4.1 Introduction

The simulation tool in Figure (3-1) is used to record experiments event with different transmission protocols to compare the results. Using the mat-lab model and parameters that correspond to each recorded file, the differences between the experimental results and analytical values (the factors of QoS for VoIP application applied the TCP,UDP and RTP protocols) are presented in this section.

The TCP is a connection oriented protocol that lies in the Transport layer. The connection is formed via a handshake between two hosts with connection requests and acknowledgments. Once the connection is formed, the data being transmitted is broken into segments. The tcl script runs over ns-2 and found a .trace file and a nam file then added some awk script and found required trace file. Then got the results and generate the graphs using Mat-Lab script to measure the VoIP QoS parameters.

The UDP is a simple protocol that passes data along from the application layer to IP to be transmitted. It performs none of the error checks that TCP does, The tcl script runs over ns-2 and found a .trace file and a nam file then added some awk script and found required trace file. Then got the results and generate the graphs using Mat-Lab script to measure the VoIP QoS parameters.

RTP-based multimedia applications similar send media data over UDP, and are subject to the unpredictable behavior of best-effort IP networks, including packet loss, reordering, and variable queuing delays. Real-time applications are tolerant to some amount of packet loss, either concealing the loss or using one of the available error resilience mechanisms The performance of RTP is identical to that of UDP in every aspect.

RTP merely attaches additional data to the UDP stream to provide valuable information to the application at the other end. It does not directly prevent issues such as jitter, however the information it provides can warn the application that such issues are present so that the application can take whatever preventative actions it needs to.

4.2 Throughput

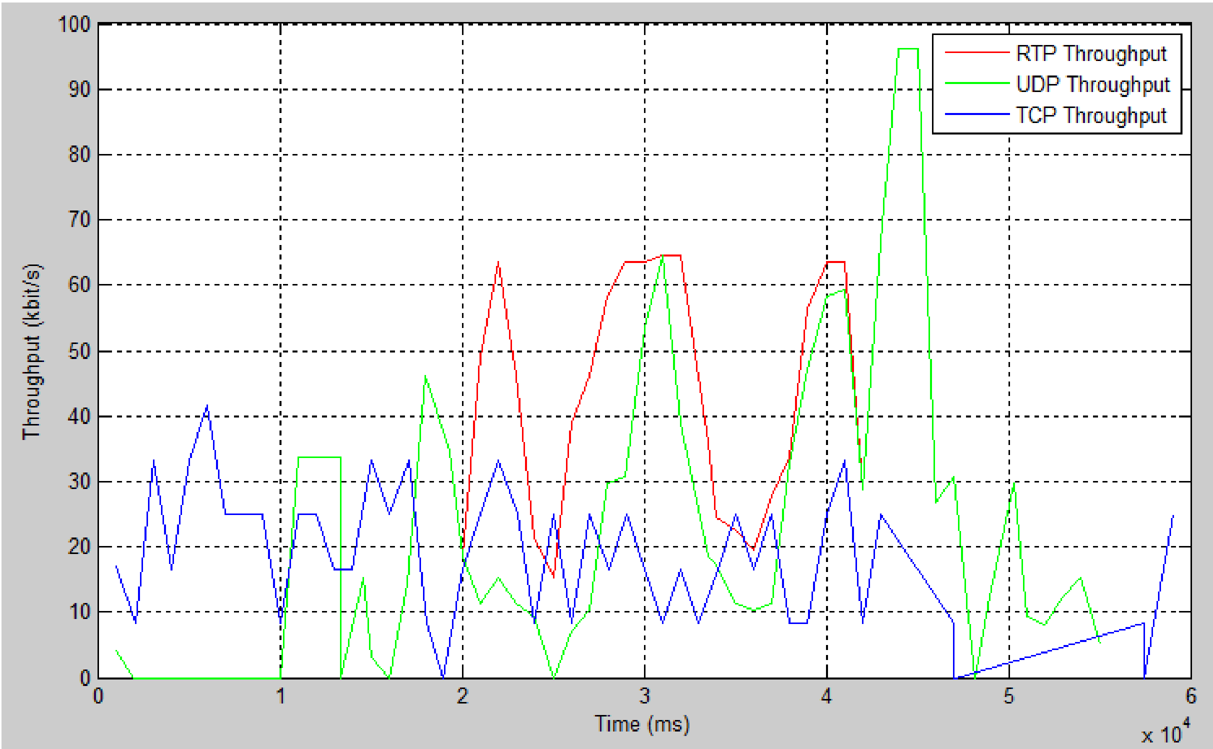

Figure (4-1): Throughput from node (0) to node (4)

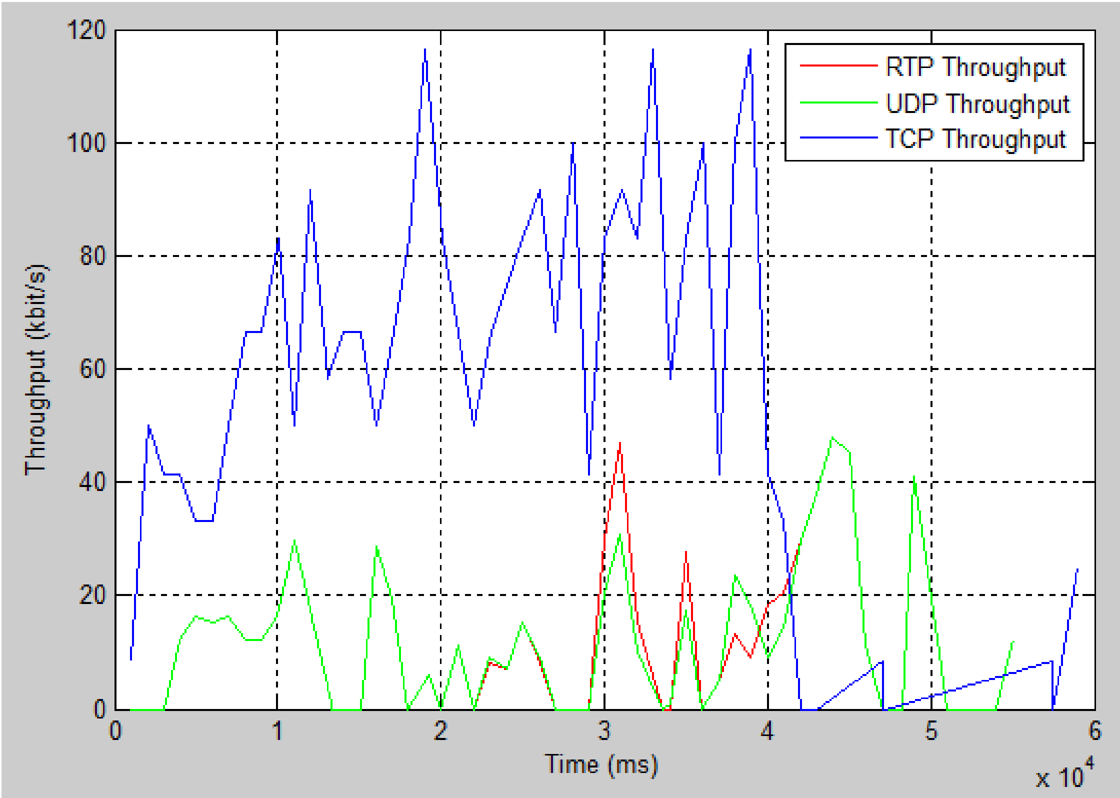

Figure (4-2): Throughput from node (4) to node (0)

In fig(4-1),(4-2) in the beginner the performance mounts with time staring from 2th second, show the throughput between node 0 to node 4 by using biue line for,TCP green line for UDP and red line for RTP , the highest packet delivered rate 42kbit/s at 0.5ms for TCP , 97Kbit/s at 0.43 to 0.45ms for UDP and 65Kbit/s at 33ms for RTP,UDP and RTP are take a similar result during a red line are hide Figure (4-1) There seems to be loss in performance less the network is under a maximal load between (40s-55s)by the other node's, from 14.5s an upper which large drops in throughput are observed from both ends. Because there is low to moderate background traffic present in the network.

The large spike on the plot to the right at 40 seconds occurs because the user at Node 0 is not talking and thus not transmitting any data. This frees up a lot of bandwidth so that the user at Node 4 is able to transmit well despite the heavy background traffic.

The traffic load is a maximum and there is no (ack) packet leads the Throughput drop occurs on some time interval in fig(4-1), reverse the bounder between (43-48) s in fig(4-2) for TCP.

4.3 Packet Loss

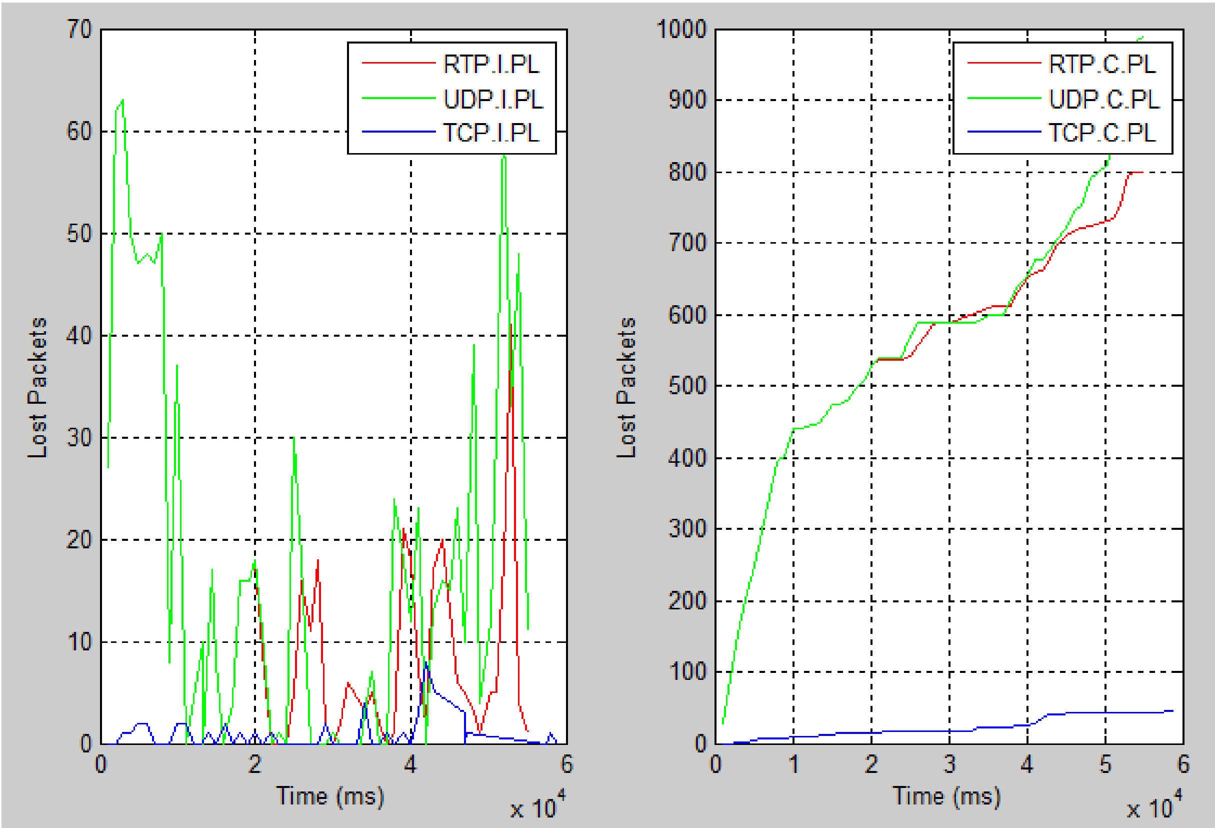

Figure (4-3): Packet loss from node (0) to node (4)

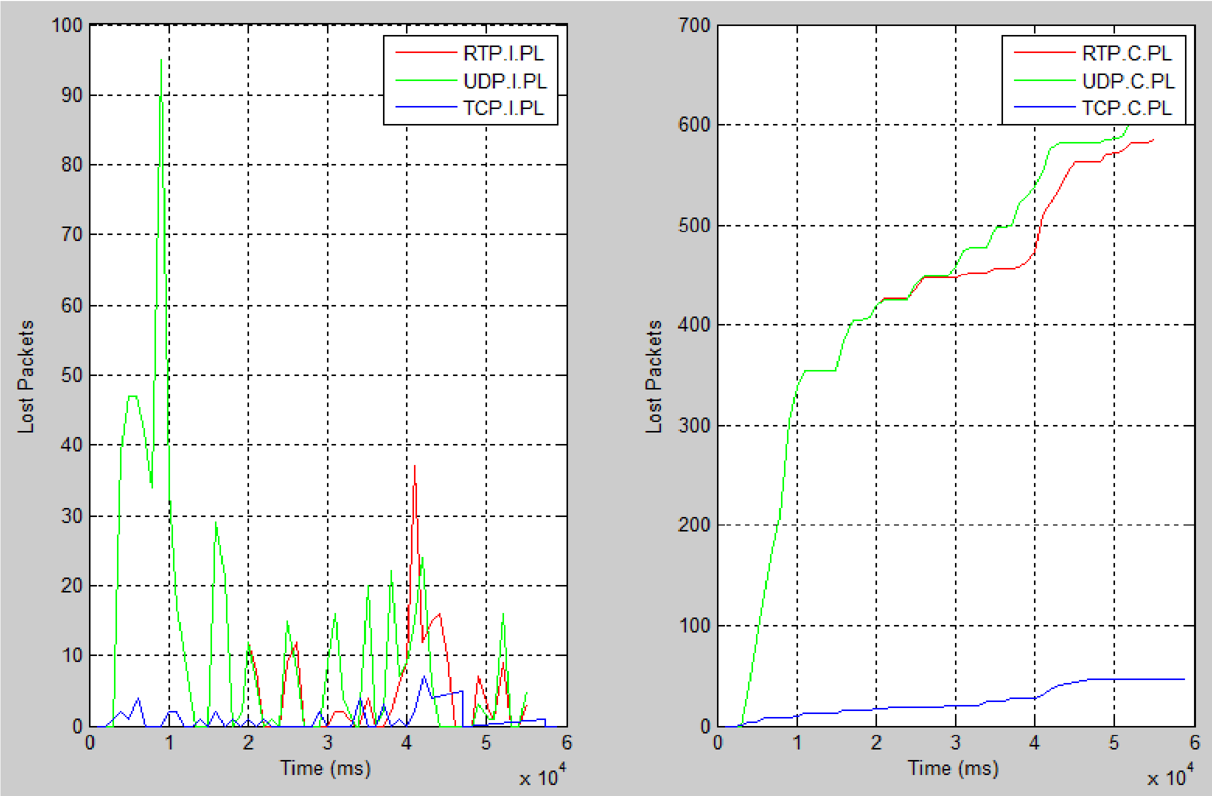

Figure (4-4): Packet loss from node (4) to node (0)

In Fig(4-3)The lower lost packet for TCP protocol, during all time TCP lost 45 packet form 126 packet , UDP lost 1384 packet from 2342 packet and RTP lost 992 packet from 2352 packet.

In Fig(4-4)The lower lost packet for TCP protocol, during all time TCP lost 47 packet form 535 packet , UDP lost 727packet from 1404 packet and RTP lost 609packet from 1468packet.

For TCP the first 40 seconds of simulation, do see a fair bit of dropped packets however, since the throughput did not drop during this interval, may conclude that the only packets lost are ACK packets, which are small in size and are not of real concern. Packets containing actual voice data are lost only after 3 seconds of simulation when the network is under a maximal load, because the packet size is more large.

The background traffic is upped at 20 seconds and beginning to see the consequences at around 30 seconds with the dramatic increase of lost packets in UDP protocol. At this point, the queues are growing exceedingly large causing many packets to be dropped. Since the background traffic remains constant, the queues don't have time to relieve themselves and as the background traffic is further increased, more packet are consistently dropped.

For the RTP in beginning of some packet losses due to the sudden congestion caused by a combination of background traffic and both users speaking at the same time. The background traffic is upped at 3th second from node 0 to node 4 and at 10th second from node 4 to node 0, the consequences at around 54 seconds with the dramatic increase of lost packets.

4.4 End-To-End Delay

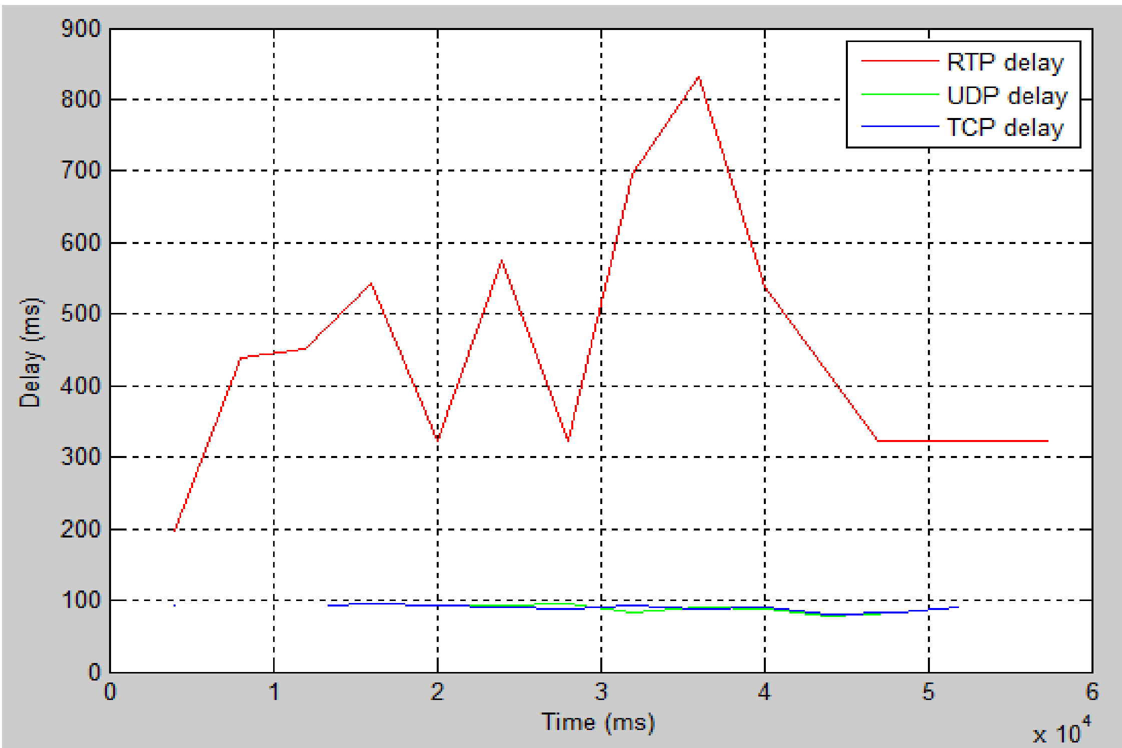

Figure (4-5): End-to-End Delay from node (0) to node (4)

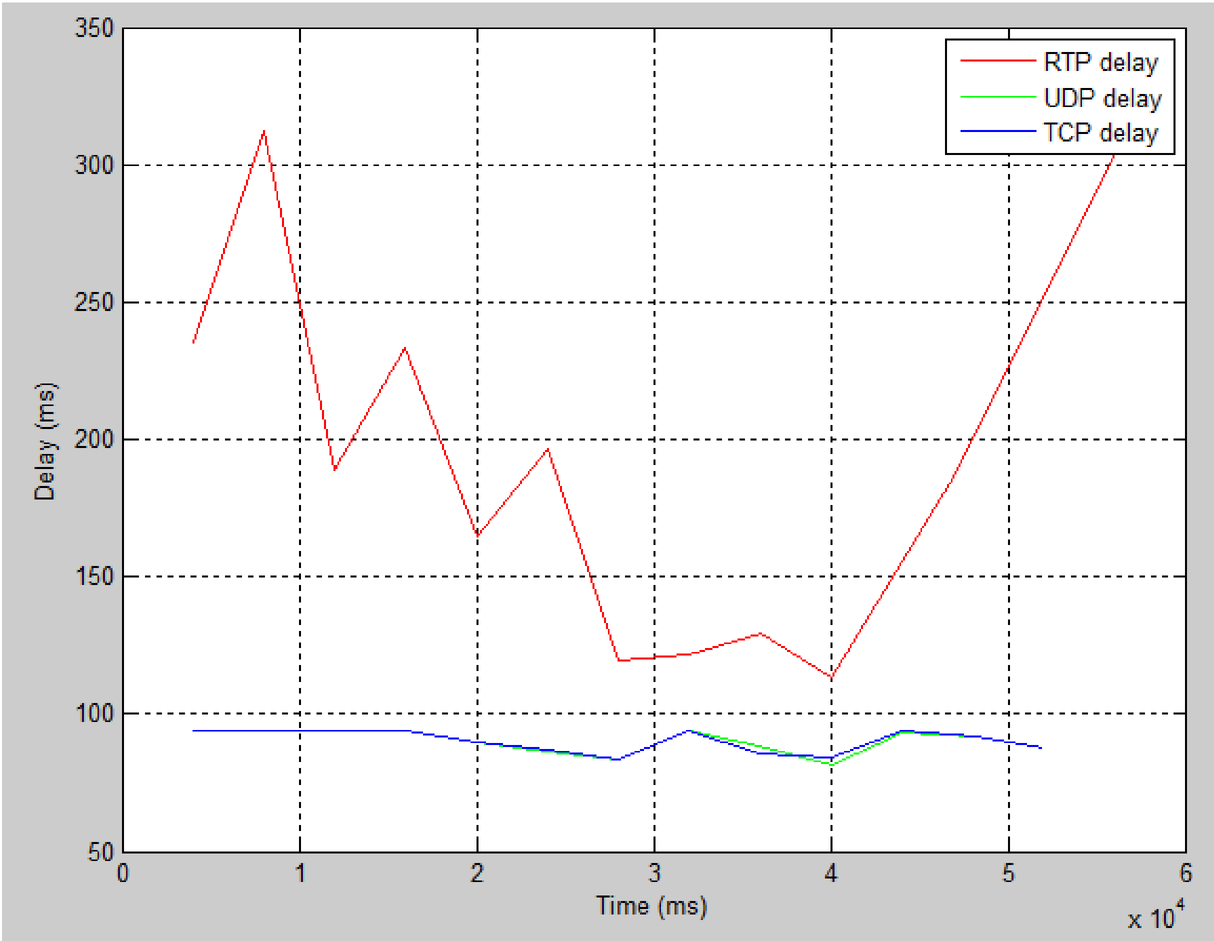

Figure (4-6): End-to-End Delay from node (4) to node (0)

Evaporate delays varying between (2.5 to 48.5) second of simulation time for TCP, the TCP delay from 0.219 to 0.810 this for node (0) to node (4) secondsin fig(4-5) and the delays from 0.110 to 0.323 seconds for node (4) to node (0) are observed in TCP in fig(4-6) , which greatly exceeds the recommendation of 150ms.

The delay for UDP in Fig (4-5) and Fig(4-6) is around 94ms which is well below the recommended limit of 150ms specified earlier in UDP. Note that the delay varies very little regardless of increasing background traffic, the delay time between 3s and 55s.

The delay for RTP in Fig (4-5)&(4-6) is around 94ms which is well below the recommended limit of 150ms specified earlier in UDP. Note that the delay varies very little regardless of increasing background traffic , the delay time between 13s and 24s .

4.5 Jitter

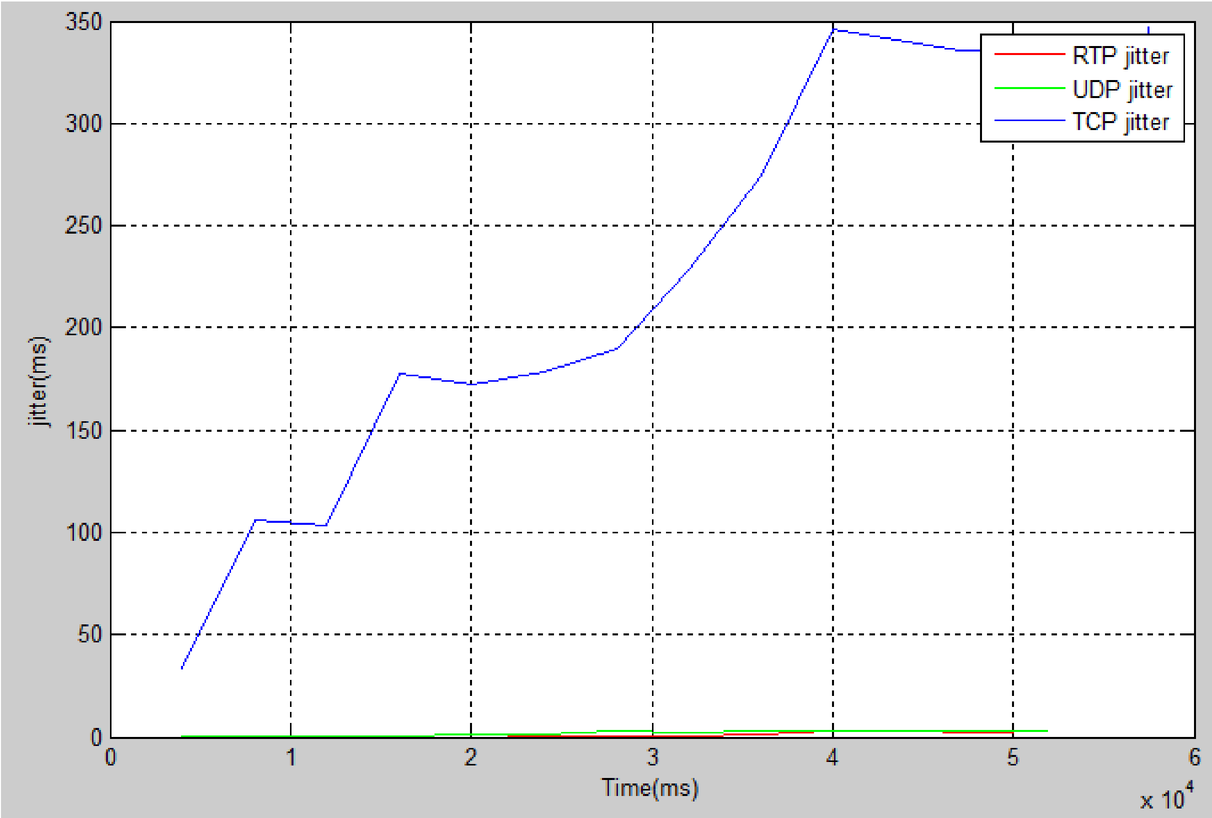

Figure (4-7): Jitter from node (0) to node (4)

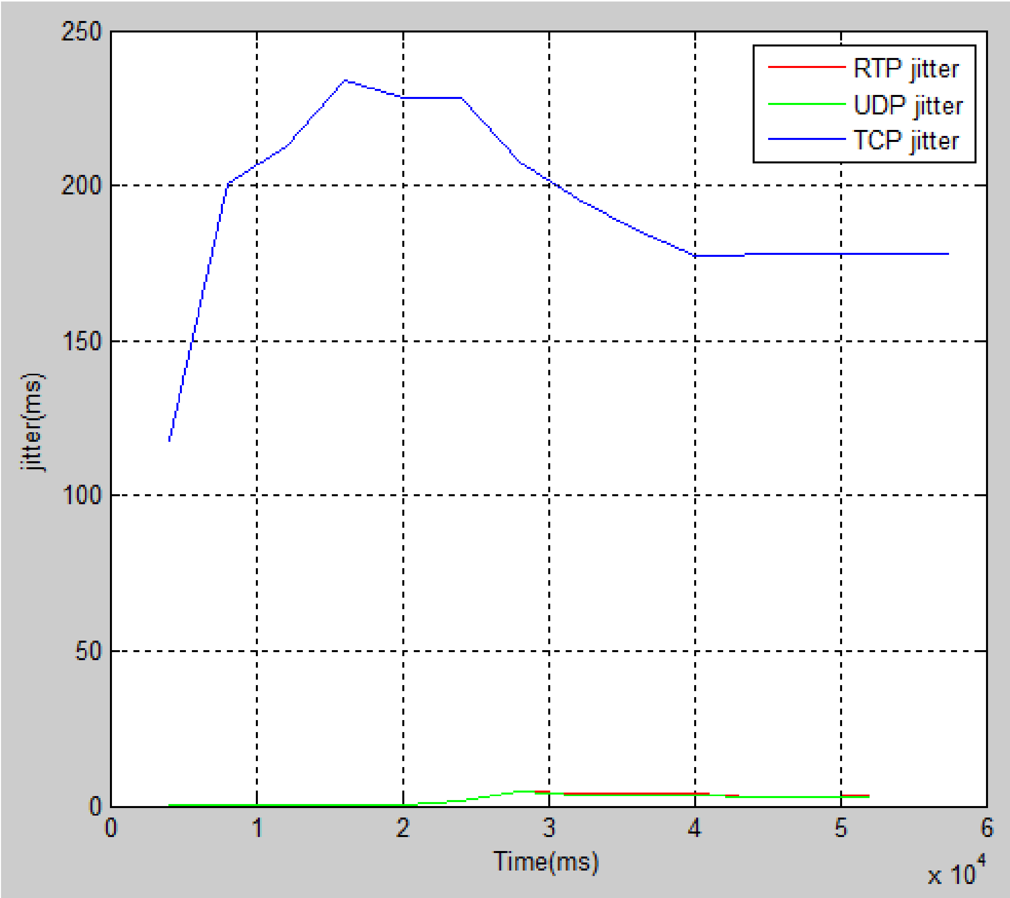

Figure (4-8): Jitter from node (4) to node (0)

The limits of variation is almost between (5^{th} to 58.5^{th})s of simulation time seconds is the time that the occurrence of the jitter, Variation in TCP figure (4-9) occurred between the (40 to 350)ms of the total delay time in the figure (4-10) note that variation between the Event (180 to 238)ms.

The small delay variation (jitter) for packet from node (0) to node (4) is observed compared with packet from node (4) to node (0) .

Observe for fig(4-7) and fig(4-8) a very low variation in the end-to-end delay times for UDP during a first 4^{th} second, which is ideal for VoIP. Stating from 4^{th} second to 13^{th} second the jitter can be increasing the high variation refer to high background traffic causes increase and decrease for end to end delay.

Very low variation in the end-to-end delay times for RTP between 4^{th} and 16^{th} second, which is ideal for VoIP. Stating from 16^{th} second to 44^{th} second the jitter can be increasing the high variation it above than 45 refer to high background traffic causes increase and decrease for end to end delay.

4.6 Summary

With regards to our criteria of low packet loss to maintain a high voice quality, note that TCP outperforms UDP/RTP. Packet losses were not observed for TCP unless the background traffic was at maximum load, Low delay and jitter are a higher priority than loss of quality. TCP have large delay and jitter is unacceptable for VoIP.

That lead the UDP/RTP to be better than TCP because the UDP/RTP have acceptable voice quality in modulate background traffics.