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Performance Evaluation of Real Time Traffic in LTE Networks

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Abstract
Long-Term Evolution, commonly known as 4G LTE, is a standard for wireless communication of high-speed data for mobile phones and data terminals. TCP protocol plays a significant role in LTE network. Congestion control algorithm is an integral module of TCP that directly determines the performance of the protocol in IP Network. Many TCP variants like TCP-Vegas, TCP-Tahoe, TCP-Reno, TCP-NewReno, TCP-SACK, and TCP-FACK have been proposed for high bandwidth real time services in LTE network [1]. These TCP variants can be used to improve quality of service parameters i.e., throughput, average delay and lost packet in LTE network [2]. Since LTE systems has high-data-rate, low-latency and packet-optimized radio access technology therefore these parameters directly affect the traffic. In this paper, complete LTE Network is simulated using the OPNET simulator to evaluate network performance in real time [3]. The simulated LTE network is tested for each type of TCP protocol and the result is monitored for throughput, end to end delay, http and FTP traffic, re-transmission response time and packets drop etc and the results are plotted. At the end, it is concluded that TCP New Reno has outperformed in terms of throughput, re-transmission rate and http uplink and downlink traffic. In medium size network, the performance of Reno is satisfactory. Similarly changes in mobility rate enhance the performance of TCP Reno in other congestion control algorithms. On the other hand, TCP Cubic is found to be the worst performer for throughput, re-transmission rate and http traffic parameters.
Keywords: LTE, QoS, EUTRAN, TCP, http, Opnet

I. INTRODUCTION

LTE is the brand name for emerging and developed technologies that comprise the existing 3G and 4G networks. The main advantages with LTE are high throughput, low latency, plug and play, scalable bandwidth capacity, an improved end-user experience and backwards compatibility with existing GSM and UMTS technology.

II. LTE NETWORK ARCHTECTURE

The network interfaces are based on IP (Internet Protocol) in LTE network. The architecture comprised of Core Network and Access Network which represent IP layer of LTE System. The non-radio componentis “System Architecture Evolution” (SAE), comprises of Evolved Packet Core (EPC) network. Together LTE and SAE comprise the Evolved Packet System (EPS). EPS uses the concept of EPS bearers to route IP traffic from a gateway in the PDN to the UE.A bearer is an IP packet flow with adefined quality of service (QoS) between the gateway and the UE. In order to provide connectivity to different PDNs or different types of QoS streams, multiple bearers are established.

For example a user can connect voice call over IP and at the same time performing FTP download or http browsing. The basic architecture of LTE network consists of five parts as shown in Fig1. They are evolved-NodeB (eNodeB), Serving Gateway, Packet Data Gateway, User Equipment, and Routing Unit.

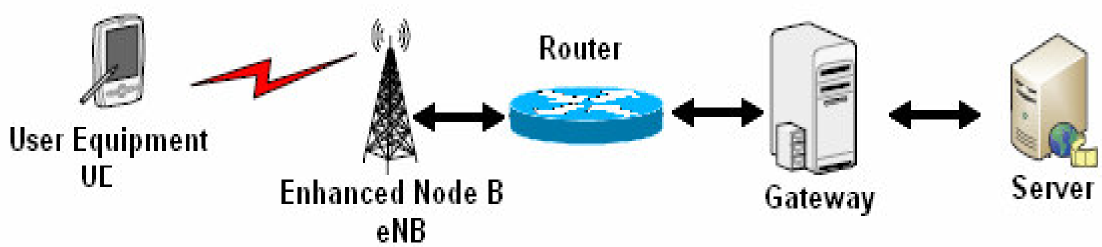


Figure.1

III. TCP OVER LTE SYSTEMS

Transmission Control Protocol (TCP) is a connection oriented protocol working at transport layer to control communication between links. There are various types of TCP variants which are developed to control congestion in network with maintaining high throughput in the network [4], [5].The following TCP flavors are used for analysis of traffic in LTE network [6], [7].

TCP Standard

Fast recovery or retransmit of packets are not used in standard TCP flavor. With standard TCP, flows with very short RTTs will receive ACKs faster and therefore have their congestion windows grow faster than other flows with longer RTTs.

TCP Tahoe

In Tahoe, fast retransmit as defined in RFC 2001 will be executed once the node receive n-th duplicate acknowledgement. The TCP Tahoe variant is implemented with slow-start, congestion avoidance, and fast retransmits algorithms.

TCP Reno

In TCP Reno, fast retransmit as defined in RFC 2001 will be executed once the node receives the n-th duplicate acknowledgement and then it will enter the fast recovery phase. When duplicate acknowledgement reached to its threshold level, the packet segment is considered to be lost. The lost segment is retransmitted when TCP enters the Fast Re-Transmit phase.

TCP New Reno

In this version, fast retransmit as described in RFC 2001 will be executed with two modifications to the algorithm:

- *Fast retransmit will never be executed twice within one window of data.*
- *If a partial acknowledgement (acknowledgement advancing) is received, the process will immediately retransmit the next unacknowledged segment.*

TCP New-Reno does not wait for retransmission timer in case of multiple packet loss. Moreover, in New-Reno,receiving partial ACKs often indicates the lossof the packets which instantly follows the acknowledged packet in the sequence space.

TCP SACK

Basically TCP SACK is upgraded version of TCP New Reno which resolves the major issues experienced by the New Reno version.TCP uses a collective acknowledgment technique which detect single lost segment per round trip time. The sender has to either retransmit the received packet or wait for a round trip time in order to discover the packet lost. Resultantly, TCP loses its ACK-based clock and thus decreases the overall throughput.

TCP CUBIC

TCP Cubic uses an optimized congestion control algorithm for high bandwidth networks with high latency. It is a less aggressive and more systematic derivative of BIC TCP, in which the window is a cubic function of time since the last congestion event, with the inflection point set to the window prior to the event.

IV. MODELING AND SIMULATION

In order to simulate the LTE Network, OPNET (Optimized Network Engineering Tool) simulator software is selected as it has several modules comprising of large number of protocols and network elements[8].The main beauty of this simulator is that it provides real-life simulation which is very near to reality. As compare to other simulator it has GUI Interface, large collection of network models, protocols and interfaces to create any network model. The following are the entities which we have selected from the *Object Palette Tree* from the OPNET Modeler.

- *Lte_attr_definer_adv,*
- *Mobility Config*
- *Application Config*
- *Profile Config*
- *Lte_wkstn_adv*
- *lte_enodeb_4ethernet_4atm_4slip_adv*
- *router_slip64_dc*
- *lte_access_gw_atm8_ethernet8_slip8_adv*
- *ethernet4_slip8_cloud_adv node*
- *ethernet4_slip8_gtwy*
- *ethernet_server model*

The simulated model as shown in Figure.2consists of LTE Core Network, LTE Access Network and remote Gateway Router. The remoteend is connected to two servers for serving FTP and HTTP request through Gateway Router to provide a source connection for the TCP link over the topology. The service provider area has three eNodeB, each serving to 10 User Equipment (UEs).Each eNodeB is connected to Service Provider’s back bone Gateway Router which further connected to remote Gateway router through IP Internet cloud. The network is simulated with the parameters given in Table.1 for analysis of the LTE traffic in real time.

TABLE 1. SIMULATION PARAMETERS

SN	Parameters	Values
1	Types of TCP Variants	Reno, New-Reno, Cubic, Tohoe, Standard
2	Simulation Area	6000m X 6000m
3	Number of eNode Bs	03
4	Number of Cells	03
5	Cell Radius (km)	1.00 km
6	No. of Nodes per cell	10
7	Speed	10meter/second
8	Area of Movement	Within Network
9	Number of UE	30
10	Nodes with Mobility Configured	30
11	TCP MSS	Auto-assigned
12	SACK-Enable	ON
13	B.Width (4G LTE eNB)	100Mbps
14	UE Tx. Power (W)	Cell Sized Based
15	eNode B Tx. Power (W)	Cell Sized Based
16	PHY Profile	LTE 20 MHz FDD
17	Bearer Name	Silver
18	Handover Parameters	Same as Serving eNodeB

IV. PERFORMANCE EEVALUATION AND ANALYSIS

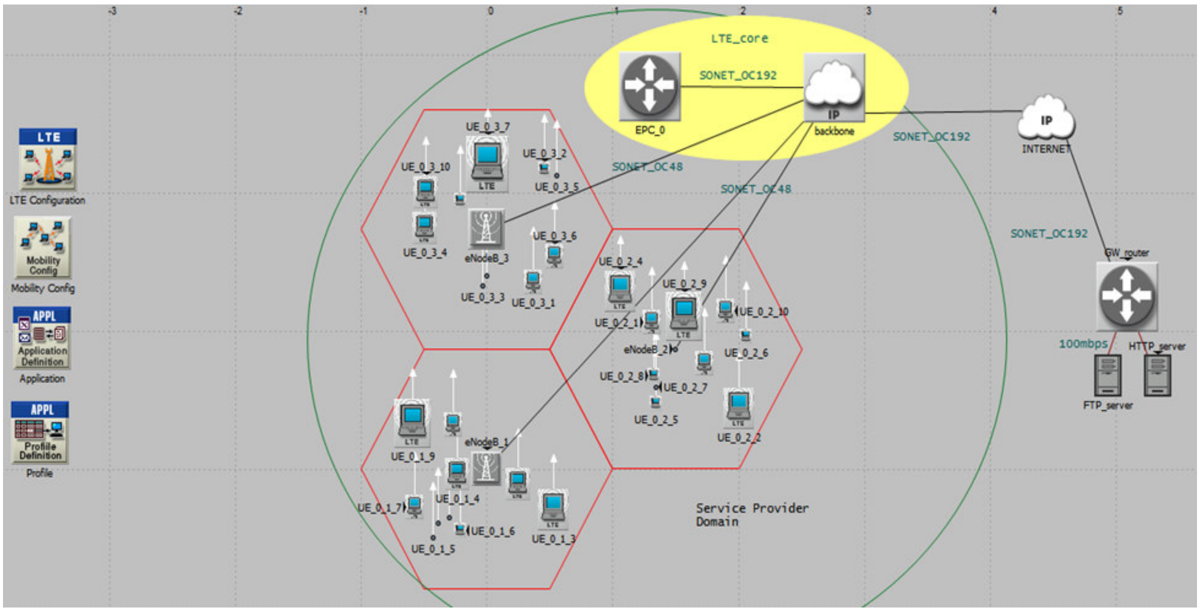
In this research, performances of the following parameters are evaluated against different TCP variants [9], [10].

Throughput Performance

The total amount of data packet which is successfully delivered to destination node from source node over communication network is called throughput. The throughput is measured in bits per second (bits/sec). Mathematically, throughput can be defined by the following formula.

$$\text{Throughput} = \frac{\text{Number of delivered packet} \, * \, \text{Packet size (Bytes)} * 8}{\text{Total duration of simulation (sec)}}$$

Refer to Figure 3, the X-Axis represents the time in minutes and Y-Axis shows the throughput in bits/seconds. Similarly the traffic is not generated for up to 100 seconds as shown in Figure . This portion is called as Warm up time. The overall throughput in LTE network remained at peak from 22 to 24 bits/sec from 4.2 minutes to 6.8 minutes for TCP Standards. Considering the performance, TCP Standard provides better throughput than other variants.Meanwhile the throughput for TCP Tohoe is found to be in between 20 to 22 bits/sec for period of 4.2 minutes to 8 minutes.It is apparent that TCP Standard keeps outperforming other than TCP Variants through



achieving a higher throughput of 24 bits/sec. On the other hand TCP Cubic and TCP New Reno achieve the lowest amount of throughput in the network approximately 20 bits/sec and 19 bits/sec respectively[11].

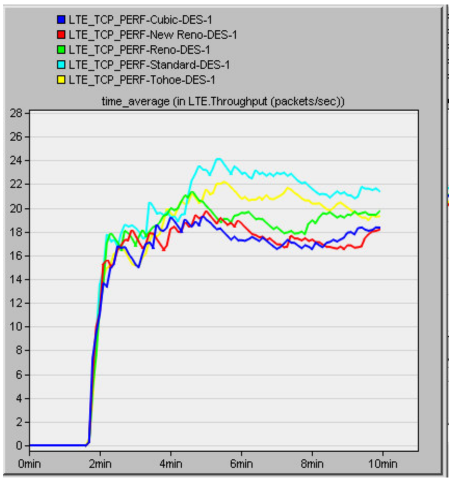


Figure 3. Throughput in LTE Network.

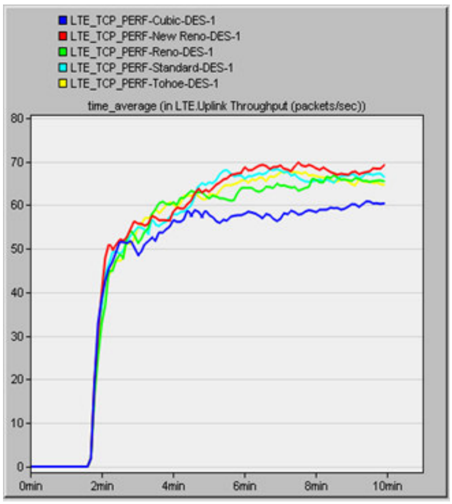


Figure 4. LTE Upload Throughput

End-to-End Delay

The End-to-End delay can be defined as the time at which it is created to the time it received. When the rate of rerouting in a network is increased the end-to-end delay is automatically increased. Generally, the end-to-end-delay is measured as per the following equation.

$$D_{end\_to\_end} = N\left[D_{trans} + D_{prop} + D_{proc} \right]$$

Where

$D_{end\_to\_end}$ = end to end delay.

D_{trans} = transmission delay.

D_{prop} = propagation delay.

D_{proc} = processing delay.

N= a scalar number.

As can be seen in figure 5, that the end-to-end is calculated for different variants on eNode level of the LTE Network. The TCP Cubic has the lowest (but steady) end-to-end delay of about 21.0 milliseconds.

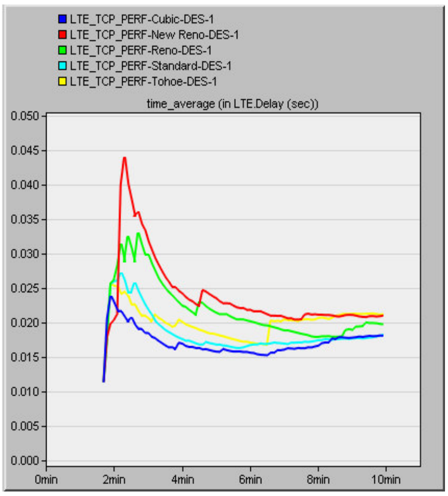


Figure5. eNode_1 Statistics.

(ms) on average, while the end-to-end delay for the TCP New-Reno is about 44.0ms, displaying a maximum delay among all the TCP Variants [12].

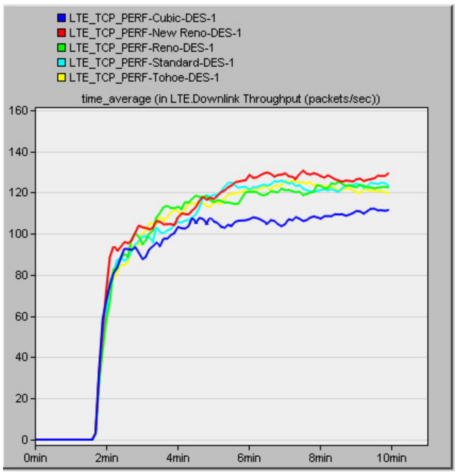


Figure 6. LTE Download Throughput.

LTE Global Parameters

When analyzing the performance globally in LTE network in Figure 7, TCP Cubic gives the maximum delay of 10.5 milli sec in Downlink at 4.2 minutes of the simulation time. Such behavior is usually happened in small networks. In case of down link delay, the performance of all TCP variants almost remained the same [13].

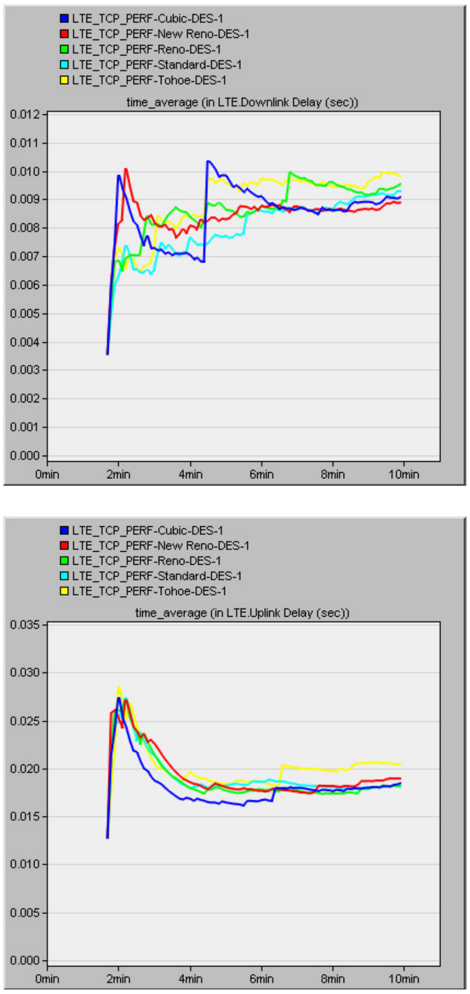


Figure7. Downlink and uplink Delay.

Point- to- Point link between Backbone and Internet

The following graph represent the point-to-point delay occurs between the backbone and internet cloud in LTE network.

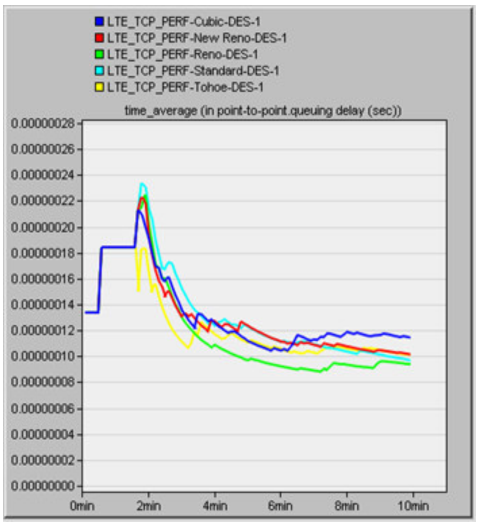


Figure 8. Uplink Delay.

As shown in figure 8, TCP Standard is at highest as compared to other variants affecting the flow of packets in network [14].

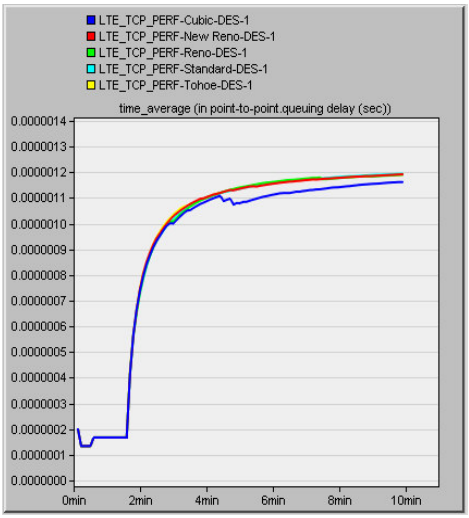


Figure9. Downlink Delay.

Down load and Upload Response Time

Download response time is the Time taken in sending the request for data packet and receiving the response, measured from the time client sends data request and it receives back the response. Upload Response time is the total time during which a file is send and their response is received [15].

In this section, the effectiveness and performance of download and upload activities are evaluated by the amount of upload response time and download response time.

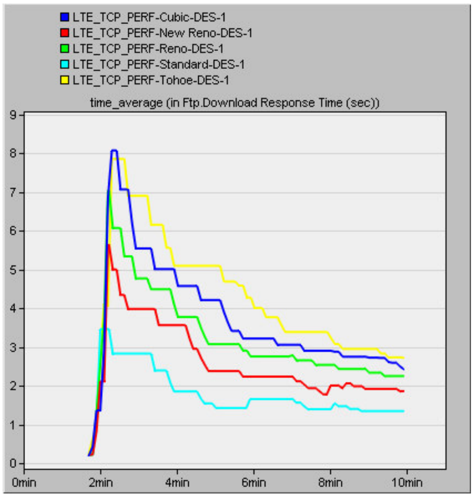


Figure10. Download Response Time.

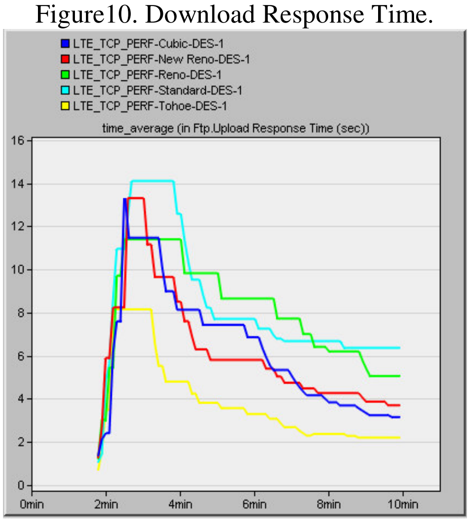


Figure11. Upload Response Time

Hence, these two parameters play an important role in data traffic measurement, where lowering their value, faster will be the task proceeded [17]. Figure 10 and 11 demonstrates the download and upload response time for transmitting an FTP file,whereas X-axis and Y-axis in each figure represent simulation time and upload/download response time. The highest download takes place at the beginning of the simulation time, approximately 1.8 minutes for all the TCP variants. Consequently, the download response time for all the TCP variants decreases abruptly within the instance of first few minutes, which is then sustained to stabilize in the long run, although a smoother drop occurs for all the TCP variants as obvious in Figure 11. The performance varies and reduces among different TCP variants As a result, more time is required to finish the data recovery activities. Similarly, more time is to be spent to download a file in the presence of high number of nodes in a network [18].

Re-Transmission Attempts

This parameter can be defined as the total number of retransmission attempts by all MACs in a network which may have been damaged or lost due to links failure. This parameter not only determines the rate of retransmission attempt, but can also figure out the number of packet drops per second, which has to be retransmitted. So, the lower isthe retransmission attempt, the more reliable is the TCP variant [16].

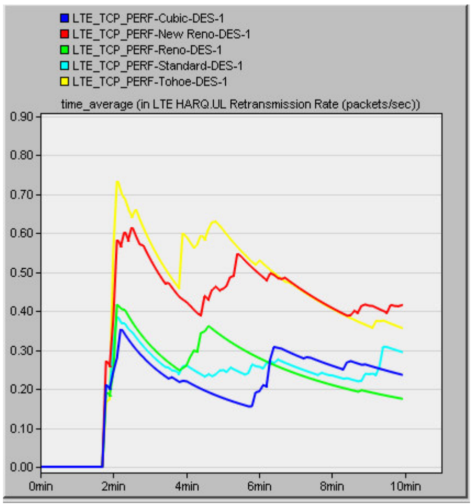


Figure12. Uplink Re-Transmission Rate.

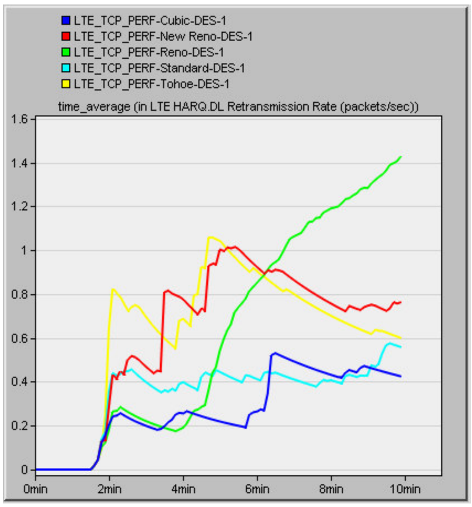


Figure13. Downlink Re-Transmission Rate

In all the scenarios of Figure 12, the highest packet drops are observed at the beginning of transmission. Hence, the maximum retransmissions are attempted at that period. TCP retransmissions are usually due to network congestion in wired media. Where as in wireless medium, physical links are more noisy for data transmissions thus when the signals propagated through these links degraded and caused noise and interference. Consequently, packet losses are increased, which increase the rate of retransmissions [18]. The numbers of retransmission are also increased due to the high number of network nodes. This is due to disconnection in physical layer as well as high rate of packet errors and high network density.

"http" Global Statistics

The http protocol is used for web browsing where FTP for file transfer. In this scenario 30 UEs devices are deployed to generate traffic byaccessing of Web page and FTP download response times. This simulation was repeated for all TCP variants and the results were compared. Figure 14 to 17 demonstrates similar results for the *http* traffic, web page response time and object response time.

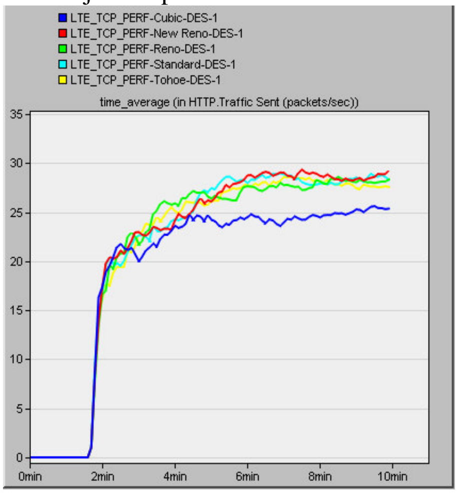


Figure14. *HTTP* Traffic Sent.

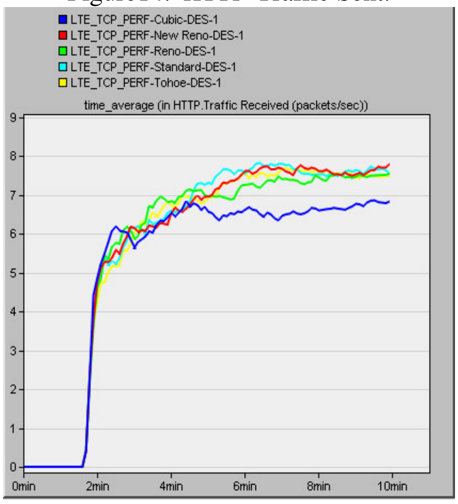


Figure15. *HTTP* Traffic Received

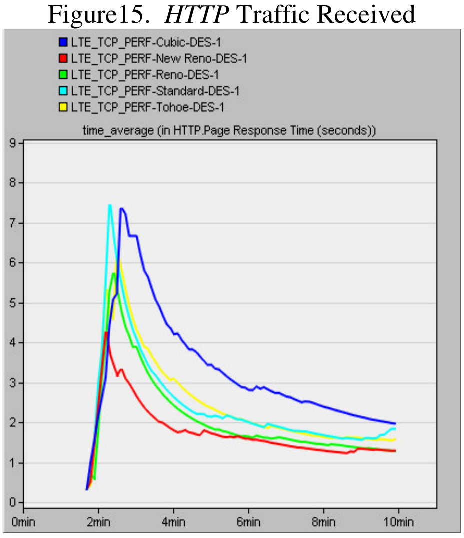


Figure16. *HTTP* Page Response Time.

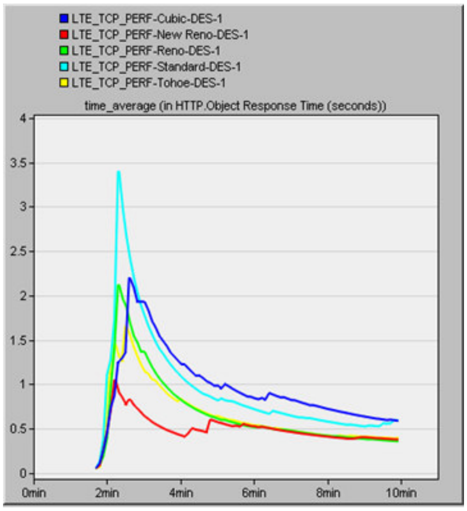


Figure17. *HTTP* Object Response Time.

The FTP download response time is 08 seconds for TCP Cubic with 30 users, whereas the upload response time is 14 seconds for TCP Cubic which remained flat from 2.2 minutes to 4 minutes.

FTP Global Statistics

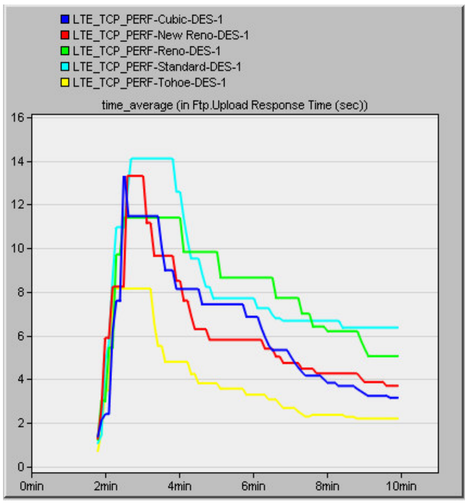


Figure18. FTP Upload Response Time.

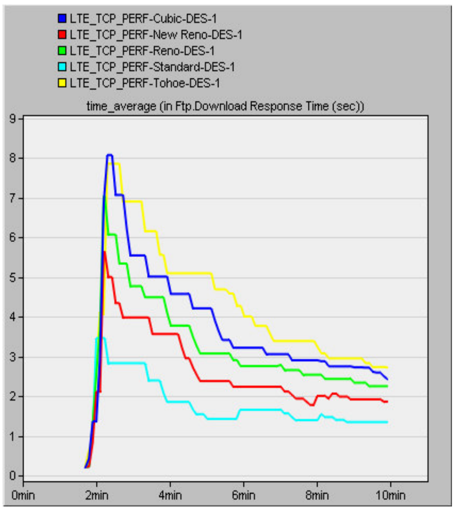


Figure19. FTP Download Response Time

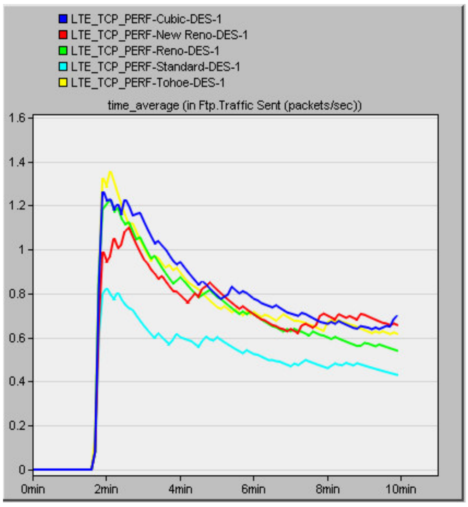


Figure 20. FTP Traffic Sent.

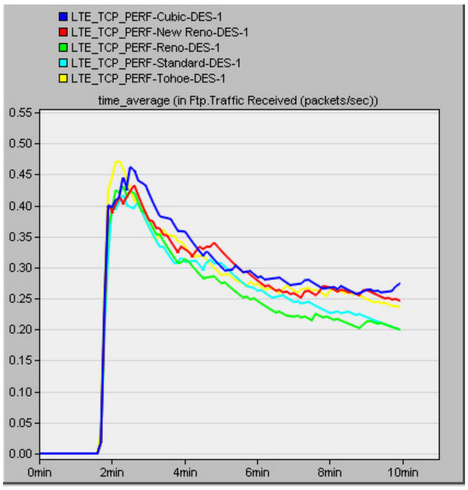


Figure 21. FTP Traffic Received

EPS Bearer

EPS being a connection-oriented requires the formation of "virtual" connection between two endpoints (e.g. a UE and a PDN-GW). This virtual connection provides "bearer service", i.e. a transport service with specific QoS attributes and is called an "EPS Bearer. Associated QoS parameters to the bearer are: QCI, ARP, GBR and AMBR.

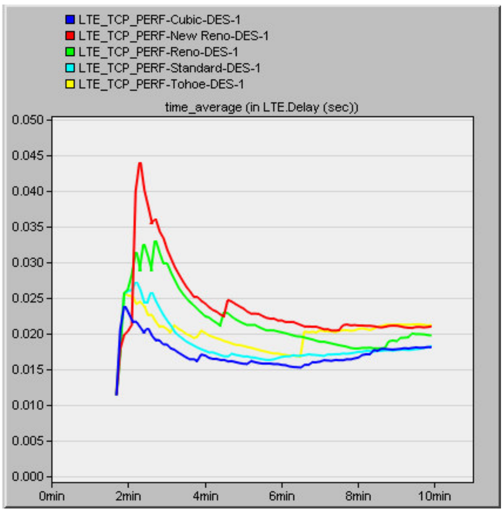


Figure22. EPS bearer delay

During the simulation for EPS bearer, TCP Reno shows the maximum value of the EPS Bearer Delay of 44 milli seconds at 2.4 minutes as shown in figure 20.

Packet Loss Performance

Packet loss refers to the failure of one or more transmitted packets to reach their destination across a network. For satisfactory performance of VoIP applications, more than 98% of the voice packets have to be delivered successfully. Packet loss is defined from the following equation.

$$Packet\ loss = \left(\frac{Send\ packet - Receive\ packet}{Send\ packet}\right) * 100$$

In this section, the packet dropped rate for different scenarios are presented. In all of the figures presented in following sections, X-axis represents simulation time in seconds whereas Y-axis represents the sent and received traffic in packet per second.

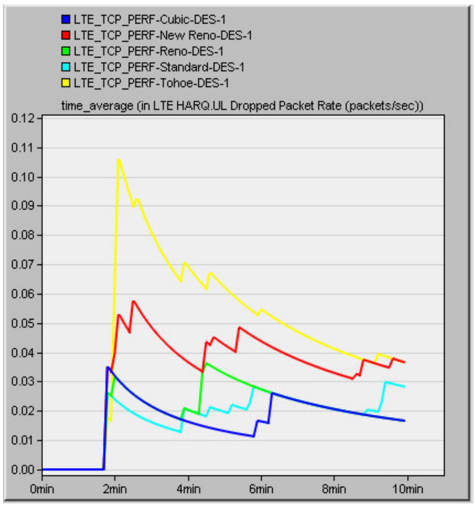


Figure 23. Packets Loss Rate in uplink

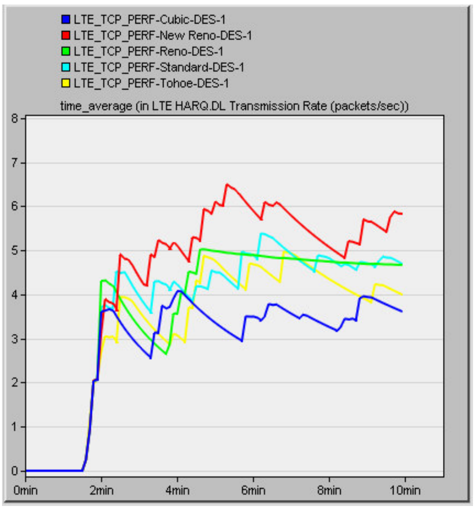


Figure 24. Packets Loss Rate in down link

The goal of this section is to evaluate the performance in terms of packet loss for the different cases of TCP variants. The graph shows that the packet drop rate is highest for TCPTohoe in uplink whereas in downlink the TCP New Reno gives maximum packets loss [19], [20].

V. COMPARATIVE ANALYSIS

The network performance is analyzed in terms of FTP Traffic sent/received, FTP download/upload response time, http traffic sent/received, throughput, end-to-end delay download response time, upload response time and retransmission attempts. Comparative analysis of each variant is summarized below in Table.2:

TABLE2. Comparative analysis of TCP Variants with respect to LTE's Traffic parameters

TCP Flavors	http Traffic (Packet/Sec.)		FTP traffic (Seconds)		Throughput (Packet/Second)		Delay (Seconds)		Response Time (Seconds)		Re-Transmission Rate (Packets/Sec.)	
	Tx.	Rx.	Tx.	Rx.	LTE Network	Pt-to-Pt	UL	DL	UL	DL	UL	DL
Cubic	25	6.8	13.5	8	19	60	10.5	25.5	13.4	8	0.35	0.5
New Reno	30	7.8	13.6	5.5	19.5	70	10.1	25.4	13.4	5.7	0.6	1
Reno	28	7.5	11.5	7.2	21.5	60.4	10	25.4	11.4	7	0.41	1.4
Standard	28	7.8	14	3.5	24	60.8	9	25.4	14	3.5	0.38	0.6
Tohoe	27	7.8	8	7.9	22	60.6	10	25.6	8	7.9	0.75	1.2

VI. CONCLUSIONS

During simulation, details of the traffic parameters have been achieved in terms of the graphical results. It has been found that by using the TCP standard flavor *(in which no Fast Retransmit/Fast Recovery has been used)* maximum value of throughput is obtained in LTE Network. Similarly TCP New Reno provided the highest throughput out of the five TCP flavors used for throughput measurement. The graphical analysis of the traffic generated through http and FTP protocols reveals that TCP flavor with standard values has highest value of the http upload/download and FTP upload and download traffic.

For calculation of end-to-end delay, transmission power of the UE and eNode B is configured according to the Cell size in the network and bandwidth is set to 100 MHz. The simulation results showed that performance of the five variants in point-to-point link between the backbone and Internet is almost the same for downlink and uplink delays. Normally E2E delay is higher when the number of nodes and mobility rates get higher. Finally, rate of packet loss in uplink and downlink have been analyzed. Packet loss is insignificant in network having fixed nodes however the rate increases when the node starts moving.

While concluding, the research reveals important findings on TCP variants when their performances are analyzed for different traffic parameters. The results summarized in Table 2 shows that TCP New Reno has outperformed in terms of throughput, re-transmission rate and http uplink and downlink traffic.

In medium size network, the performance of Reno is satisfactory. Similarly changes in mobility rate enhance the performance of TCP Reno in other congestion control algorithms. On the other hand, TCP Cubic is found to be the worst performer for throughput, re-transmission rate and http traffic parameters.

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