

Internet Protocols HomeWork Assignment 3

1. P1

(a) Possible source and destination port numbers for the segment sent from A to S.

The ports 1024 and below are used for standard services. However ports b/w 1024 - 65535 can be used for connection by clients.

So, Port for Server S = 23

A → S

Client A port can be b/w 1024 to 65535 (Eq 1555).

(b) Possible source and destination ports for segments sent from B to S.

Server port is 23

B → S

Client B port will be b/w 1024 to 65535 (Eq 1556)

(c) Segment sent from S to A.

Server Port 23

S → A.

Client A Port 1555 (1024 to 65535)

(d) The segments sent from S to B.

Server Port 23,

Client B port 1556 (1024 to 65535)

S → B.

(e) If A and B are different hosts it is possible that the source port number in segments from A to S is same that from B to S.

The TCP connection will have different IP address so they can be easily differentiated.

(f) A TCP connection is comprised of source IP address, source port number, destination IP address & destination port number.

So its not possible that, If Client A & Client B are on the same host they will never get assigned the same port by the underlying operating system.

P3 UDP and TCP use 1's complement for their checksums.

01010011
01100110
01110100

1's complement of these 8-bit bytes →
Adding the first 2 8-bit bytes -

$$\begin{array}{r} 01010011 \\ 01100110 \\ \hline 10111001 \end{array} \quad \begin{array}{l} \text{Adding this sum with the 3rd} \\ \text{bit byte} \end{array} \quad \begin{array}{r} 10111001 \\ 01110100 \\ \hline 10010110 \end{array} \quad \begin{array}{r} 00101101 \\ \hline 00101110 \end{array}$$

To get 1's complement of this sum invert of 1's

11010001

Why does UDP take the 1's complement -

Because the receiver adds up the checksum sent with the 8-bit bytes that will be sent. The sum of all this should be 11111111. If there is a zero somewhere in the middle. The receiver gets to know that there was an error.

No a one bit error can never go undetected while sending the data to the receiver.

The sum will reflect the 1 to become 0, or 0 to become 1

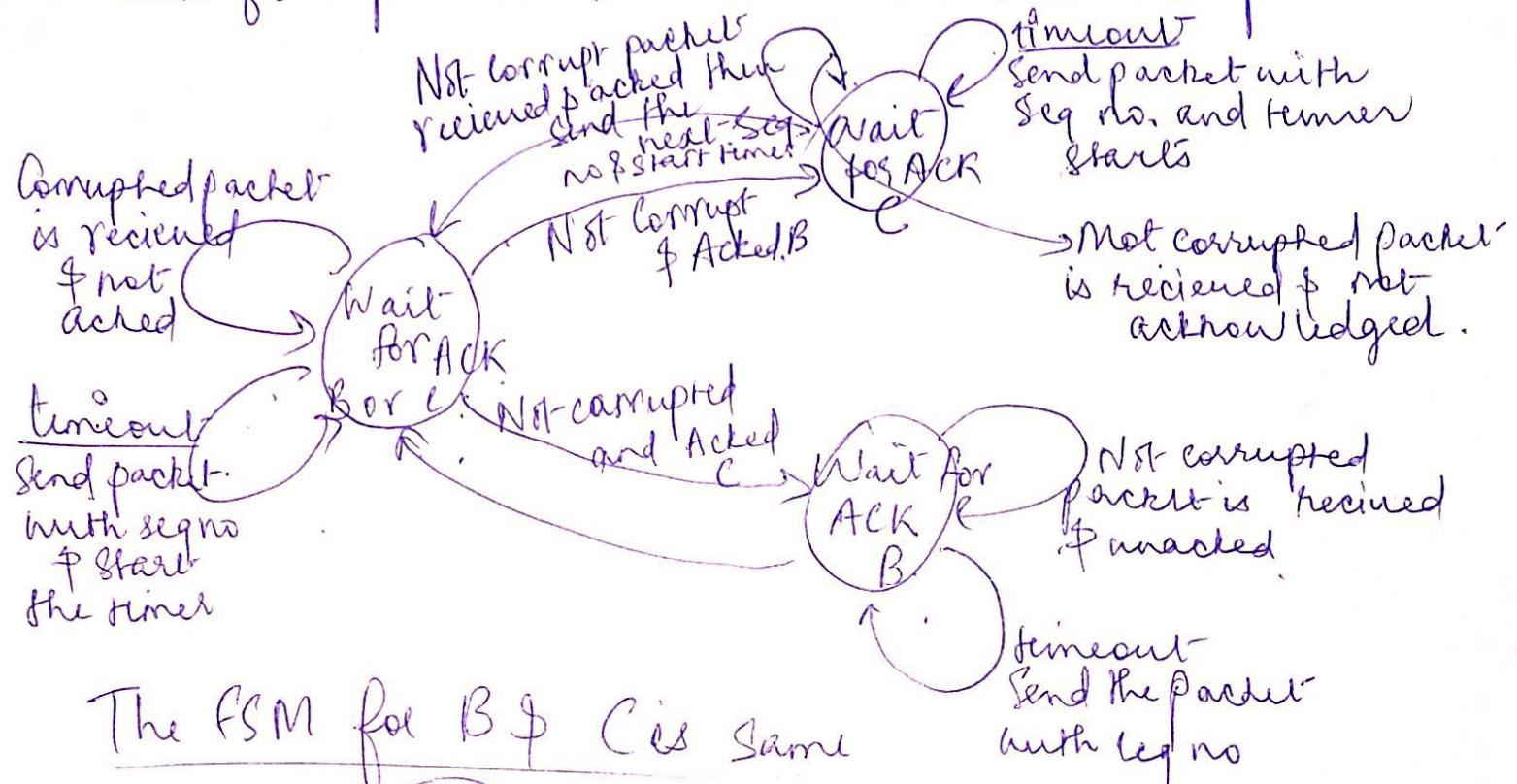
* Yes a 2 bit error can go undetected, when 2 bits are changed in such a way that their sum is the same. Eg -

$$\begin{array}{r} 1001 \\ 1011 \\ \hline 0100 \\ 101 \\ \hline 0101 \end{array} \quad \begin{array}{r} 1011 \\ 1001 \\ \hline 0100 \\ 0101 \\ \hline 0101 \end{array}$$

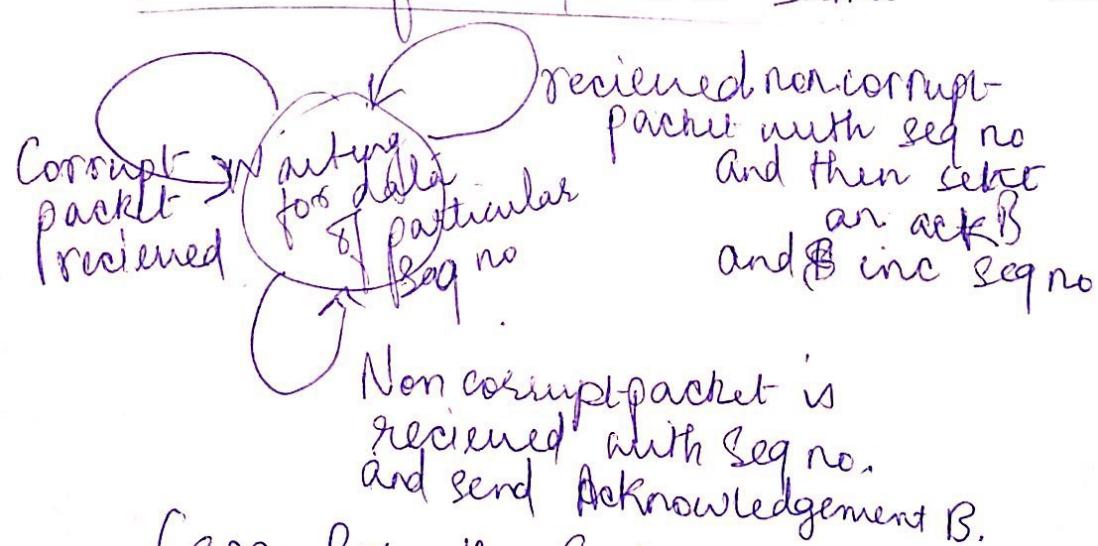
In this case we can see that the bits have changed but the checksum is the same

P6. let us consider that the sender is in the state "Wait for call 1 from above" Receiver is in the state "Wait for 1 from below" When the first packet will be sent from sender with seq number 1. The sender will go into "Wait for Ack or NAK". When the receiver will receive the packet it will send an acknowledgement. The receiver after sending the acknowledgement will move into the "Wait 0 from below" state. This means that the receiver is waiting for packet sequence number. Let's say that the acknowledgement which was sent by the receiver was corrupted. The sender on receiving a corrupted acknowledgement will resend the packet with sequence of 1. Now the receiver gets the packet with seq no 1 and sends a NAK. The sender will keep sending a packet with seq 1 and the receiver will keep sending NAK hence leading to a deadlock.

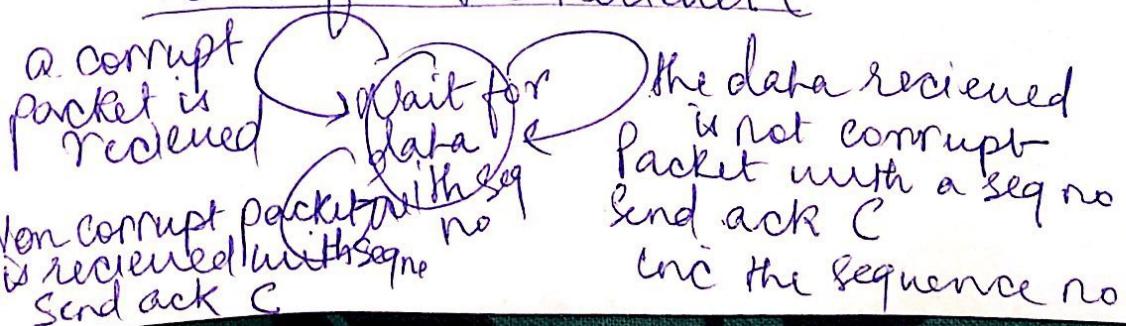
(P19) So when the data is being transmitted over a channel, the messages can be lost in the middle so it is an important thing to use sequence numbers for packets to retransmit the lost packets.



The FSM for B & C is same



FSM for the Receiver C



P19 FSM explanation →

SenderFSM

- Starts with Seq no 0
- Starts the timer and sends the packet to B & C
- Then wait for ACK
- If there is no ack in a particular time then broadcast the packet to B & C again.

ReceiverFSM -

- Wait to receive an uncorrupted data packet from B & C with the same seq no as the acknowledgement no.
- If a packet's seq no is not ack no, then loose the ackno and make the packet with ack |
- If the packle seq no matches with the acknowledgement no. Then send the ack no. to the sender

Q

P22 Consider the GBN protocol
Sender window size = 4
Sequence number range of 1024
at time t, the next inorder packet that receiver is expecting has a sequence number of K

(a) The possible sets of sequence numbers inside the sender's window at time t?

* Let's consider a case where all the acknowledgements have been received by the sender. In this case the sender window size will be $[K, K+1, K+2, K+3]$

* Now let's say in the other case none of the acknowledgements are received at the sender.

In this case the sender window will have N packets upto $K-1$ so it will have in its window

$$[K-4, K-3, K-2, K-1]$$

* Hence the sender window size is of length 4 & it has the following possibilities -

$$[K-4, K-3, K-2, K-1] \quad [K-3, K-2, K-1, K] \quad [K-2, K, K+1, K+2]$$

$$[K-1, K, K+1, K+2] \quad [K, K+1, K+2, K+3]$$

(b) If the receiver is waiting for packet K. Then it has acknowledged $K-1, K-2, K-3, K-4$

If none of the acknowledgements are received by the sender, then the ACKs $K-4, K-3, K-2, K-1$ might still be propagating.

If it's a case that the sender has sent $K-4, K-3, K-2, K-1$ then it has received an ack for $K-5$.

Once the receiver has sent an ack for $K-5$, it will never send an ack less than that.

Hence the possible values of ack that can be sent by the receiver are

$K-5, K-4, K-3, K-2, K-1$.

P26 Transferring an enormous file L Bytes from Host A to Host B. MSS = 536 Bytes TCP seq num field has 4 Bytes.

① What is the max val of L that TCP Seq no.'s are not exhausted?

The possible sequence numbers are 2^{32} .
The maximum file size that can be sent from A to B is simply 2^{32} Bytes as the Seq no increments by the bytes of data sent.

⑥ The number of segments will be -

$$\left\lceil \frac{2^{32}}{536} \right\rceil = 8012999$$

How long it takes to transmit the file -

- header size getting added = 66 Bytes per segment
- $66 \times 8012999 = 528857934$
- Size of file = 2^{32} Bytes.

$$\text{time} = \frac{4.824 \times 10^9}{155 \times 10^6} = 249 \text{ seconds}$$

P27

A & B communicating over a TCP Connection
B has received all bytes up to byte 126

1st & 2nd segments have 80 & 40 bytes

first Seg Seq no 127.

Source Port 302

Destination Port 80.

(a) Seq no = $127 + 80 = 207$

Source port = 302

Destination port = 80

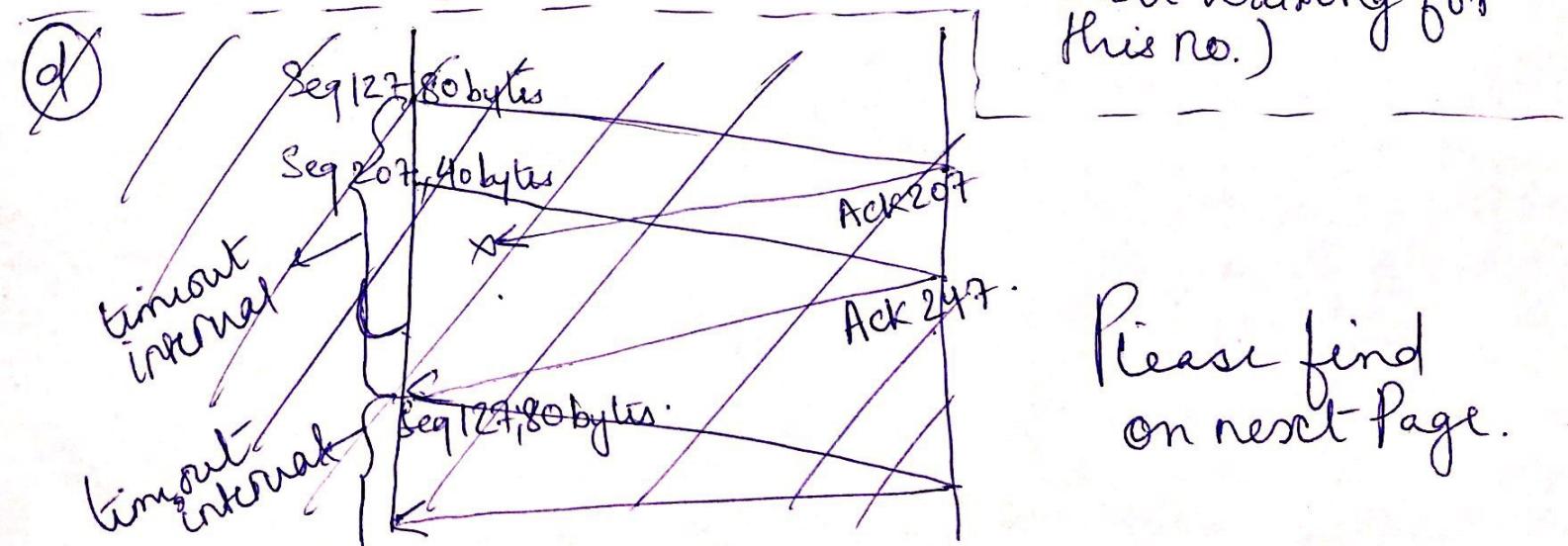
(b) The Acknowledgement no. = 207.

Source Port = 80

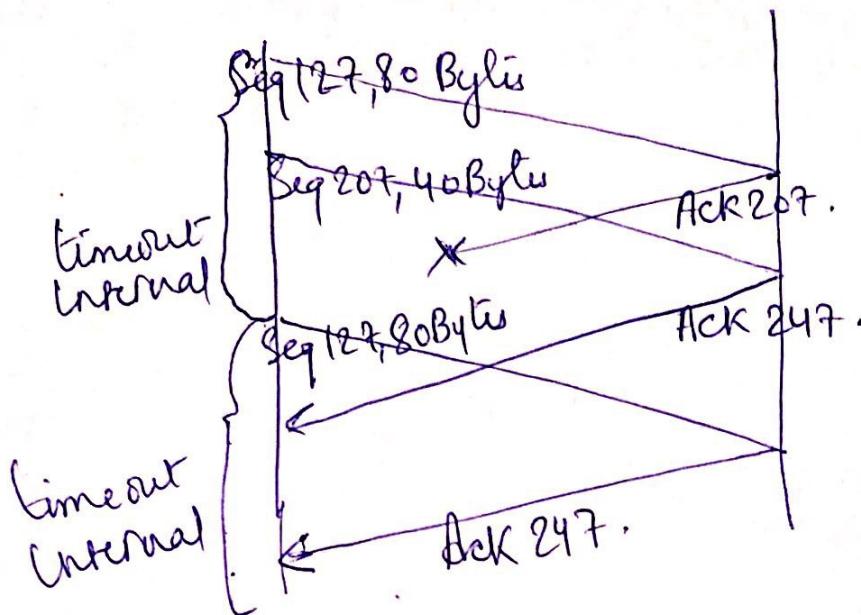
Destination Port 302.

(c) If second segment arrives before -

Acknowledgement no = 127. (Indicating it still waiting for this no.)



Please find
on next Page.



P32 ① Estimated RTT⁽¹⁾ = Sample RTT, $\boxed{\alpha = 0.1}$

Estimated RTT⁽²⁾ = α Sample RTT₁ + $(1-\alpha)$ Sample RTT₂.

Estimated RTT⁽³⁾ =

$$\alpha \times \text{Sample RTT}_1 + (1-\alpha) [\alpha \text{Sample RTT}_2 + (1-\alpha) \text{Sample RTT}_3]$$

$$= \alpha \text{Sample RTT}_1 + (1-\alpha) \alpha \text{Sample RTT}_2 \\ + (1-\alpha)^2 \text{Sample RTT}_3$$

Estimated RTT⁽⁴⁾ =

$$\alpha \text{Sample RTT}_1 + (1-\alpha) \text{Estimated RTT}^3.$$

$$= \alpha \text{Sample RTT}_1 + (1-\alpha) \alpha \text{Sample RTT}_2 \\ + (1-\alpha)^2 \alpha \text{Sample RTT}_3 + (1-\alpha)^3 \text{Sample RTT}_4$$

(b)

$$\text{Estimated RTT}^n = \alpha \sum_{j=1}^{n-1} (1-\alpha)^j \text{Sample RTT}_j$$

(c) Estimated RTT^(∞) =

$$= \frac{\alpha}{1-\alpha} \sum_{j=1}^{\infty} (1-\alpha)^j \text{Sample RTT}_j$$

$$= \frac{1}{9} \sum_{j=1}^{\infty} (0.9)^j \text{Sample RTT}_j$$

P 4 (a) TCP slow start is operating from intervals $[1, 6]$ and $[83, 26]$

The client doubles the amount of packets sent each time till the receiver receives an ACK meaning congestion detected or client's window limit is reached.

(b) Congestion avoidance is from $[6, 16]$ & $[17, 22]$

As we can see that the window is increasing additively.

(c) After the 16th transmission round, a segment loss is detected by a triple duplicate ack. If in case a timeout occurs the congestion window size is dropped to 1.

(d) After 22nd transmission round, segment loss is detected due to a timeout and in this case as well the congestion window will be set to 1.

(e) The value of threshold is set to half of the value of congestion window in case packet loss is detected. Hence when loss is detected at 16 congestion window size is 42, which is reduced to 21 at 18

(e) The threshold is initially set to 32. Since it is at this size of window slow start stops and Congestion avoidance phase begins.

(g) The threshold is set to half the value of the Congestion window when packet loss is determined.

~~At~~ At 22 \rightarrow Window Size = 26

At 24 \rightarrow Window Size = 13.

(h)	1 st Round.	1 packet
	2 nd	2, 3
	3 rd	4, 5, 6, 7
	4 th	8, 9, 10, 11, 12, 13, 14, 15
	5 th	16 to 31.
	6 th	32 to 63
	7 th	64 to 96

Thus 70th packet in 7th round.

(i) The cwnd & threshold will be $1/2$ the current cwnd (8). So new values of Threshold & cwnd = 4

(j) The cwnd size will be set to 1 first-
Then in 19th round it will increase to 4.

(k) 17 to 22. packets sent.

17 - 1

18 - 2

19 - 4.

20 - 8.

21 - 16

22 - 21. So it will be total 52 packets.