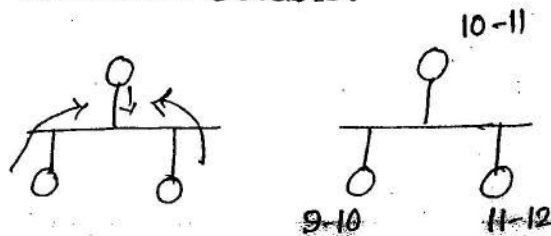
* Based on the server capability client sends the data.

sliding window protocol: To control the flow among client and server.

- * stop & wait
- * Go-back N.
- * select- Reject.

3. Access control:

* Time slot mechanism is allotted to all the stations since lot of collision occurs.



Sophisticated access control mechanisms:

* ALOHA

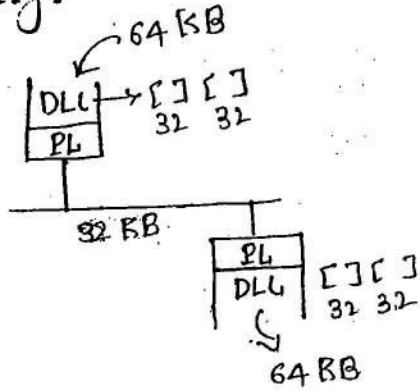
* CSMA/CD

* CSMA/CA

* TP.

* user can generate any sized data.

Framing:



* Framing - LAN

* Segmentation - WAN

* The link capacity is 32-bits,

* PL converts any sized PKT into 64 KB.

3. Network layer:

Responsibilities:-

* Logical addressing system

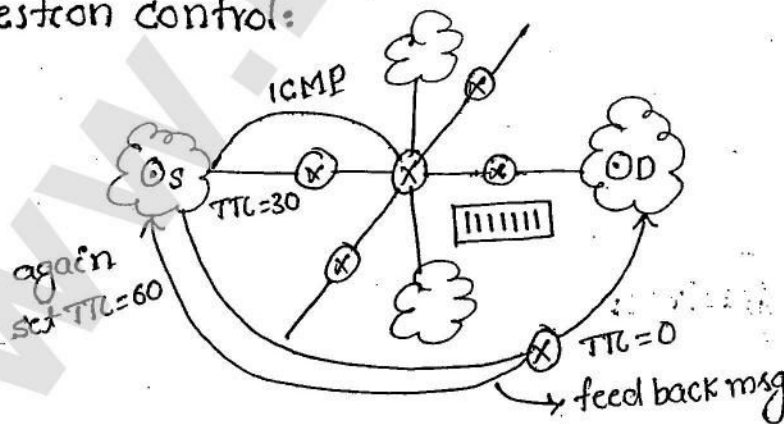
* congestion control

* Routing

* Feedback message => PING

ICMP => Internet congestion message protocol

congestion control:



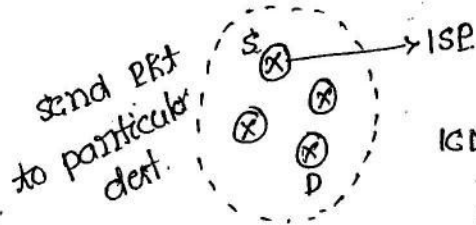
* if the data is full in all the buffers of the routers, then it is as congestion.

* To control congestion, routers send the message to the source to stop transmission of packets by knowing

it's IP address, rather than to all the adjacent routes.

Feedback message:

- * The receiver sends ICMP to the source.



ICMP (it's not exact path, if source gives shorter IP address, then it reject it's higher IP)

- * logical address system is temporary.
- * Physical address system is permanent.

4. Transport Layer:

Responsibilities:-

- * Application identification
 - * client-side-entity identification
 - * segmentation and Re-assembly.
 - * Multiplexing & De-multiplexing
 - * service-point addressing system.
 - * Error control.
 - * Transmission error detection.
 - * Flow control.
- * TL offers end-to-end communication between end devices through a network.
 - * combining all the different protocols data and sending is called multiplexing.
 - * There are equally partitioned and sent through the media, at the received side, all these are combined and received which is known as de-multiplexing.

* Error verification is done by 2 aspects

→ Link level (Data link layer)

→ end-to-end (Transport layer)

it have the extra field called "checksum" along with the data.

Data	checksum
------	----------

* CRC ⇒ check link error ⇒ DLL.

* checksum ⇒ check n/w error ⇒ Transport layer.

* Logical address of the pkt: permanent

* physical address of the pkt: Temporary.

⇒ source and destination generates CRC code and checks, if correct send ack to source, if ack received by source the packet is remove.

* if a problem occurs in Transport layer of receiver, then it send the -ve ack, to the sender. then retransmit.

5. Session Layer:

* This session layer allows applications, functioning, on devices to establish, manage, and terminate a dialog through a network.

Responsibilities :-

→ virtual connection b/w application entities

→ synchronization of data flow

→ creation of dialog units

→ connection parameter negotiation

→ partitioning of services into functional groups.

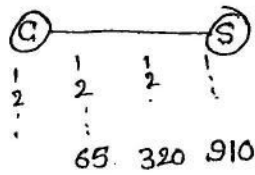
→ ack of data received during a session.

→ Maintaining checkpoints / synchronization points

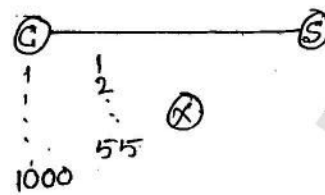
→ Grouping of operations.

checkpoint maintainance :

Eg: Downloading files.



DAB supports checkpoints



* when a file is being downloading, then if connection is discarded in the download, then the user have to start from the first.

* in order to overcome this problem checkpoints, are introduced which specifies, up to completed file and continues at the same point when download again.

6. presentation Layer:

* The presentation layer is responsible for how application formats the data to be sent out on the network.

Responsibilities:-

- Encryption and decryption of msg for security.
- compression and expansion
- Graphics formatting
- content translation
- system specific translation.

7. Application Layer:

An interface for the end user operating a device connected to a network.

Maintaining harmony among protocols:

→ Depending on user's preference one protocol is converted into others and after completion of the task again converted to original protocol.

Eg: ATM for checking a/c details, http protocol is used and for transaction http is used

- User interface design
- for file transfers
- Electronic mail
- Electronic messaging
- Browsing the web.

↓
Security

Maintaining harmony:
Information flow from one protocol to another (eg. HTTP to SMTP & back)
Understanding user scenarios (HTTP & HTTPS back etc. must be main roles & responsibilities of application layer

Application	DNS, FTP, TFTP, BOOTP SNMP, SMTP, telnet, FINGER	Gateway
Presentation	HTTP, SMTP, SNMP	Gateway, Redirector
Session	RPC, Ripes, ASP	Gateway
Transport	TCP, UDP	Gateway, Brouter.
Network	IP, IGMP, ICMP	Brouter, Router
Data link	LLC, MAC	Bridge, Switch
Presentation	IEEE 802, IEEE 802.2 ISDN	Repeater, Hubs, Multiplexer, Amplifier.

Advantages of layering systems:

- * it uses divide-and-conquer principle. Therefore maintenance and administration is simple.
- * it uses Object-oriented principles like Abstraction & encapsulation.
 - ∴ Security is high, extensibility is simpler.
- * Abstraction ⇒ hiding the elements.
- * independently, the layers are changed without impact on others.

Dis Adv:

- * interdependency among layering systems ⇒ TCP/IP
 - TL NL
- * Duplication of functionality.

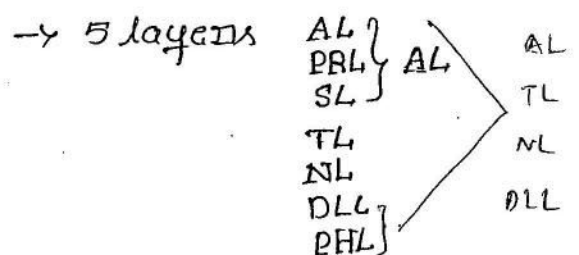
OSI- Reference

→ 7 (seven) layers

- No definition for multicasting
- standards are ideal
- No flexibility

(Max-size of Pkt = 64KB only)

TCP/IP



- it is clearly defined in TCP/IP
- standards are practical.
- Lot of flexibility.

(depends on characteristic of each layer.)

Sliding Window Protocol

Characteristics:

- * it is used in connection-oriented communication
- * it offers flow control and packet-level error control.
- * it is used in both Transport layer and Datalink layer.
- * it is a theoretical concept, practically implemented as,
 - Stop-and-wait
 - Go-Back-N
 - selective-Repeat protocol.

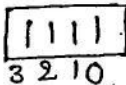
Different types of Delays:

- * Queuing delay
 - * processing delay
 - * Transmission delay
 - * propagation delay.
- * The amount of time taken by the process to be in queue before entering into the router.

Queuing delay:

The amount time packet is waiting in queue before being taken up for processing is known as "queuing delay"



 → Buffer, if buffer size = 4 then 5th pkt is discarded.

- * it is varying from 0 to infinite.
- * it depends on router processing speed and buffer capacity.

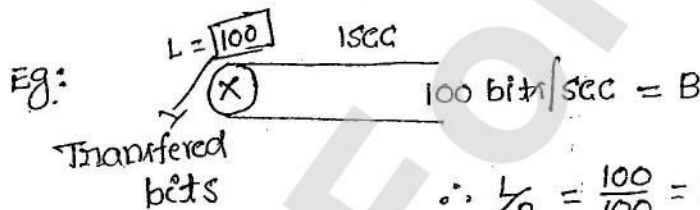
Processing delay:

The amount of time taken by router to process a packet [looking at destination IP, extracting new ID, searching in the routing table, identifying destination route] is known as "processing delay".

* it depends on router processing speed, but not size of the packets.

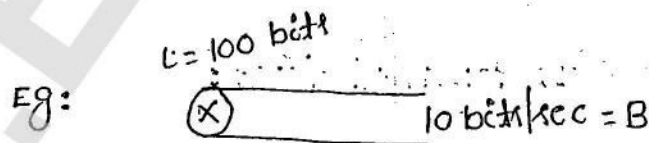
* Transmission Delay: (L/B)

The amount of the time taken by the router to transfer the packets to outgoing link is known as the "Transmission delay".



$$\text{Transmission delay} = \frac{L}{B}$$

L = length of the packet
B: capacity of the channel
or
Bandwidth of the link



$$\Rightarrow L/B = \frac{100}{10} = 10 \text{ sec for sending 100 bits.}$$

Propagation Delay:

Amount of the time taken by the packet to make a physical journey from one router to another, is known as "propagation delay".

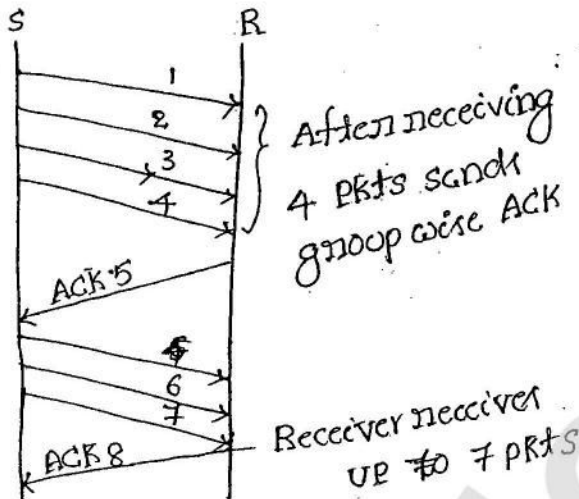
$$\text{Propagation delay} = \frac{d}{v}$$

d = distance b/w routers
v = velocity of

$$RTT = 2 * [\text{propagation delay} + N (\text{queuing delay} + \text{transmission delay} + \text{processing delay})]$$

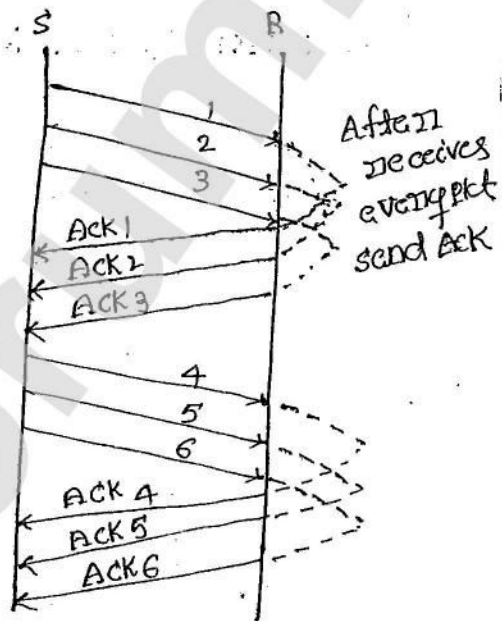
Time out = 2 * RTT.
TTL = 2 * Timeout

Cumulative:



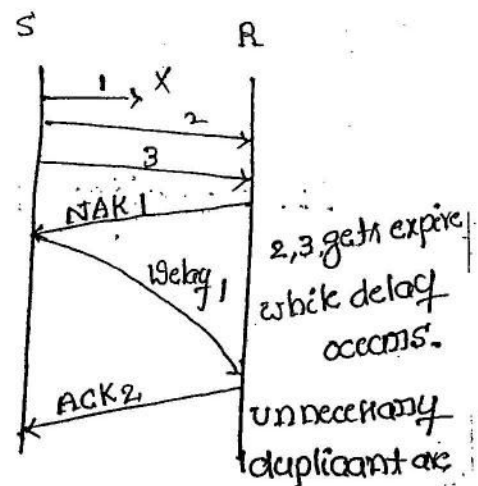
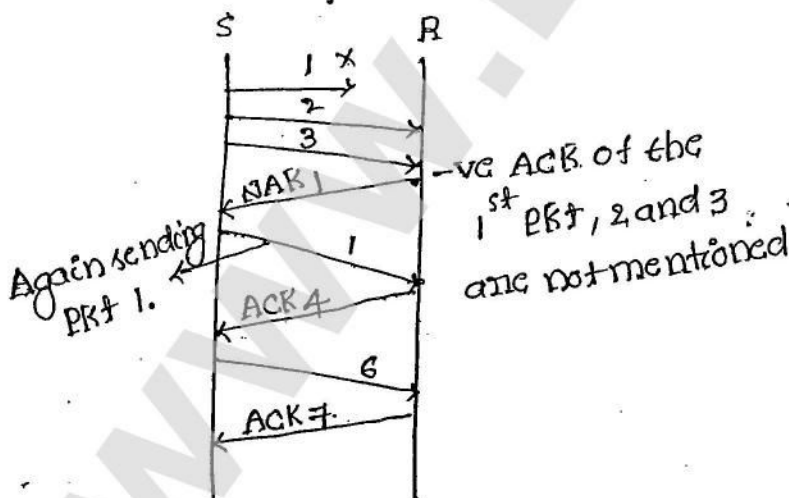
- => Network traffic is low
- => Reliability is low.

Independent:



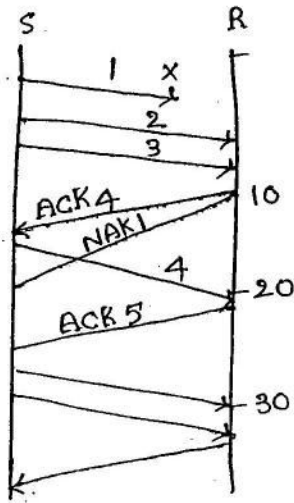
- => Network traffic high
- => Reliability is high.

Case study for cumulative:



if the delay is high to send pkt 1, then at the time of receiving pkt 1, the packets 2,3 get expired, so again actual packets of 2,3 is sent.

combination of cumulative and independent (Real time):
 i.e. Maintaining certain time slots.



suppose, consider 1,00,000 packets are transferred.

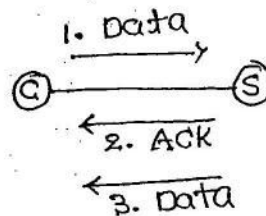
if first packet is lost at receiver side in cumulative after all packets are transmitted it send ack. At the time the sender knows that first packet is lost.

To overcome this problem repeat checking the pkts, improve reliability.

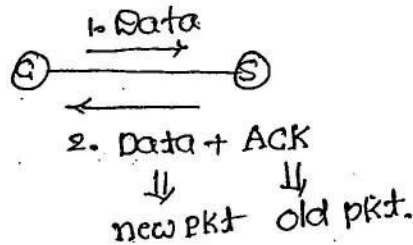
Piggy backing (web):

- Mainly used in web services
- To reduce the net traffic.

General Approach

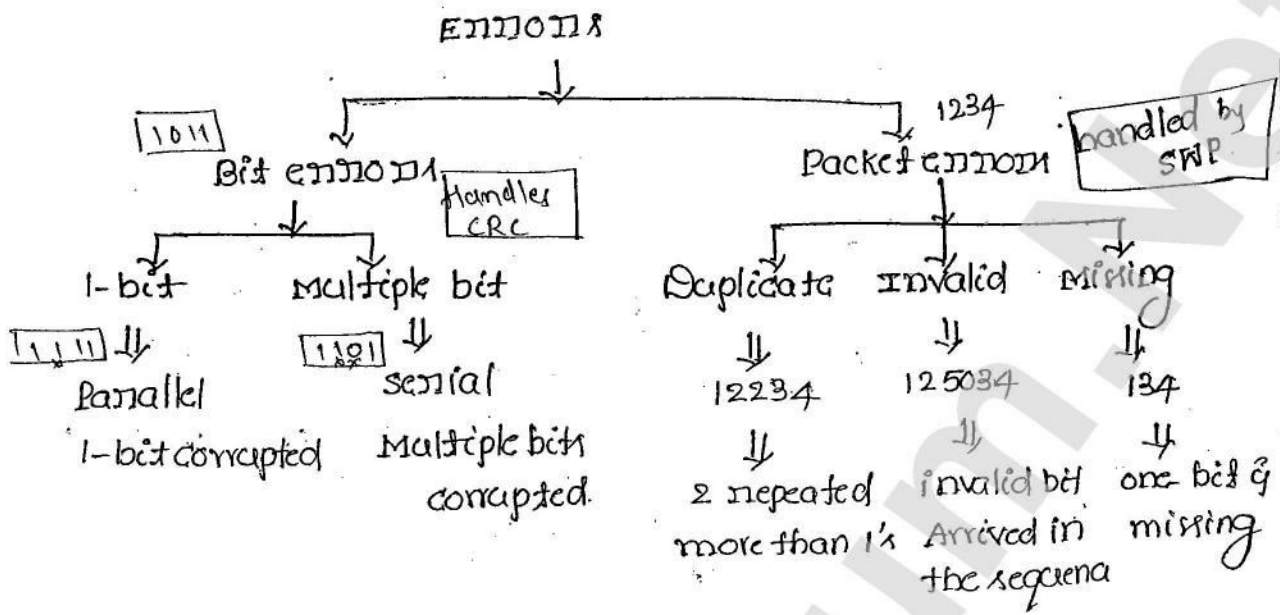


Piggy backing:



=

Different types of errors:



$$\text{Bit delay} = \frac{1}{B}$$

B: Bandwidth.

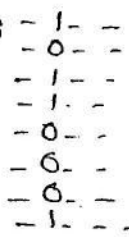
$$B = 10 \text{ bits/msec}$$

$$= \frac{1}{10 \times 10^6} = 0.1 \text{ usec.}$$

serial data transferred is: 10110001

* if noise is present in serial transmission that all the bits are corrupted.

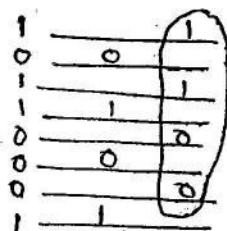
Parallel data transferred:



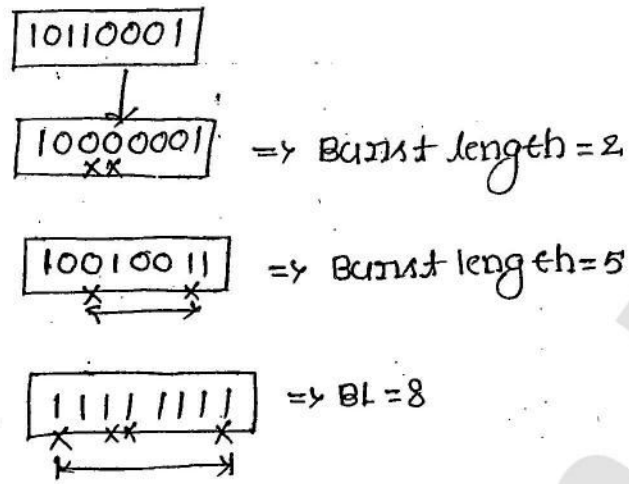
* in parallel transmission only that particular bit gets corrupted.

* Serial transmission is considered in computer n/w.

* if length of the communication is long (>1 meter) we use serial transmission, else we use parallel transmission.



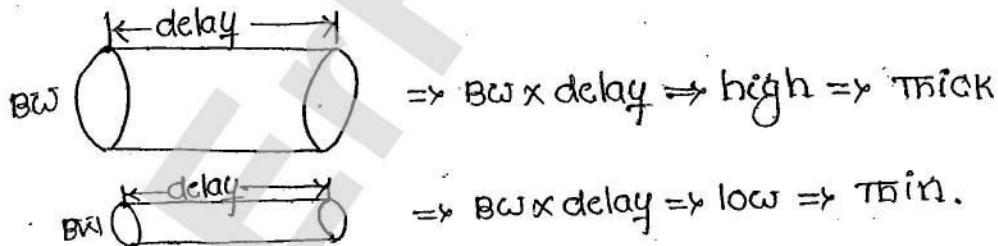
For synchronization and collecting the data bits a group. so for long distances it not support.



- * Burst length is calculated a bits present between starting and ending corrupted bits.
- * Burst length depends up on the type of OS i.e 32-bit OS the max BL = 32, (or) 64-bit OS then max BL = 64.

* Based on the max BL we develop polynomial for error checking with x^{32} or x^{64} . $FCS = x^{32} + \dots$ (if OS is of 32 bit) (Frame check seq (FCS))

Bandwidth delay:

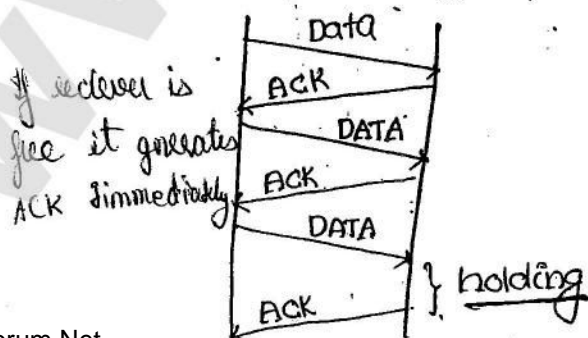


Stop-and-wait protocol:

STOP & USES 2 rules at sender side:

Rule: 1: Transfer only one packet at a time

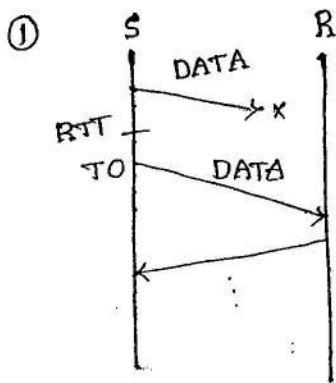
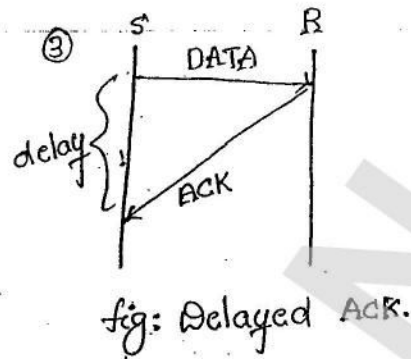
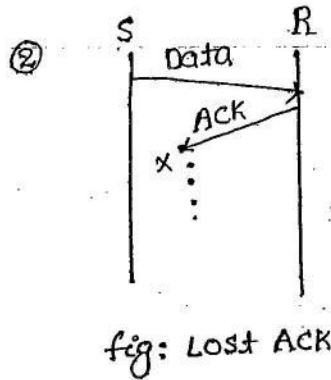
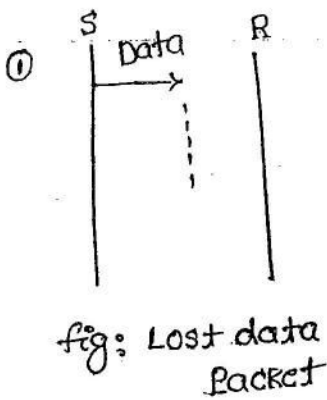
Rule: 2: After receiving the ACK only the other packet is transferred
 If source gets ACK for the pkt



For maintaining control

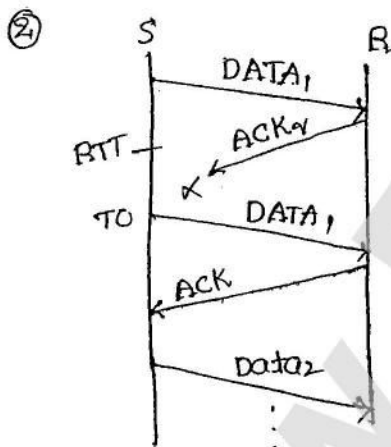
- * 2 principles in sender
- * 2 " " Receiver, it will hold ACK for a when receiver busy, it can't accept while & then releases the ACK, new PKTs, so at that time it won't send any ack to sender

Drawbacks:



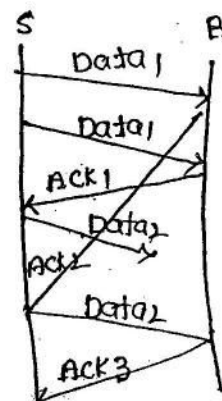
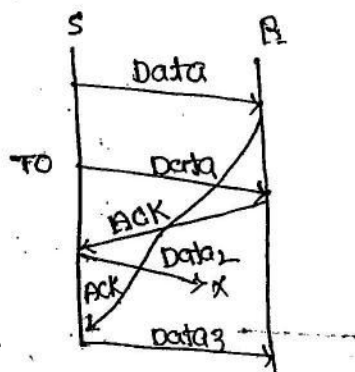
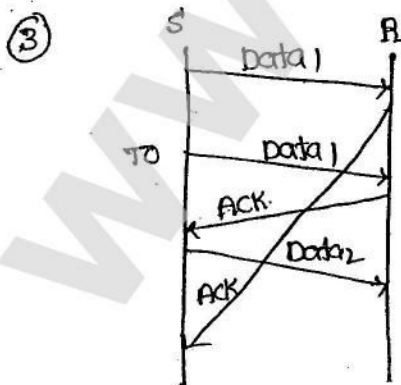
Either the lost data or ACK only the time out is considered. If within the given time ACK is not received then next data is sent as represented it by timeout.

stop and wait + Time out



if within time ACK of data is not received and after data is received at receiver end and sender received ACK of after TO data. It means to have ACK of Data 1, so not be get confused, sequence num is represented.

stop & wait + Timeout + Sequence num (Data)



in order to not to get confused about the ack of particular data packet, then the Ack sequence num also represented.

$$\boxed{\text{stop \& wait} + \text{Timeout} + \text{sequence num(Data)} + \text{sequence num(Ack)}} \\ \Downarrow = \text{Stop \& wait ARQ}$$

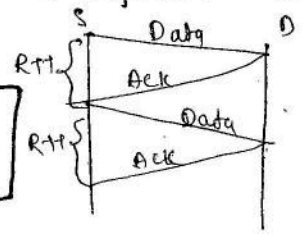
Automatic Repeat Request \Rightarrow it controls the packet level error control.

characteristics of stop & wait:

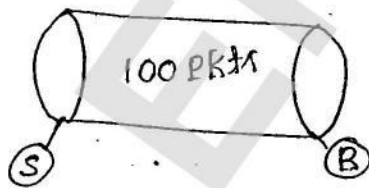
* it uses the link between sender and receiver as a half-duplex link.

* Throughput of stop and wait protocol:

$$\boxed{\text{Throughput}(T) = \frac{1 \text{Data}}{RTT}}$$



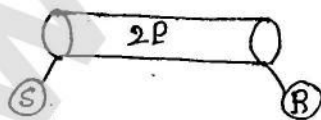
* if Bandwidth x delay product is very high then stop & wait protocol becomes useless.



capacity = 100 pkt

filling pipe with 1 pkt then

$$\text{Efficiency} = \frac{1}{100} = 1\%$$

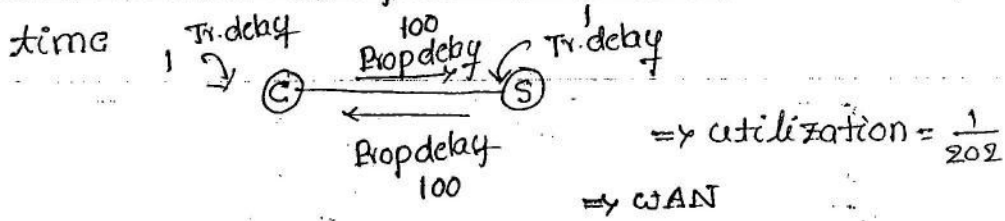


$$\Rightarrow \text{Efficiency} = \frac{1}{2} = 50\%$$

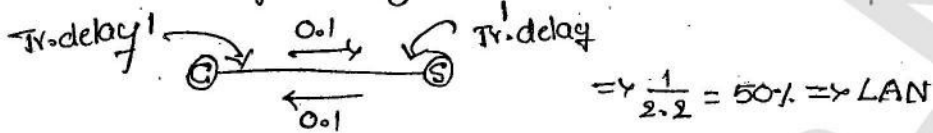
* it is not suitable for WAN because delay is high only suitable for LAN.

* if Propagation delay is high compared to transmission delay then, stop & wait protocol becomes useless. efficiency is less.

* Transmission delay is 1 mean up to 202 ms only 1 unit of

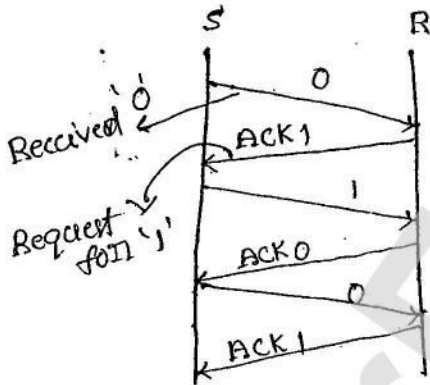


* Transmission delay is high compared to prop. delay



* stop and wait an example of closed loop protocol (connection-oriented)

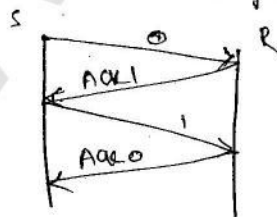
* stop & wait protocol use only two sequence numbers.



* in special category of stop and wait protocol with window size is 1.

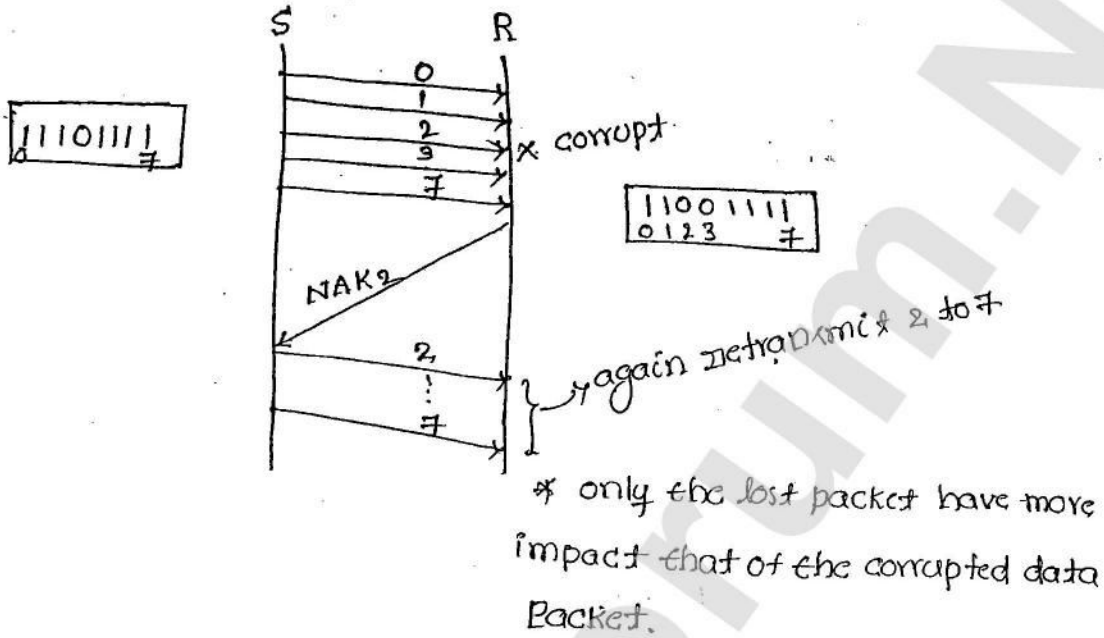
stop & wait $\Rightarrow S_w = R_w = 1$

* Stop & wait protocol req. only two seq. numbers (0, 1) irrespective of no. of packets sender is having.

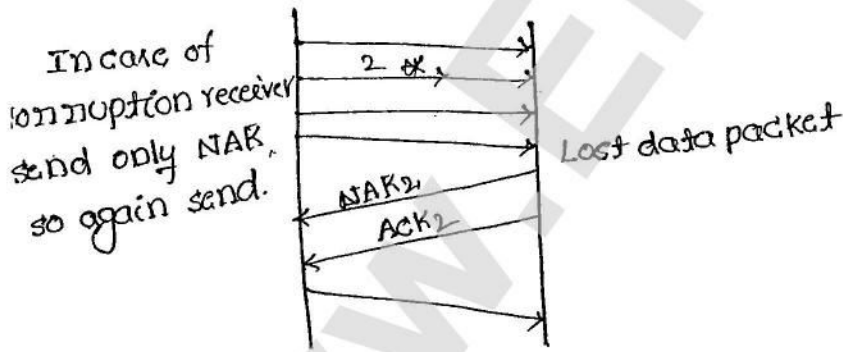


www.ErForum.Net

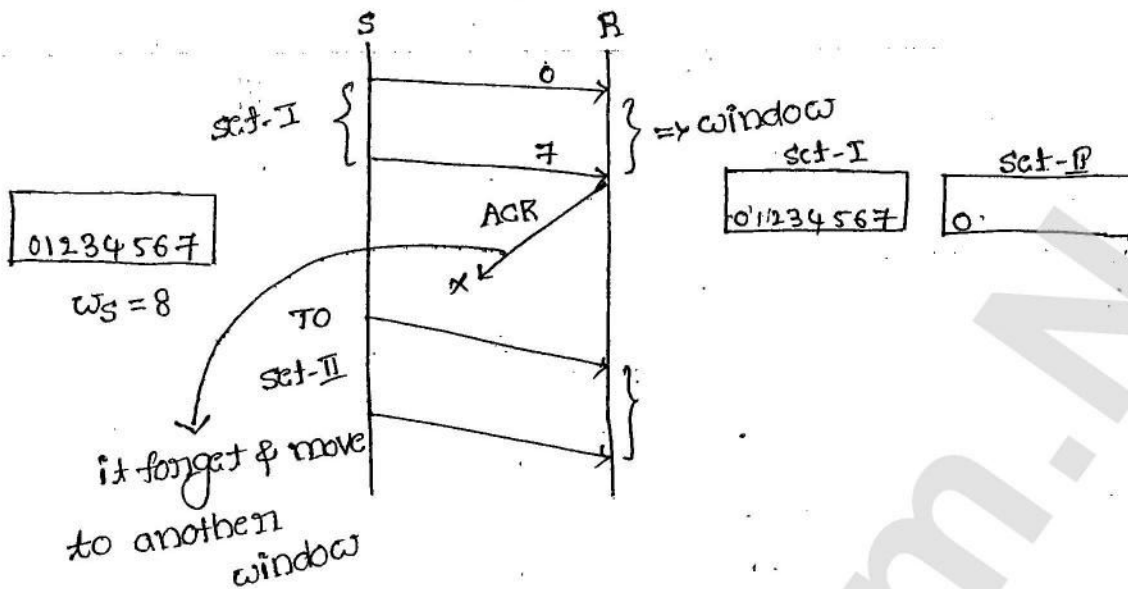
In this series sending packets if pk_2 is missing and all the seven packets are being sent then NAK_2 is sent. Then 3 to 7 packets are being discarded and pk_2 is retransmitted corrupted data packets:



- * Here the data packet is being corrupted then also the remaining all the packets are also get discarded.
- * high retransivity problem.



so in order to overcome above problem of data lost while sending the NAK_2 & ACK_2 is also being sent. so that it represents that pk_2 is not received and then after sending pk_2 it can continue from ACK_2 .

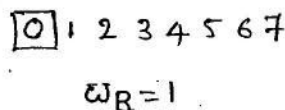
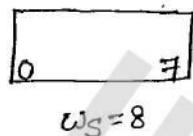


After time out, sender again send first window.

* To solve the above problem we have to re-adjust w_S, w_R size.

Case (a): w_R size:

it is equal to 1 always irrespective of w_S size



so here it is waiting for 1 after selection

'0' and if any other num other than 1 comes they are discard.

Case (b) w_S size:

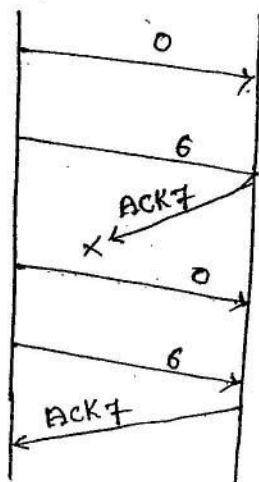
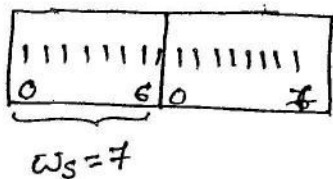
w_S size is calculated based on following formula

$$w_S + w_R \leq \text{Available sequence number (ASN)}$$

$$w_S = \text{ASN} - w_R$$

$$w_S = \text{ASN} - 1.$$

Adjusting window size:



0 1 2 3 4 5 6 7 0 1 2 3 4 5 6 7

eg: 0 15

Max available sequence num = 16

$W_S = 15$
 $W_B = 1$

eg: Total sequence numbers = 10

then $W_S = 9, W_B = 1$.

case(1): Assume $N =$ Max available sequence numbers

$$\therefore W_S = N - 1$$

$$W_B = 1$$

case(2): if N is defined as max sequence numbers

$$W_S = N$$

$$W_B = 1$$

case(3): if K is no. of bits available in sequence number P

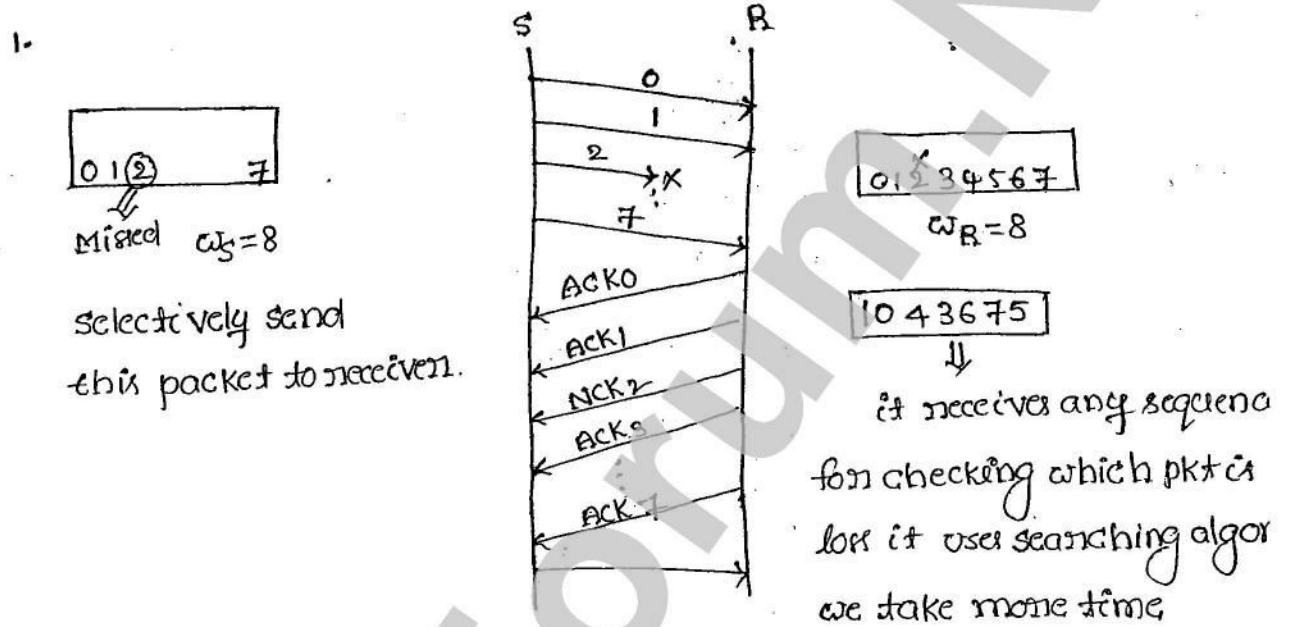
$$W_S = 2^K - 1$$

$$W_B = 1$$

=

Selective Repeat:

- * selective Repeat receives receiver out of order packets.
- * it's natural choice is independent ack (if possible, it will also use piggy backing)



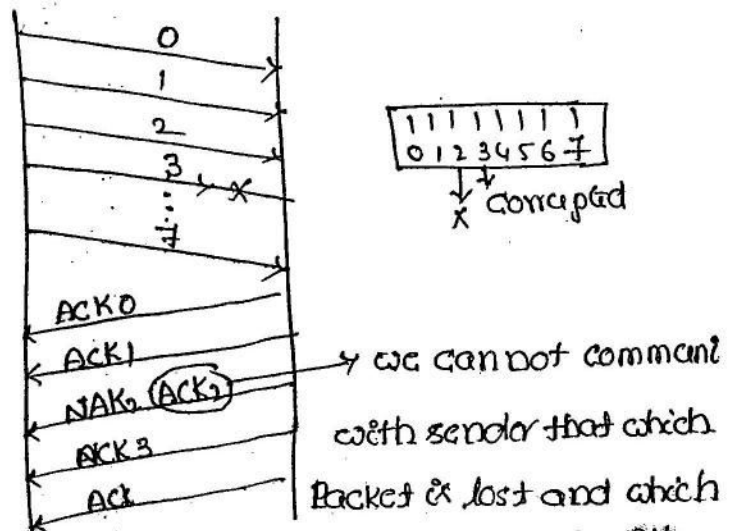
- * A searching algorithm is used.

Dis Adv:

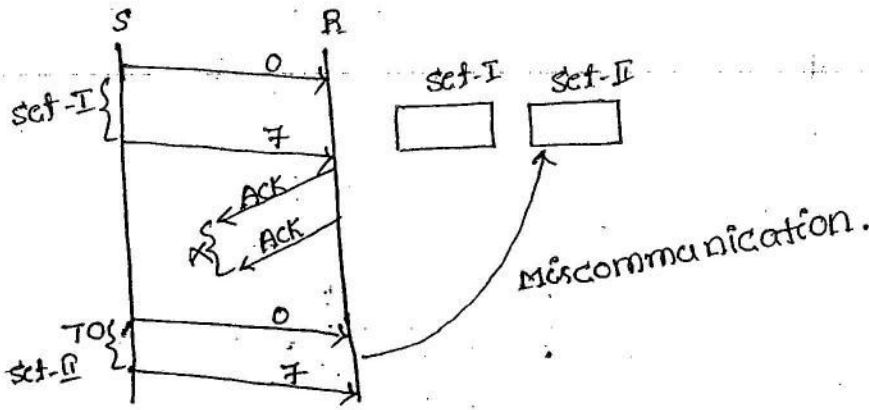
Time increases as it requires searching and sorting of num. of packets to be transmitted explicitly.

② corrupted:

we won't resolve the condition either packet is lost (or) corrupted because every time it send only NAR



Lost Acknowledgement:



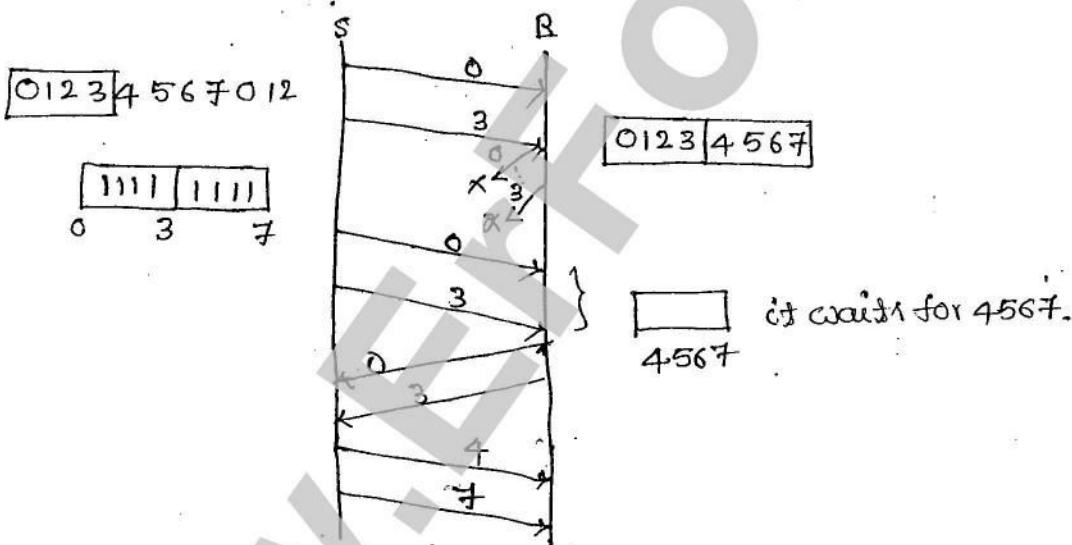
To solve the above problem, we re-adjust ω_S and ω_R based on the following formula

$$\omega_S + \omega_R \leq ASNT$$

* $\omega_S = \omega_R$

* $\omega_S > \omega_R$

* $\omega_S < \omega_R$ ✗



case (i): if N is defined as max available sequence num then

$$\omega_S = \frac{N}{2}, \quad \omega_R = \frac{N}{2}$$

case (ii): if N is defined as max sequence no. then

$$\omega_S = \frac{N+1}{2}, \quad \omega_R = \frac{N+1}{2}$$

case (iii): if R is defined as no. of bits in sequence number

$$\omega_S = 2^{R-1}, \quad \omega_R = 2^{R-1}$$

eg: 8-bit sequence number.

Go-Back-N $\Rightarrow \omega_S = 7, \omega_R = 1$

selective Repeat $\Rightarrow \omega_S = 4, \omega_R = 1$

For same available sequence number, GBN can transfer more packets

GBN $\omega_S = 7, \omega_R = 1 \Rightarrow 8$

SR $\omega_S = 7, \omega_R = 7 \Rightarrow 14$

Comparison:

characteristic	stop&wait	GBN	Selective Repeat
1. operation	simple	Medium	complex
2. Requirement of sequence num	Low	Medium	High
3. Bandwidth utilization	Low	Medium	High
4. Buffer requirement	Low	Medium	High
5. Efficiency	Low	Medium	High

Stop&wait formula:

$$\text{Efficiency} = \frac{\text{Tran.delay}}{\text{Tran.delay} + 2 * \text{propdelay}}$$

$$= \frac{1}{1 + 2 \frac{\text{Propdelay}}{\text{Tran.delay}}}$$

$$= \frac{1}{1 + 2a} \quad \therefore a = \frac{\text{Propdelay}}{\text{Tran.delay}}$$

$$= \frac{L/B}{L/B + R}$$

$\therefore L = BR \quad \eta = 50\%$

$L > BR \quad \eta > 50\%$

$$= \frac{L}{L + BR}$$

$L < BR \quad \eta < 50\%$

Steps to be solved in s/w process:

1. Calculate the RTT
2. Based on given bandwidth and RTT calculate no. of bits, we are able to transfer within RTT and equate it as "window of bits" (w_{bits})

$$3. w_{pkt} = \frac{w_{bits}}{(pkt\ size)_{bits}} \approx w_p$$

4. sequence numbers required = w_p

5. $2^k = w_p$ where $k =$ no. of bits in sequence num. p.

Problems:

① $B = 1.5 \text{ Mbps}$

RTT = 45 ms

$L = 1 \text{ KB}$

link utilization = ?

= 12.1%

$$T = \frac{1 \text{ Data}}{\text{RTT}} = \frac{1024 \times 8}{45 \times 10^{-3}} = 182 \text{ Kbps}$$

$$\eta = \frac{T}{B} = \frac{182 \text{ Kbps}}{1.5 \text{ Mbps}}$$

$$= \frac{182 \times 10^3}{1.5 \times 10^6}$$

$$= 0.121$$

$$= 12.1\%$$

② Packet size = 1000 bytes.

(A) $d = 10 \text{ km}$

(B) $d = 5000 \text{ BM}$

$$v = 70\% \times 3 \times 10^8 \text{ m/sec}$$

$$= 0.7 \times 3 \times 10^5 \text{ km/sec}$$

$$v = 2.1 \times 10^5 \text{ km/sec}$$

(A) Propagation delay = $\frac{d}{v}$

$$= \frac{10 \text{ km}}{2.1 \times 10^5 \text{ km/sec}} = 0.476 \text{ } \mu\text{sec}$$

$$\begin{aligned}
 RTT &= 2 * \text{Propagation delay} \\
 &= 2 * 476 \\
 &= 952 \text{ } \mu\text{sec}
 \end{aligned}$$

$$\text{Throughput} = \frac{1 \text{ Data}}{RTT} = \frac{1000 * 8}{0.952 * 10^{-6}} = \frac{8}{.952} * 10^3 \Rightarrow$$

$$\textcircled{B} = T_{\text{prop}} * \frac{d}{v} = \frac{5000 \text{ KM}}{2.1 * 10^5 \text{ km/sec}} = 95.2 * 500$$

$$RTT = 95.2 * 500 \text{ } \mu\text{sec}$$

$$T = \frac{1000 * 8}{95.2 * 500 \text{ } \mu\text{sec}} = \frac{80 \text{ mbps}}{500} = 0.16 \text{ mbps}$$

$$\textcircled{3} \text{ Packet size} = 1 \text{ KB}$$

$$\text{Propagation time} = 15 \text{ ms.}$$

$$B = 10^9 \text{ bits/sec}$$

$$RTT = 30 \text{ msec} \quad \text{Transmission time} = \frac{L}{B} = \frac{1024 * 8}{10^9} = 0.008 \text{ msec}$$

$$\text{utilization} = \frac{1}{30 + (0.008)^2}$$

$$= \frac{1}{30.016}$$

$$= 0.033$$

$$= 3.3\%$$

$$\textcircled{4} \quad B = 4 \text{ kbps}$$

$$\text{propagation delay} = 20 \text{ m/sec}$$

$$\eta = 50\%$$

$$\text{RTT} = 2 * \text{propagation}$$

$$= 40 \text{ msec}$$

$$L = BR$$

$$= 4 \times 10^3 \times 40 \times 10^{-3}$$

$$= 160 \text{ bps}$$

$$\eta = 50 \text{ then } L = BR$$

⑤

$$\text{propagation delay} = 100 \text{ } \mu\text{s}$$

$$d = 20 \text{ KM}$$

$$L = 1 \text{ KB} = 1024 \times 8$$

$$B = ?$$

$$\text{RTT} = \text{Transmission delay}$$

$$\text{RTT} = 200 \text{ } \mu\text{s}$$

$$\text{Transmission delay} = \frac{L}{B}$$

$$B = \frac{1024 \times 8}{200 \times 10^{-6}} =$$

$$= 40 \text{ mbps}$$

⑥ $B = 1 \text{ mbps}$

$$\text{latency delay (or) propagation delay} = 1.25 \text{ sec}$$

$$L = 1 \text{ KB}$$

$$1. \quad \text{RTT} = 2 \times 1.25$$

$$= 2.5 \text{ sec}$$

$$2. \quad 1 \text{ sec} \quad 1 \times 10^6 \text{ bits}$$

$$2.5 \text{ sec} \quad ?$$

$$\omega_{\text{bits}} = 2.5 \times 1 \times 10^6 \Rightarrow 2.5 \times 10^6$$

$$3. \quad \omega_p = \frac{\omega_{\text{bits}}}{(\text{Pkt size})}$$

$$= \frac{2.5 \times 10^6}{1024 \times 8}$$

$$= 305$$

$$\therefore 2^k = 305$$

$$\therefore k = 9$$

Gate:

window size = 5 packets

Data packets = 1000 bytes

Transmission time for such packets = 50 μ s

Propagation delay = 200 μ s

Throughput = $\frac{1 \text{ Data}}{\text{RTT}}$

RTT = Tr delay + 2 * propagation

$$= \frac{5 \times 1000 \times 8 \times 10^{-6}}{50 + 400 \times 10^{-6}}$$

$$= \frac{5 \times 1000 \times 8}{450 \times 10^6}$$

$$= 88.88 \times 10^{-6} \text{ bits}$$

$$= 11.11 \times 10^{-6} \text{ bytes}$$

9

3960 40



$$\frac{396000}{396000 + 8040} = 98\%$$

$$\frac{8040}{396000 + 8040} = 2\%$$

10



WS = 4 T = 2

At t=0 packets are released at A and immediately available at R.

"0" starts leaving at R.

∴ 123 are in Queue.

$t=1$, 0 arrives at B, ack, is made, meanwhile, 1 starts leaving "R" therefore 2 & 3 are in the queue.

$t=2$, ack, arrives at R and then at A. Therefore 4 is released from A and immediately available at R.

meanwhile arrives at B Hence Ack, is made at the same time 2 starts leaving "R" and therefore 3 & 4 in the queue.

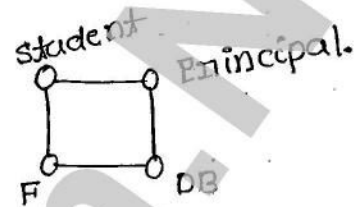
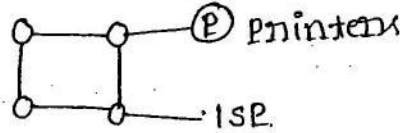
At $t=5$, 6 & 7 are in the queue

$t=10$, 11 & 12 are in the queue.

Lan Technologies

Advantages of LAN:

- * Resource sharing on resource utilization (H/W & S/W).
- * Information sharing.



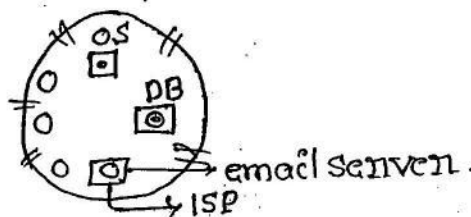
Types of LANs:

- * Dedicated server LAN
- * peer-to-peer-LAN
- * Zero slot LAN.

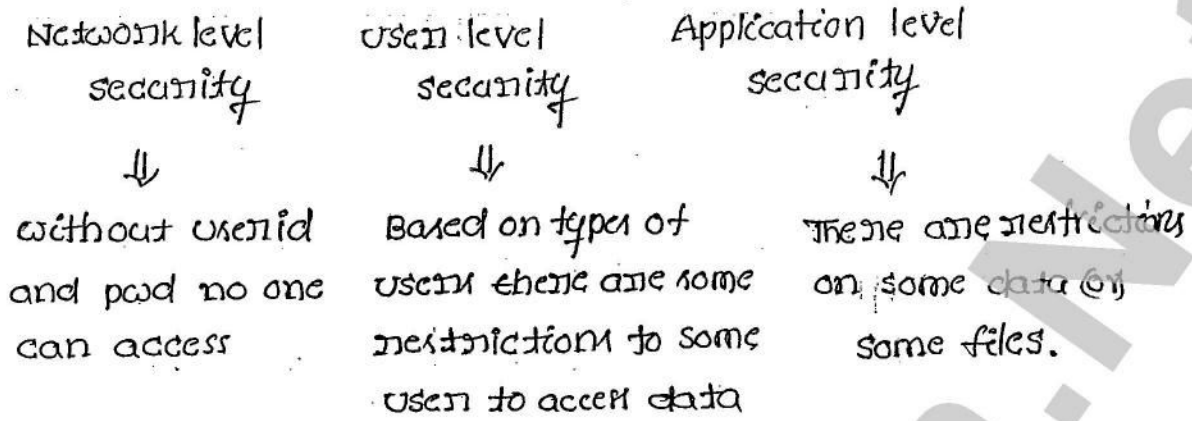
	Dedicated Server LAN	peer-to-peer LAN	zero-slot LAN.
Packet share	80%	10-15%	5%
security	high	Moderate	low
No. of system	Any	50-60	< 10
Application	Any	few	very few
cost	High	Moderate	low
equipment	IP, NOS, NIC	NIC	X

Dedicated Server LAN's:

- * To access OS (or) database, username and password must be submitted and then again to access internet, username, & password are again needed to be submitted provided by ISP



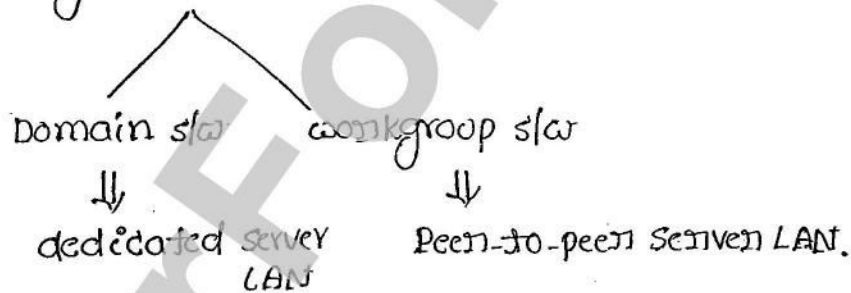
Security is provided at three levels:



Peer-to-peer LAN:

As in dedicated server LAN's, there is no requirement of network operating system and IP address.

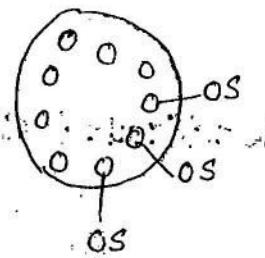
Right click ⇒ TCP/IP



ACE → workgroup



names of individual systems within a workgroup.



communication done among the system within a workgroup.

since, names are being included to individual systems, only a limited no. of systems are get connected, because there is a change of occurring "naming conventions"

workgroup: All the system are being connected for the purpose of communication among them.

* There is no particular leader in the workgroup. All the systems are considered equal. Hence it is called "peer to peer"

Eg: Browsing Centre

(contains no. of clients & only one server)

* They have less security and more flexibility

Zero-slot LAN:

No slot is necessary for connection

NIC is not necessary

Eg: Home networks (using USB ports communication is done)

* No slot is necessary to insert the NIC into mother board.

At the most we get 2-serial

2-parallel

4 USB ports so it has < 10 systems

* No real-time application, and Oracle application are possible.

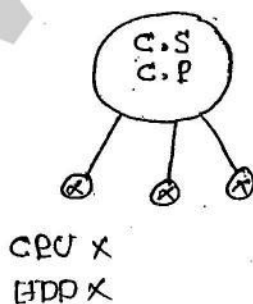
LAN components:

1. Network operating system.

2. cable

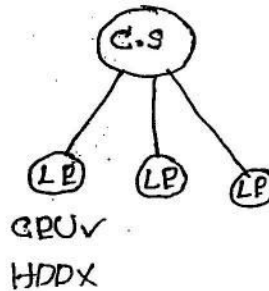
3. NIC

1. First generation



Eg: UNIX

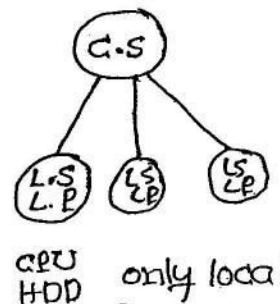
2. second generation



Thin client

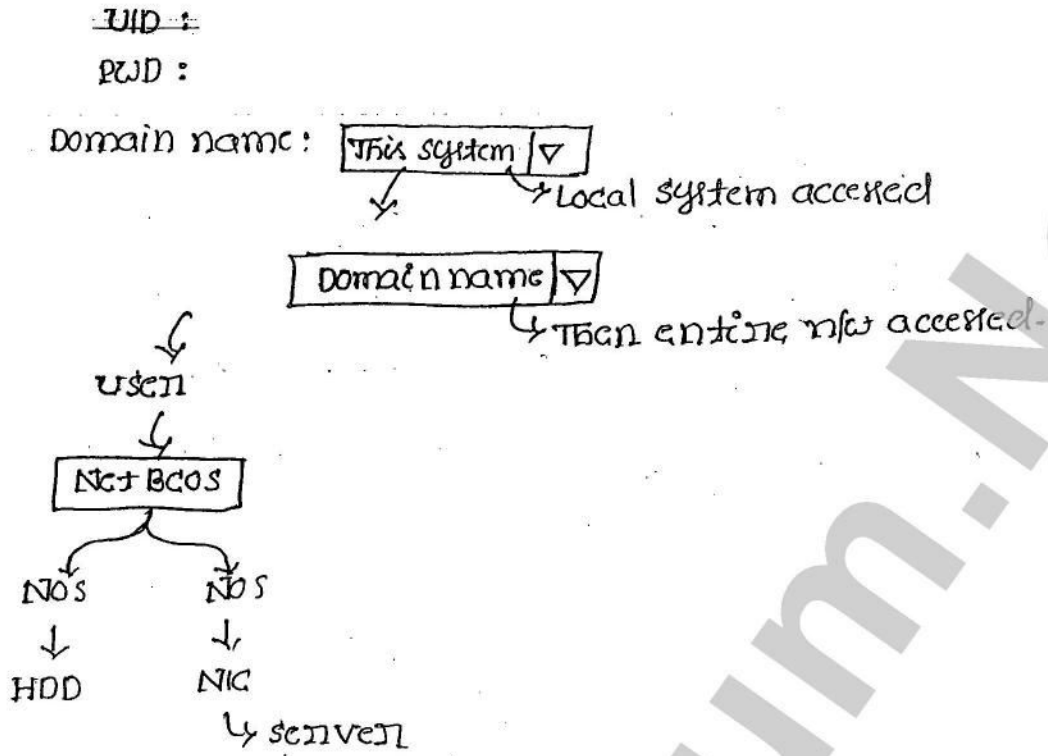
Eg: Novell network

3. Third generation

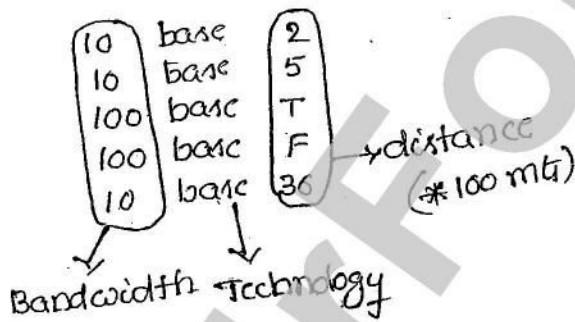


only local Process execute

Eg: windows



2. Cable: Types of cables



Baseband LAN

↳ single type of frequency Base₂ ⇒ up to 200 mt without signal br.

Broadband:

WAN ⇒ Directly connected with
1SP ⇒ Dial-up connection

⇒ All types of frequencies are carried out.

* In cable TV networks "Booster" are used within limited distance.

Broadband 200 mt etc use repeaters.



- * Twisted pair \Rightarrow 100 mts 10mbps
- * Fibre optic cable \Rightarrow 2000 mts
- * 100 base T \Rightarrow category 5 cables \Rightarrow bulky \Rightarrow RJ45 \Rightarrow connectors are used.
- * category 3 \Rightarrow incoming signals. to repost the signals repeaters are used.

3. NIC: Hardware \rightarrow physical layer
 PL + DLC \Rightarrow (combination of HW & SW)
- * New technology systems use the NIC cards.

IEEE 802:

These are exclusively meant for LAN.

Main layers: Transport layer.
 Network layer.

Segmentation & Re-assembly:

- * In local network, no need of it.
- * For LAN's Transport & network layers are not needed.
- * Main focus on data link layer and physical layer.

LLC \Rightarrow framing

Medium Access Control (MAC):

Error control
 flow control
 Access control
 Physical address.

Different types of LAN's for different applications:

For real time applications MAC is replaced with another MAC

IEEE 802.1

• 2. ethernet

• 4 Token bus

• 5 Token ring

• 11 wireless LAN

• 16 wireless MAN

eg: For real time applications

MAC is replaced with 802.5.

Ethernet

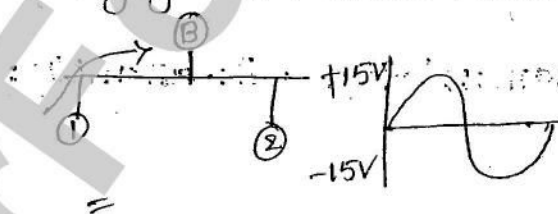
Characteristics:

- * Ethernet offers connectionless communication
- * no flow control & packet level error control
- * no acknowledgment. [either (VE) or (VVE)]
- * it uses bus topology.
- * it uses CSMA/CD as an access control method.

CSMA/CD:

- * The channel, whether communication is taking place or not if there then, wait for the another channel to complete it's transfer of data packet or else transmit the data packet from the current channel only.

- * The channel is sensed in term of voltage if $V=0$ not wave form, i.e no channel is engaged to transfer packet (Media is free)



Multiple Access:

- * if more than one channel, sense the medium, both the channels try to access the medium and want to transfer data packets simultaneously then it is called "multiple access".

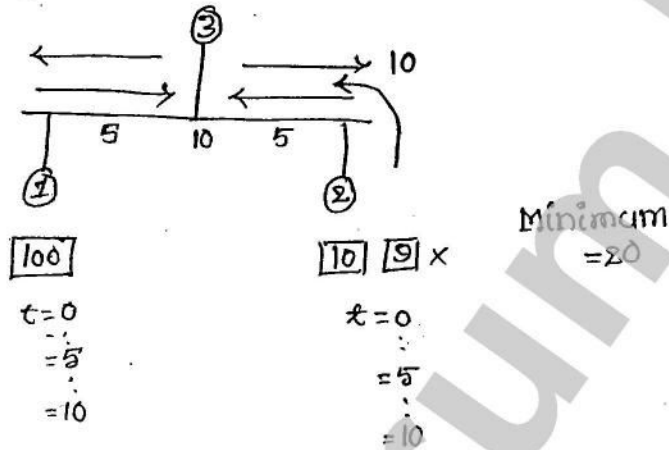
Collision Detection:

At the situation of multiple access of channel collision occurs while transferring data packet. so a JAM signal is used to detect the occurrence of collision and it is sent to both channels



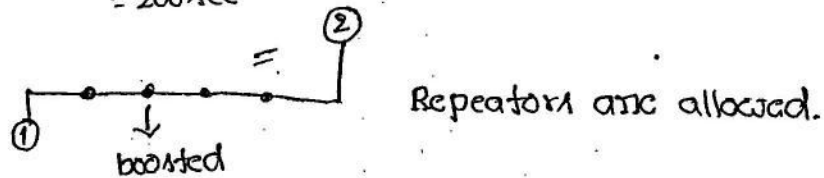
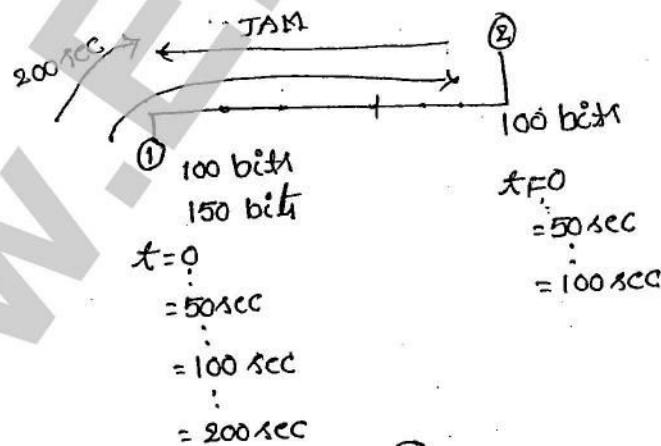
* All channels are having different frequency and jam signals have different frequency so there is no chance of collision occurrence

* Since channels have different frequency \Rightarrow Bandwidth increases.



s_1 system have 100 bits, s_2 have 10 bits assume that they take the channel at a time then collision occurs. for recognizing collision every system maintains min. frame size based on channel length.

RTT = Transmission Delay. $\Rightarrow 2 * \frac{d}{v} = \frac{L}{B}$

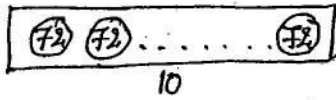


$\Rightarrow 2 * \frac{d}{v} = \frac{L}{B} \Rightarrow 2 * (\frac{d}{v} + 4 * \text{Repeater delay})$

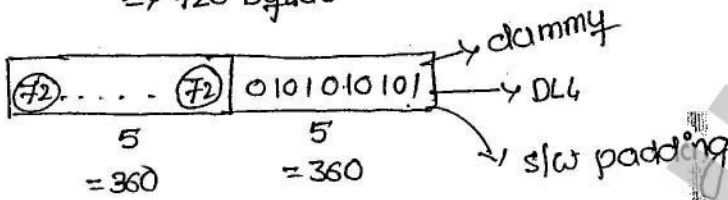
$57.6 \text{ msec} = \frac{L}{10 \text{ Mbps}}$

Basic Ethernet Fast Ethernet Gigabit Ethernet

10 mbps	100 mbps	1 Gbps
2500 mts	250 mts	250 mts
72 bytes	72 bytes	72 bytes
		Basically 25 mts 720 bytes



=> 720 bytes



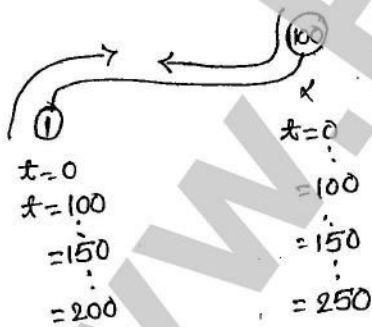
Backoff Algorithm:

it gives waiting time for stations that are involved in collision

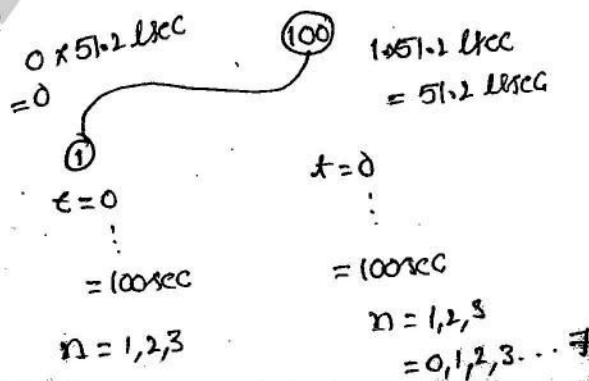
$$\text{waiting time} = K \times 51.2 \text{ usec}$$

where $K \rightarrow$ randomly derived from 0 to $2^n - 1$.

where $n =$ collision number.



Case study:



Let $n=1$
 $k = 0$ to $2^n - 1$
 $\Rightarrow 0$ to $2 - 1$
 $\Rightarrow 0, 1$

Let $n=1$
 $k = 0$ to $2^n - 1$
 $\Rightarrow 0, 1$

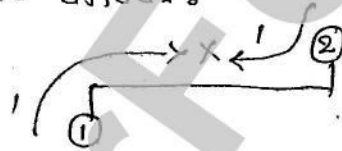
* if let us consider $k=0$ to channel 1 and $k=1$ to channel 100
 then waiting time of channel 1 = $0 \times 51.2 \mu\text{sec}$
 $= 0$

waiting time of channel 2 = $51.2 \mu\text{sec}$.

∴ so channel 1 have waiting time = 0, then it can transfer the data packets but channel 2 must wait up to $51.2 \mu\text{sec}$.
 then again back-off algorithm is applied to proceed the transmission through either channel 2 or others.

Limitations of Back-off Algorithm:

capture effect:



Let $n=1$
 $\therefore k = 0$ to $2^n - 1$
 $= 0, 1$

Then $WT = 0 \times 51.2 \mu\text{sec}$
 $= 0$

$n=2$
 $k = 0$ to $2^n - 1$
 $= 0$ to $2^2 - 1$
 $= 0, 1, 2, 3$

Then repeat same for $n=3$

$k = 0$ to $2^3 - 1$
 $\Rightarrow 0, 1, 2, 3, 4, 5, 6, 7$

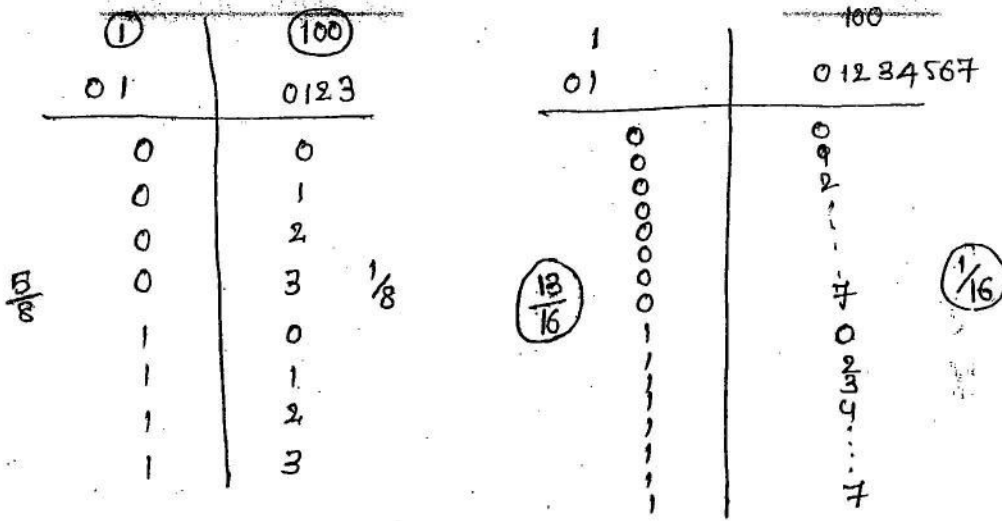
Let $n=1$
 $k = 0$ to $2^n - 1$
 $= 0, 1$

$WT = 1 \times 51.2 \mu\text{sec}$
 $= 51.2 \mu\text{sec}$

∴

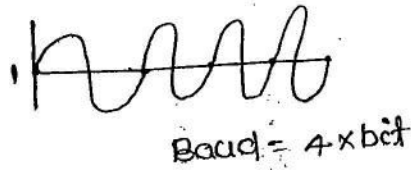
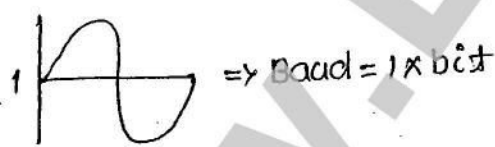
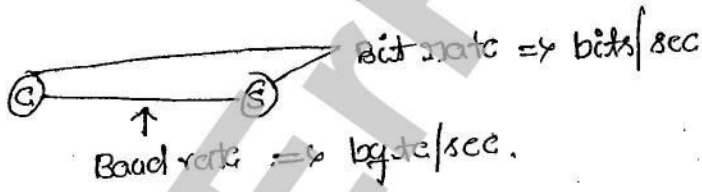
if after $51.2 \mu\text{sec}$ again another channel also wants to access the medium, then again backoff algorithm is applied then $n=2$.

Then the probability of channel 1 and channel 100 to access the



Data Specifications:

- ① Data rate
 - ⇒ 10 mbps
 - ⇒ 100 mbps
 - ⇒ 1 gbps
- ② signal : :
 - Manchester Encoded signal
- ③ Addressing system
 - 48-bit physical address.



Ethernet:

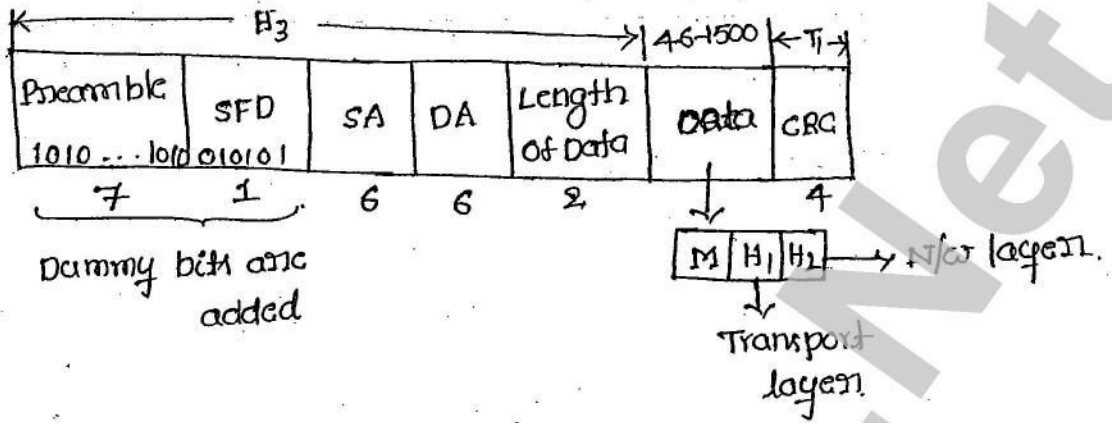
For a 10mbps ethernet

Baud rate = 20 megabaud.

For a 100 mbps ethernet

Baud rate = 200 megabaud.

Frame format of 802.3:



Preamble:

- * it contains continuously 1's & 0's for seven bits
- * it is used for synchronization purpose.

SFD (Start of frame Delimiter)

- * its signal actual start of the frame.

* Dummy bits are represented by preamble 1111010111 ① 1010101010 ②

Source & Destination Address:

- * They are 48 bit physical addresses representing source & destination.
- * Maximum size of data = 46 bytes, so that we make out 76 byte frame.
- * Maximum size of data = 1500 bytes. → To avoid monopolization
 1/
 change given to other channels also.

Min	Max
46	1500
72	1526 → frame
64	1518 → frame from source address

⇒ Preamble & SFD are neglected

Length of the data:

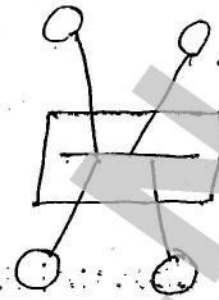
since data is varying from 46 to 1500 to keep track correct size of the data in packet we need "length of data" field

CRC:

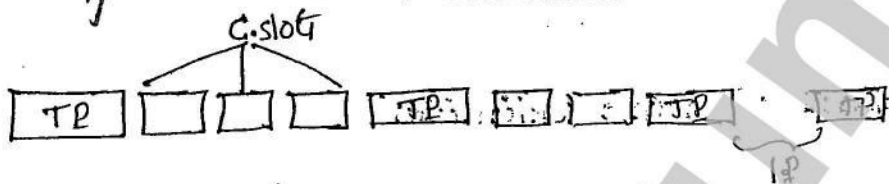
- * it is added only at the tail to identify bit errors
- * To avoid more no. of transmissions we use CRC at tail end.

Implementation :

Physical Addressing - Star topology
 Logical Addressing - Bus topology



Efficiency calculation of Ethernet:



T.P = Transmission period

C.P = collision period

I.P = ideal period

$$\text{Efficiency} = \frac{T.P}{T.P + C.P + I.P} \quad (\because I.P = 0)$$

$$\eta = \frac{T.P}{T.P + C.P}$$

Let $N \rightarrow$ total no. of systems in network.

$P_s \rightarrow$ Probability of a station to transfer data pkt

$1 - P_s \rightarrow$ Probability of a station not to transfer data pkt

- * To get a successful transmission for a station remaining $(N-1)$ station shouldn't transfer the data pkt.

- * $(1 - P_s)^{N-1} \Rightarrow$ Probability for the remaining $(N-1)$ stations not to transfer data packets.

- * $P_s (1 - P_s)^{N-1} \Rightarrow$ Probability of the success for a single station.

$$\boxed{N P_s (1 - P_s)^{N-1} = A} \Rightarrow \text{it is the probability of success}$$

for any arbitrary station among "N" stations

$$\text{No. of contention slots} = 1/A$$

$$\text{if } N \rightarrow \infty \Rightarrow A = 1/e$$

$$\text{No. of contention slots} = 1/A = 1/(1/e) = e$$

Contention period (C.P):

$$= \text{No. of contention slots} * \text{slot duration}$$

$$\text{C.P} = e * \Sigma \text{prop delay}$$

$$\text{Transmission period} = L/B$$

$$\eta = \frac{T.P}{T.P + C.P}$$

$$= \frac{L/B}{L/B + 2 * \frac{d}{v} * e}$$

$$\boxed{\eta = \frac{1}{1 + \frac{2dBLe}{LV}}}$$

$$\uparrow \eta = \frac{1}{1 + \frac{2dBLe}{LV}}$$

* if load increases, efficiency decreases

* if pkt size increases, then efficiency also increases

$$1. \quad \eta = \frac{TP}{TP + CP} = \frac{t_{\text{trans}}}{t_{\text{trans}} + 2t_{\text{prop}} * e}$$

$$= \frac{1}{1 + 2 \frac{t_{\text{prop}}}{t_{\text{trans}}} * e}$$

$$= \frac{1}{1 + 2 \frac{d}{v} * e} = \frac{1}{1 + \frac{2dBLe}{LV}}$$

$$\begin{aligned}
 2. \quad \eta &= \frac{T \cdot P}{T \cdot P + C \cdot P + \epsilon_{prop}} \\
 &= \frac{T_{trans}}{t_{trans} + 2 \cdot \epsilon_{prop} \cdot C + \epsilon_{prop}} \\
 &= \frac{1}{1 + 2 \cdot \frac{\epsilon_{prop}}{t_{trans}} \cdot C + \frac{\epsilon_{prop}}{t_{trans}}} \\
 &= \frac{1}{1 + 2aC + a}
 \end{aligned}$$

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

Every station must be waited for one fraction time, after collision aborting bits from collision point

Advantages of Ethernet:

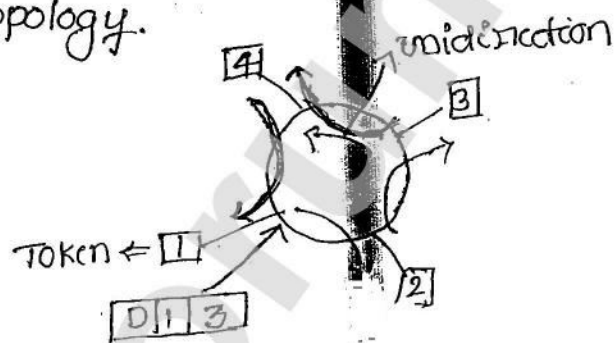
- * cost of ethernet is less.
- * Ethernet cables are robust to noise
- * simple operation

DisAdv:

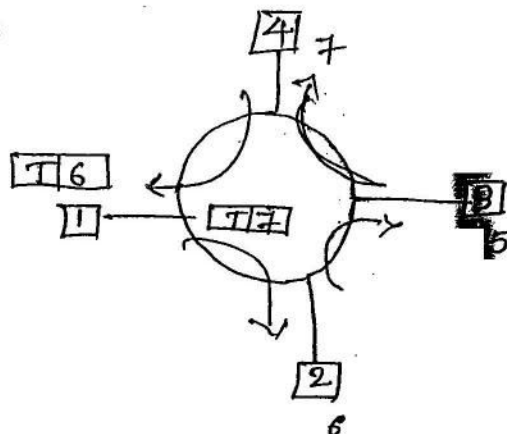
- * Ethernet offers non-deterministic service. Therefore, it is not suitable for real time applications.
eg: CNC machines.
- * There are ~~no~~ no priorities in ethernet. Therefore not suitable for client server applications.
- * There is a restriction on the min size of pkt, Hence it is not suitable for interactive applications.
eg: interactive applications needs 1 or 2 bytes.
eg: ATM.
- * if load increases, efficiency decreases.

Token Ring

- * characteristics:
- * it also offers connectionless communication
- * it uses piggybacking acknowledgement system
- * No restriction on number, min size of data, priorities are possible & deterministic service is possible
- * it uses token-passing system as an access control method and ring topology.

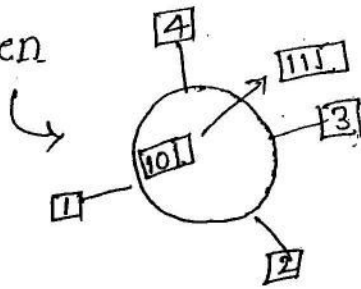


- * Token is maintained at one station only then the data is sent to the other station, where it checks for source & destination. if same then it copies only and then sent the original data to other, therefore there is no collision.
- * if any station have high priority than the token or equal then it can access and low priority cannot access. therefore no collision



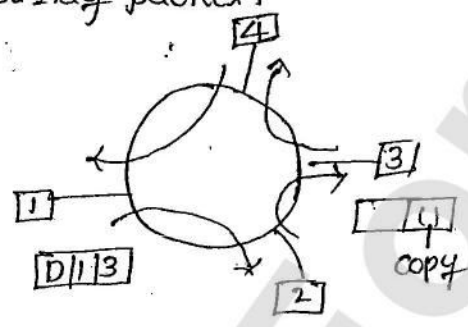
Problems with Token Ring::

1. (A) vanished Token
- (B) corrupted token



2. Source: partial packet is produced and the station is recognized by this partial pkts.

- A. Orphan packets
- B. stray packets



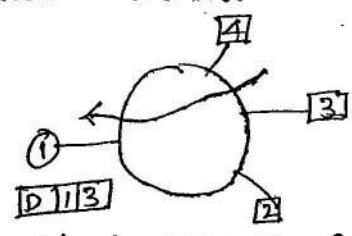
orphan pkts => source trash rotate upto a time
 stray pkts => no one understands.

(C) Monopolization (Single man rule)

one station is being accessed whole the time which cause problem to other stations.

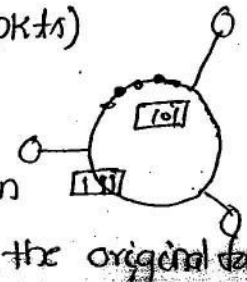
3. Destination:

- A. safe operation
- B. Busy destination (not able to copy the frame)
- C. crash destination (drop the pkts)



4. Ring:

- * Major cut in the ring, stops operation
- * unhealthy token => no station can use the original token



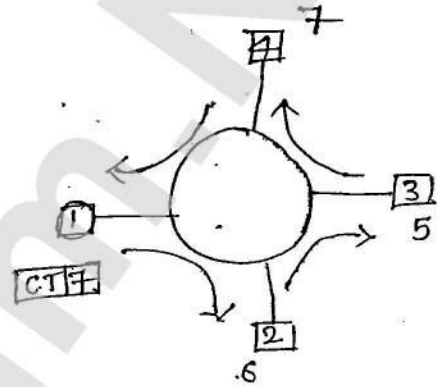
To overcome token ring problems we have the following

1. Token Holding Time:

one station must release the token within 10 msec of time, it can transfer 20,000 pkts in 10 msec.

2.1. Monitor: (leader for the ring)

To become a monitor station it must release claim token, &c based on the priority of station



Minimum TRT = Propagation delay in the ring

100 μ sec

+
No. of active stations * Delay at each station.

Maximum TRT = Propagation delay in the ring

+
No. of active stations * Token holding time

* Monitor station expect this token within these 200sec. if packet not arrives then it reproduce the token thinking that it is missing so it solves the vanishing token problem
it also waits for 200sec more than 200sec.

* corrupted tokens are recorrected by the monitor station within the 2nd cycle no other station have 3-bits.

2. Orphan:

when a packet crosses a monitor station it makes a cron stamp on it stamped pkts are not allowed.

3. Stray:

checking the validity of pkt while crossing the monitor station.

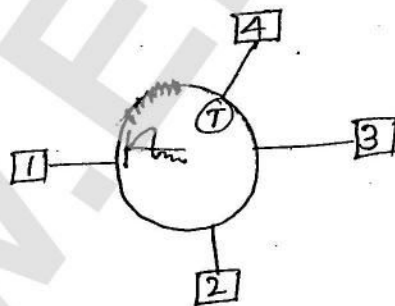
3. Destination:

Two fields 'A' & 'C' are attached:

A=0	C=0	
A=1	C=0	=> Safe
A=1	C=0	=> Destination Busy
A=0	C=1	=> Destination not available but frame is copied & represents that other than destination station has copied the frame.

4. Ring: A special frame is introduced.

* if continuously produce the pkt and no response, then cut it & produce a wave form at $t=0$ and if response within $t=1 \mu\text{sec}$ then it identifies as the major cuts



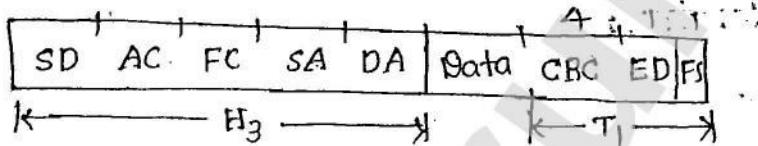
* A special frame is sent by monitor station for every 10 sec saying that it is available, if any frame is not received then it is assumed that monitor is crashed and there is a change of other stations to become a monitor station.

Specifications of Token Ring:

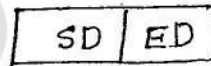
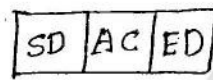
1. Data rate
 - * 4 Mbps
 - * 16 Mbps
2. signal: DME
3. Addressing system: 48-bit physical Address.

Frame format:

1. Data frame

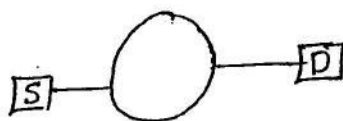
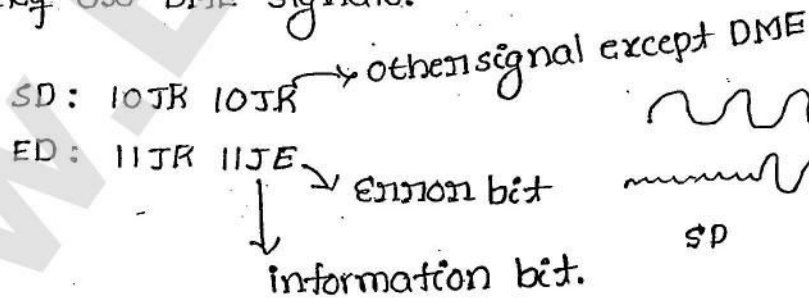


2. Token frame
3. Abort frame

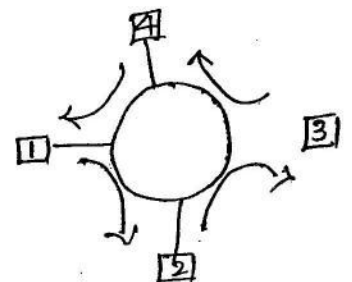


Start Delimiten } used to indicate two extreme ends of
End Delimiten } the packets.

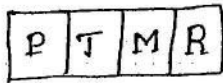
* They use DME signals.



1=0
2=0
3=0
4=1 ⇒ it gives indication of ending on last packet



if E=1 then it simply transfer



M \Rightarrow Monitor bit

T=0 \Rightarrow Data

=1 \Rightarrow Token

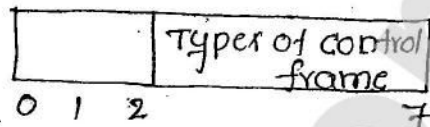
M=0 \Rightarrow Before crossing monitor bit

=1 \Rightarrow After crossing the monitor bit

\Downarrow
if again requested it just eliminate it.

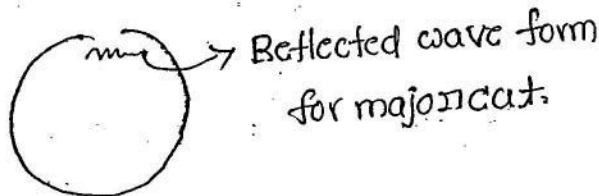
Frame control:

6 types of frame control

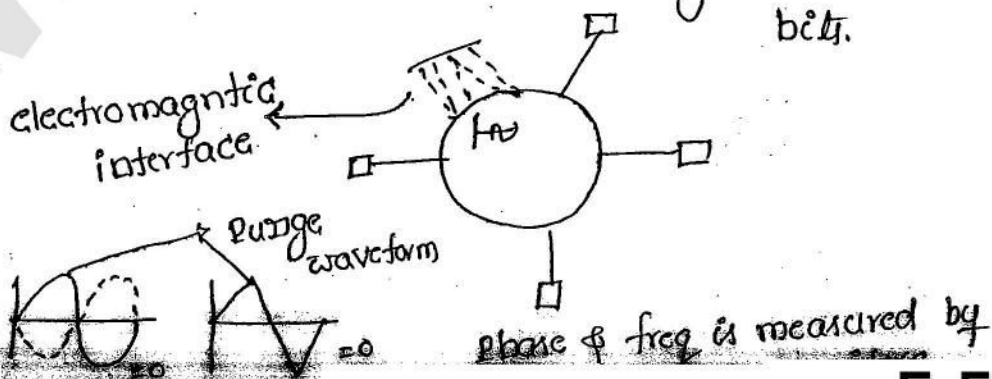


00 - Data
11 - Control.

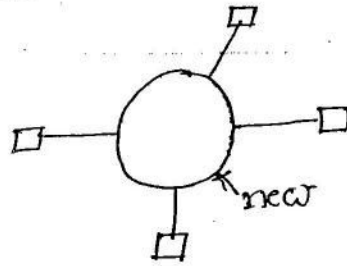
1. client token: it is used in the election process of monitor.
2. Active monitor presence:
it is issued by the monitor in equal intervals to make its presence.
3. Beacon: it is used to identify major cut in the ring.



4. purge frame: it is used to clear the ring from unwanted bits.



5. Duplicate Address test frames:



SD AC, DAT 4 11...11 Data CRC, ED FS

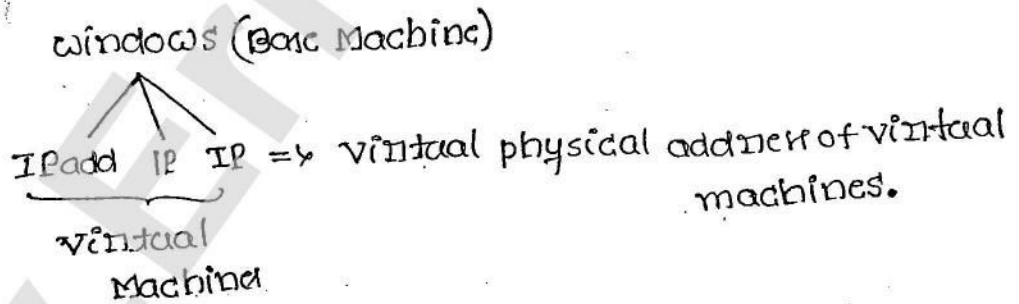
⇓
new channel address.

⇓
it is being checked with all other channel who have this address. if any other address is given & again checked.

* In specific conditions only DAT is used not for all the physical addresses.

Virtualisation:

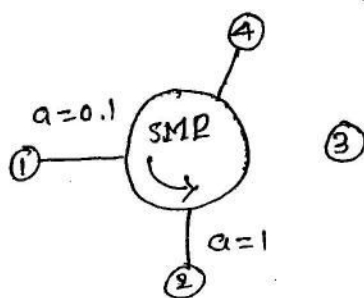
* "DAT" case is only used in virtual physical addresses.



* Proxy physical addresses are also considered under the "DAT"

6. SMP (Standard by Monitor preserve):

it is used to carryout neighbour identification.



Upstream: From whom the channel receives the data packet.

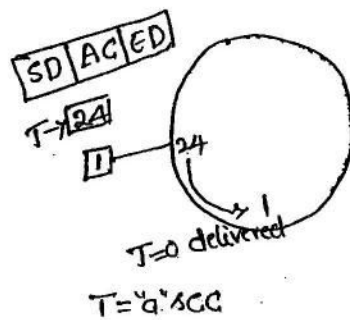
Down stream: To whom the

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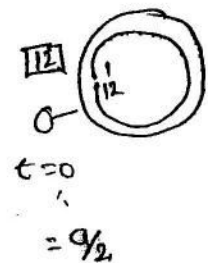
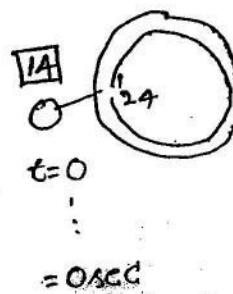
Modes of operation:

- * Transmission mode
- * Listen mode
- * Receiving mode
- * Bypass mode.

calculation of minimum size of token ring:



Min token rotation time



coming back

$$T_{prop} = "a" \text{ sec} = \text{trans. delay}$$

$$T_{prop} = T_{trans} \Rightarrow \frac{T_{prop}}{T_{trans}} = 1$$

$$\frac{T_{prop}}{T_{trans}} = 1 \Rightarrow \text{min.}$$

$$> 1 \Rightarrow \text{Max}$$

$$< 1 \Rightarrow \text{overlap.}$$

* Token rings used for big networks.

* Token ring: propagation delay = Transmission delay

$$\frac{d}{v} = \frac{L}{B}$$

eg: $v = 2 \times 10^8 \text{ m/sec}$

$$L = 24 \text{ bits}$$

$$B = 4 \text{ mbps}$$

$$d = ?$$

$$\Rightarrow \frac{d}{v} = \frac{L}{B}$$

$$d = \frac{L}{B} * v$$

$$= \frac{24 \times 2 \times 10^8}{4 \times 10^6}$$

$$d = 1.25 \text{ km.}$$

eg: if Bandwidth of ring is 10 mbps, frame size 200 bits
velocity = $2 \times 10^8 \text{ m/sec}$ find min size token ring.

$$d = \frac{L}{B} * v$$

$$= \frac{200 \times 2 \times 10^8}{10 \times 10^6}$$

$$= 4000 \text{ km.}$$

=

$$\text{Ring latency} = \text{Min TRT} = \text{Propagation delay in the ring} + \text{no. of active stations} * \text{Delay at each station}$$

$$\text{Propagation delay} = \frac{d}{v} + mb$$

\downarrow \downarrow
 sec bits

$$\text{Bit delay } b = \frac{1}{B}$$

d = Total length of ring
 B = bit delay at each station

$$L = \frac{d}{v} + \frac{mb}{R} \text{ sec}$$

$$RL = \frac{dR}{v} + mb \text{ bits}$$

v = velocity propagation
 R = bandwidth of ring
 L = latency.

Various Token re-insertion strategies:

Delayed Token strategies

* Token is released after getting entire data packet.

* Efficiency is low

* Reliability is high

* it is used under load condition.

$$\text{cycle time} = (a + b + c + d) \text{ sec}$$

where $a \rightarrow$ data transmission

$b \rightarrow$ Ring latency

$c \rightarrow$ Token Transmission time.

$d \rightarrow$ propagation delay b/w station.

Early Token strategy.

* Token is released after data is transferred.

* Efficiency is high

* Reliability is low.

* it is used under high load conditions.

$$\text{cycle time} = (a + c + d) \text{ sec}$$

Problems:

① $d = 2500 \text{ mts}$

$$V = \frac{60}{100} \times 3 \times 10^8 \text{ m/sec}$$

$$B = 10 \text{ mbps}$$

$$L = ?$$

RTT = Transmission delay

$$2 \times \frac{d}{V} = \frac{L}{B}$$

$$L = 2 \times \frac{d}{V} \times B$$

$$= \frac{2 \times 2500 \times 10 \times 10^6}{1.8 \times 10^8}$$

$$= 277 \text{ bits.}$$

②

$$B = 1 \text{ Gbps}$$

$$d = 1 \text{ km}$$

$$V = 200,000 \text{ km/sec}$$

$$L = ?$$

$$\frac{L}{B} = 2 \times \frac{d}{V}$$

$$L = 2 \times \frac{1}{200,000} \times 1 \times 10^9$$

$$= 10,000 \text{ bits or } 1250 \text{ bytes.}$$

③

$$B = 10 \text{ Mbps}$$

$$\text{Propagation delay} = \frac{d}{V} = 225 \text{ bit times}$$

$$\text{Bit delay} = \frac{1}{B} = \frac{1}{10 \times 10^6} = 0.1 \mu\text{sec}$$

Transmission can be considered as either 1 bit delay or 0.1 μsec

at $t=0$ A & B started their communication.

At $t = 225 \frac{1}{2}$ there is a collision

At $t = 225$ the A & B will come to know about the collision

Assume 'A' started producing jam signal.

∴ At $t = 273$ ($225 + 48$), A finishes producing jam signals
 \Downarrow
 Jam.

④

⑤

$B = 10 \text{ Mbps}$

slot time = $51.2 \text{ } \mu\text{scc}$

$L = 512 \text{ bytes}$

No. of slots = 1.716

$$\eta = \frac{T \cdot P}{T \cdot P + C \cdot P + I \cdot P} \quad (I \cdot P = 0)$$

Transmission time = L/B

$$= \frac{512 \times 8}{10 \times 10^6}$$

$$= 40 \text{ } \mu\text{scc}$$

C.P = no. of slots * slot time

$$= 1.716 \times 51.2$$

$$= 87.8 \text{ } \mu\text{scc}$$

$$\eta = \frac{40 \times 10^{-3}}{40 \times 10^{-3} + 87.8 \times 10^{-6}}$$

⑥

$$B = 10 \text{ Mbps}$$

$$d = 2.5 \text{ km}$$

$$v = 2.3 \times 10^8 \text{ m/sec}$$

$$L = 128 \text{ bytes} \Rightarrow \boxed{98+30}$$

$$\text{overhead} = 30 \text{ bytes}$$

$$\eta = \frac{1}{46444} = \frac{1}{1+6.44(0.10)} \therefore \alpha = \frac{T_{\text{prop}}}{T_{\text{tran}}} = \frac{1.086 \times 10^{-5}}{1.024 \times 10^{-4}} = 0.10$$

$$= 57\%$$

$$T_{\text{prop}} = \frac{d}{v} = \frac{2.5 \text{ km}}{2.3 \times 10^8} = \frac{2.5 \text{ km}}{2.3 \times 10^5 \text{ km/sec}} = 1.086 \times 10^{-5}$$

$$T_{\text{tran}} = \frac{L}{B} = \frac{128 \times 8}{10 \times 10^6} = 1.024 \times 10^{-4}$$

⑦

$$B = 4 \text{ Mbps}$$

$$\text{Total holding time} = 10 \text{ msec}$$

$$4 \text{ bits} - 1 \text{ sec} - 4 \times 10^6$$

$$? \quad 10 \text{ msec} - 10 \text{ msec} - ?$$

$$\text{longest frame} = 4 \times 10^{+5} \times 10^{-3}$$

$$= 4 \text{ Kbps.}$$

⑧

$$R = 4 \text{ Mbps}$$

$$m = 20$$

$$d = 20 \times 100$$

$$b = 2.5 \text{ bits}$$

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

$$RL = \frac{dR}{v} + mb$$

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

$$= 80 \text{ bits}$$

$$R = 16 \text{ Mbps}$$

$$m = 80$$

$$RL = \frac{dR}{v} + mb$$

$$= \frac{80 \times 100 \times 16 \times 10^6}{2 \times 10^8} + 80 \times 2.5$$

$$= 840 \text{ bits}$$

10

station 1 = m

$d = m \times 100$

$b = 8 \text{ bits}$

$L = 1250 \text{ bytes}$

$R = 25 \text{ Mbps}$

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

Transfer time = $\frac{L}{B} = \frac{1250 \times 8}{25 \times 10^6}$

= 400 μsec

$\frac{RL}{\text{Tr.time}} = 1$

$\Rightarrow 3.33 \times 10^{-3} m = 1$

$m = 300$

1 sec $\rightarrow 25 \times 10^6$

? 1250×8

$\Rightarrow \frac{1250 \times 8}{25 \times 10^6} =$

$RL = \frac{d}{v} + \frac{mb}{R} \text{ sec}$

$= \frac{m \times 200}{2 \times 10^8} + \frac{m \times 8}{25 \times 10^6}$

$= m \left(\frac{200}{2 \times 10^8} + \frac{8}{25 \times 10^6} \right)$

~~---~~

11

$m = 32$

$L = 1000 \text{ bit packet}$

$B = 10 \text{ Mbps}$

latency/adapter = 2.5 bit

$d = 50 \text{ mtr}$

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

Data transmission = $a = \frac{L \cdot D}{B}$

$= \frac{1000}{10 \times 10^6} = 10^{-4}$

$b = RL = \frac{d}{v} + \frac{md}{R} \text{ sec}$

$= \frac{32 \times 50}{2 \times 10^8} + \frac{32 \times 2.5}{10 \times 10^6}$

$= 880 \times 10^{-8}$

A. Early token = $(a + c + d) \text{ sec}$

$= 10^{-4} + 24 \times 10^{-7} + 25 \times 10^{-8}$

= 10 msec

Token transmission = $c = \frac{LT}{B} = \frac{24}{10 \times 10^6}$

$= 24 \times 10^{-7}$

Propagation delay = $d = \frac{d}{v}$

$= \frac{50}{2 \times 10^8}$

$= 25 \times 10^{-8}$

B. Delay token = $(a + b + c + d) \text{ sec}$

$= 10^{-4} + 880 \times 10^{-8} + 24 \times 10^{-7} + 25 \times 10^{-8}$

= 11 msec

2) it is heavily loaded, we suppose to use early token strategy

$$G.T = a + c + d \quad a = \frac{L_D}{B} = \frac{256}{10 \times 10^6} = 2.56 \mu\text{sec}$$

$$d = 1 \text{ km}$$

$$B = 10 \text{ Mbps}$$

$$L = 256 \text{ bits}$$

$$\text{bit delay} = 8 \text{ bits}$$

$$\text{Propagation speed} = 200 \text{ m}/\mu\text{sec}$$

$$a = \frac{L_T}{B} = \frac{8}{10 \times 10^6} = 0.8 \mu\text{sec}$$

$$d = \frac{200 \text{ m}/\mu\text{sec}}{2 \times 10^8 \text{ m}/\text{sec}} = 2.5 \mu\text{sec}$$

$$G.T = 26.5 \mu\text{sec}$$

$$26.5 \mu\text{sec} \Rightarrow 224$$

$$1 \text{ sec} = \frac{224}{26.5 \times 10^{-6}} = 8.5 \text{ Mbps}$$

$$\eta = \frac{8.5}{10} \times 100 = 85\%$$

3) $d = 200 \text{ km}$
 $B = 100 \text{ Mbps}$
 $V = 200,000$
 $L = 1024 \text{ bytes}$

$$a = \frac{L_D}{B} = \frac{1024 \times 8}{100 \times 10^6} = 81.24 \mu\text{sec}$$

$$b = R_L = \frac{d}{V} + \frac{mb}{R} \text{ sec} \quad \left(\because \frac{mb}{R} = 0 \right)$$

$$= \frac{200 \text{ km}}{200,000} = 1 \text{ msec}$$

$$c = \frac{L_{\text{token}}}{B} = \frac{24}{100 \times 10^6} = 0.24 \mu\text{sec}$$

$d \Rightarrow$ not given ignore

$$1.08 \text{ msec} - 1024 \times 8 \text{ bits}$$

$$1 \text{ sec} - ?$$

$$= \frac{1024 \times 8}{1.08 \times 10^{-3}} = 7.6 \text{ mbps}$$

$$\eta = \frac{7.6 \text{ mbps}}{100 \text{ Mbps}} \times 100 = 7.6\%$$

$$G.T = a + b + c$$

$$= 81.24 \mu\text{sec} + 1 \text{ msec} + 0.24 \mu\text{sec}$$

$$= 1.08 \text{ msec}$$

(14)

propagation speed = 200 m/ μ sec

$$B = 1 \text{ Mbps}$$

$$\text{bit delay} = \frac{1}{B} = \frac{1}{10 \times 10^6} = 1 \mu\text{sec}$$

$$1 \mu\text{sec} \Rightarrow 200 \text{ mts}$$

$$B = 40 \text{ Mbps}$$

$$\text{Bit delay} = \frac{1}{B} = \frac{1}{40 \times 10^6} = 0.025 \mu\text{sec}$$

$$1 \mu\text{sec} - 200 \text{ mts}$$

$$0.025 \mu\text{sec} - ?$$

$$\Rightarrow 200 \times 0.025$$

$$= 5 \text{ mts.}$$

(15)

$$B = 5 \text{ Mbps}$$

propagation speed = 200 m/ μ sec

$$\text{bit delay} = \frac{1}{B} = \frac{1}{5 \times 10^6} = 0.2 \mu\text{sec}$$

$$1 \mu\text{sec} - 200 \text{ mts}$$

$$0.2 \mu\text{sec} - ?$$

$$\Rightarrow 200 \times 0.2$$

$$\Rightarrow 40 \text{ mts.}$$

$$=$$

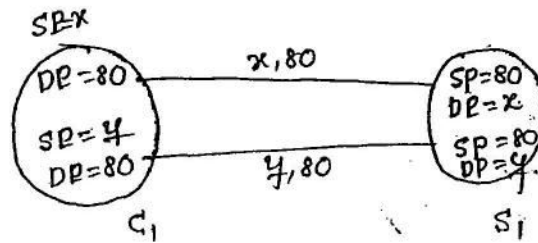
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Transmission Control Protocol.

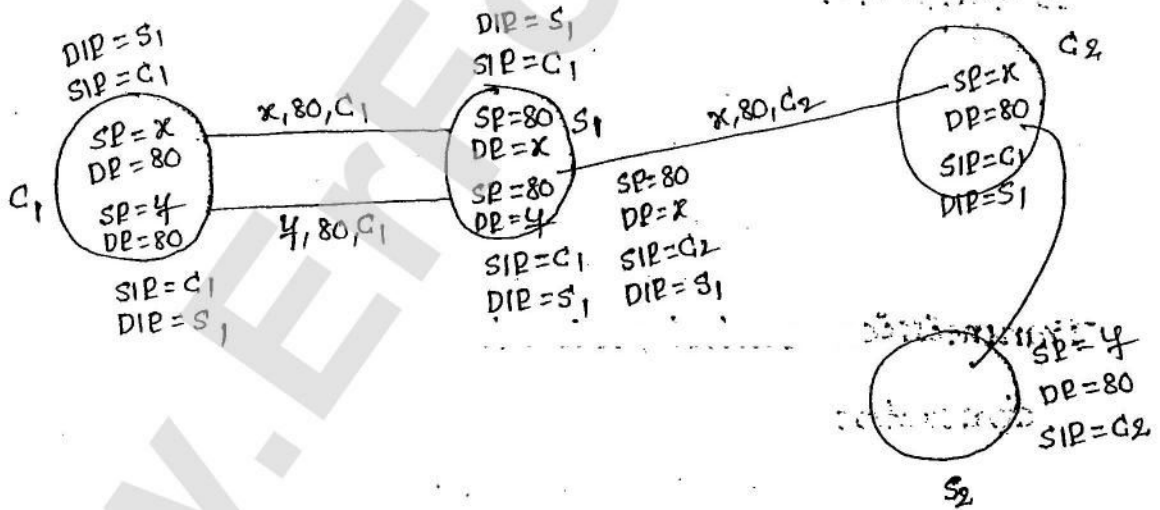
why we call network as TCP/IP networks

(on)

Relationship between TCP & IP.



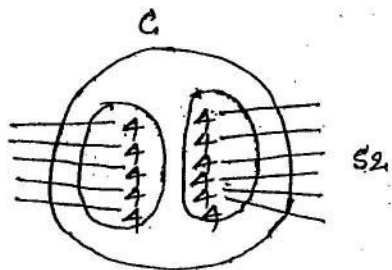
- * $ctrl + N \Rightarrow$ another connection.
- * while establishing a new connection source port & destination port is addressed.
- * server can handle more no. of clients, so introducing a new client involves both source port and destination port.



- * In order to differentiate the destination IP address either for server₁ (or) server₂, the DIP addressed. where "80" represent the http and sip represents the user's own address.
- * The source port & destination ports are handled by TCP protocol in the Transport layer. \Rightarrow 2 ports parameters.
- * The source IP & destination IP are handled by IP protocol

4 parameters } source port
 Destination port
 source IP
 Destination IP

Socket:



* socket is a logical component which groups a set of parameters for communication.

$$5 \times 4 = 20$$

$$6 \times 4 = 24$$

$$5 \times 2 = 10 + 2 = 12$$

$$6 \times 2 = 12 + 2 = 14$$

Advantages:

* Resource utilization

* Maintenance & Administration

→ Allows certain num-of connections for socket.

Transmission control protocol:

characteristics:

* it is reliable, byte oriented, point-to-point, transport



layer protocol.

connection-oriented.

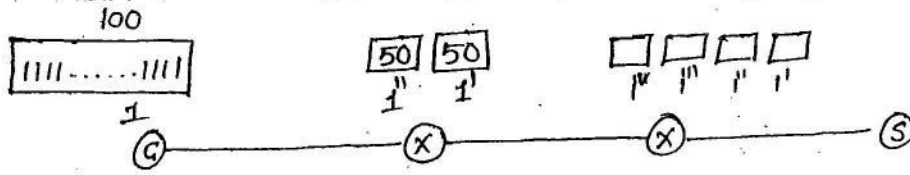
* Message oriented (UDP)

* Packet oriented (SWP)

* Byte oriented (TCP)

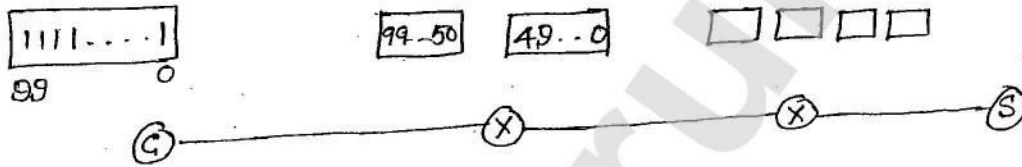
* Bit-oriented (HDLC)

Byte-oriented (TCP) / stream oriented



Burden on intermediate routers.

- * The bits are splitted by half to router, as it can handle only certain set, so it have a problem to exactly split, (if not, there is a chance of missing a packet)



- * The above problem can be overcome by not splitting the pkts but just only dividing the stream of bits and transfer so it has no burden on the intermediate routers.
- * since the bits are transferred as byte (or) a stream, the TCP is considered as "stream oriented".
- * TCP uses "cumulative acknowledgment".
- * The connections are "full duplex". Therefore it is having two half-duplex connections.

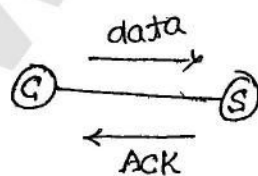


fig: full duplex

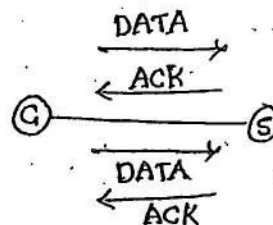


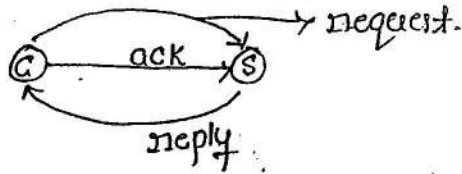
fig: Half duplex.

- * TCP, uses, sliding window protocol (swp) for its flow control. Therefore each TCP connection has 4 windows.



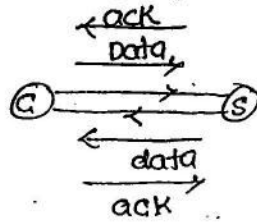
* TCP connections are having 3 phases.

1. connection establishment phase: (negotiation phase)

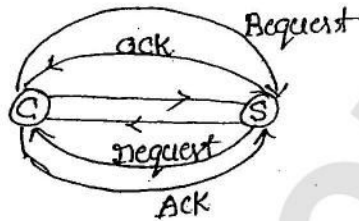


* it is a single step process.

2. Data transmission phase



3. connection termination phase.

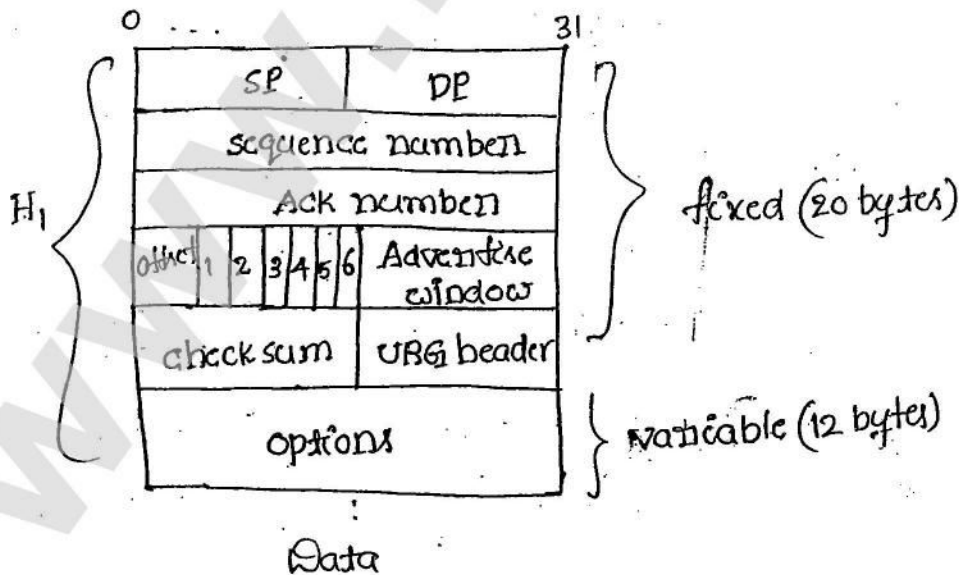


* it is not single step process even though it requested

for connection termination. The termination must done in both the systems (not only one)

TCP operations:

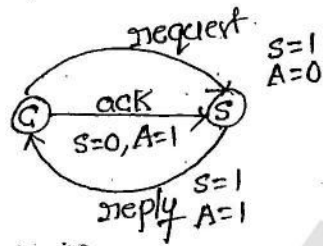
TCP header:



Flags:

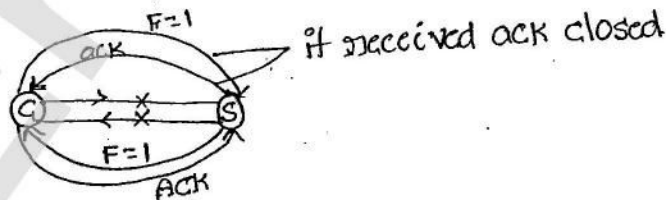
1. SYN & ACK
2. FIN
3. RST
4. PSH
5. URG

1. SYN & ACK: SYN & ACK flags are used in connection establish phase of different request and reply packets

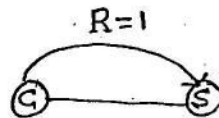


- S=1 A=0 => Request
- S=1 A=1 => Reply
- S=0 A=1 => ACK
- S=0 A=0 => Data

2. FIN : it is used in connection termination phase.



3. RST Flag: it is used to reset the connections.

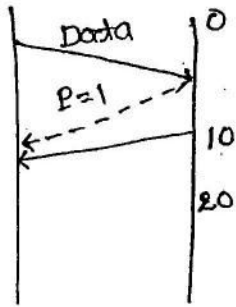


* New connection is established just refreshing the window

* while transferring the data if any problem arises, then the complete connection is cancelled and a new connection is made, so it is not suitable for every time to have new connection. so we use RST. to reset.

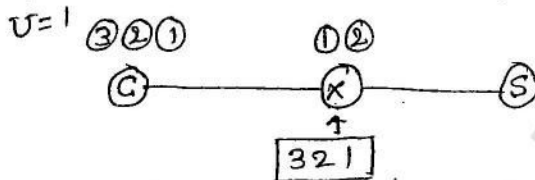
4. PSH flag:

* it is used for high priority packets to push the packet to upper layer without waiting for time interval.



if the data is transferred and of high priority it needs a fast ack from sender. so by using PSH flag sender sends the ack fastly as soon as data reaches without waiting for the time interval it have.

5. URG: it is used take care of "out of band rate"



sequential order

Intend: 1, 2, 3

output: 3 2 1

↓
using URG Pkt.

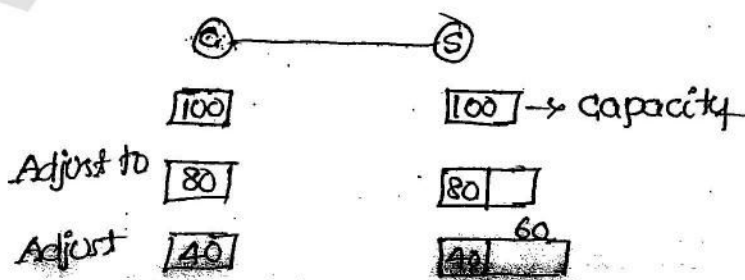
* URGENT pointer indicates the amount of the data that is important in the packets

* it is valid only if URG=1

* if the packets 1 2 & 3 sent to sender by representing URG=1 to 3 packet then instead of 1 and 2, 3 packet is reached firstly, it represents an URGENT packet.

Advertise window:

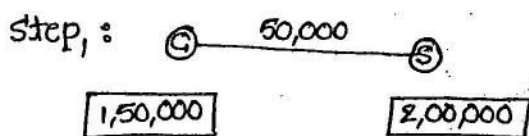
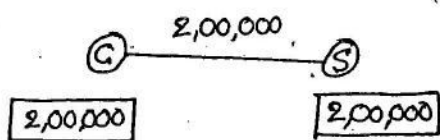
* it is used to implement flow control



- * If the server have no empty space, it can't take further data. so it must represents that it has no empty space, but it is difficult to send to each and every client. so we can solve this problem as follows.
- * At $t=0$, client sends the packet and server have no empty space so simply discard it. it checks for all the time intervals.
- * if suppose server have empty space, at particular time interval, then it can accept the data packet.

Silly window syndrome (sWS):

- * when sWS occurs efficiency is "0".
- * There are three reasons for silly window syndrome.
 - > when server announces it's empty space is "0"
 - > when client is able to generate only one byte at a time.
 - > when server consumes only one byte at a time.
- * Always ensure to transfer only one byte among all the bytes of data in order to reduce viruses.
- * Always server needs to consume only one byte transfer. possible to reduce the sWS value, so that efficiency increase.



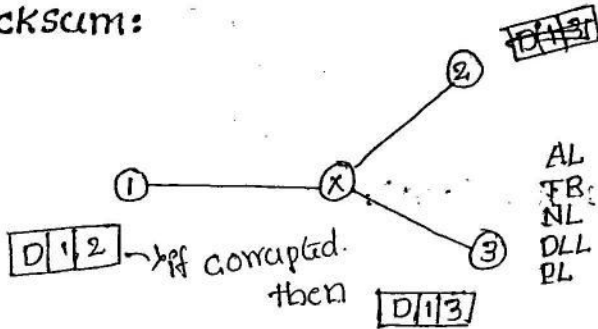
- * There is a chance of sending all the 1,50,000 bits, at both the sides but we cannot transfer them because only 50,000 bits are allowed to be transferred.

* if empty space in the server is more than 2^{16} then use scale factor in the - "option" field.

eg: if empty space is 1,50,000 advertisement window = 50,000
 & scale factor = 3.

eg: if empty space = 1,00,000 & advertisement window = 50,000
 then scale factor = 2

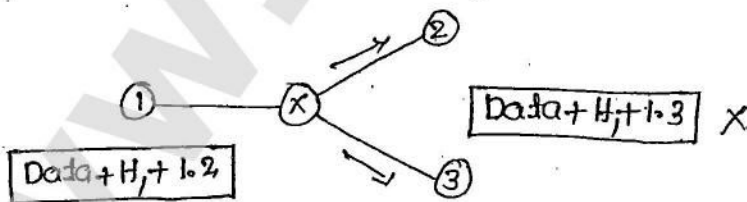
checksum:



* The transport layer transfers the corrupted data to the application layer. Then application layer recognizes as a corrupted bit.

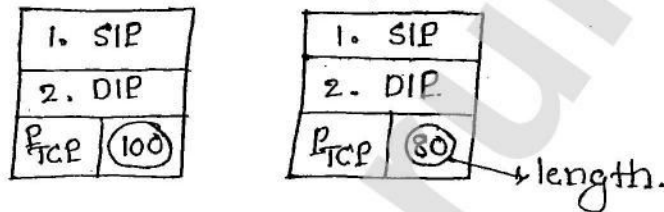
* so in order not to have burden on application layer, before a packet reaches application layer, it gets corrected at the transport layer by including the concept of "checksum".

* checksum includes "data + H₁ + pseudo-header" in its calculation.



* calculate the checksum $c_c(\text{Data} + H_1 + (1.2))$ at the sender side and send the checksum. at the receiver the checksum is again calculated. $[\text{Data} + H_1 + (1.3)] \rightarrow$ corrupted, so discard it by transport layer.

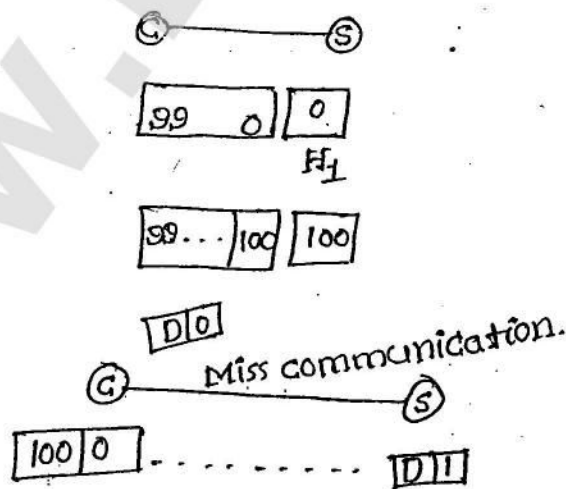
- * Pseudo-header is used to check whether data packet has been received by the correct destination or not.
- * it is prepared at the source and included in checksum calculations.
- * once packet is received by the destination again it is prepared by the destination with destination values.
- * if incoming checksum is similar to calculated checksum, then packet is consumed, else it is discarded.



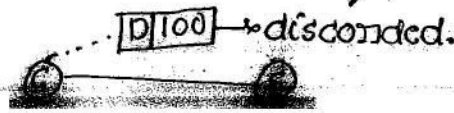
- * sequence no. for the packet is first data byte sequence num. in the packet.

characteristics for Sequence num & Ack:

1. sequence no. for the packet is first data byte sequence num. in the packet.



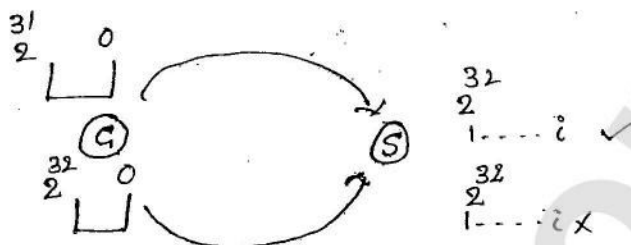
TCP uses random initial sequence number.



* The sequence numbers always not start with 0 if selects a random number among the set of $0-2^{32}$ and then sends the random number. if corrupted random num generates simply discard it.

3. Get randomly sequence numbers:

if two different packets of the same sequence num are generated simultaneously then the server thinks that one is the duplication of another packet and just discard one of the packet, which leads to a loss of packet which is different from the other packet.



* In order to overcome this problem, increases the value from 2^{32} to 2^{64} .

Eg: $0 \xrightarrow{1.536 \text{ Mbps}} 0$

1 sec = 1.536×10^6 bits \Rightarrow 1st pkt

1 sec = $\frac{1.536 \times 10^6}{8}$ bytes \Rightarrow 2nd packet

1 sec = $\frac{1.536 \times 10^6}{8}$ sequence no. of 2nd packet

1.536 Mbps - 64 hrs

10 Mbps - 57 min

100 Mbps - 6 min

1.2 Gbps - 28 sec.

Eg: consider bandwidth of link = 100 mbps
 sequence no. field = 24 bits

Find the "wrap around" of sequence numbers

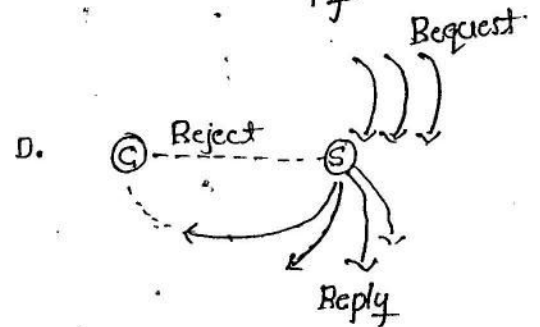
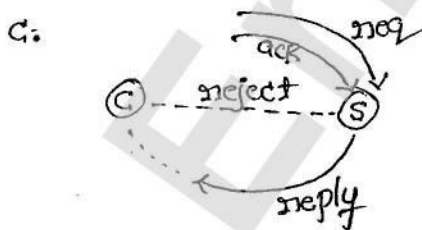
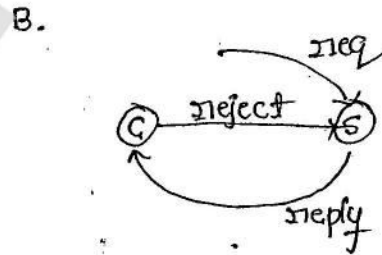
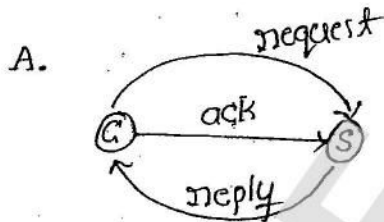
$$\hookrightarrow 1 \text{ sec} = 100 \times 10^6 \text{ bits}$$

$$= \frac{100 \times 10^6}{8} \text{ bytes}$$

$$1 \text{ sec} = \frac{100 \times 10^6}{8} \text{ sequence num's}$$

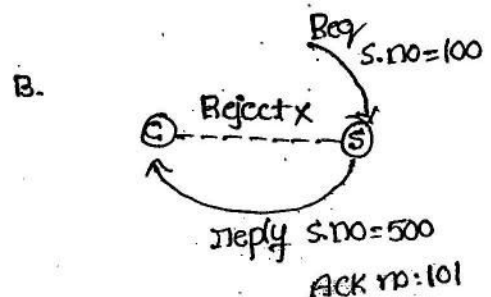
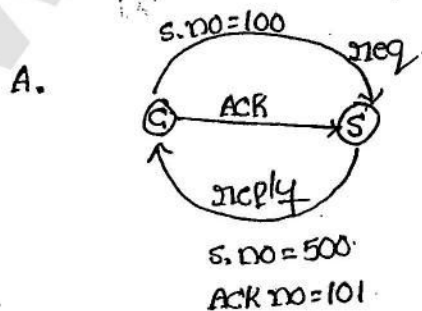
$$? \quad 2^{24} \Rightarrow \frac{2^{24} \times 8}{100 \times 10^6} = 1.3 \text{ sec.}$$

Applications:

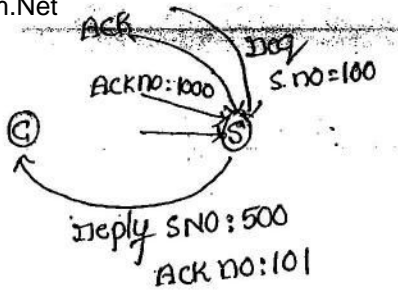


Denial of service attack:

* server denies the client to server for a certain time interval
 To overcome above problem we use "sequence numbers".



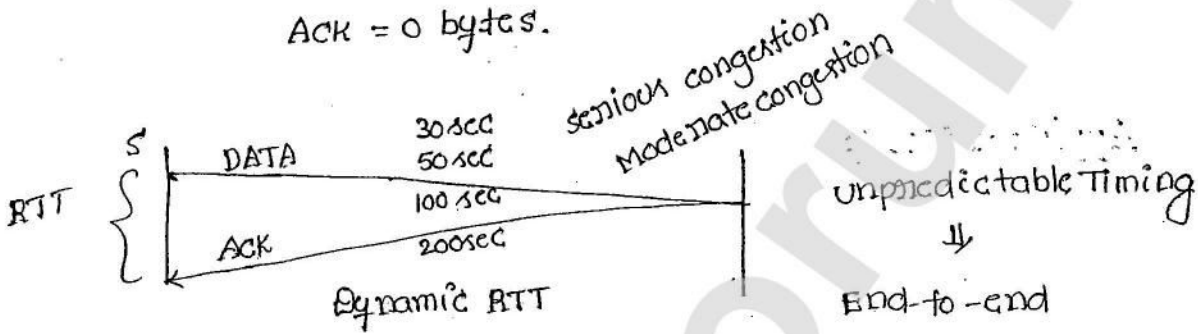
C.



Hacker can't know the random num generated by server, so he guess simply rejected. if guess = exact no then the server cannot generate huge ack so it discard to send ack

syn } 1 byte
FIN }

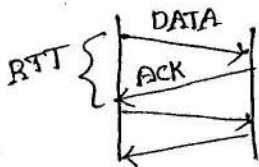
ACK = 0 bytes.



End-to-end

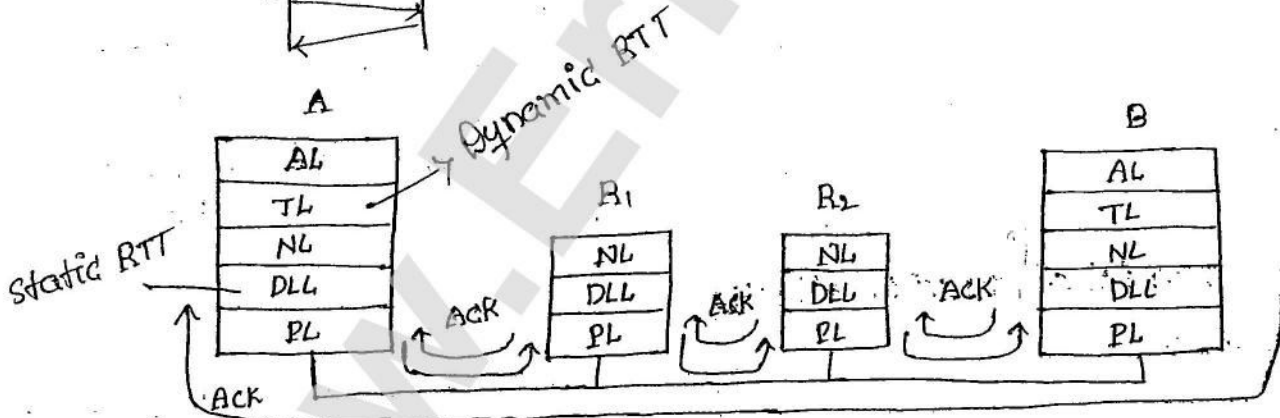
Link-to-link

Predictable Timing.



A

B



For the calculation of RTT we have certain Algorithms.

1. Basic Algorithm:

(IRTT) initial Round Trip Time = 30 sec

(NRTT) New Round Trip Time = 40 sec

$$\alpha = 0.9$$

$$\begin{aligned} 2. \text{ Estimated RTT} &= \alpha \text{IRTT} + (1-\alpha) \text{NRTT} \\ &= 0.9 \times 30 + (1-0.9) 40 \\ &= 31 \text{ sec} \end{aligned}$$

$$\begin{aligned} \text{Time out (T}_0) &= 2 \times \text{ERTT} \\ &= 2 \times 31 \\ &= 62 \text{ sec} \end{aligned}$$

$$\begin{aligned} \textcircled{3} \quad \text{IRTT} &= 31 \text{ sec} \\ \text{NRTT} &= 50 \text{ sec} \end{aligned}$$

$$\begin{aligned} \text{ERTT} &= \alpha \text{IRTT} + (1-\alpha) \text{NRTT} \\ &= 0.9 \times 31 + (1-0.9) 50 \\ &= 32.9 \text{ sec} \end{aligned}$$

$$\begin{aligned} \text{Time out} &= 2 \times \text{ERTT} \\ &= 2 \times 32.9 \\ &= 65.8 \text{ sec} \end{aligned}$$

$$\begin{aligned} \textcircled{4} \quad \text{IRTT} &= 32.9 \text{ sec} \\ \text{NRTT} &= 45 \text{ sec} \end{aligned}$$

$$\begin{aligned} \text{ERTT} &= \alpha \text{IRTT} + (1-\alpha) \text{NRTT} \\ &= 0.9 \times 32.9 + (1-0.9) 45 \\ &= 34.11 \end{aligned}$$

$$\begin{aligned} \text{Time out} &= 2 \times \text{ERTT} \\ &= 2 \times 34.11 \\ &= 68.22 \text{ sec} \end{aligned}$$

2. Jacobson's Algorithm:

IRTT = 30 sec

NRTT = 40 sec

$$\alpha = 0.9$$

initial deviation (D_i) = 5

$$\begin{aligned} \textcircled{2} \text{ New Deviation } (D_N) &= |IRT - NRTT| \\ &= |30 - 40| \\ &= 10 \end{aligned}$$

$$\begin{aligned} \text{estimate deviation } (D_E) &= \alpha D_i + (1-\alpha) D_N \\ &= 0.9 \times 5 + (1-0.9) \times 10 \\ &= 5.5 \end{aligned}$$

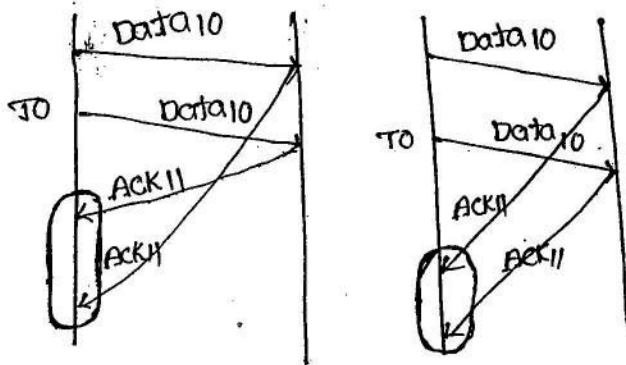
$$\begin{aligned} ERTT &= \alpha IRT + (1-\alpha) NRTT \\ &= 0.9 \times 30 + (1-0.9) \times 40 \\ &= 31 \end{aligned}$$

$$\begin{aligned} \text{Time out} &= 4 * D_E + ERTT \\ &= 4 \times 5.5 + 31 \\ &= 53 \end{aligned}$$

$$\begin{aligned} \textcircled{3} \quad IRT &= 31 \\ NRTT &= 50 \\ \alpha &= 0.9 \\ D_i &= 5.5 \\ D_{New} &= |31 - 50| \\ &= 19 \\ D_E &= 0.9 \times 5.5 + (1-0.9) \times 19 \\ &= 6.2 \\ ERTT &= 0.9 \times 31 + 0.1 \times 50 \\ &= 32.9 \\ TO &= 4 * D_E + ERTT \\ &= 4 \times 6.2 + 32.9 \\ &= 57 \end{aligned}$$

* "Time out" is high under Basic algorithm then the Jacobson Algorithm.

Korn's Algorithm:



* if there is a time out, there is a possibility to receive 2 data packets

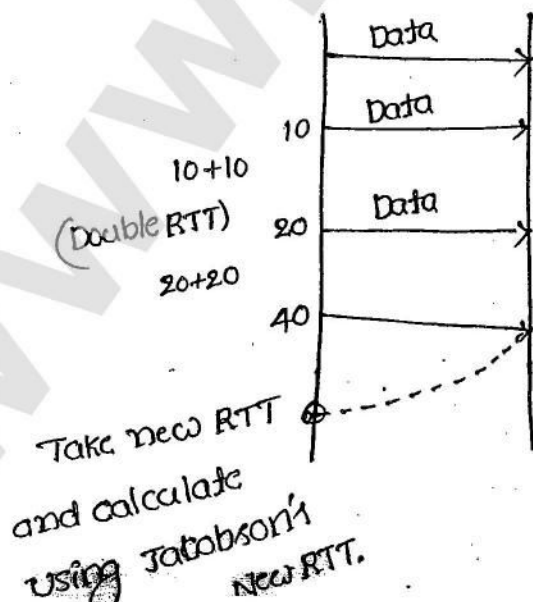
1. From original packet
2. From re-transmitted packet.

* Then there is an ambiguity that which ACK must be considered for next calculation.

∴ Therefore, Korn's has resolved this ambiguity by proposing the follows.

⇒ For every timeout double the timeout for the next transmission and continue this till to get a proper ACK.

⇒ Then we will go back to the Jacobson's algorithm.



State Transition Diagram:

Need for state transition dig:

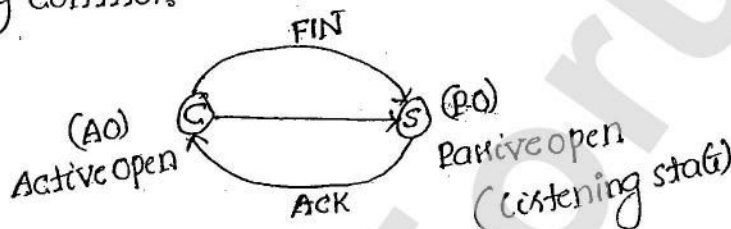
* To evaluate any protocol, we use one of the following 2 methods.

1. Get the specification of the protocol develop a software for it, integrate with network operating system and evaluate it's features. It is time consuming process and costly.

2. Get the specification of protocol develop state transition dig for it and then evaluate it's features.

it is a shortcut method but not so powerful.

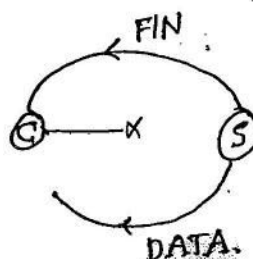
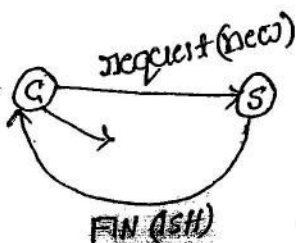
Dialog control:



Purpose of Time wait state:

* As soon as Acknowledgement is generated client will go to time wait state instead of closed state by suspecting problem with acknowledgment. Even if ACK is lost and "FIN" server is re-transmitted.

* It is treated for some connection as client is maintaining the connection in timed-wait state. If there is no such state, then retransmitted FIN is treated for new connection.



- * Server wants to terminate the connection by using FIN/ But it denies, then before the time exceeds client wanted to establish a new connection, so it requests a new connection to server. After the request reaches server, client gets the FIN.
- * Then client thinks, that it is the termination of newly established connection which is not True.
- * In order to avoid such confusion, "Sequence numbers" are given to FIN at the same time, some time it is also allotted for "FIN" which is called as "Time wait state".

Limitations of state Transition dig:

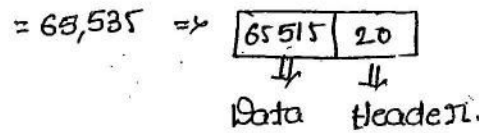
- * Error control procedures are not depicted in the dig
- * Re-transmission are not shown in the dig.

Nagle's Algorithm: (Used in wide Area Network):

- * checking the server that it works connect or not, using remote system for checking it send characters.
- * one character is sent at a time which cause silly window syndrome efficiency reduces drastically, for checking 100 characters, one at a time for which RTT is more for character typing.
- * At such conditions, add the header for all the bits & send it which improves performance.
- * But it is not applicable in LAN technologies, but support WAN tech because of RTT and typing speed capabilities
- * Round trip time is less but input speed is high.

Problem:

① Maximum data = 64 KB



② RTT = 30 msec

$\alpha = 0.9$

NRTT = 26

Basic algorithm = $\alpha (RTT) + (1-\alpha) (NRTT)$

= $0.9 \times 30 + (1-0.9) (26)$

= ~~29.6~~ 29.6 msec

T.O = $2 * 29.6 = 59.2$ msec

$D_{new} = |30 - 26| = 4$

DE = $0.9 * 4 + (0.1) * 4$

= 4

T.O = $4 * D.E + ERTT$

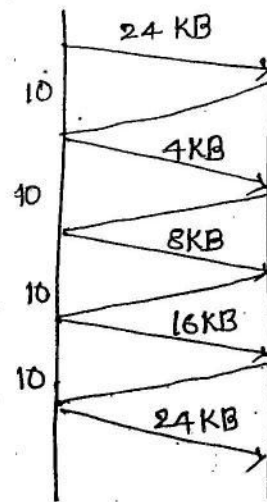
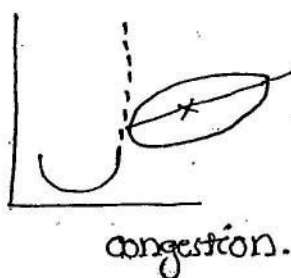
= $4 * 4 + 29.6$

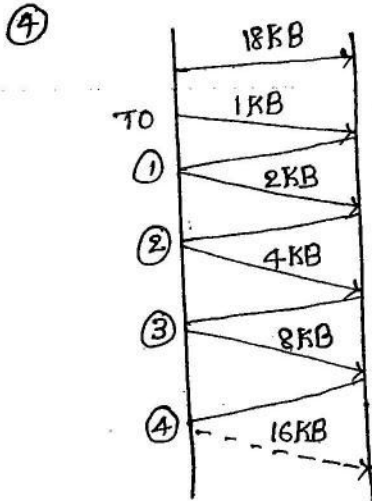
= 45.6 msec.

③



After 40 ms a full window is transmitted





After time out occurs go back to

window size = 1

Ans: 16KB.

⑤. TCP uses SWP

$$\begin{aligned} \text{Throughput} &= \frac{1 \text{ window}}{\text{RTT}} \\ &= \frac{65,535 \times 8}{20 \text{ msec}} \\ &= 26.5 \text{ mbps} \end{aligned}$$

$$\begin{aligned} \eta &= \frac{26.5 \text{ mbps}}{1 \text{ Gbps}} \\ &= 2.6\% \end{aligned}$$

⑥. Transform data unit

Total numbers available = $2^8 = 256$ and they should be consumed in 30sec.

$$\text{Data rate per connection} = \frac{128 \times 8 \times 256}{30}$$

⑨ Probability of sequence of 1 random number falls in 10^6

$$\text{total no} = 2^{32}$$

$$= \frac{10^6}{2^{32}} = 2.3 \times 10^{-4}$$

⑩ A typist can type 600 characters per minute i.e. to type a character takes 100 msec.

case:1: since RTT is very much less than typing speed, this algorithm cannot be implemented.

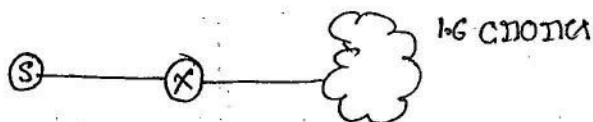
case:2: Since RTT is exactly equivalent to typing speed, this algorithm cannot be implemented.

if RTT is more than 200ms then we are able to implement this algorithm Nagle's algorithm.

User Datagram Protocol [UDP]

Need for UDP;

For multicasting and broad casting applications, TCP can't be used Hence we need UDP.



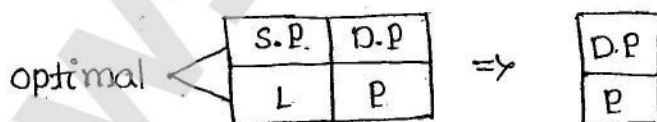
Server communicates to 16 node systems in TCP, we require 16 node connections which cannot support a huge connection.

* Applications that requires constant data flow, cannot use TCP, Hence UDP is being used
eg: Rocket launching.

* Applications that requires bulk data transfer cannot use TCP. Regulated flow in TCP, fluctuated flow in UDP.

* Applications that requires fastness than reliability cannot use UDP.

Since UDP is a connectionless, many fields in TCP are not needed in it.



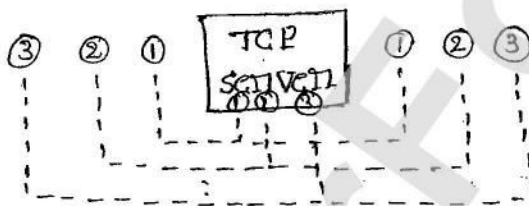
TCP

- * connection-oriented
- * slow
- * Reliable
- * overhead is high
- * HTTP, FTP, SMTP, Telnet are used

Applications

- * web applications
- * Mail, BSA applications

concurrent process:



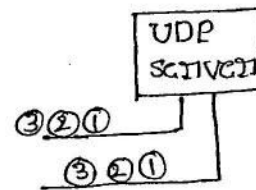
UDP

- * connection less
- * fast
- * un-reliable
- * overhead low.
- * DNS, TFTP, NFS, SNMP, multimedia & Realtime.

Applications.

- * Name transfer Application
- * Network management applications
- * Multimedia & Realtime appl's

iterative process:



- * TCP and UDP port numbers are different.

Domain Name System (DNS):

- * it is using UDP, its purpose is to keep track computers and services in a network environment.

It has four applications

- > Name Translation
- > Host Aliasing
- > Mail Aliasing
- > Load balancing.

It is using 4 types of servers

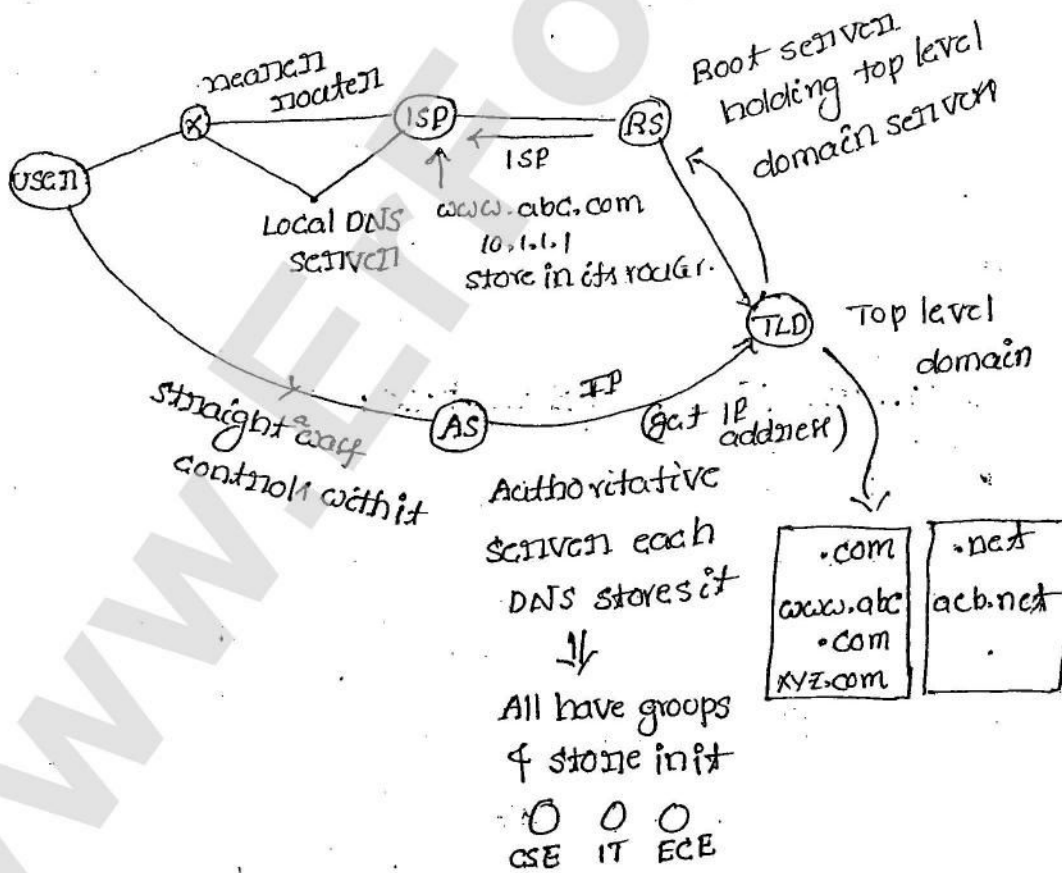
- Root name server
- Top-level domain server
- Authoritative server
- Local DNS server.

* it uses distributed database to perform its applications.

Information about computers and services are stored in these servers in terms of resource records.

* Each resource records consists 5 attributes.

- Name
- TTL
- class
- type
- value.

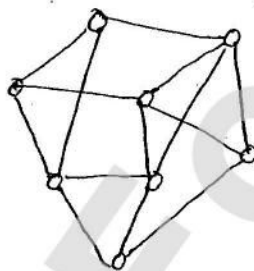


Fault gateway: IP routers specifies the default address, suggesting address that it can't know particular address.

* once a request for a website is sent, it gets stored in ISP and the local router, whenever again a request of same website is sent then there is no need to visit all the servers, and there is only subsequently request simply it shows the website without visiting root server.

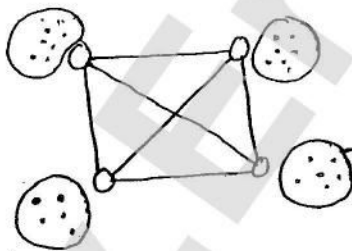
* Mainly 70% of internet services are provided by ISP & local routers.

* when top level Domain servers maintained in terms of clusters. These are connected with MESH topology. it helps to improve efficiency and reliability.



Mesh connection

Rs maintains many routers and get connected not only single router.



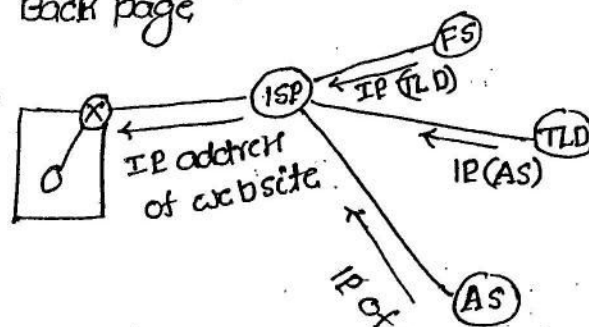
TLD server in the form of clusters and distributed data base.

* it will help for fast searching [effective usage].

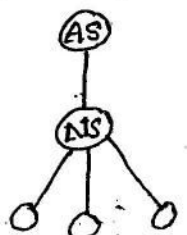
=> Getting IP address can be done in two ways.

1. Back page

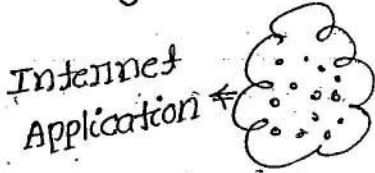
2.



NS: Name Server



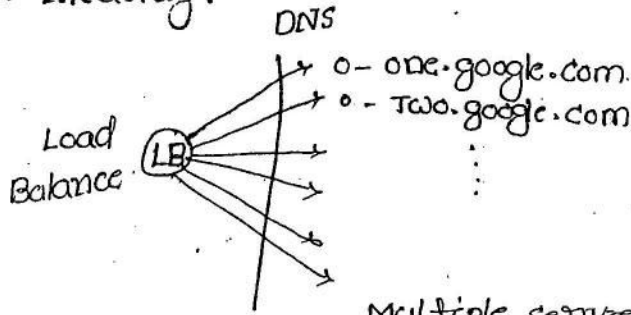
⇒ Giving names to systems in a network (Host aliasing):



L1. one.abc.com ⇒ by seeing name, one can understand position of the system.

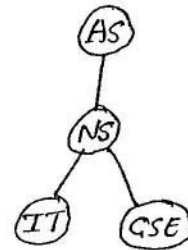
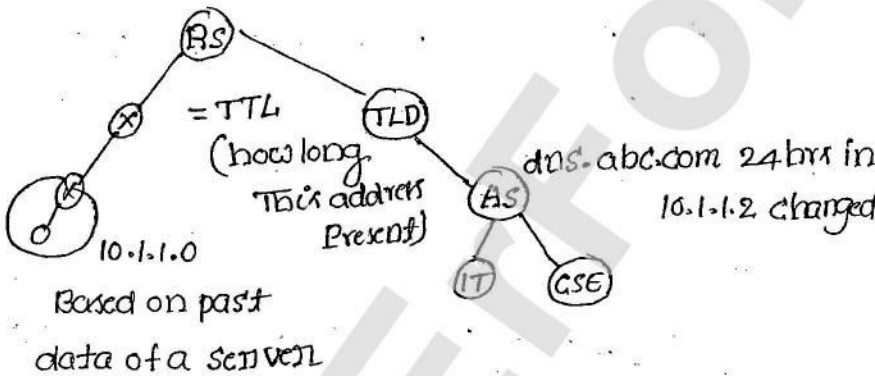
In "abc" college building, "one" lab, "one" first system

Mail Aliasing:



Multiple servers placed in different places

Distributive Database:



consider a request of page for which the IP address is stored in router. if again the same page is requested with little time gap. it is restricted from nearest router.

Canonical dependency ⇒ glbc record.

Replication ⇒ sharing data.

More replication ⇒ use TCP

Translation ⇒ use UDP

IT.abc.com 24 hrs in 10.1.1.3, www.abc.com 24hrs NS, dns.abc.com.

www.abc.com 2hrs canonical backup.abc.com.

canonical name

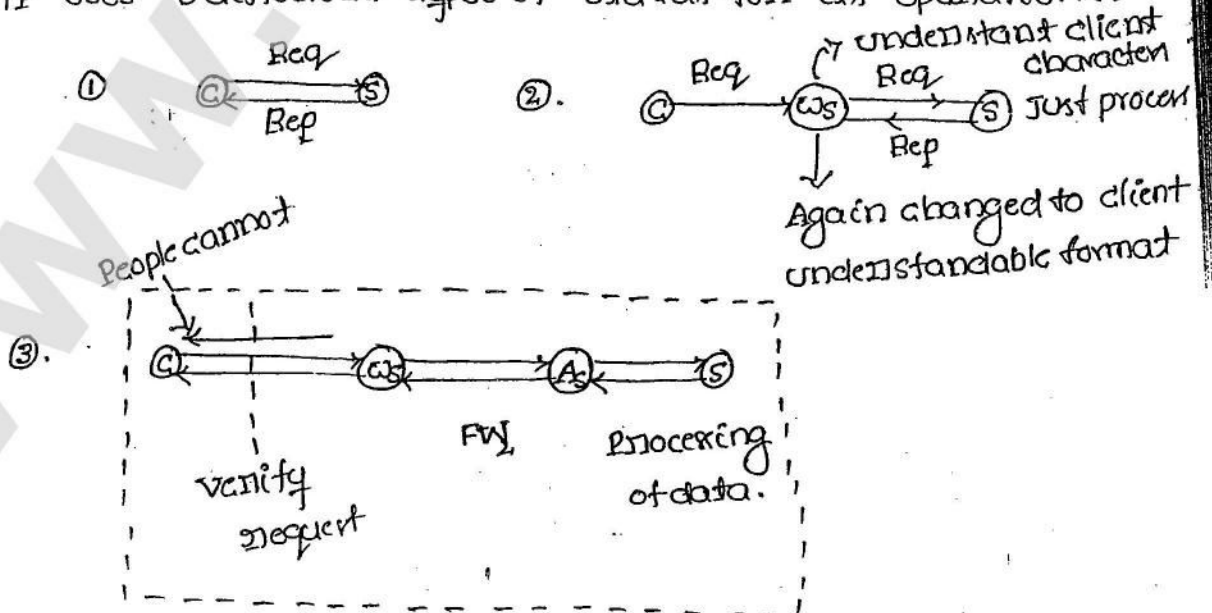


Alternative names for websites.

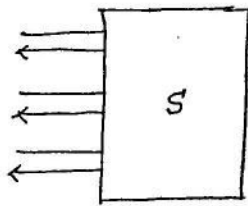
HTTP:

- * it is a client server protocol using port 80 in TCP.
- * it is stateless protocol.
- * There are 2 types of HTTP protocols
 - > persistent
 - > Non-persistent.
- * it have two types of messages -> Request
-> Response.
- * HTTP will perform its operations by using 8 different methods
 - => Head: HTTP developed through web browser, version, os (all html pages stored in webserver)
 - => get method.
 - => put => creation (create an object in server)
 - => Post => changing (modification)
 - => delete => delete
 - => Trace => debugging
 - => options => optimization.
 - => connect => Through the channel and use its security
Perform transactions.

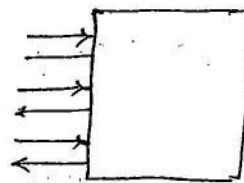
HTTP uses 3 different types of status for its operations:



* On-persistent connection in 1.0 (in 2.0 is changed to persiste)



Every time new connection is established



Persistent

using some time, limit connection is placed.

2 - successful

3 - redirecting

4 } Error Method.

* Each & every request express head method (client).

Display status error (stateless protocol).

File Transfer protocol (FTP):

* it is a client server protocol, user's port numbers

20 & 21 on TCP

* it have two types of connections:

→ Data connection (using port-20)

→ control connection (using port-21)

* There are 3 modes of operations.

→ Active mode

→ passive mode

→ Extended passive mode.

* There are 2 flavours of FTP

FTP ⇒ Authorized users

FTPE ⇒ Anonymous users.

* To keep track data transmission, FTP uses wide varieties of status codes and also it is supported with many no of commands.

* TFTP: never requires username, and password. All the users within the applications can access the data.

Eg: LIC Policy application



fig: Active mode

*



fig: Passive mode

* in passive mode, server generates a dynamic address and it is being get connected within the client.

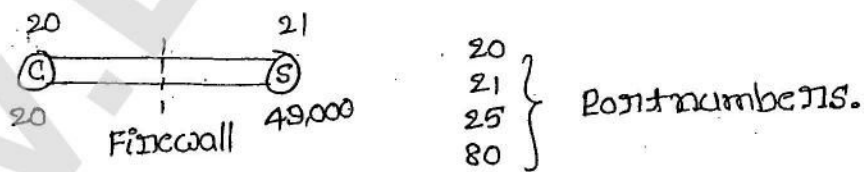


fig: Extended passive mode.

* in this mode, a firewall is being placed b/w client and server, where the time is assigned to packet. if packet is received after the time then firewall discards the packet.

- * To support the data connection perfectly, where the FTP must be monitored constantly in these time, so for this we use a no. of commands.
- * The monitoring of FTP constantly assumes the checking of status codes.

HTTP + SSL \Rightarrow HTTPS

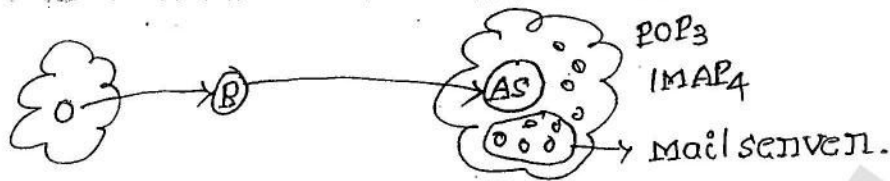
FTP + SSH \Rightarrow security.

- * By using SSH \Rightarrow a secured pipeline is established which is used in FTP.

SMTP:

- * it uses port 25 at TCP.
- * it is host-to-host transport protocol.
- * it is text-based protocol, but enabled with multimedia with MIME extension.
- * it is having two components.
 - \rightarrow User agent
 - \rightarrow Mail transfer agent.
- * it is a part of push-full mechanism in the mail comm.
 - \therefore SMTP is used to push the mail.
- * POP-3 and IMAP₄ are used for pulling the message.
- * it is an example for asynchronisation communication.
 - \therefore client and server are indirectly connected.
- * if client and server are directly communicated (connected) \Rightarrow synchronous commands

* it is connected to DNS server also.



* if a host is needed to send the data, then it gets connected to the router and then the router gets connected to the authoritative server through SMTP protocol. The AS identifies particular host in the mail server and "push" the data into that hosts. and if any other host requires the data within the network, then the data is being "pulled" by the server.

* Hence the mechanism is being considered as the "PUSH-PULL" mechanism, for this mechanism POP₃ and IMAP₄ are used.

* IMAP₄ is more advantageous than POP₃ for this consider an example:

Before downloading a file, a msg is delivered to check it is SAFE (or) associated with any virus. if user is interested to continue (either it is SAFE or UNSAFE) then only it proceeds and if he is not interested simply "CANCEL" it => it is

advantages of IMAP₄ over POP₃

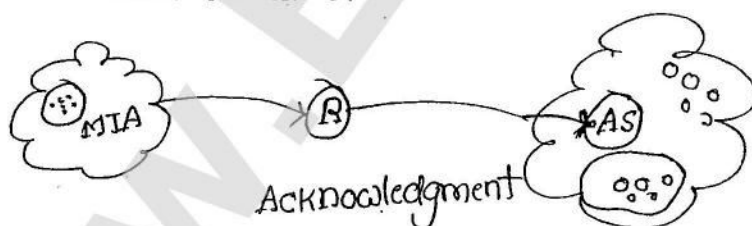
* IMAP₄ Push some of the dangerous mails to junk mails where as POP₃ cannot do.

~~Hierarchies in the inbox:~~

- * A folder is created in the inbox and are configured with some mail address. so that there is no chance of missing important information.

Two components:

- * User agent
- * Mail Transfer agent (In charge to negotiate with TCP for connection SMTP)
- * Forward message => attachments are available with the message
- * Reply message => all the attachments are automatically dropped.
- * Read Receipt component (if set) => Then user (who send a mail to other can able to know either the other user who received a mail) has received or not.
- * While user reads the mail an acknowledgment is sent to the sending user that the recipient had read it.



- * User agent is used for handling the attachments and the disattachments and mail transfer agent is in charge to have a connection with mail of SMTP through TCP.

Internet Protocol.

Different special IP Addresses:

S.NO	source IP Addresses	Destination IP Address.
1	X	X
2	X	✓
3	X	✓
4	✓	X
5	✓	✓
6	X	✓

* The above 6 IP addresses are used only for special purpose within the internet protocol.

* some of the IP addresses are used to represent only source IP addressing and some are destination IP addresses. e.g. it represents to have NID and some HID.

1.

✓	X	=> filled by ds
NID	HID	
X	X	

A: 10.0.0.0

B: 150.157.0.0

C: 192.168.1.0

Name: "This network" address.

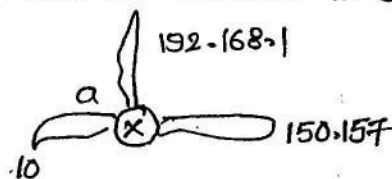
Purpose: used by Router.

=> Represents different network

D	10.0.0.0	150.157.0.0
---	----------	-------------

 => not valid

* They cannot be used as source IP address & dest IP address.



10.0.0.0	a
150.157.0.0	b
192.168.1.0	c

* This type of IP addressing system are used by routers to identify the network, for which it belongs to.

~~NID~~ ~~110~~ \Rightarrow ~~filled by 1's~~
 X ✓

A: 10.255.255.255

B: 150.157.255.255

C: 192.168.1.255

name: Directed Broadcast system // Delivering packets to all system in some other network.

Purpose: To all in sender network

D	10.1.1.1	150.157.255.255	✓
---	----------	-----------------	---

D	150.157.255.255	10.1.1.1	X
---	-----------------	----------	---

* Host ID is appended with all 1's and network ID can be any other value.

3. 1's 1's
 NID HID

255.255.255.255.

Name: Limited Broadcast address. // Delivering packets to all system in our own n/c's

Purpose: To all in the local network.

* Both the host ID and network ID's are being appended with 1's

4. 0's 0's
 NID HID

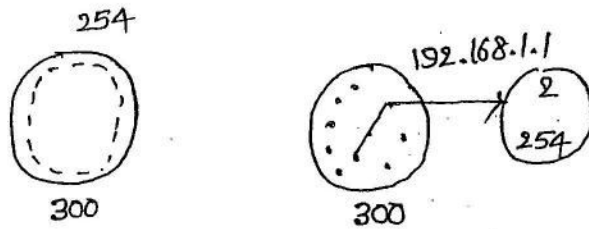
0.0.0.0

Name: Dynamic IP address.

Purpose: Dynamic Host configuration protocol (DHCP).

* In dynamic more no. of systems, so it can be limited IP addresses.

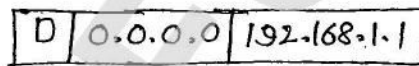
* More IP addresses & less no. of systems



* For a particular duration of time an IP address is permanently given to the user. After performing the "logout" by any user on particular system, then the IP address is assigned to the other user.

* Like wise 254 IP addresses are managed by the server (but not client) in the FCFS basis. if there are more request an "Queue" is maintained.

* so it is used only for the source but not destination



Dynamic:

The operating system on the basis of administrator, assigns the IP address to the system.

Auto: The operating system directly assigns the IP addresses to all the systems without the intervention of administrator automatically.

5.

0	✓
NTID	HID

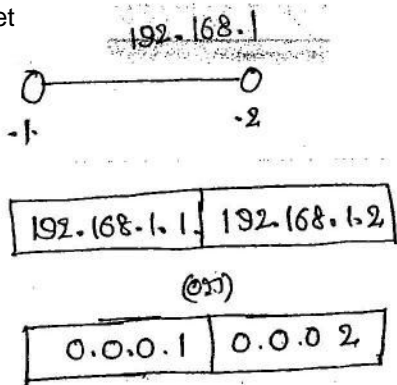
A: 0.1.1.1

B: 0.0.100.1

C: 0.0.0.100

Name: Host in this network

Purpose: local communication



- * The communication is done among the two host systems within the same network => local communication.
- * The communication is not possible for host in one network with the other host in other network.
- * Therefore, the network ID is fixed and unique, and only the host ID alternates for every communication.

6. 127. . . Any

127.1.1.1 or 127.100.0.255

Name : Loop back address

Purpose: interprocess communication.

- * self checking or self connectivity with check

127.0.0.0 X

127.255.255.255 X

- * Both IP addresses are not valid other than these two IP address all the others are valid, which starts with ID = 127.

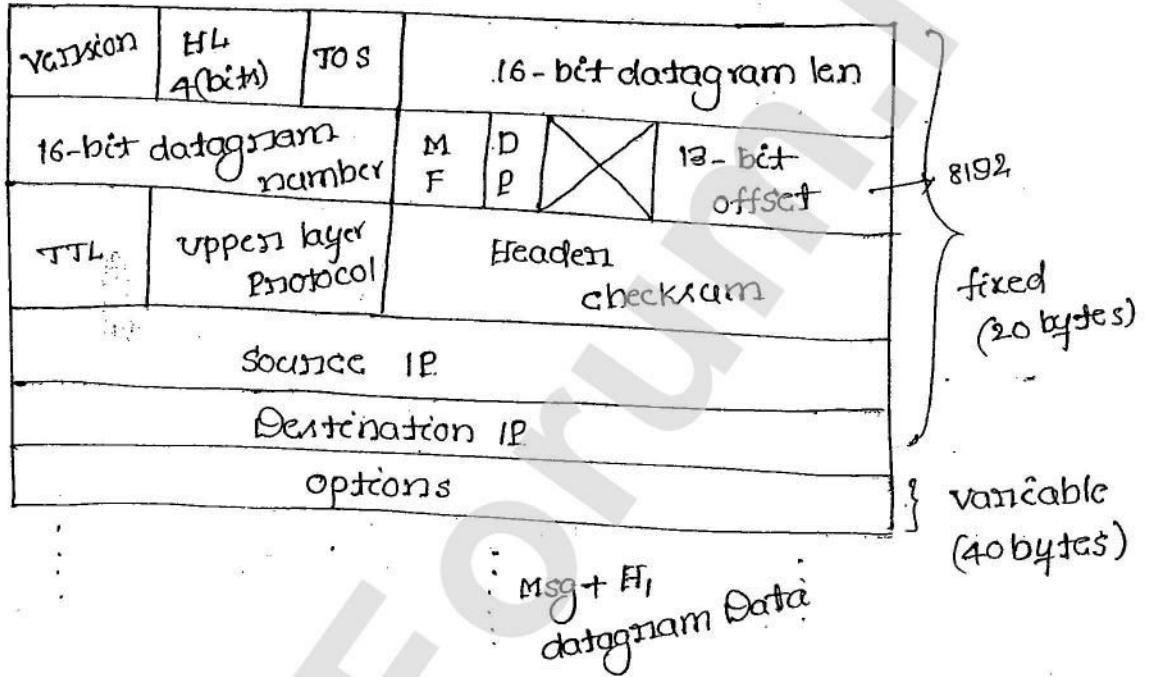
=> which of the following IP addresses are used as only source IP addresses

- A. 10.1.1.1 **B. 0.0.0.0** C. 0.0.0.1 D. 127.1.1.1

⇒ which one of the following IP addresses are used as both source and destination addresses.

- A. 10.1.1.1 B. 255.255.255.255 C. 10.255.255.255 D. 0.0.0.0

IP operations:



* Version ⇒ to indicate either IP₄ (32) or IP₆ Packet (4-bits)
IP₆ less complicated than IP₄.

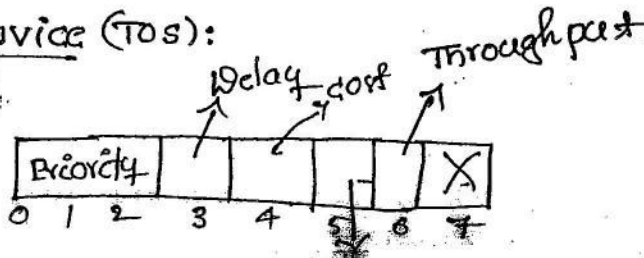
* Header Length: Maximum size ⇒ 60 bytes (40+20)
Minimum size ⇒ 20 bytes.

$$2^4 = 0 \dots 15$$

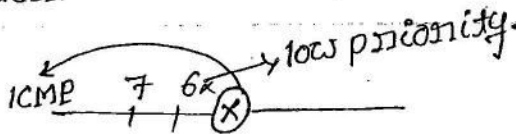
constant scale factor = 4

$$\frac{\text{Actual header length}}{8} = \text{Available header length in PKT (4-bit)}$$

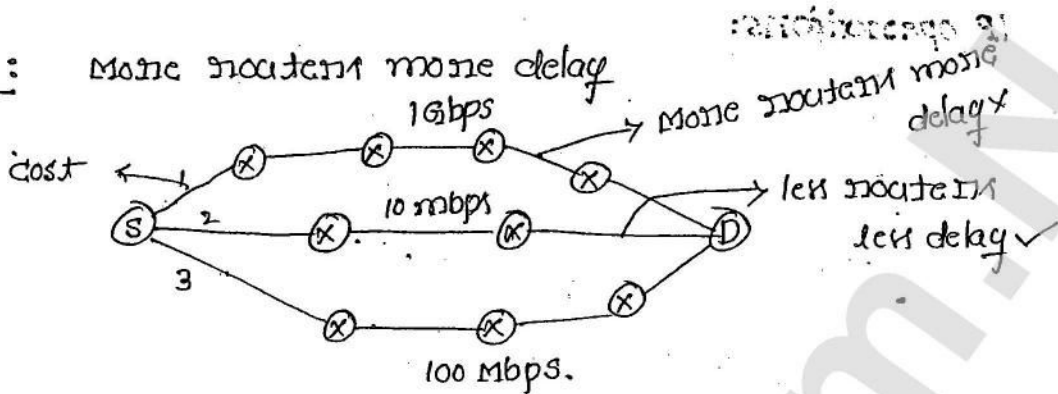
Types of service (TOS):



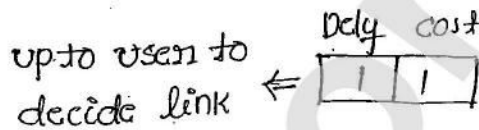
* Based on priority, packet is transferred to intermediate router.



Delay:



cost: considering bandwidth, num. of routers, error rate, distance among the routers, the cost is being calculated.



Reliability: it represents the "error rate".

Throughput: it depends on bandwidth, if high bandwidth for a link => then allows

$$2^{16} = 64 \text{ KB} \Rightarrow \text{the maximum size of the total packet.}$$

16-bit datagram numbers:

Every datagram associated with the sequence numbers starting with '0'.

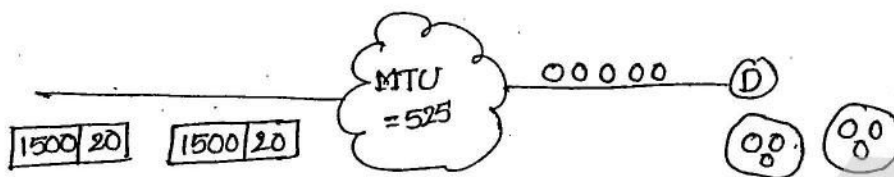
More fragments (MF).

Maximum Transfer Unit (MTU):

Fragmentation is applicable to only datagram

packet but not for header.

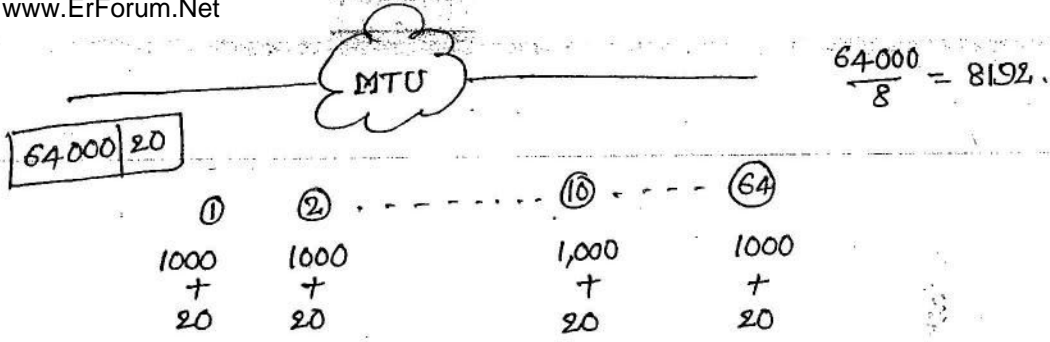
* OFFSET indicates no. of databytes ahead of this fragment in that particular packet.



	①	②	③	④	⑤	⑥	
504	505	504	491	505	505	490	
	+	+	+	+	+	+	
	20	20	20	20	20	20	
	5	5	5	6	6	6	
MF	1	1	0	1	1	0	
			end				
Offset	0/8	505/8	1010/8	0/8	505/8	1010/8	CSF = 8
		504	1003				

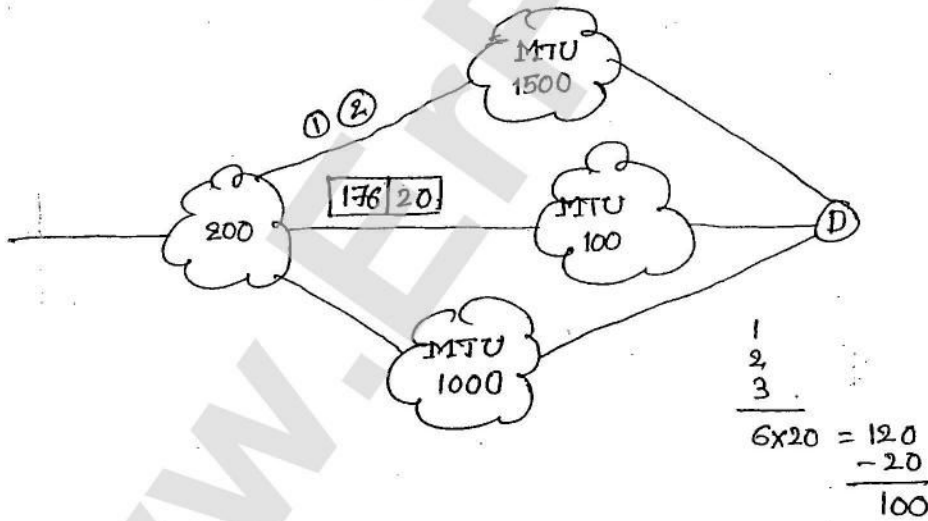
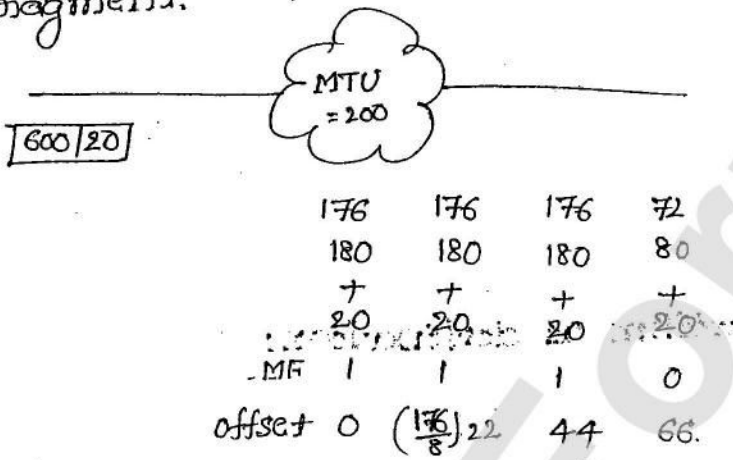
Re-assembly Algorithm at destination:

- * classify, fragments based on 16-bit datagram numbers.
- * identify the fragment with offset=0 and designate it as a first fragment.
- * identify the fragment with MF=0 and designate it as a last fragment.
- * identify data in the first fragment and look for the fragment with same offset value and designate it as second fragments.
- * Repeat previous step as many times as possible to cover all the fragment.



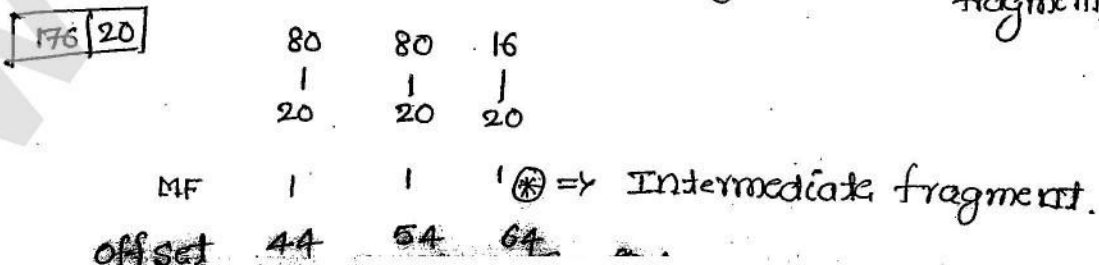
* Datagram data (or) fragment data must be divisible with '8' if not adjust it's number so that it is divisible with '8'.

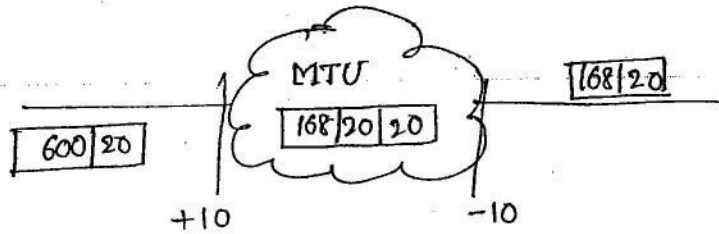
* This rule is applicable for all the fragments except for last fragment.



* if offset = MF = 0 => Original packet.

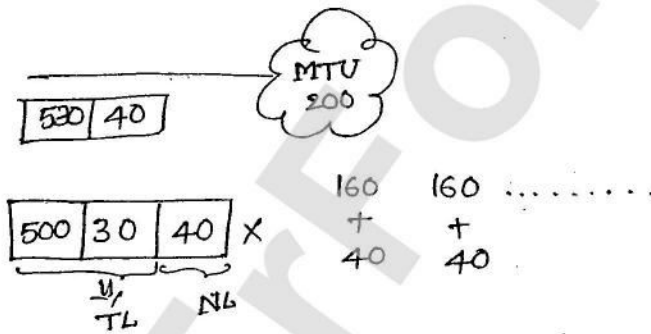
* if any one of them is non-zero => fragment (intermediate fragment)





168	168	168	96
170	170	170	96
+	+	+	+
20	20	20	20
+	+	+	+
10	10	10	10
MF 1	1	1	0
offset 0	168/8	168+168/8

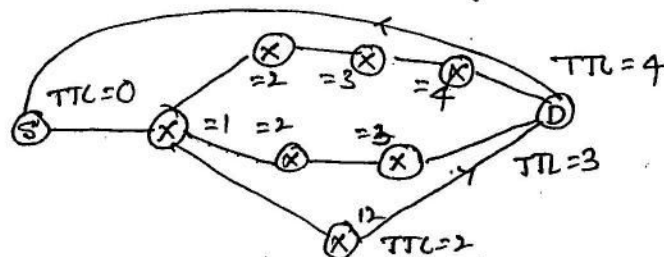
Msg : 500 } Datagram data.
 TCPH : 30 }
 IPH : 40
 MTU : 200



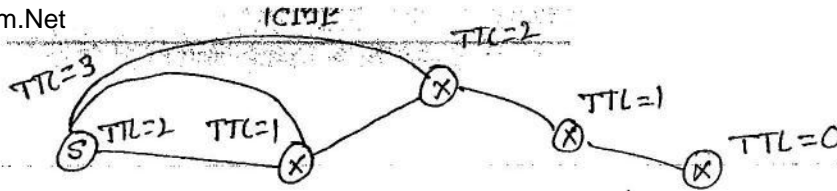
Time To Live (TTL):

* it have four applications:-

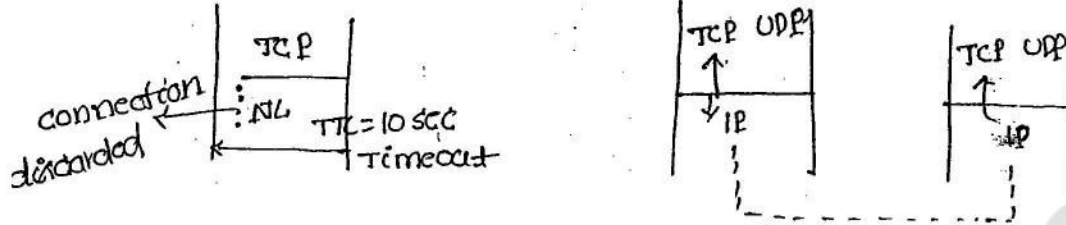
- > To avoid infinite looping
- > To identify no. of routers between source & destina
- > To debug the network
- > To help upper layers in timer management



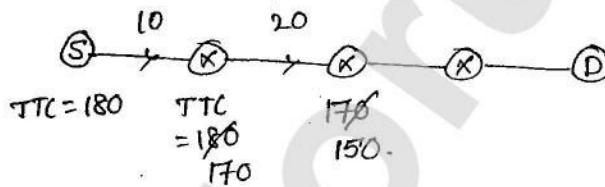
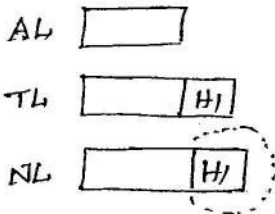
* It is possible to find no. of routers in a route by using



Time Management:



* Headers checksum is carried out at every router and only at Header.

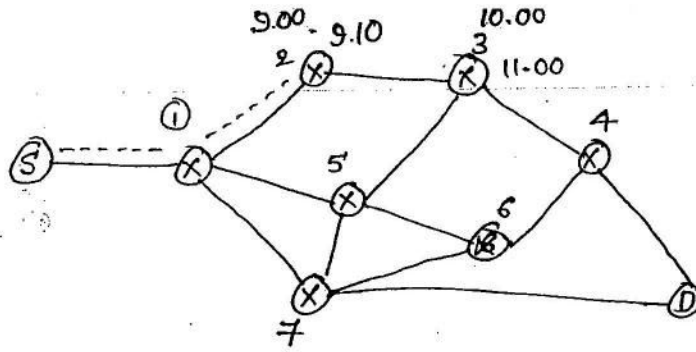


- > TTL
 - > MF
 - > Offset
 - > 16-bit datagram length
 - > Options
- => Tend to change at every router : (variable)

source destination } => fixed.

Options:

- * Strict source routing } source will decide the route
- * loose source routing }
- * Record routing => Router decides the route
- * time stamp
- * Security



Strict source routing: Each and every route is specified

0 1, 2, 3, 4

Loose source routing: only the important routers are specified and the route is generated based upon those routers => practical

1	2	3	4
1	5	3	4
1	7	6	4
1	5	6	4

Record Routing:

0 1, 7, 5, 6

* Packet can be transferred as it wishes among all the routers

Time stamp: Arrival time and Departure time of each and every packet is stored.

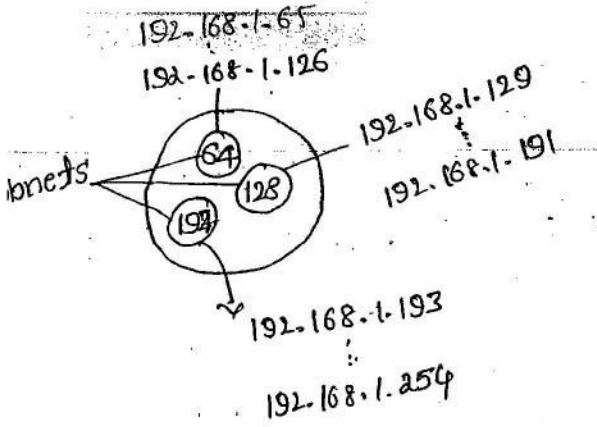
Security: mails are sent along with certification which provides secured access for a page.



then which of the options are most available in all the fragments and which of them are necessary to present in any one of the fragments?

Ans: strict source routing } most available in all the fragments.
 loose source routing }

* Record routing } must be necessary in any one of
 * Time stamp } the following.

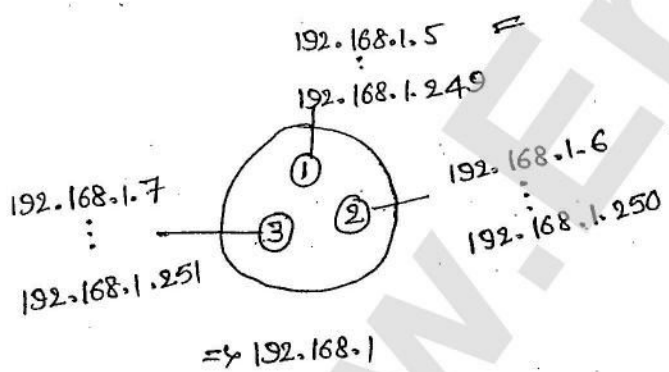


eg: 192.168.1

01
 01 000 001 - 65
 01 000 010 - 66
 ⋮
01 111 110 - 126
 10
 10 000 001 - 129
 10 000 010 - 130
 ⋮
10 111 110 - 191
 11 000 001 - 193
 ⋮
 11 000 010 - 254

C: NID HID ⇒ borrowed 2 bits
 24 2,6
 SID HID
 $2^2 = 4$
 $2^6 - 2 = 62$

128	64	32	16	8	4	2	1
0	0	= 0					
0	1	= 64					
1	0	= 128					
1	1	= 192					



..... 01
 00000101 = 5
 00001001 = 9
 ⋮
11111001 - 249

NID HID
 24 6,2
 128, 64, 16, 8, 4, 2, 1
 00 - 0
 01 - 1
 10 - 2
 11 - 3

..... 10
 00000110 - 6
 00001010 - 10
 ⋮
11111010 - 250

* Subnet id's and IP address of the network are changing

* If the destination address of data packet, which follows the routing table, is not checked, then the packet will not be forwarded and is lost.

to some specific subnet and available on that network.

→ If not available then the packet is directly assigned to that.

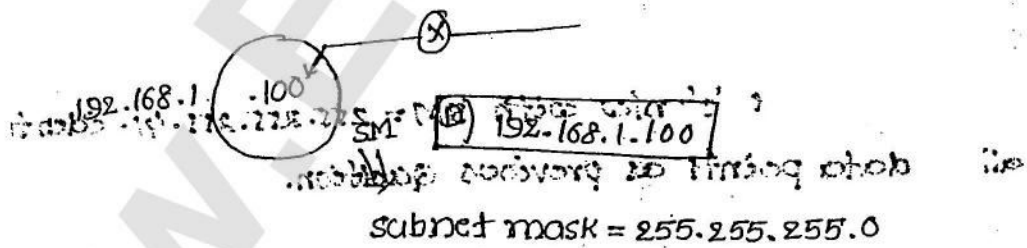
→ If available then identify, the packet belongs to which subnet for that we use subnet mask (SM)

Subnet Masking:

* It is also a 32-bit system and it is used to indicate whether a subnet are available (on) not in the network.

* If available, it will give the information about no. of bits borrowed from host id and their position based on the following 2 rules.

1. No. of 1's in the subnet mask, indicates n/w id plus subnet id.
2. No. of 0's indicates Host Id part.



SM: 11111111.11111111.11111111.00000000

Rule (1): $\frac{24}{NID} + \frac{0}{SID} = 24$ (since, class C)

(2): $HID = 8$ (since, no subnet, nothing is borrowed from HID)

NID = 192.168.1

HID = 100.

Q. Consider a class C network with SM: 255.255.255.192. identify no. of bits borrowed from HID and their position, possible subnets and their ID's possible no. of systems for subnet and range of IP addresses in each and every subnet.



SM: 255.255.255.192
 .11000000

Rule (1): $\overset{24}{NID} + \overset{2}{SID} = 26$

HID = 6

No. of bits borrowed = 2

Their position is = 128th bit, 64th

Possible subnets = $2^2 = 4$

Their subnet ID's = 11000000

- 00 - 0
- 01 - 64
- 10 - 128
- 11 - 192.

No. of systems per subnet = $2^6 - 2$

Range of IP addresses = 62.

eg: Consider a class 'c' net with SM = 255.255.255.41. identify all data points as previous question.

SM: 11111111. 11111111. 11111111. 00101001

$\overset{24}{NID} + \overset{3}{SID} = 27$

HID = 5

No. of bits borrowed = 3

Their position is 32, 8, 1

Possible subnets = 2^3 (0, 1, 8, 32, 9, 40, 41, 33)

No. of systems per subnet = $2^5 - 2$

Their subnet ID's =

32	8	1	= 0	011	= 9	32	8	1	= 40
000			= 1	100	= 32	111			= 41

Eg: Consider a class B net with SM = 255.255.255.0, identify all the data points as proposed ques.

↳ subnets are available

Entire 3rd packet are borrowed for subnet IDs

$$\therefore \text{no. of subnets} = 2^8$$

$$\text{Their ID's} = (0-255)$$

$$\text{No. of systems} = 2^8 - 2$$

$$\text{Their ID's} = (1 \text{ to } 254)$$

Eg: Consider a class C network with SM = 255.255.255.15.

$$\text{SM: } 11111111.11111111.11111111.00001111$$

$$\begin{array}{cc} 24 & 4 \\ \text{NID} + \text{PID} & = 28 \end{array}$$

$$\text{No. of bits borrowed} = 4$$

$$\text{Their position is } = 1, 2, 4, 8$$

$$\text{Possible subnets} = 2^4 = 16$$

$$\therefore \text{No. of system per subnet} = 2^4 - 2 = 14$$

Eg: Consider a class C net proposed an appropriate subnet mask to have 7 subnets each with 25 systems.

$$7 * 25 = 256 \text{ (since, it is class C)}$$

3 bits must be borrowed (since, 7-subnets)

$$24 \quad \begin{array}{cc} 8 & \\ \hline 3 & 5 \\ 5 & 3 \end{array}$$

$$3\text{-bits} \Rightarrow 2^3 = 8$$

$$\left. \begin{array}{l} 255.255.255.224 \checkmark \\ 255.255.255.7 \checkmark \\ 255.255.255.41 \checkmark \\ 255.255.255.67 \checkmark \end{array} \right\} \begin{array}{l} \text{all are possible but} \\ \text{left to right is} \\ \text{appropriate} \end{array}$$

eg: consider a class B n/w and propose an appropriate SM to have 150 subnets each with 200 systems.

$$150 * 200 <= 64,000$$

$$150 \Rightarrow 8 \Rightarrow 2^8 - 2 = 254$$

(required) 200

$$255.255.255.0$$

$$255.255.0.255$$

$$255.255.240.240$$

$$255.255.192.252$$

eg: consider a class C network. propose an appropriate SM to have 20 subnets each with 15 systems

$$20 * 15 <= 256$$

$$300 \neq 256$$

not possible.

eg: consider a class C network, propose an appropriate SM of ~~60~~ 60.60.120

$$\text{class C} \Rightarrow 8$$

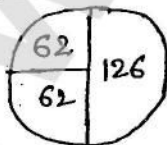
$$\begin{array}{c} \overline{2} \quad \overline{6} \quad \overline{1} \quad \overline{7} \\ 2^2 = 4 \quad \times \quad 2^1 = 2 \end{array}$$

$$\times \quad 2^6 - 2 = 62, \quad 2^7 - 2 = 126$$

(not possible) (not possible)

so in order to propose appropriate SM we use the concept of VLSM.

$$255.255.255.192$$



$$255.255.255.128$$

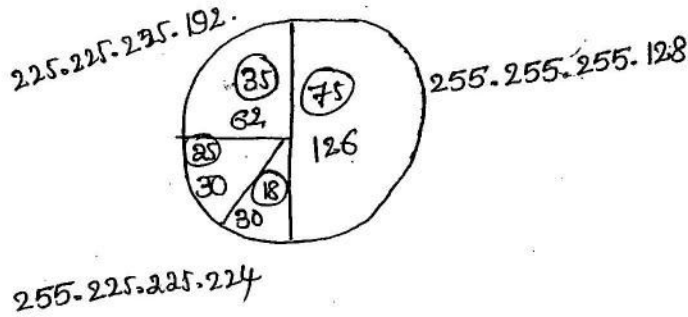
$$255.255.255.192$$

$$\begin{array}{c} 2^1, 2^6 - 2 \\ = 2 = 62 \end{array}$$

$$\underline{1 \quad 6}$$

$$1 \quad 7$$

eg: consider a class 'c' network propose an appropriate SM to have 4 subnets of 75, 35, 25, 18.



$$2^6 = 64$$

$$2^7 = 128$$

$$2^6 - 2 = 62$$

$$2^7 - 2 = 126$$

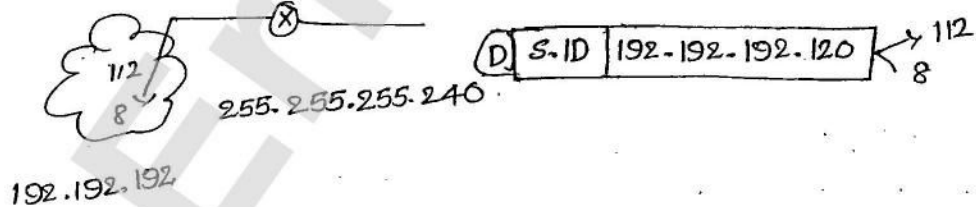
eg: consider a class c network, propose an appropriate SM to have 6 subnets of 30, 25, 22, 20, 18, 15.

class c = 8

$$2^3 = 8$$

$$2^6 = 64$$

eg:



Identify subnet ID, Host ID and directed broadcast address.

$$IP: \underbrace{11000000. 11000000. 11000000. 01111000}_{NID \quad \quad \quad SID \quad HID}$$

192.192.192

$$SM: \underbrace{11111111. 11111111. 11111111. 11110000}_{NID \quad \quad \quad SID \quad NID}$$

112 8

SID = 112

HID = 8

120

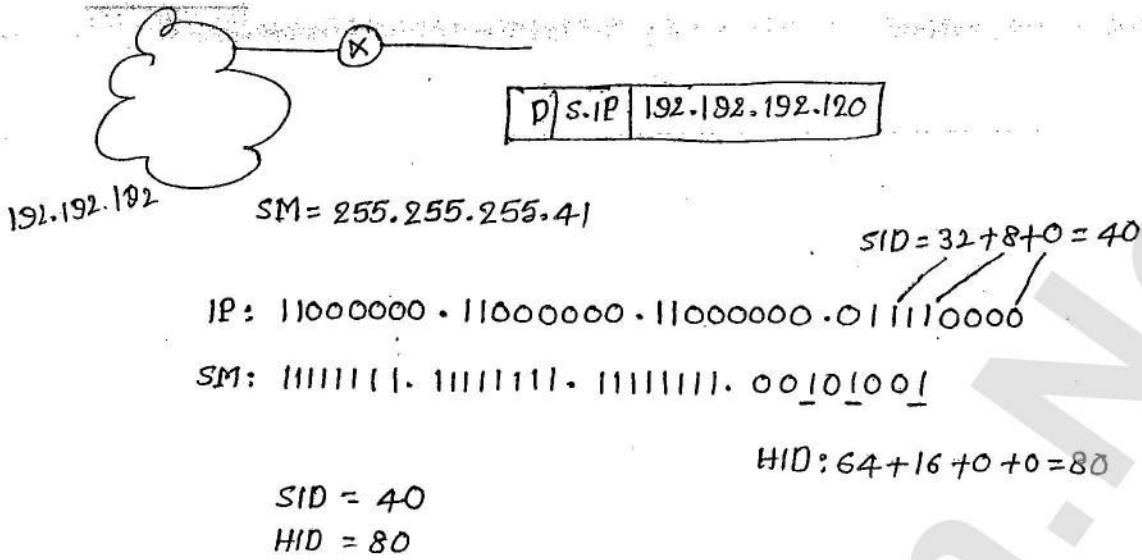
192.192.192.127

11000000. 11000000. 11000000. 01111000

7680 32 10

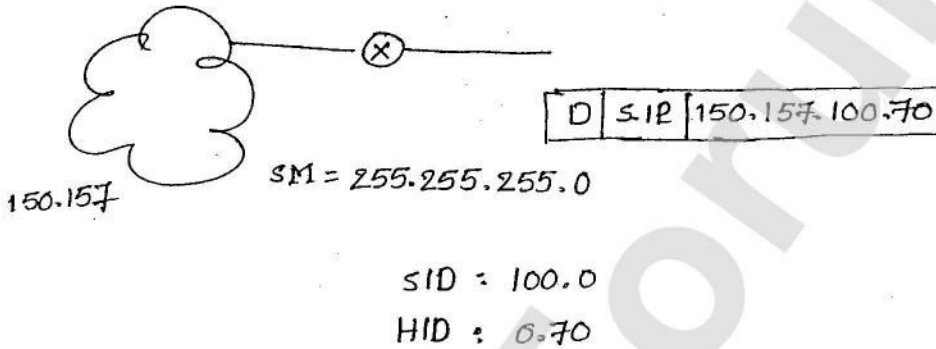
Broad Address

Eg:



↳ 192.192.192.254 ⇒ Broadcast address.

Eg:



Denied Broadcast address = 150.157.100.255.

Eg:

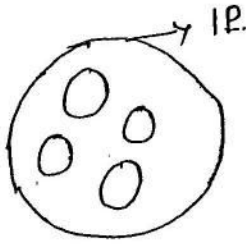


IP	SM	ε/δ
10.0.0.0	255.0.0.0	a
157.157.0.0	255.255.0.0	b
192.168.1.0	255.255.255.0	c
0.0.0.0	0.0.0.0	
0.0.0.0	0.0.0.0	

192.168.1.100	192.168.1.100	192.168.1.100
AND	AND	AND
255.0.0.0	255.255.0.0	255.255.255.0
-----	-----	-----
192.0.0.0	192.168.0.0	192.168.1.0

* 0.0.0.0 ⇒ Default route subnet mask

Supernet Mask:



$192.192.0.0 : 11000000.11000000.00000000.00000000$
 $192.192.1.0 : 11000000.11000000.00000001.00000000$
 $192.192.2.0 : 11000000.11000000.00000010.00000000$
 $192.192.3.0 : \frac{11000000}{8} . \frac{11000000}{8} . \frac{00000011}{6} \frac{00000000}{2} . \frac{00000000}{8}$

* It is a 32-bit system used to generate a single IP address of group of networks based on following 2 rules.

1. num. of one's in supernet mask indicates fixed part
2. num. of 0's indicates variable part.

11111111.11111111.11111100.00000000

SM = 255.255.252.0

(1). 192.192.0/22 :

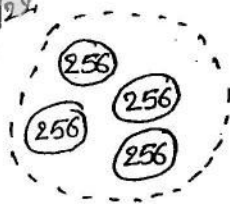
↳ Represents NID.

class C:

$24 - 22 = 2$

$2^2 = 4$

192.192.0/22



$256 * 4 = 1024$

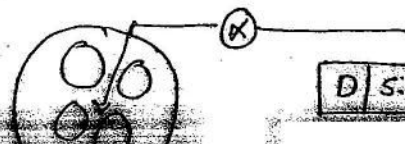
- /24 = class C
- /16 = class B
- /8 = class A

32	
22	10
NID	IID

$2^{10} = 1024$

(2). 192.192.0

255.255.255.0



192
↓
11000000

Difference Between Subnet mask & Supernet mask

SUBNET MASK

- * No. of 1s in the subnet mask is either equal to network id (or) more than network id bits.

- * Bits are borrowed from HID

- * it is applicable to single network

SUPERNET MASK

- * No. of 1s in supernet mask is always less than NID bits

- * Bits are borrowed from NID.

- * it is applicable for two or more networks.

⇒ CIDR aggregation (Classless Inter Domain Routing) ⇒ another name for supernet

	A	B	C
255.0.0.0	subnet	subnet	supernet
255.255.0.0	subnet	subnet	supernet
255.255.255.0	subnet	subnet	subnet

192.192.0/22 ⇒ NID

1/24 ⇒ class C

1/16 ⇒ class B

1/8 ⇒ class A

1/22 - ?

1/20 - ?

CIDR.

Routing Algorithms:

Routing is the process of forwarding packets from one network to another. All the information needed for a router to forward packets to a hop can be found in the router's routing table.

⇒ Static Routing: it occurs when you manually add route in each router's routing table.

⇒ Dynamic Routing: it is the process of using protocol to find and update routing tables on routers and to maintain a loop-free, single path to each network.

⇒ common fields in a routing table:

- * Mask
- * Network address
- * Next hop address
- * Interface
- * Flags
 - U: up
 - G: Gateway
 - H: host specific
 - D: added by redirection
 - M: Modified by redirection.

⇒ Delivery semantics:

- unicast: delivers a msg to a single specified node.
- Broadcast: " " " to all nodes in the network.
- Multicast: Deliver a msg to a group of nodes that have expressed interest in receiving it.

DEF: An autonomous system (AS) is a group of networks & routers under the authority of a single administration.

- * Routing inside AS is called "intradomain routing"
- * Routing between AS is called "interdomain routing".

