

Excercise 1.1

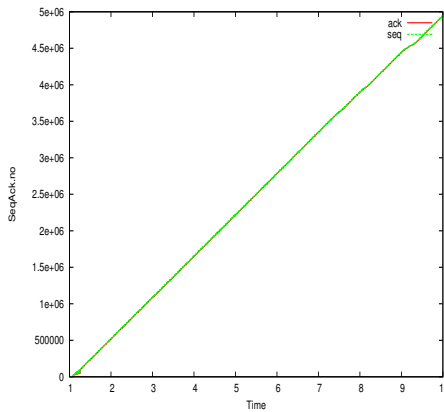


Figure 1: Time Sequence Graph

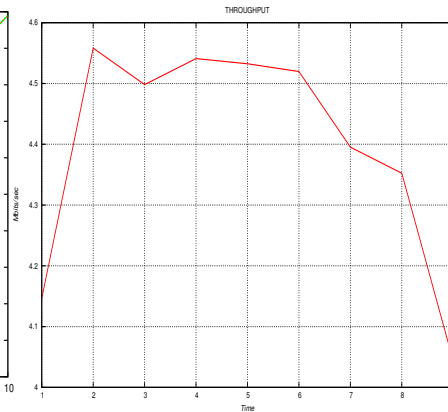


Figure 2: Throughput

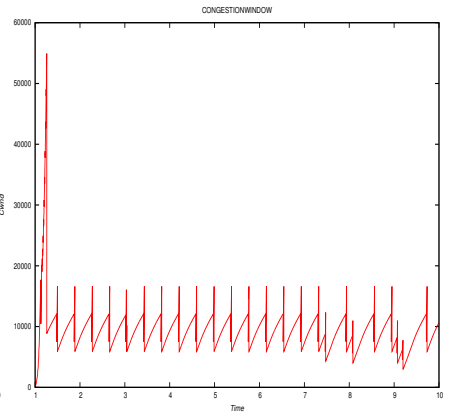


Figure 3: cwnd

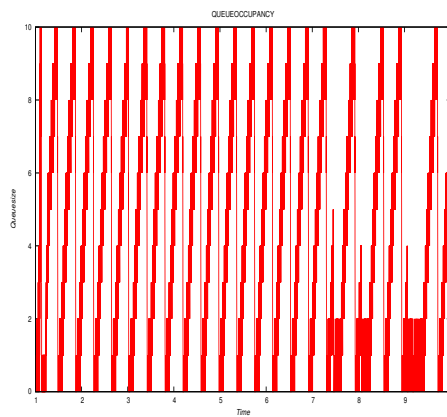


Figure 4: Queueoccupancy

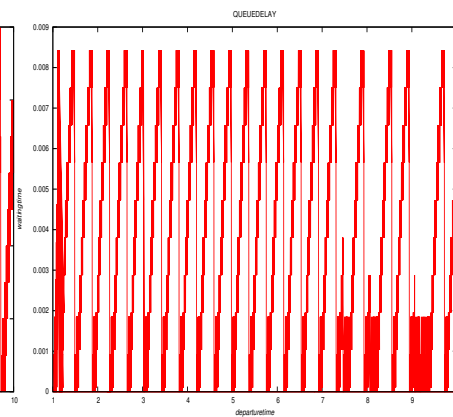


Figure 5: Waitingtime

- Here, we are using TCP new reno, it uses fast recovery when duplicate acknowledgements are received. We can see one big step and some small steps in sequence/ack no. graph. During that step, sender is receiving Dup. acks(same ack. nos.) from receiver. Accordingly, we can see that cwnd increases by 536Bytes(1 MSS) in Cwnd graph for each dupack. This is cwnd inflation. Once new ack. received, seq/ack no. increases normally in seq/ack no.graph. So, everytime when it completes fast recovery, it halves the congestion window.
- From throughput graph, Avg. throughput = 4.40 Mbits/sec Throughput is below bottleneck link (5 Mbps) because of acknowledgements, overhead transmitted with packets, and error rate of the channel.

Excercise 1.2

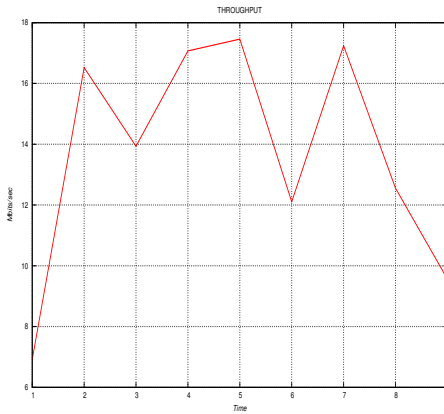


Figure 6: Average throughput for 5ms delay

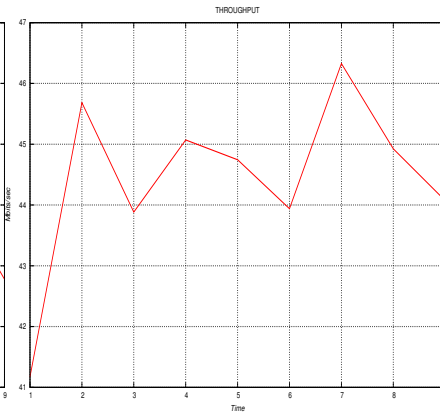


Figure 7: Average throughput for 1ms delay

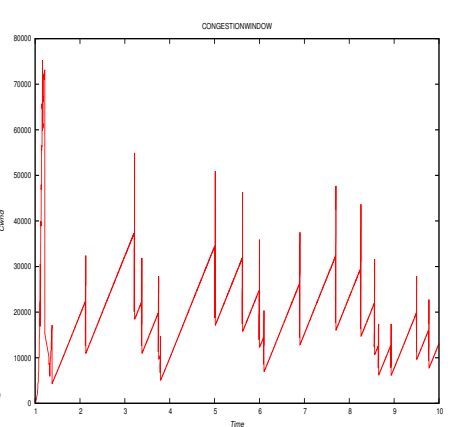


Figure 8: cwnd

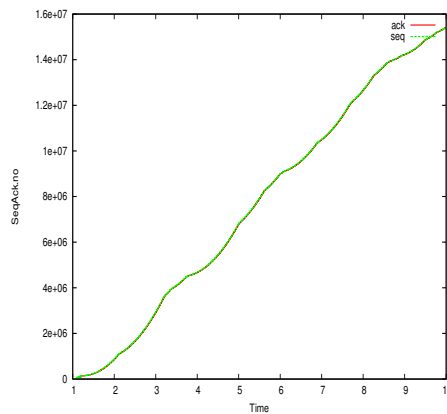


Figure 9: Time sequence graph

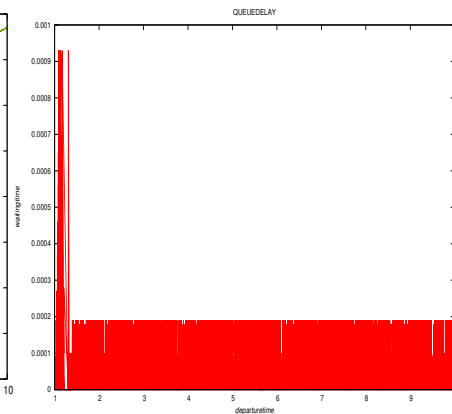


Figure 10: Waitingtime

- Average throughput = 13.70 Mbits/sec Which is way below then bottleneck link (50 Mbps). We know that to get idle value for throughput, sender's window size should be close to Bandwidth*Delay product. As sender's window size gets close to BDP throughput increases. Once it reaches BDP, from there on throughput won't increase anymore. But if sender's window is much lesser than BDP then throughput will be less as channel has more capacity to carry then what sender is sending.
- In our example, BW=50Mbps, delay=5ms \Rightarrow BDP= 250000bits = 31250Bytes = $(31250/536) = 59$ MSS But, here sender is sending much lesser load then BDP, so not enough packets to fill the channel. We can observe this from queue occupancy graph. After slowstart, queue occupancy never increases beyond 2.
- If we make Delay=1ms, then BDP= 50000bits = 6250Bytes = 12MSS Now, we made BDP close to sender's window size. So that channel will get filled with what sender in sending. Hence, throughput increases drastically. Here, Average throughput = 44.92 Mbits/sec, which is very much close to 50Mbps.

Excercise 1.3

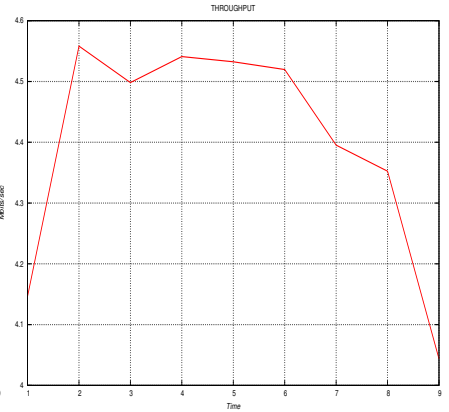
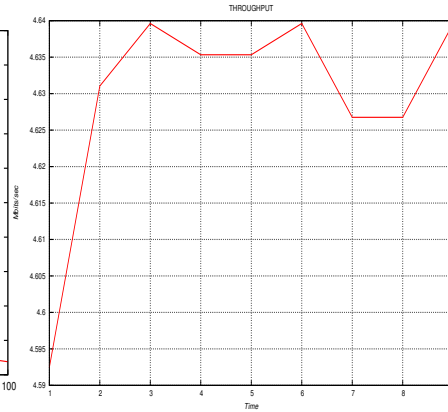
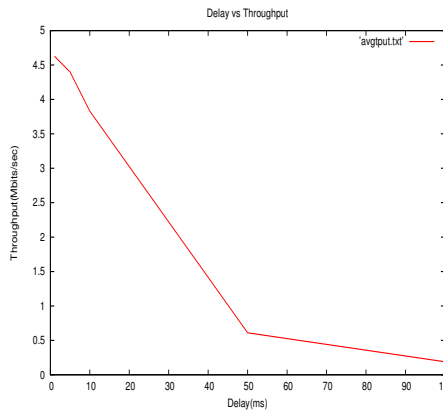


Figure 11: Delay vs Avg throughput Figure 12: Throughput for delay 1ms Figure 13: Throughput for delay 5ms

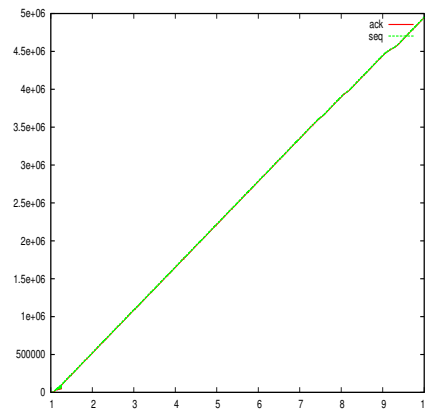
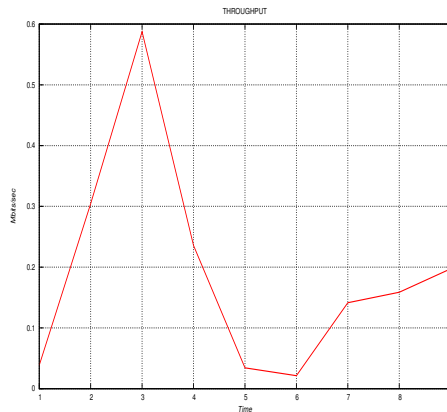


Figure 14: Throughput for delay 100ms

Figure 15: Time sequence graph

- Throughput of tcp decreases as delay increases.
- As we can see from graph of delay vs avgtput, when delay is below 5ms throughput is in range of 4.40 - 4.70 Mbits/sec, very close to link rate. But as we increase delay throughput falls as much as close to 0 Mbits/sec for delays above 50ms.
- Reason for this is, as link delay increases data send by sender in RTT decreases. Because RTT increases. and as we know, Throughput is inversely proportional to RTT.
- We can observe from tcp congestion window graph that as we increase the delay, gap between two cwnd updation increases. And as per delay, queue size should also be increased to match BDP, ideally equals to BDP. Otherwise, packets will get dropped from queue due to congestion and sender will receive more and more dup. acks for those packets. We can see that from seq/ack no. graph.

Excercise 1.4

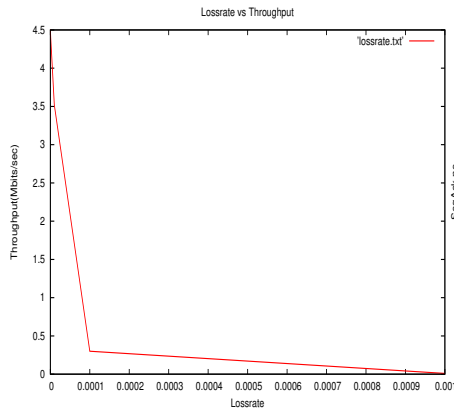


Figure 16: Lossrate vs Throughput

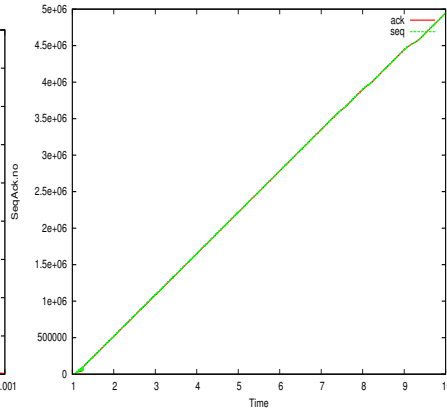


Figure 17: Time Sequence graph

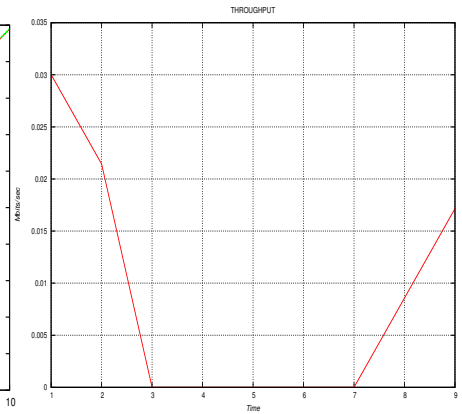


Figure 18: Throughput for loss rate 0.001

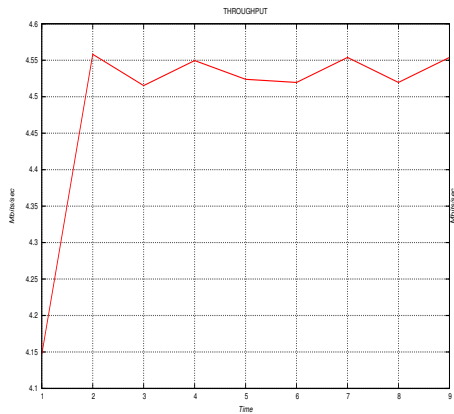


Figure 19: Throughput for loss rate 0.0000001

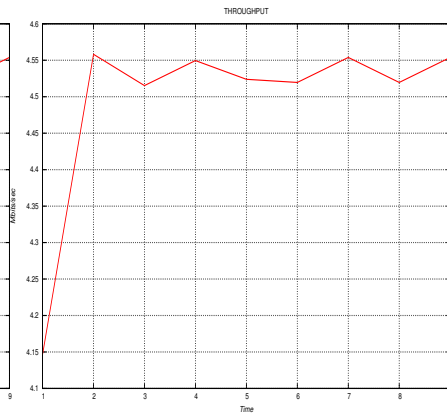


Figure 20: Throughput for loss rate 0.00000001

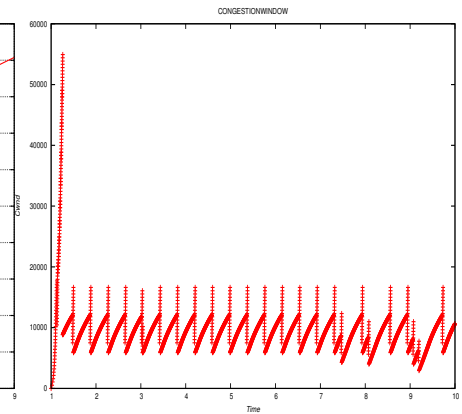


Figure 21: CWND when delay is 0.000001

- We can observe that Tcp throughput decreases as packet loss rate increases. It is quite clear from the graph as well as from formula for average throughput where average throughput is inversely proportional to square root(packet loss).
- From lossrate vs throughput graph, when packet loss rate is 0.000001, throughput is 4.40 Mbits/sec. When lossrate is less than that throughput will still be around 4.50 Mbits/sec. That is because, throughput can never be more than bottleneck link bandwidth (5Mbps), whatever packet loss rate is there.
- One more thing to observe is, as packet loss rate increases more and more dup. acks, retransmissions are needed. We can see that from sequence no. /ack no. graph. And tcp will take more time in fast recovery phase. Because of that cwnd is also updated very less no. of times, which we can see from tcp cwnd graph.

Excercise 1.5

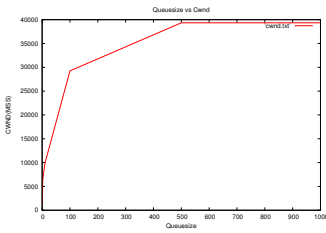


Figure 22: Queue size vs Avg CWND

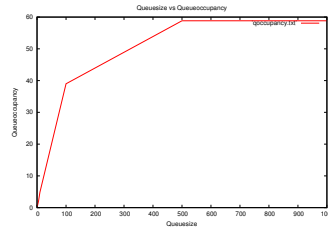


Figure 23: Queue size vs Avg Queue Occupancy

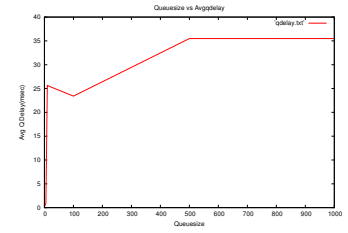


Figure 24: Queue size vs Avg delay

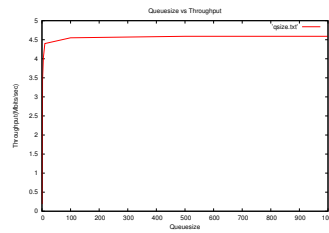


Figure 25: Queue Size vs Figure 26: Throughput for Avg Throughput

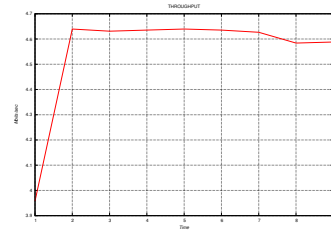
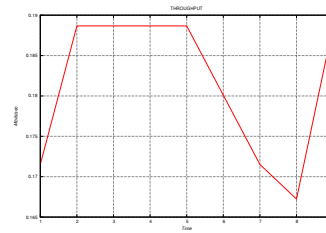


Figure 27: Throughput for Queue size 100

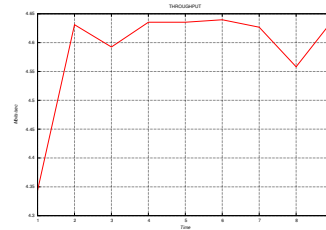


Figure 28: Throughput for Queue size 500

Queue Length	Avg Throughput	Avg. Queueing Delay	Avg. cwnd	Avg Queue Occupancy
1	0.19	0.45	0.256	1347.69
2	3.45	0.78	0.410	6140.70
5	4.03	1.99	0.947	7650.70
10	4.40	4.84	25.63	9842.91
100	4.55	39.0	23.40	29246.29
500	4.59	58.83	35.49	39336.18
1000	4.59	58.83	35.49	39336.18

- We can observe from various graphs that as queue size increases average queueing delay, avg queue occupancy, avg cwnd, avg throughput increases. But after some threshold value, all the thing will be constant with more increase in queue size. For ex, throughput can be maximum upto BDP. It can't go beyond that. Like this other metrics will also be constant after some threshold value.
- Here, suitable queue size is more than 6. After a certain value like after 10, average throughput will be almost constant. Ideally, our BDP is 6 MSS. So, once queue size increases beyond that throughput will not change much. So, queue size should be in the range 6 - 10 (atmost 100) as per graphs we obtained.

Excercise 1.6

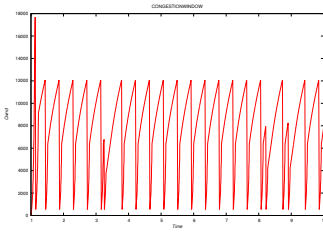


Figure 29: CWND for TCP TAHOE

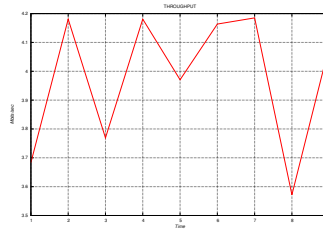


Figure 30: Avg Throughput for TCP TAHOE

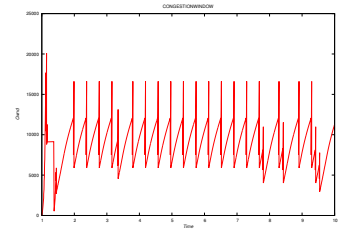


Figure 31: CWND for TCP RENO

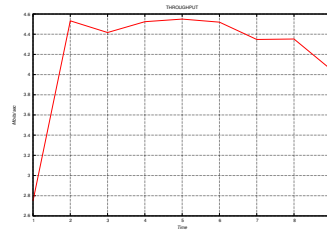


Figure 32: Avg Throughput for TCP RENO

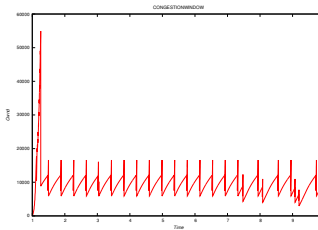


Figure 33: CWND for TCP NEW RENO

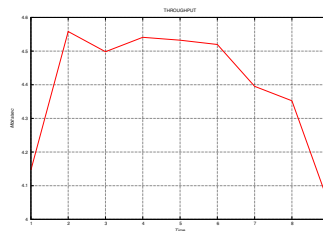


Figure 34: Avg Throughput for TCP NEW RENO

- We can compare throughput for different versions of TCP from average throughput graph. TCP tahoe has the least throughput among all the variants. It has no fast recovery mechanism. We can see that from cwnd graph. So, when packet is lost sender will wait for time out for retransmission. Thus during that time pipeline will be dry and overall throughput decreases. TCP reno has almost same throughput as TCP new reno as it implements fast recovery mechanism, we can see that in cwnd graph. Still TCP new reno has little edge as it can recover from multiple packet loss.
- In TCP tahoe's cwnd graph, cwnd starts from an initial value, then it increases exponentially (slow start) \rightarrow when it reaches to $ssthresh$ it increases additively (congestion avoidance) \rightarrow when time out occurs, cwnd falls to initial value and Tahoe will again start slowstart phase with $ssthresh = cwnd/2$. It has no fast recovery mechanism.
- In TCP reno's cwnd graph, It is almost same as TCP new reno except in slow start phase cwnd of tcp new reno goes much higher than cwnd of reno. TCP reno also has additional fast recovery + fast retransmit mechanism than TCP tahoe. So, we can observe cwnd inflation in cwnd graph, during begining of fast recovery phase.

Excercise 2.1

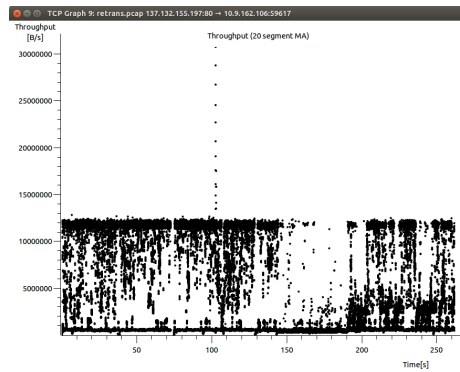


Figure 35: Avg Throughput for entire transfer

1. Average TCP throughput of entire transfer = 6.5 Mbit/sec.

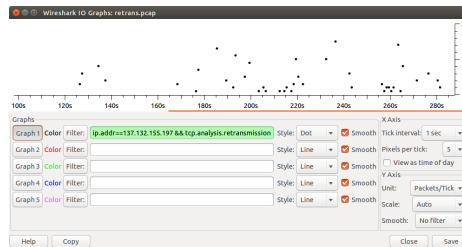


Figure 36: TCP retransmission rate

2. No. of retransmissions due to duplicate acknowledgements = 984
No. of retransmissions due to retransmits = 189

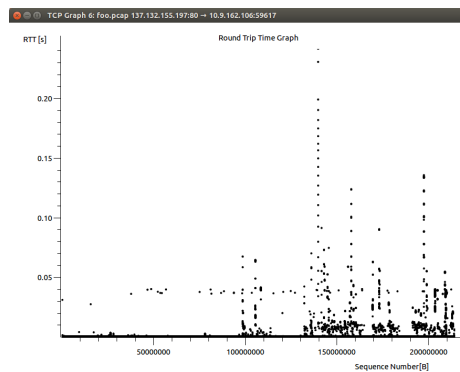


Figure 37: TCP RTT

3. Minimum RTT = 250ms
Maximum RTT = 1us
Queueing delay plays major part in RTT. Queueing delay is random for packet transmission every time where else transmission delay and propogation delay don't change much in every RTT.

Exercise 2.2

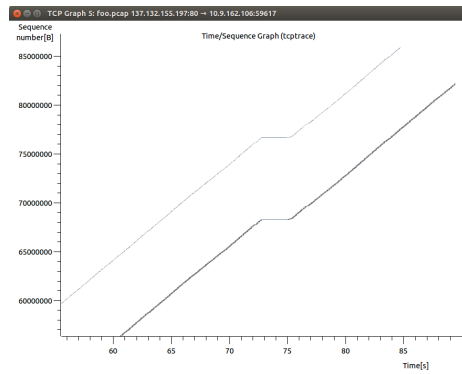


Figure 38: Time sequence graph

We can see from below screen shot that tcp uses Fast recovery mechanism, as it send a packet for every dup ack received. So, clearly sender follows the TCP Reno style. And I have not observed any sack based recovery for this transfer.

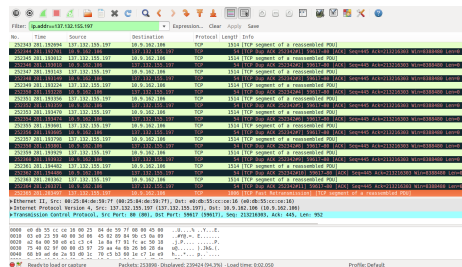


Figure 39: Fast Recovery