

## Transport layer

- \* Provides logical communication b/w application process running on diff host.
- \* [from application Perspective], Not directly connected host
- \* Transport layer protocol implemented in end systems not in routers.
- \* breaks into segments, add a transport layer header to each chunk.
- \* Why do we need transport layer?
- \* Why do transport layer need error correction, error control when we have this facility in data link layer?
- \* N/W layer provides logical communication b/w host.
- \* Message delivery
- \* Message given (1) end to end connectivity
- \* Landline      Telephone line
- \* N/W layer

Traditional Indian Telephone usage

→ Now after sometime you are frustrated occasionally when phone ring you say, person is not there at their house, means dropping message occasionally.

- \* IP protocol is based on "best effort delivery service"  
guarantee of segment delivery, no orderly delivery. of seg.  
no guarantee of integrity of data, IP unreliable  
service.

- \* integrity checking by error detection field.

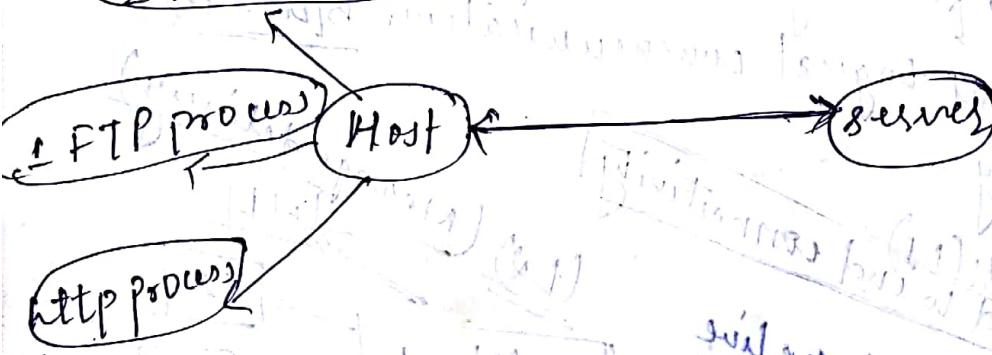
## TCP

- \* reliable data transfer → flow control, sequence number,  
acknowledgements, times.

- \* congestion control → byte stream oriented transport protocol.

- \* TCP uses sliding window protocol.

### 2. Telnet Process



- \* Transport layer delivers data to socket of incoming host.  
Each socket has unique identifier.

- \* Job of delivering the data in a transport layer segment to the correct socket is called demultiplexing.

- \* Multiplexing - Job of gathering data chunks at the source host from diff sockets, encapsulating diff each data chunk at the source host from diff sockets encapsulating each data chunks with headers info. to create segments and passing the segment to the NW layer is called multiplexing.

Port Number - 16 bit, 0 to 65535

0 to 1023  $\rightarrow$  well known port numbers.

source port numbers and destination port numbers in header.

\* UDP as transport protocol does multiplexing/demultiplexing function and some light error checking.

\* UDP - connection less, no handshaking, process to process delivery and error checking.

\* Why UDP is needed when TCP is already there?  
→ finer application level control over what data is send and when - UDP takes data and send, TCP uses congestion control mechanisms, that throttles the transport layer & TCP uses resend mechanism regardless of how long it takes. Not suited for real time application.

→ No connection establishment - No delay for connection establishment.

→ No connection state - buffers, congestion control para seq no, ack no.

→ small packet header overhead -

\* Rippled Reliable data transfer Protocol

### TCP connection

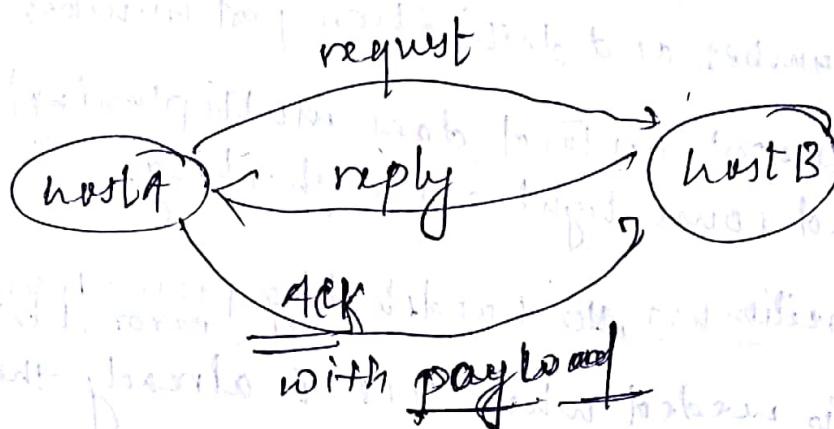
\* for connection establishment two process must handshake

\* TCP protocol runs on end systems, the intermediate networks do not maintain TCP connection state.

\* full duplex service

\* TCP connection are always point to point, multicasting is not possible.

## \* Three way handshake



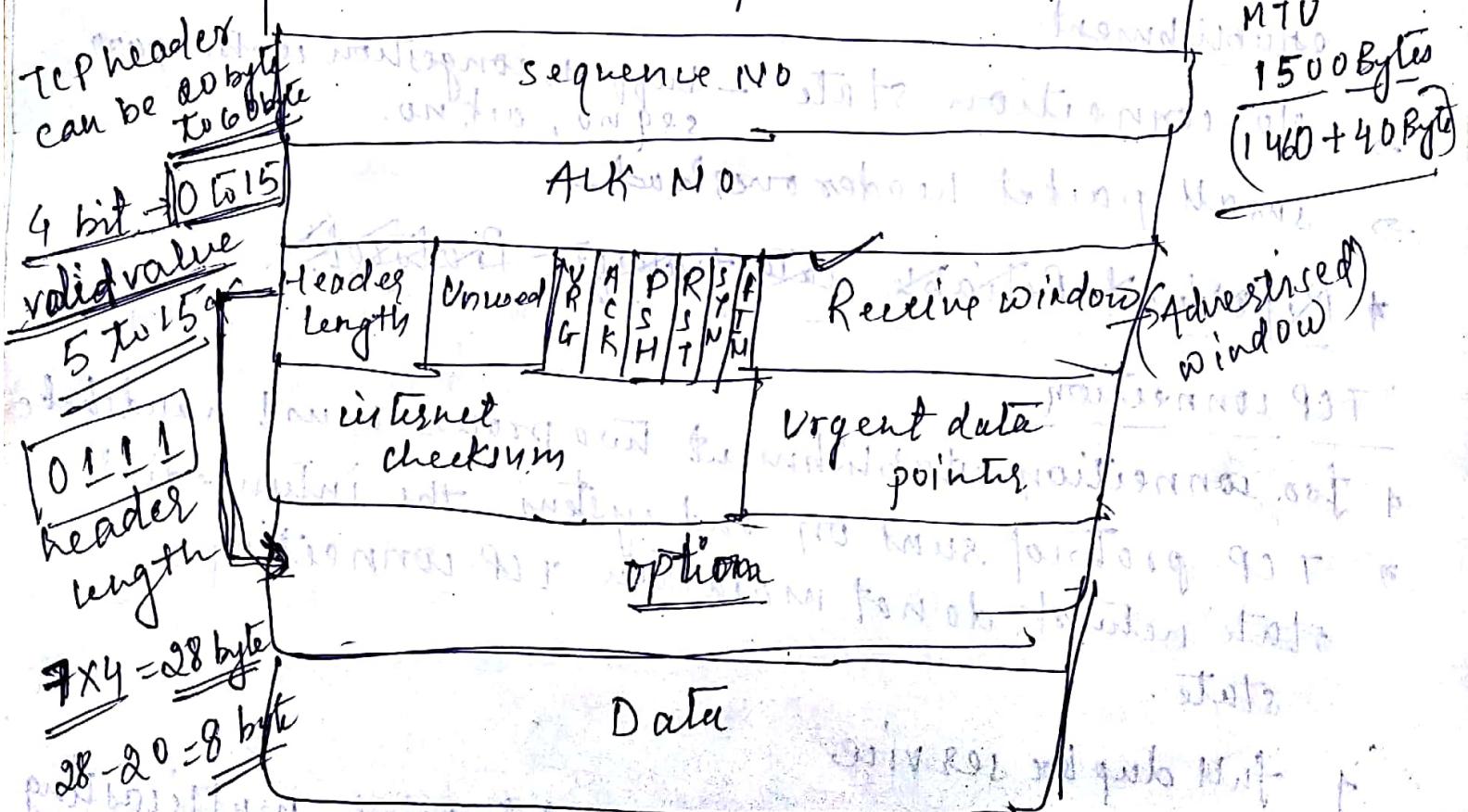
- \* MSS = is the Maximum amount of application layer data in the segment, not the maximum size of TCP segment including headers.

## TCP segment structure

(Max packet size of TCP connection 64 KB), but not practical

because of MTU

$$1500 \text{ Bytes} \\ (1460 + 40 \text{ Bytes})$$



sequence no & ACK no  $\rightarrow$  to implement a reliable data transfer service and ordered delivery.

- \* Receive window (16 bit) - used for flow control.
- \* header length field (4bit) - specifies the length of TCP header in 132 bit words. The TCP header can be variable length due to TCP option field.
- \* The option ~~an~~ field is used when a sender and receiver negotiate the Maximum segment size or as a window scaling factor for use in high speed networks.

Flag field - 6 bits -

ACK - the segment contain ACK for a segment

ACK = 1  
data + ACK that has been successfully received.

ACK = 0  
invalid

connection setup & teardown.

PSH - receiver should pass the data to upper layer immediately.

URG - there is a data in segment that the

sending side upper layer entity masked as urgent

\* 16 bit Urgent Pointer - The location of the last byte of urgent data is indicated by 16 bit urgent data pointer field. TCP must inform the receiving side upper layer entity when urgent data exist and pass it a pointer to the end of urgent data.

## Packet recognition:-

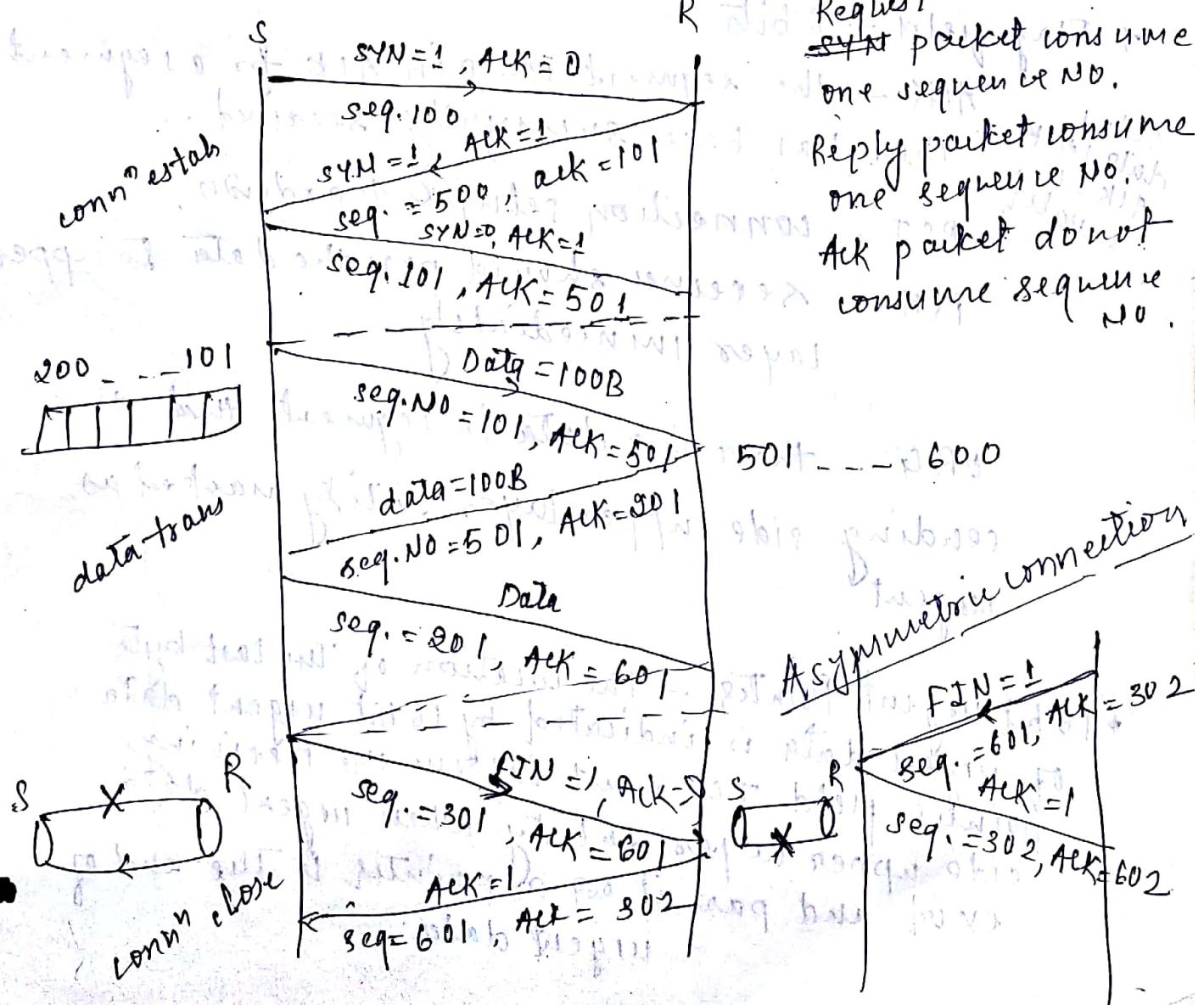
	SYN	ACK
request packet	1	0
reply	1	1
<del>deflated ACK</del>	0	1
purely data packet	0	0

FIN - used to terminate the connection.

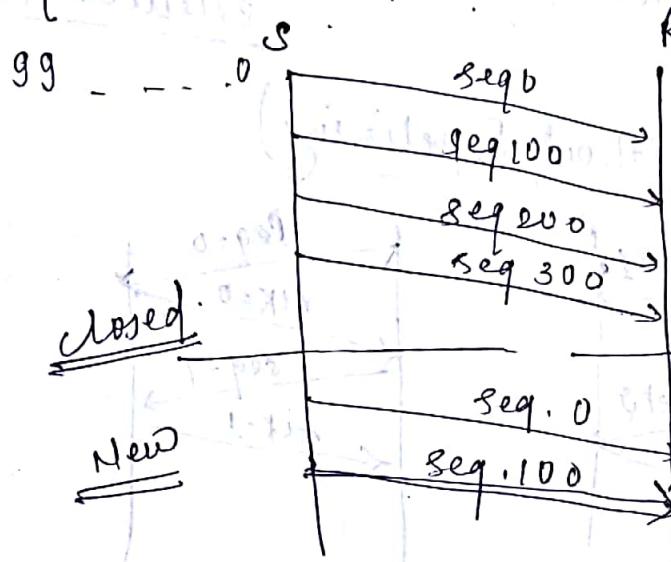
RST Flag - it is used to refresh the connection.

~~sequence~~

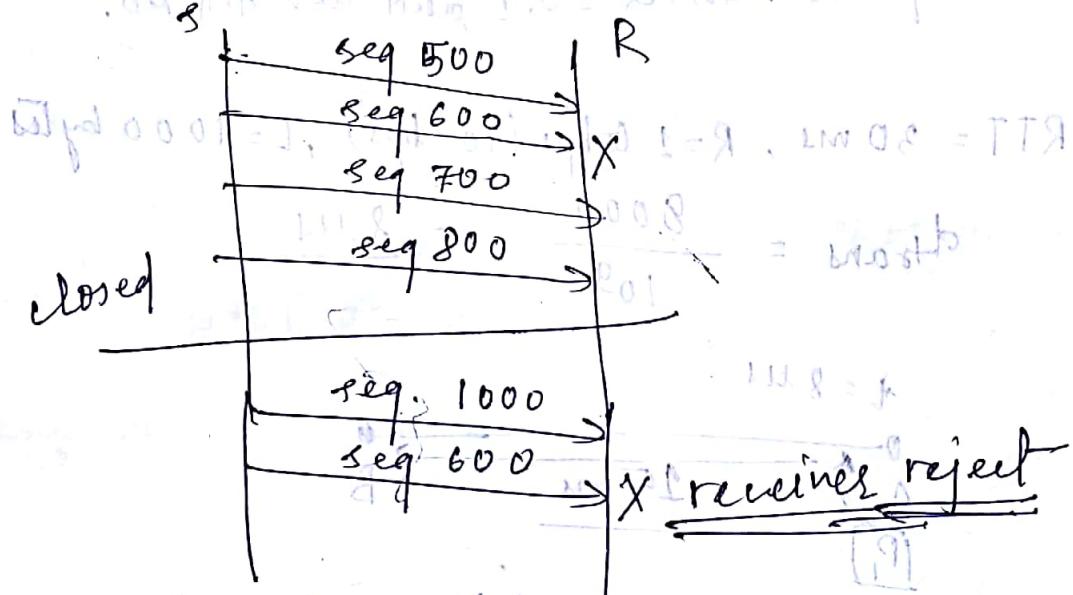
Sequence Number and Acknowledgement No.



Why each connection starts with random initial sequence numbers.



What happens if host S receives sequence numbers it did not expect? The receiver needs to reject but did not.



~~Security issue~~ = problem now 100

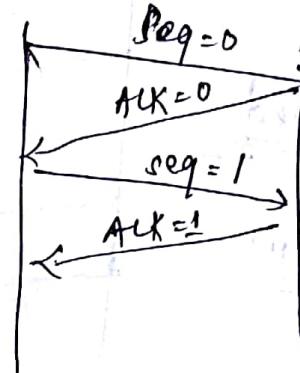
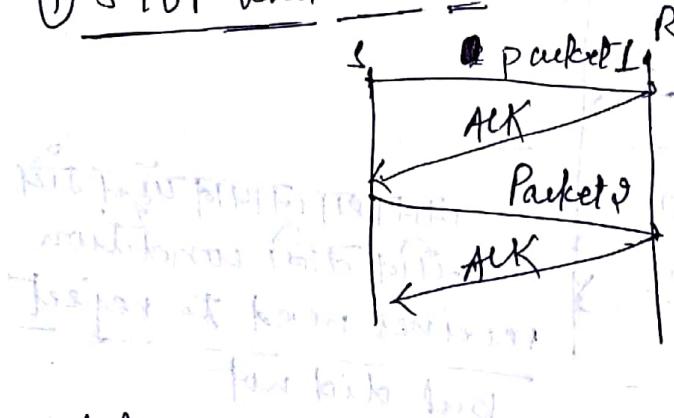
1. TCP views data as ordered stream of bytes. Sequence no reflects this view as the stream of transmitted bytes and not over the series of transmitted segments.

2. The acknowledgement number that receiver puts in the segment is the sequence number of the next byte receiver is expecting from host A.

Pipelined reliable data transfer protocol  
 → Also called sliding window protocol  $\Rightarrow$  Go Back N & Selective Repeat

Non-pipelined protocol

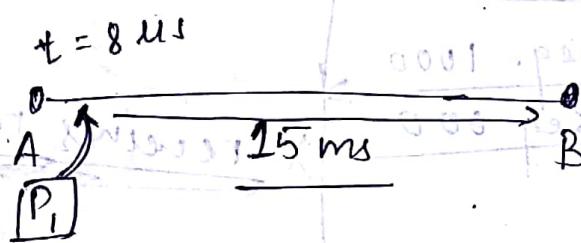
① STOP and Wait (Without Pipelining)



\* of sequence number = 0, 1, and two ACK NO.

\* RTT = 30 ms, R = 1 Gbps ( $10^9$  bps), L = 1000 bytes

$$d_{trans} = \frac{8000}{10^9} = 8 \mu s$$



One way delay = 15.008 ms

No ACK is very small, ACK will take 15 ms to reach to A.

$$t = RTT + \frac{L}{R} = 30.008 \text{ ms}$$

in 30.008 ms, host is sending for only = 0.008 ms

$$\text{Efficiency utilization} = \frac{0.008}{RTT + \frac{L}{R}} = \frac{0.008}{30.008} = 0.00027$$

$$\text{throughput} = \frac{1000}{30.008} = 26.7 \text{ kbps} = \boxed{26.7 \text{ kbps}}$$

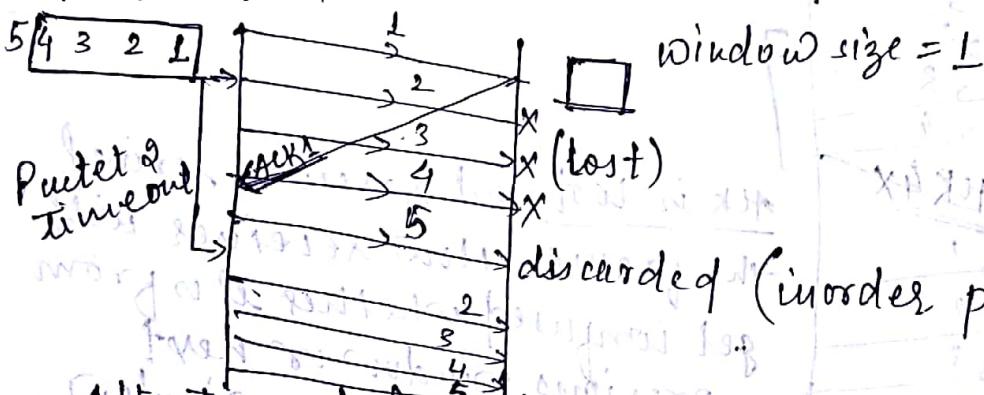
## To Back N :-

5

A sender is allowed to transmit multiple packets without waiting for an acknowledgement.

- \* Not more than N packet, unacknowledged packet is allowed.
- \* It requires large buffer size.
- \* More sequence numbers are needed.

window size = 4



Window size = 1  
window size = 1

- \* After timeout for 2nd packet sender will know that packets in windows 2, 3, 4, 5 are lost so resend.

- \* ~~Acknowledgement~~ acknowledgement are two types:-

(1) independent

(2) cumulative

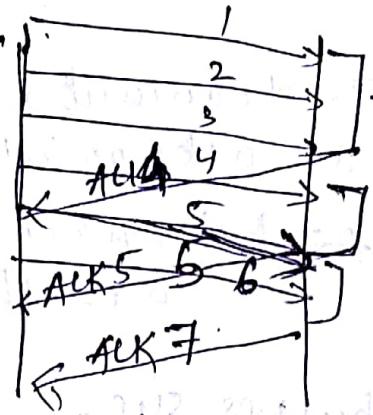
- \* For GBN cumulative ACK is used :-

GBN maintains a timer called acknowledgement timer which starts when any packet is received. When ACK timer expires a single acknowledgement will be sent for all the packets within this time.

If ACK timer is too big, it leads to timeout & retransmission if ACK timer is too small, it becomes independent ACK & more traffic.

8 7 6 5 4 3 2 1

Timeout  
Timer



ACK timer

if partition function

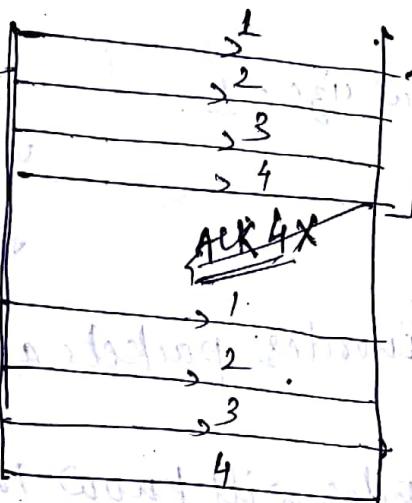
is handling poll +

backoff

Relation between window size and sequence number:

GB21

4 3 2 1



Window size = 4  
sequence no = 4

(lost frame in sequence)

ACKing frame with 4

so if window size is N, then min no of seq. no required =  $N+1$ .

If k is the max no of bits available in sequence number of field, then what is max window size

$$N = 2^k - 1$$

$W_S + W_R \leq \text{Available sequence No.}$

Suppose you are designing a sliding window protocol for a 1 Mbps point to point link to stationary satellite revolving around the earth at  $3 \times 10^4$  km altitude. Assuming that each frame carries 1 kB of data, what is the minimum number of bits that you need for the sequence number in the following case? Assuming speed of light  $3 \times 10^8$  m/s

(a)  $RWS = 1$

$[W_S + W_R \leq \text{Available Sequence No}]$

(b)  $RWS = SWS$

$$\underline{\text{soln}} \rightarrow \text{One way latency} = \frac{3 \times 10^4 \times 10^3}{3 \times 10^8} \\ = 100 \text{ ms}$$

$$RTT = 200 \text{ ms} = 0.2 \text{ s}$$

$$\text{data send in one RTT} = BW \times RTT \\ = \cancel{300} \quad 1 \text{ Mb} \times 0.2 \\ = 0.2 \text{ Mb}$$

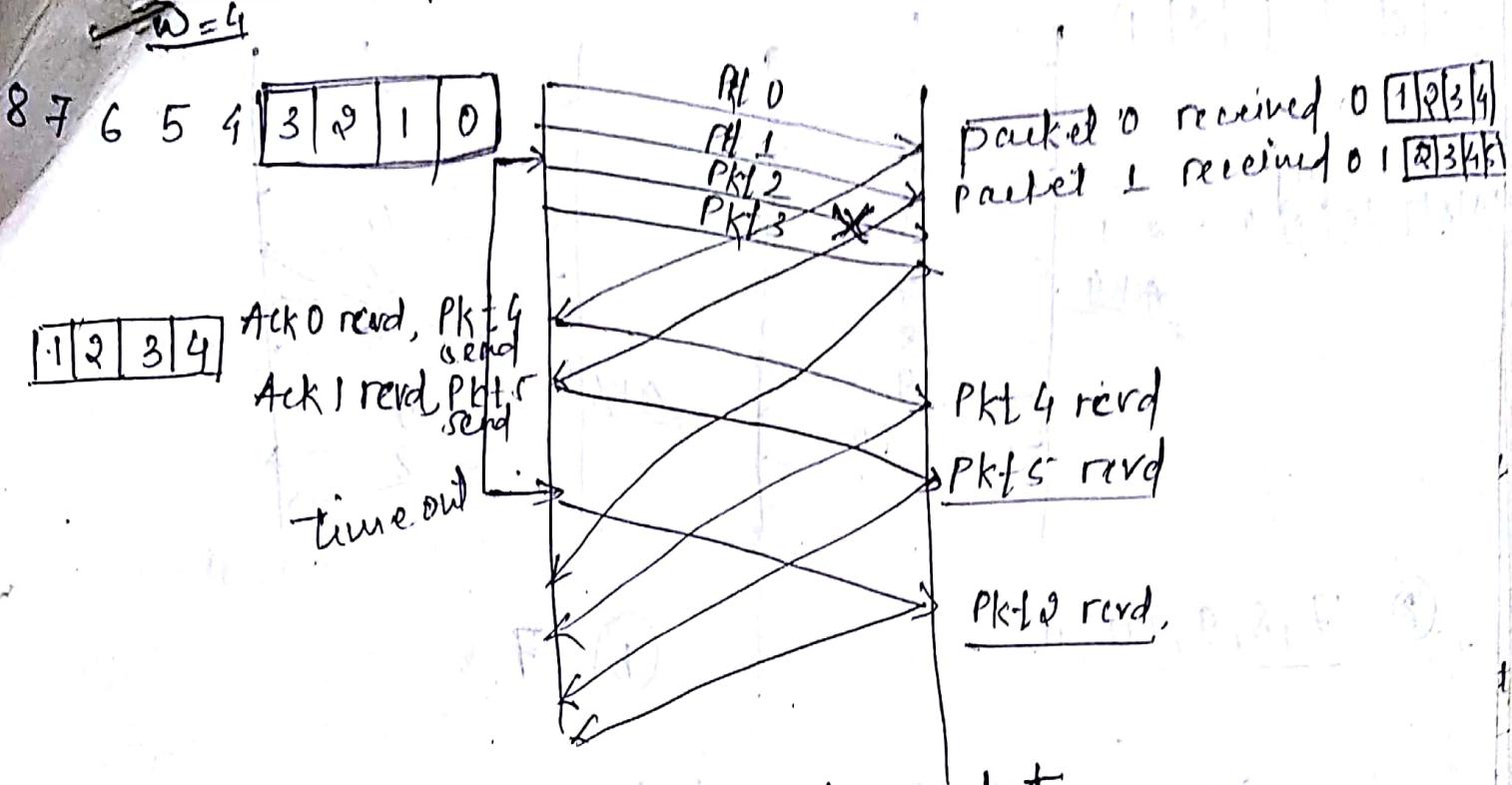
NO of packets simultaneously send ~~for that~~

$$= \frac{0.2 \times 10^6}{8 \times 10^3} = 25 \text{ per frame}$$

(a)  $RWS = 1$ , necessary sequence number =  $25 + 1 = 26$   
NO of bits = 5

(b)  $RWS = SWS$ , necessary seq. no required  
=  $25 + 25 = 50$ , NO of bits = 6

## Selective Repeat operation



1 Receives can accept out of order packets.

2 sender will send only lost packet.

3 Here if sender window = receiver window

4 If  $k$  is no of bits in sequence numbers then window size:

$$W_s = \frac{q^k}{q} = q^{k-1}, \quad W_R = \frac{q^k - 1}{q} = q^{k-1}$$

$$2^m \geq s+r \\ 2^m \geq 2x$$

### Ques 1

① Go back N protocol, window size = 7, Frames sent = 1, 2, 3, 4, 5, 6, sender received an ACK for frame 4. Frames 7, 8, 9, 10, 11, 12 waiting to be sent. Draw Timeline.

② Which frames can sender send before it must wait for the next ACK from the receiver?

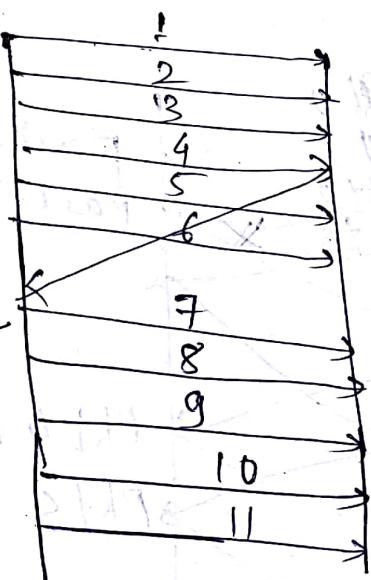
③ Repeat for Selective Repeat.

Go back N

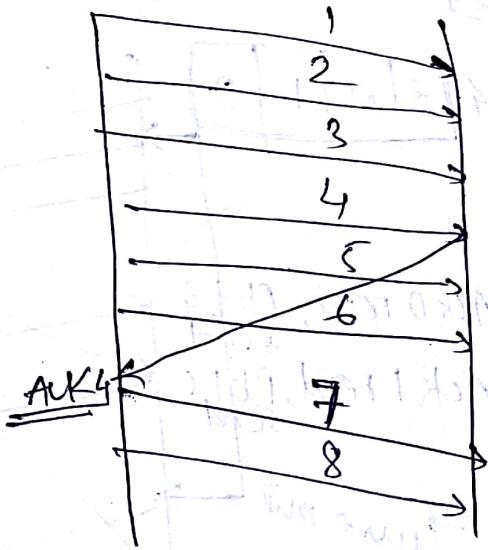
8	7	6	5	4	3	2	1
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9	8	7	10	11	5
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ACK 4



Selective Repeat



②

7, 8, 9, 10, 11

③

7, 8