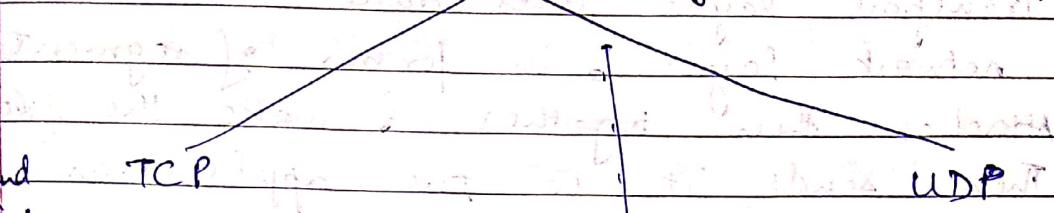


Transport Layer



- Found in TCP
- (Error detection) Reliable
- (3-way handshaking) Connⁿ oriented
- High overhead
- slow
- flow control (tries to synchronise speed of sender & receiver)
 - ↳ lessens data loss
 - ↳ lessens network traffic
- congestion control
 - ↳ so that the packet reaches to the receiver within the specific time.
 - high RTT / packet loss
 - ↳ applied by sender to reduce the ~~no~~ traffic on the network.
- Byte stream protocol
 - ↳ each & every byte identified through sequence no.

→ (Multiplexing) : Server side

- Application layer sends the file in form of a message to transport layer.
- Then it divides the message into segments & puts header info ~~or attaches~~ with all the segments & sends it to network layer.

Multiplexing

Port No. : To recognize different applications.

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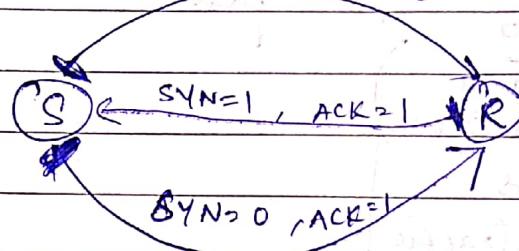
→ Receiver side : (Demultiplexing)

Transport layer takes data from network layer, in forms of segments.

Attaches them together & make the file. Then sends it to the application layer.

→ 3-way handshaking :

SYN=1, ACK=0 ()



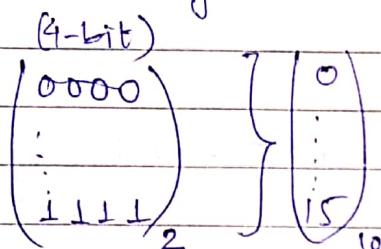
→ TCP header :

(16 bit)	(16 bit)	
source port	destination port	- 4 bytes }
sequence no. (32 bit)		- 4 bytes
Acknowledgement no. (32 bit)		- 4 bytes }
header length (4 bits)	Unused R C S S Y I G K H T N N	Advertised window (16 bit)
Internet check sum (16 bit)		20 bytes fixed
option (0-40 byte)		- variable
data		

$$\text{port no.} = (0 - (2^6)) \rightarrow 0 - 1023$$

- Sequence no. : starts with random initial sequence no.
- To differentiate b/w connections.
 - Suppose random sequence no. matches then, the connⁿ which reaches later would get dropped.
 - Every byte would have different sequence no.
- Acknowledgment no. :
- what receiver is expecting to receive the next sequence no.

→ Header length :



NOTE: Why do we need header length in TCP header?

Sol: Option field is a variable field, so length of TCP header can vary from (20 - 60) bytes

To overcome the prob. of representation of header, we use scaling factor = 4.

$$\text{Header length field max} = 15 \quad \left[\begin{array}{l} \text{Header length field max} = 15 \\ \text{TCP Header length size max} = 60 \text{ bytes} \end{array} \right] \quad \frac{60}{15} = 4 \quad \text{scaling factor}$$

$$\text{Header length field} = (x)_2 = (y)_{10}$$

$$\text{Then Header length of TCP} = (y)_{10} \times 4 \text{ bytes}$$

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$$\therefore \text{Length of option field} = [(y)_{10} \times 4] - 20 \text{ bytes}$$

NOTE: Since min value for header length of TCP is $20 = (5)_{10} \times 4$. Thus, possible values for header length field can be from $(5-15)_{10}$.

→ Unused (6-bits) This field is still unused. Reserved for future usage.

→ Flags :

URG - Urgent

ACK - Acknowledgement - receiving acknowledgement

PSH - Push

RST - Reset

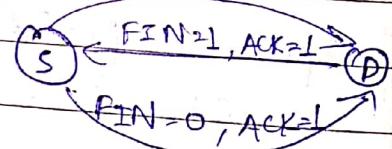
SYN - Syn - used for connⁿ establishment.

FIN - Fin - used for closing the connⁿ.

Urgent pointer

Urgent data \Rightarrow URG = 1
else URG = 0.

FIN=1, ACK=0



PSH : Push

A fixed size of data is sent to TCP as sending small packets with 20 byte

header size with each packet would be very inefficient.

→ RST - Reset :

Resetting the conn'

• Err.

→ Internet Checksum :

Characteristic of error detection and error correction.

$$\text{Total data bytes}_{\text{sender side}} = \text{Total data bytes}_{\text{receiver side}}$$

↓
no error

else, error is there.

correction : resend the segment.

→ Urgent data Pointer : ^{contains} ending point of ~~the~~ urgent data segments
A pointer that points to urgent data points

→ Option field :- Contains meta-data info.

like Max segment size

- variable sized (0-40 bytes)

- incase, no meta data info is there
the option field contains 10 bytes.

→ Advertised window : Receiver window

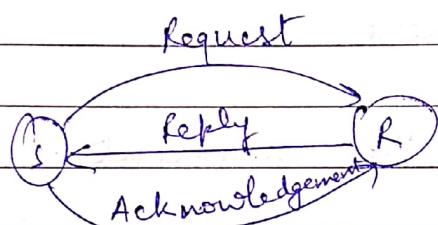
• flow control : Don't overwhelm the receiver through too much of data.

• Implemented through advertised window.

Contains buffer size of receiver - capacity

left at the receiver to take the data.

- Receiver sends it with the ACK packet.
- At handshaking time, receiver sends the info, in accordance with that sender starts sending data.
- Then as the receiver acknowledges current data, in accordance with that sender adjusts the flow of remaining packets.



- Most of the time acknowledgement is sent with data packets. This is known as piggybacking.

group

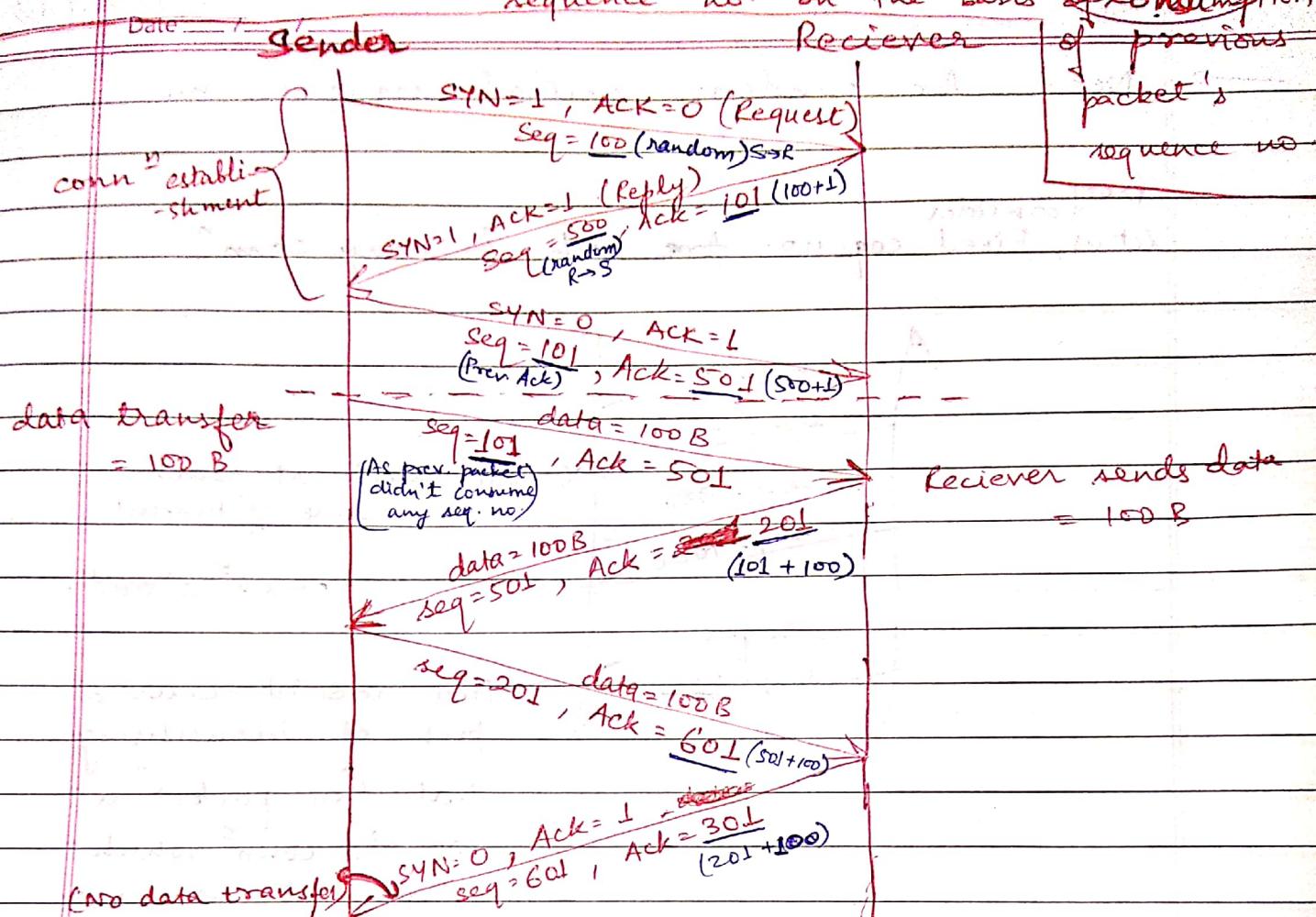
* Flow of sequence number ($0 - 2^{32} - 1$)

- FACT:
- ① Request packet consumes one sequence no.
 - ② reply packet consumes one sequence no,
 - ③ acknowledgement packet doesn't consume any sequence no.

NOTE:

In TCP, each & every byte is identified by its own seq. no. i.e. consumes one sequence no.

★ Receiver sends an ACK which it is expecting as a sequence no. next time. (Not random) it calculates the next sequence no. on the basis of ~~sequence~~ of previous packet's sequence no.

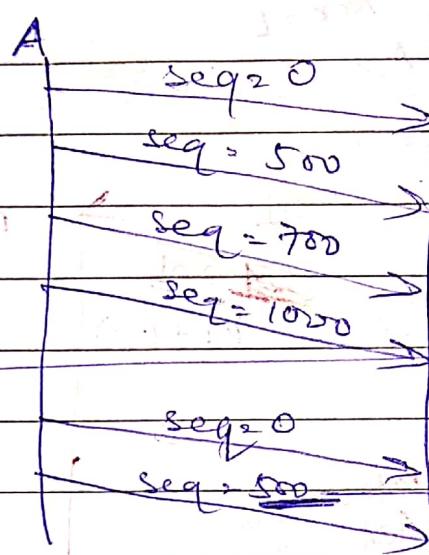


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→ Need for 'random initial' sequence no.

Proof: consider

Let us, Fixed seq. no. for each & every connⁿ.



→ This could not reach because of timeout
connⁿ closed.

→ This would occur a prob. of identifying that this packet is of old connⁿ which got lost or new connⁿ.

This is why random initial sequence no. is needed.

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Q- The following is a dump of TCP header in a hexadecimal format:

~~05320017 00000001 00000000~~

500207FF 00000000

- i - destination port
 - ii - seq. no.
 - iii - length of header (size)
 - iv - Type of segment
 - v - Adv. window

0532 0017
source destination

0 0 0 0 0 0 0 1,
sequence

0000 0000)
Ack

The diagram shows a 32-bit register labeled "Window". The bits are numbered from 31 to 0. The most significant bit (bit 31) is labeled "Advertised". Below the register, the binary value **10000000000000000000000000000010** is shown. The "Advertised" bit is highlighted in blue. The entire register is enclosed in a blue rectangular box.

$$5 \times 4 = 20 = \text{header length}$$

2

Ques - In a segment of TCP, option field is 16 bytes long. Find out the value of HLEN, HLEN header length field.

$$20 + 16 \rightarrow 36 / 4 \rightarrow (1001)_2$$

Ques - In a TCP connⁿ, the initial sequence no from client side is 2200 & server side is 5289, the client opens a connⁿ & sends two segments carrying 1000 bytes of data each. & closes the connⁿ. What is the value of sequence no. in each of the following cases?

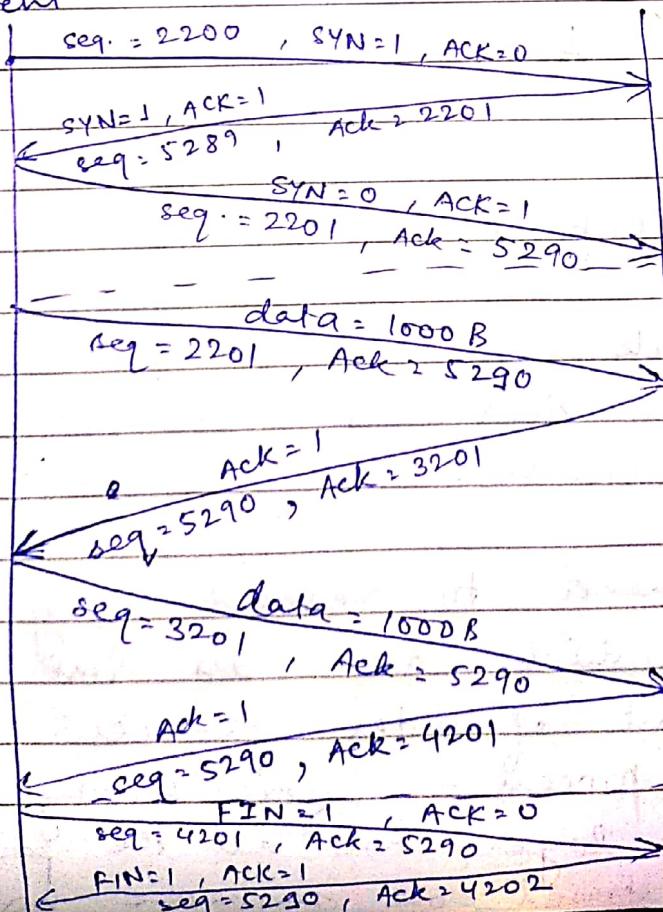
i - Handshaking

ii - Data Transfer

(Use timing dig. to show your ans.)

Client

Server



Page No. 1
FIN=0, ACK=1
Seq = 4202, ACK = 5291

Connⁿ closed

Sliding Window Protocol

→ Go back N :

- inorder delivery
- cumulative acknowledgement
- discard out of order packets
- sender window size = N ,
- receiver window size = 1
- $W_s + W_r \leq$ Available seq. no.

- $N+1$ = total sequence no. in GBN

→ Suppose K is max. no. of "bits" available for sequence no. then sender window size will be =

$$W_s + W_r = 2^K$$

$$\Rightarrow W_s + 1 = 2^K$$

$$\Rightarrow W_s = 2^K - 1$$

→ Selective Repeat

• GBN drawback :

Throughput decreases, as network traffic increases.

• You ~~only~~ need to resend only those packets which are ~~also~~ lost and accept out of order packets.

but extra process - before sending them to application layer, sort all the packets ~~as~~ based on sequence no.

* Selective Repeat

GBN drawback : Repetition of packets due to discard

- accept out of order packet

- only resend those packets which got lost in N/W.

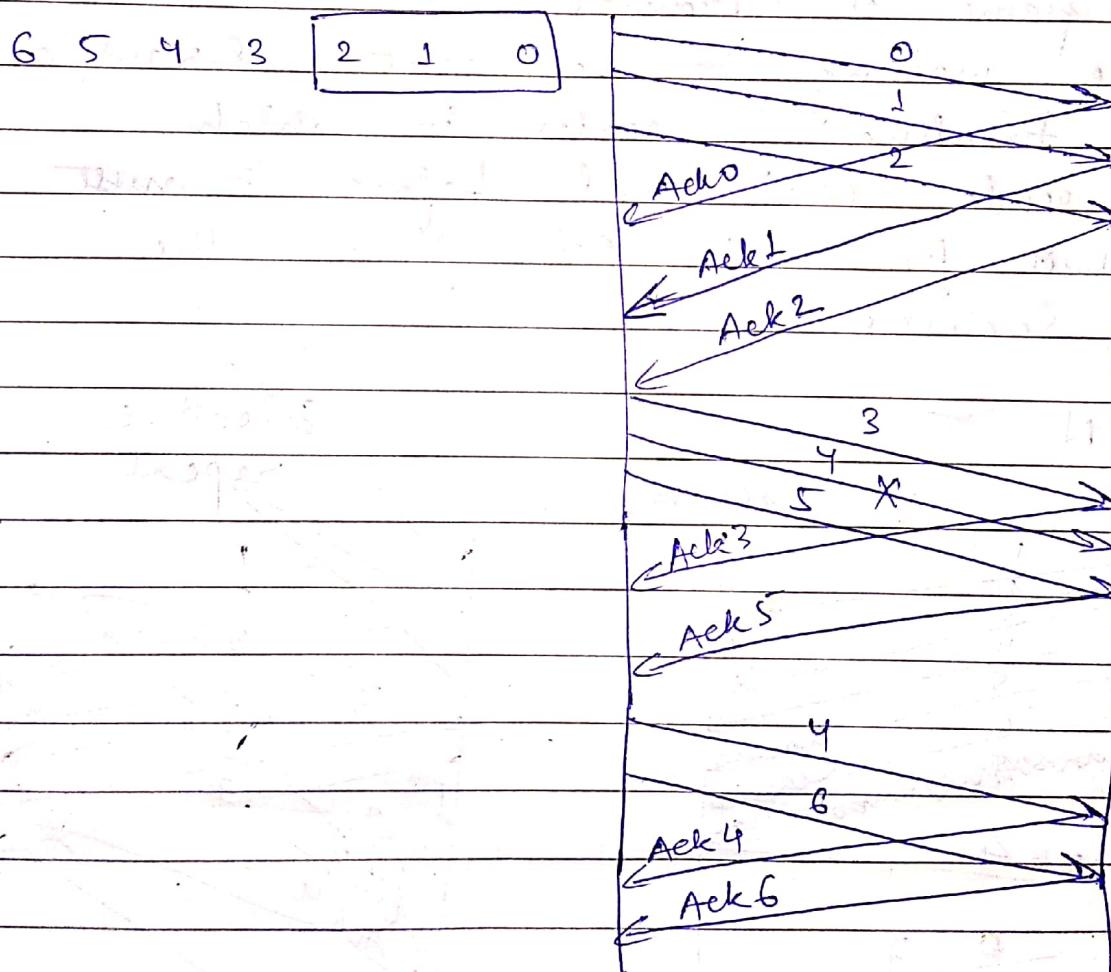
- sort all the packet after receiving at receiver side.

- use independent acknowledgement.

- sender window size = receiver window size

$$W_S = W_R = N$$

$\# N=3$



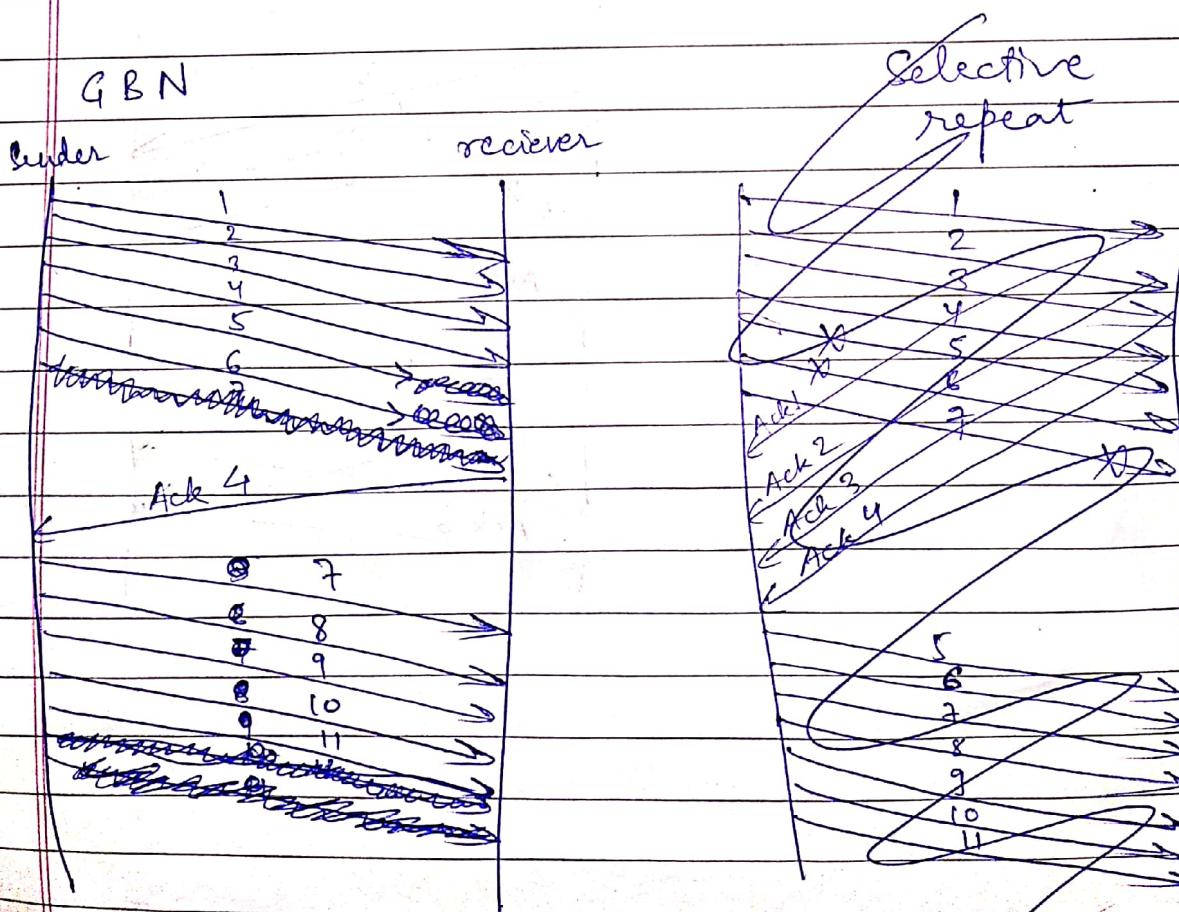
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Suppose K max. no. of bits available
for seq. nos.

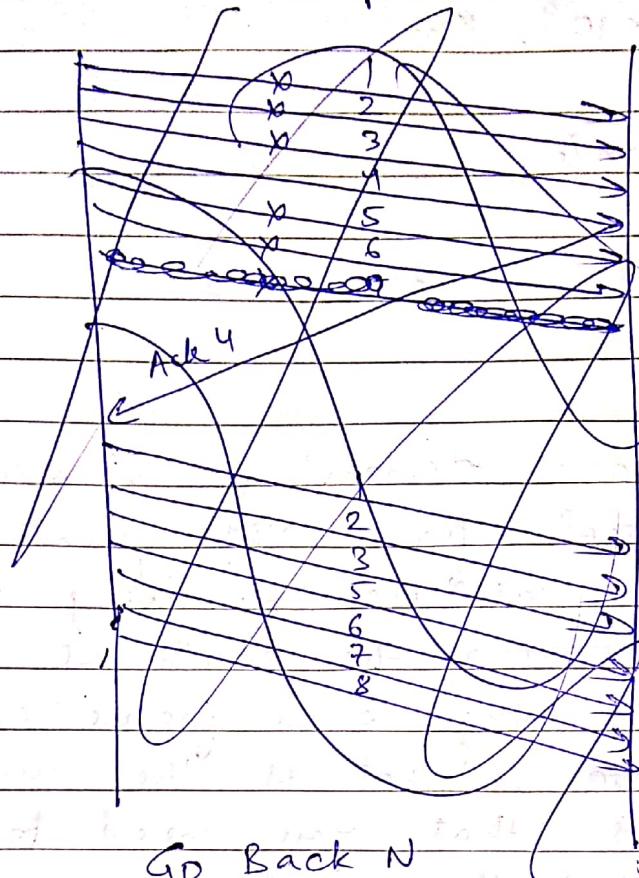
$$\text{then } W_s + W_r = 2^K$$

$$\begin{aligned} \therefore 2W_s &= 2^K \\ \therefore W_s &= 2^{K-1} \end{aligned}$$

Ques - Go back N protocol having a window size of 7, sending frames as 1 2 3 4 5 6, after sending the frame sender received an Ack for frame 4. Frames 7 8 9 10 11 12 are waiting to be sent - Draw a timeline for sender in which sender can send before it must wait for next Ack from the receiver.

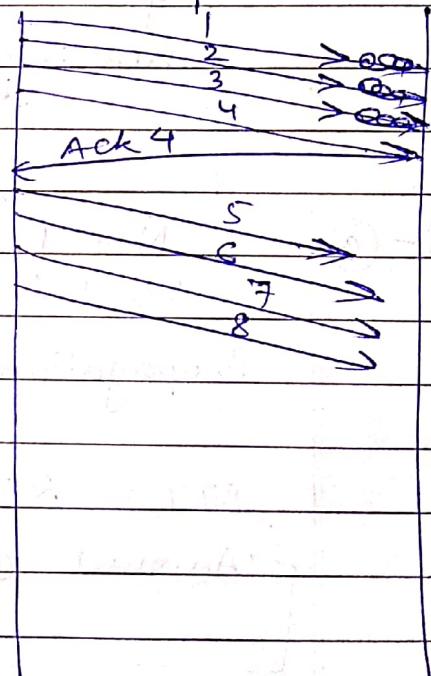
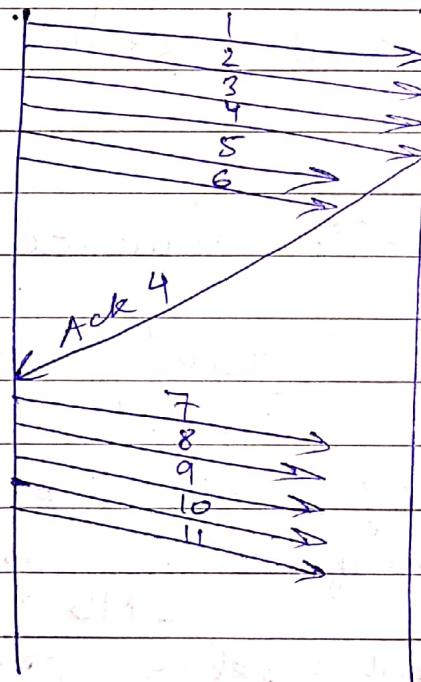


Selective repeat



Go Back N

Selective repeat



NOTE: Until & unless it is mentioned that packet got lost, we cannot assume it.

Date / /

Q- $BW = 1 \text{ Mbps}$

Altitude = $3 \times 10^4 \text{ Km}$

$L = 1 \text{ KB}$

$S = 3 \times 10^8 \text{ m/s}$

P a) Receiver window size = 1

b) $W_R = W_S$

Suppose you are designing a sliding window protocol for 1Mbps point to point to stationary satellite revolving at around altitude. Assuming the earth at $3 \times 10^4 \text{ KM}$ altitude. Assuming each frame carrying 1KB of data, what is the min. no. of bits that you need for the seq. no. in the following cases. assuming signal speed = $3 \times 10^8 \text{ m/s}$.

Sol - a) $N + 1 \leq \text{Available sequence no.}$
 $= W_S + 1$

Propagation delay = $10^{-1} \frac{3 \times 10^4 \times 10^3}{3 \times 10^8} s = 0.1s.$

RTT = $2 \times \text{Prop. delay} = 2 \times 0.1 s = 0.2 s.$

Amount of data transferred in 0.2 s. =

$$1 \text{ Mb} \times 0.2 = 0.2 \text{ Mb.}$$

b) ~~$W_S + W_R \leq \text{Available seq. no.}$~~

~~$\leq 2W_S \leq \text{Available seq. no.}$~~

Date -

$$\text{NBB} \times 10^5 \\ 1 \times 10^3 \times 8$$

$$1.2 \times 10^5 \\ 4.8 \times 10^3$$

$$\Rightarrow W_S = 25$$

(a) $\therefore \text{Available seq. no.} = 25 + 1$
 $= 26$

$$\therefore \text{No. of bits} = 26 \quad 5$$

(b) $W_S = 25 = W_R$

$$\therefore W_S + W_R = 50$$

$$\therefore \text{No. of bits} = 6$$

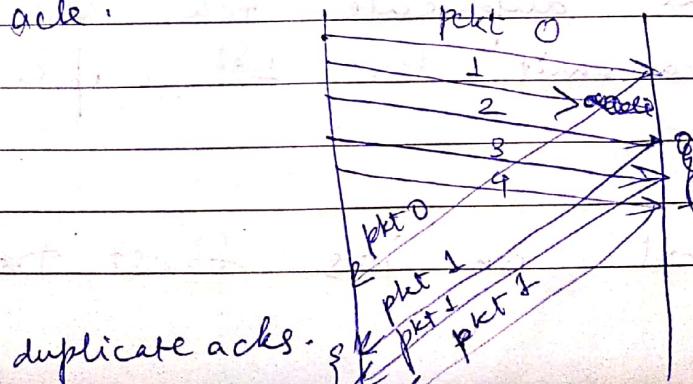
* Fast Retransmission (3 duplicate Ack)

We have different timers in TCP . e.g. timeout.

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

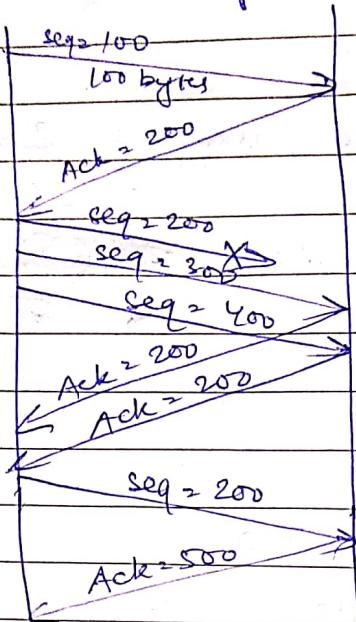
- If we don't get acknowledgement even after timeout , then we would resend the packet .

- See whether I am getting a duplicate ack .



duplicate acks .

- As soon as we receive 3 duplicate acks, then we retransmit the packet irrespective of the timeout.



NOTE: If last packet got lost then we need to wait for the timeout.

Q - Draw a Timeline diagram for sliding window algo with $W_S = W_R = 3$, for the following -

Use a timeout interval of $2 \times RTT$

Assume fast retransmission with

selective retransmission is implemented

& that packet with seq. no. 1 - 6 will

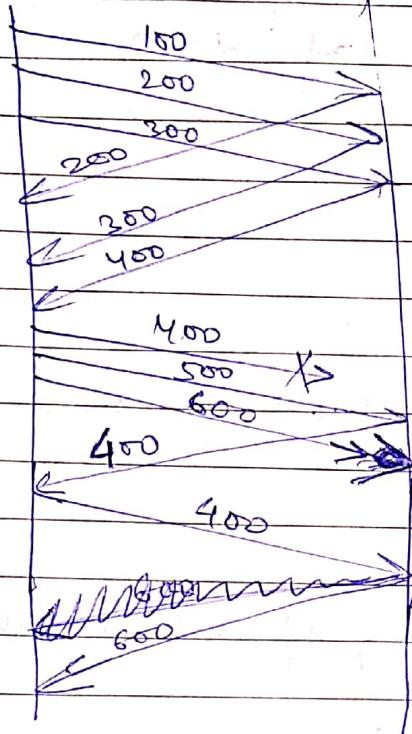
be sent for fast retransmission if more than one duplicate ACK is received.

Only retransmit for 1st duplicate ACK.

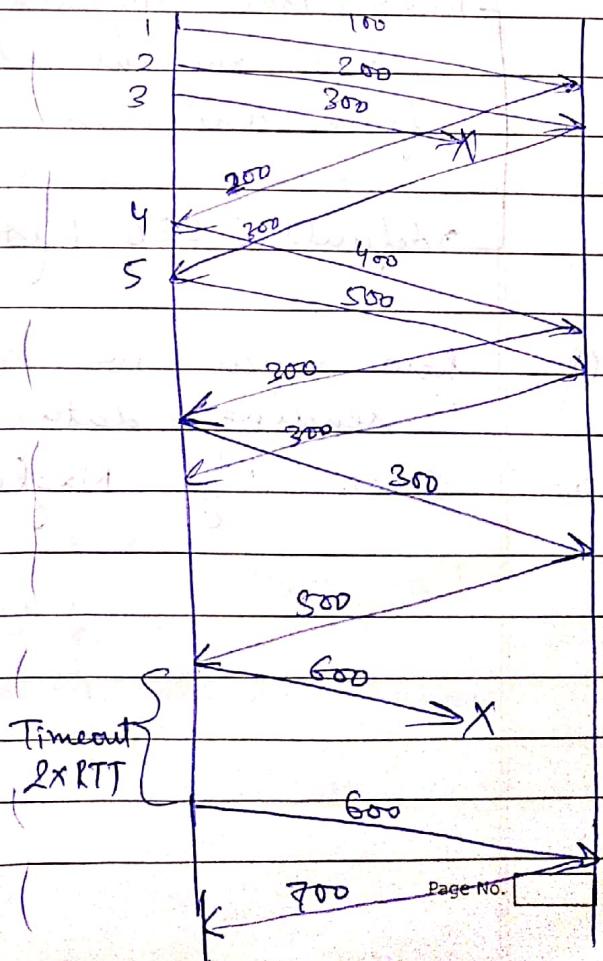
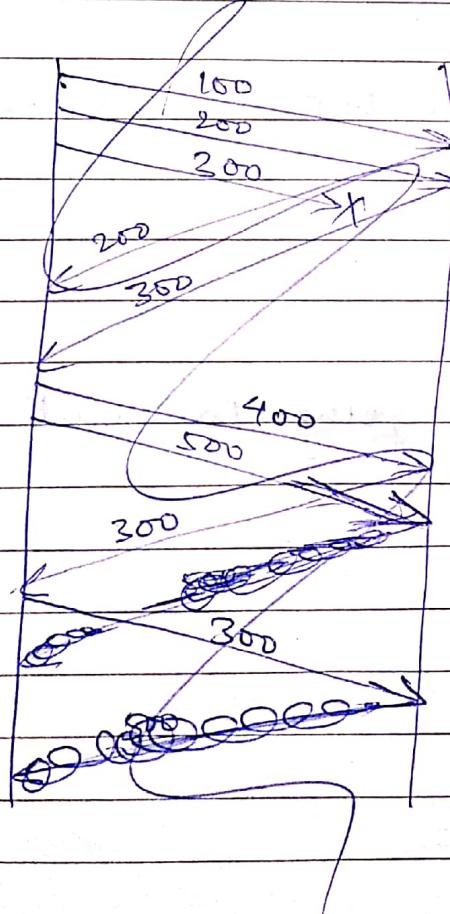
Q - 4 is lost on its first transmission

b) 3 & 6 got lost in their first transmission.

(a)



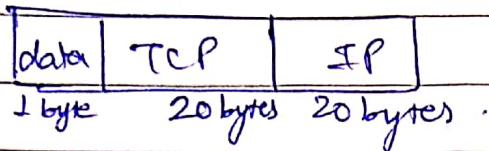
(b)



Flow control

Silly window syndrome :

sending 41 bytes of data for
sending 1 byte of data.



→ Sender : sending speed is slow.

→ Receiver : receiving speed is slow.

Sender side

MSS (Max. segment size)

max. amount of data that can be sent
from sender to receiver.

→ default : 536 bytes

Q - how much we are going to wait for
sending data ?

A - Given by Nagle's algo.

NAGLE's ALGORITHM

- # for first segment send the data, irrespective of segment size.
- # If there is an unacknowledged data or data in flight, then wait for either ack or for TMSS of data.

if (1st transmission)

send packet

else if (unacknowledged data)

wait for 1 MCS

else

send data.

first

NOTE: if a packet got lost, then wait for retransmission (timeout)

Receiver side

1KB | 1KB | 1KB

(S)

1 byte at a time

(R)



Receiver buffer full

Advertisement window = 1



wait until it becomes larger
(Clark's solution)

Date / /

→ Clark's Sol. :

delay acknowledgement if advertisement window is = near zero.
 until either to accomodate 1MSS or
 until atleast half of receiver buffer
 is empty.

→ Advantage of Clark's & Nagle's sol. :

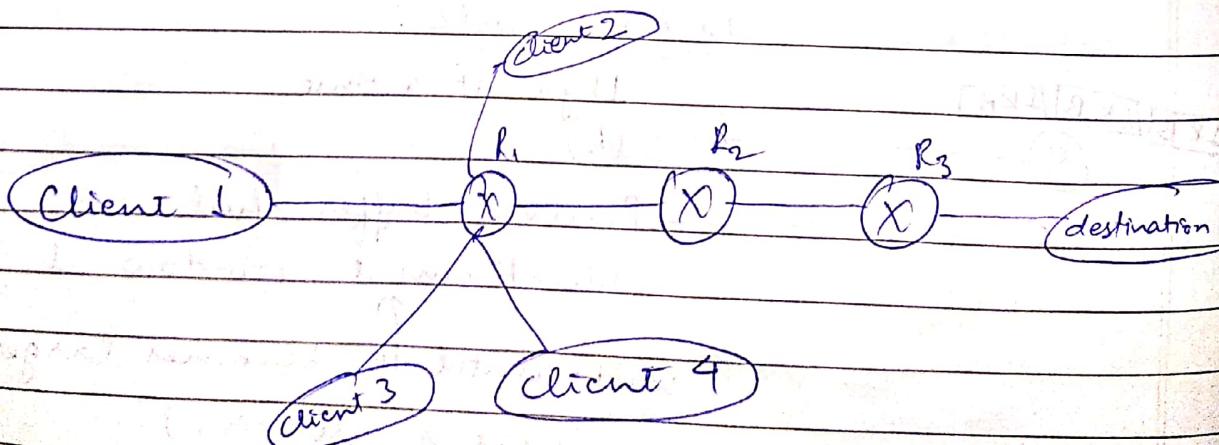
- reduces traffic.
-

→ Disadvantage :

- if timeout happens , then you need to retransmit , although receiver has received the packets but delayed ack.

Congestion control

- duplicate acks . (retransmission)
- ack time for acks . increases .



Header doesn't contain any field to guess congestion on the traffic.

Date 1/1

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processing delay : checks for error + seq check
check hai ya nahi packet mein

strategy : reduce no. of packets

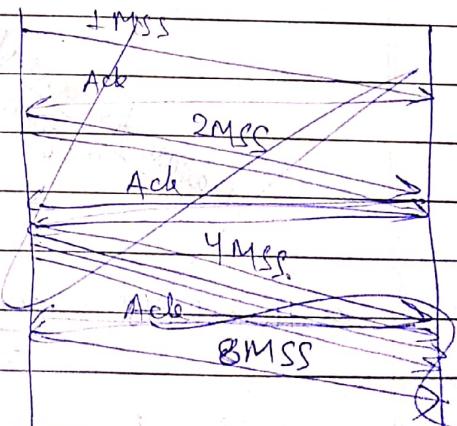
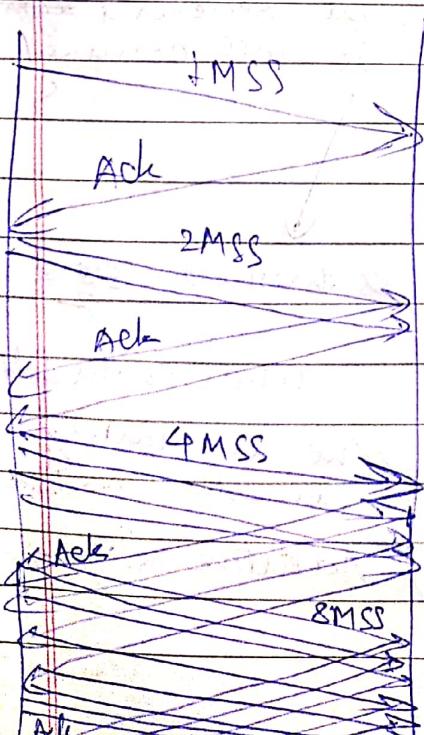
congestion window - resides at the sender.

perception of congestion in the network.

- 3 phases :-
- ① slow start
 - ② Congestion avoidance
 - ③ Congestion detection.

→ upto a certain threshold, we incⁿ congestion window size exponentially.

a. cwnd = 1MSS (initially)
congestion window size = 1MSS (max 536 bytes)



ss threshold = 8 MSS (here)

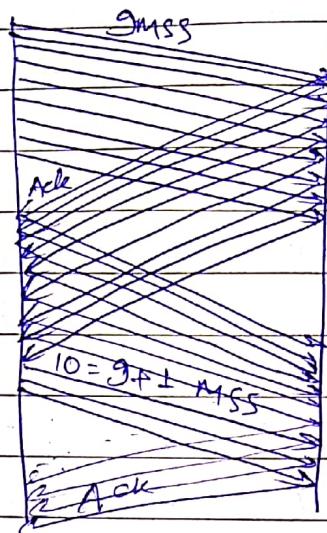
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⇒ As soon as we reach threshold, second phase starts.

Now cwnd increases one-by-one.

(additive increase)
(until any packet gets lost)

So now, cwnd becomes $8 + 1 = 9$.



Retransmission → RTO timeout (stronger reason for congestion control)

→ Fast Retransmission

(sending segment ~~now~~ gets lost, but receiver's ~~ack~~ ack is still coming)

measure for controlling congestion

$$ss\text{ threshold} = \frac{cwnd}{2}$$

$$cwnd = 1$$

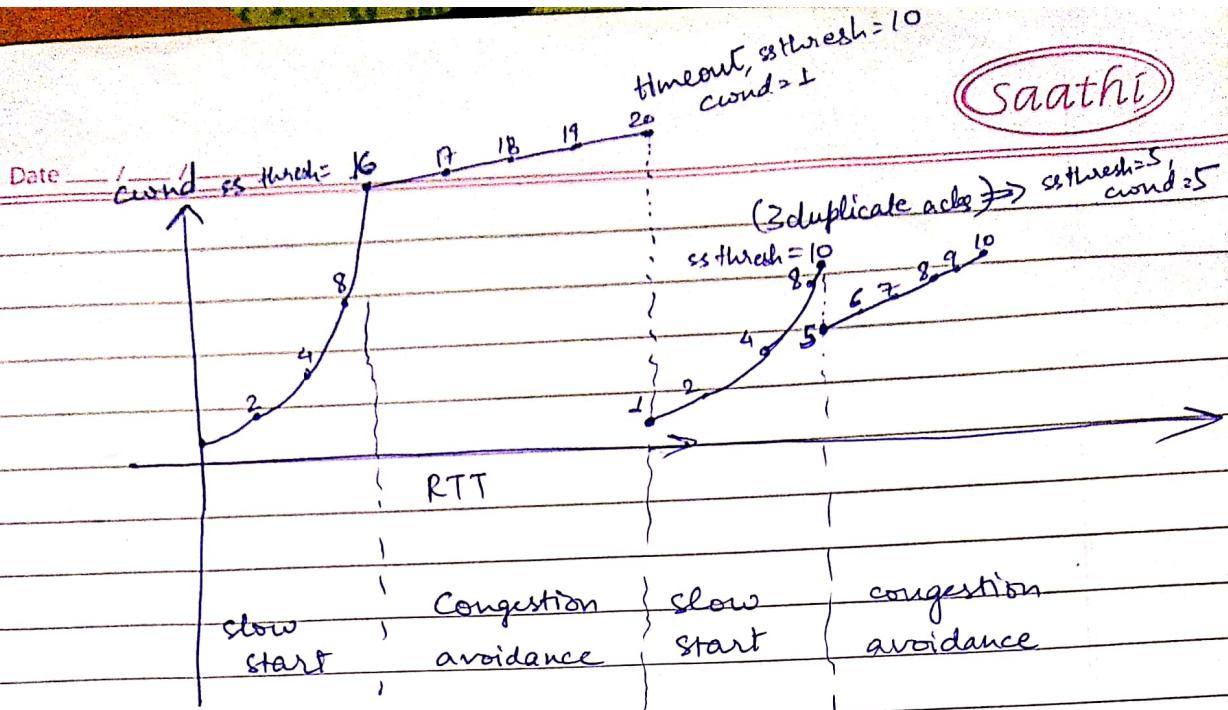
// we are starting slow start phase again

// exponential inc↑

$$ss\text{ threshold} = \frac{cwnd}{2}$$

$$cwnd = ss\text{ threshold}$$

// we are starting congestion avoidance
// Additive inc↑ only



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