Performance Evaluation of TCP NewReno and TCP Vegas over LTE Network

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***Abstract*—Long Term Evolution (LTE) is a mobile broadband technology that can provide the implementation of wireless networks. It offers a variety of advantages such as better access speed, bandwidth capacity, architectural simplicity and ease of implementation, as well as breadth of type of user equipment (UE) that can access the LTE. The majority of Internet connections in the world are implemented using one of the transport protocols, i.e. TCP (Transmission Control Protocol) due to its reliability in transmitting packets in the network. TCP reliability lies on the ability to control congestion in the network. Formerly, TCP was originally designed for wired media which is more stable compared to wireless medium like LTE. Currently, there are many variants of TCP designed for better performance depending on its usage and network scenarios. In this paper, the performance evaluation is conducted to compare the performance of TCP NewReno and TCP Vegas based on simulation using NS-2. The TCP performance is analyzed in terms of throughput, RTT, packet loss and end-to-end delay. In comparing the performance of TCP NewReno and TCP Vegas, the simulation result shows that the throughput of TCP NewReno is a bit better than TCP Vegas, whereas TCP Vegas shows significantly better end-to-end delay and packet loss.**

***Index Terms*—Long Term Evolution (LTE); Transmission Control Protocol (TCP); TCP NewReno; TCP Vegas.**

I. INTRODUCTION

Long Term Evolution (LTE) is an evolution of mobile network technology which offers variety of advantages, especially in terms of access speed. It can provide a capacity level of at least 100 Mbps downlink, 50 Mbps uplink and Round-Trip Time (RTT) of less than 10 ms [1]. In telecommunication system network, the inevitable exchange of data involves data exchange protocols at transport layer, which is the layer 4 in the Open Systems Interconnection (OSI). Transmission Control Protocol (TCP) is a transport protocol that works at the layer 4 of the OSI model which serves to transmit data per segment, meaning that the data packets are sent in burst in an amount corresponding to the amount of the package then sent one by one until finish. In order for data transmission to work well, TCP will include serial number, known as sequence number for every packet transmission. In addition, LTE supports deployment on different frequency bandwidths, where the current specification outlines the following bandwidth blocks: 1.4MHz, 3MHz, 5MHz, 10MHz, 15MHz, and 20MHz [2].

LTE comes from the standardization of 3GPP (Third Generation Partnership Project) as a method for high-speed access to the continuing development of new wireless telecommunications move towards fixed-mobile convergence (FMC). The evolution towards LTE technology offers a significant increase in network capacity in terms of data throughput in the mobile terminal, thus providing mobile broadband services much better [3]. The main purpose of the evolution of this technology is to provide mobile services with minimum quality equal to fixed broadband access today, as well as reduce operational expenditures by using IP flat architecture. The factors that cause developing 3GPP LTE technology among others is the demand of the users to increase the speed of data access and quality of service as well as ensuring the continued competitiveness of the 3G system in the future. Built by the 3GPP family which had previously been successful in establishing GSM technology, GPRS, EDGE, and WCDMA and HSDPA now and then HSPA +, LTE offers a smooth evolution towards more speed high with low latency [4]. Telecommunications development by the 3GPP standards (third generation partnership project) shown in Figure 1.

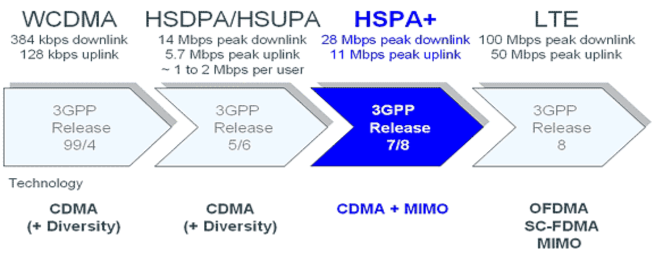

Figure 1: Evolution of 3GPP

The growing popularity of LTE networks has led to cases of heavy utilization and congestion. Network congestion is a phenomenon in which the burden exceeds the capacity of the network [3]. In overcoming these issues, TCP is the most suitable transport protocol to be used because it is a connection oriented protocol that has congestion avoidance mechanisms to ensure the delivery of data packets to the destination.

Transmission Control Protocol (TCP) is a protocol that is on layer transport layer of the TCP/IP. TCP is a protocol that is a byte stream, connection-oriented and reliable in data transmission. TCP uses byte stream communication, which means that the data is expressed as a sequence of bytes. The Connection-oriented means that before the exchange process data between computers first must set up a relationship. It can analogous to the dial on the phone number and finally formed a relationship.

The reliability of TCP in the data sends supported by mechanisms called Positive Acknowledgment with Re-transmission [5]. The data sent from the application layer will be broken up into small parts and given a serial number (sequence number) before being delivered to the next layer. The unit of data that have been broken earlier called the segment (segment). TCP is always asking for confirmation every time after sending the data, whether the data arrive at the destination computer and not damaged [6]. If data has to reach the destination, TCP will transmit data next order. If not successful, then TCP will attempt to retransmit the sequence data is lost or damaged. In fact, TCP uses an acknowledgment (ACK) as a notification between the sender and the recipient computer. Data received at the receiver side will be prepared based on the serial number given by the sender. To cope with damage to the data received, TCP uses a checksum to ensure that the data is not corrupted [7].

The models of two-way communication between two computers send and receive side before the data transmission process is called handshake. The type used TCP handshake is three-way handshake, because using three segments. The purpose of this three-way handshake is for the establishment of a connection, synchronization and notification segment of the data can be received at any time between the send and receive sides [8]. The simple process of the three-way handshake is shown in Figure 2.

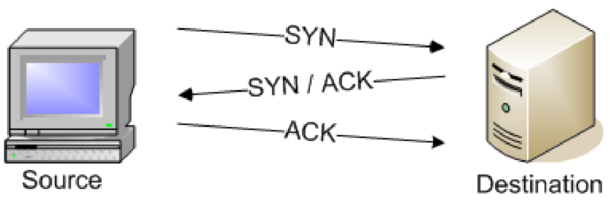

Figure 2: TCP Three-way handshake [9]

The computer is called a source start the relationship by sending a synchronization segment sequence numbers (SYN) on Destination. These segments constitute notice on the computer and that the “Source” wants to conduct a relationship and ask how the serial number will be used as the initial segment of the sequence to be sent. (The serial number is used for data remain in the correct order). The “Destination” responds on the “Source” with a segment that provides ACK and SYN. A computer thus will know the serial number information that is used the “Destination”.
Finally, the “Source” sends a segment as segment posted a reply from “Destination”, as well as doing the actual delivery of the data first. After the process, the “Source found that “Destination” is ready to receive data and soon after relationships occur, the data was sent entirely to “Destination”. At the time all the data has been completed is sent, the three-way handshake is end the relationships to ensure that there is no more data is sent, the connection was established.

The most popular version TCP and most widely used today is TCP NewReno. However, TCP NewReno is considered less effective in terms of media utility when congestion happens in the network especially for wireless environment. Therefore, the other variants of TCP with different congestion control algorithms should be considered as well in order to get the best TCP variant that can work better in a wireless environment. One of the TCP variant that has been studied here is TCP Vegas.

Congestion can be defined as a network congestion from a user’s perspective if the quality of service perceived by the user decreases as the increase in network load. If the time allocation for each user has reached the minimum threshold, but the load is still increasing, the allocation will be smaller. If this is the case, the allocation will reach a value small enough such that the perceived user cannot perform data communication [10].

At today's TCP, the core of congestion control is to adjust the variable congestion window (cwnd), which determines how many packages are not recognized by sender can be sent. Congestion control algorithms which differ primarily determine how the congestion window should be increased for each incoming ACK (acknowledgement) packet and how the congestion window should be decreased to every event of congestion. TCP congestion control was first proposed by Jacobson as a means to prevent "congestion collapse", a condition in which too much traffic on the network led to excessive packet loss [11].

TCP NewReno is most widely studied as the basic congestion control algorithm, which is the base algorithm implemented in the Linux TCP stack. It uses the traditional additive-increase, multiplicative-decrease (AIMD) to control the cwnd. In other hands, NewReno increases the cwnd linearly by one packet for every round-trip time and decreases it by half for every congestion event. One of the advantages of AIMD algorithm is that it allows the using of the cwnd for multiple flow through a link to converge to a fair value [12].

TCP Vegas was the first algorithm that proposed using packet delay or RTT over packet loss as the main signal for congestion. It records the minimum RTT value and uses it to calculate an expected rate. The expected rate is then compared with the actual rate and the cwnd is additively increased, kept constant, or additively decreased [13]. Several studies establish that TCP Vegas does achieve higher efficiency than NewReno, causes the end-to-end delay in TCP Vegas to become better than NewReno, and it is not biased against the connections with longer RTTs [14].

II. EXPERIMENTAL DESIGN

A simple LTE architecture that has been shown in Figure 3 consists of one server for serving FTP (File Transfer Protocol) and provide source connection for the TCP link over the topology [15]. In LTE system, the main job of aGW router is to control the flow rate of the streaming data from server to user equipment (UE) called evolved-NodeB (eNB), where these nodes responsible for buffering the data packets for UE over the network. Each eNB is connected to the corresponding aGW through wireless of 11 MHz bandwidth.

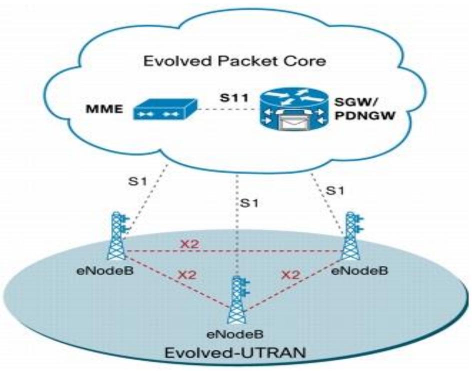

Figure 3: LTE representation

The proposed topology has shown two UEs are used, and connected to eNB within constant bandwidth 11 MHz. In this study, the type of TCP which are used are TCP NewReno and TCP Vegas with FTP as a traffic used. The goal of the experiment is to evaluate the performance of TCP NewReno and TCP Vegas over a network topology based on LTE system. Network Simulator Version 2 (NS-2) [15] is used to evaluated the performance of the proposed model based on TCP/IP. With a wide variety of media, network protocols such as TCP, UDP (User Datagram Protocol) and RTP (Real-time transport protocol) can be simulated. The traffics for the simulation can be from FTP, Telnet, CBR (Constant Bit Rate), multimedia applications such as video layer, quality of service, and audio-video transcoding. The parameters of modeling and simulation are presented in Table 1 [16].

Table 1
Simulation Parameters

Parameter	Value
TCP Protocol	TCP NewReno, TCP Vegas
Bandwidth	11 MHz
Propagation Model	Two Ray Ground
Packet Size	1500 Bytes
Simulation time	50 Seconds
Traffic	FTP

III. RESULTS AND ANALYSIS

In the simulation model shown in Figure 1, the nodes 0 to 5 are established with the same parameters and behavior. Figures 2, 3 and 4 represent the comparison of NewReno and Vegas under similar network conditions, where the bandwidth, propagation model, packet size, simulation time, and the traffic are kept the same as Table 1.

Figure 4 shows the relationship packet delivery and increasing throughput occurs at the beginning of sending packet and relatively stable when it is sending packets. The simulation result shows that the throughput of TCP NewReno is slightly higher than TCP Vegas. The average throughput value of TCP NewReno is 1033.34 kbps and the average throughput value of TCP Vegas is 912.624 kbps.

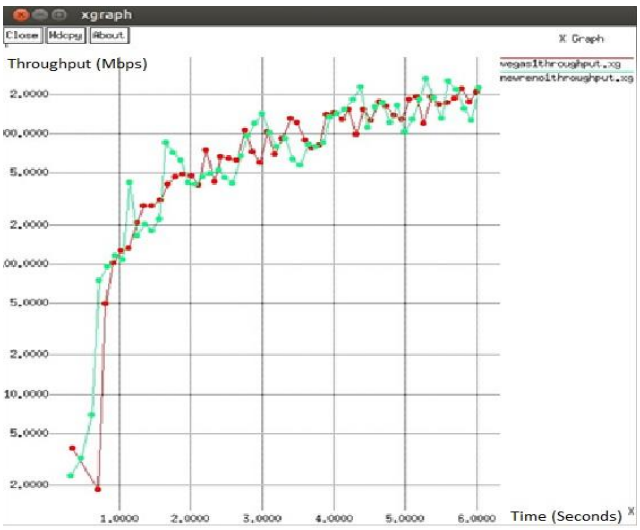

Figure 4: Throughput of TCP NewReno and TCP Vegas

On the other hand, Figure 5 shows the result of the relationship between the RTT and the sending packet size. RTT becomes higher as the packet size increases. It is found that at the beginning of packet transfer, the RTT is measured more than 0.8 seconds. Figure 5 also gives the comparison of TCP NewReno and TCP Vegas in terms of RTT with different packet size. The RTT of TCP Vegas is mostly half than the RTT of TCP NewReno, meaning that TCP Vegas performs better in terms of RTT.

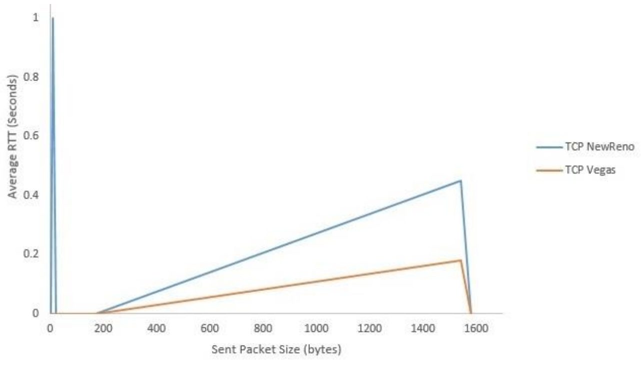

Figure 5: Packet size vs. RTT for TCP NewReno and TCP Vegas

Table 3 depicts the comparison of the packet loss and average end-to-end delay of TCP NewReno and TCP Vegas. The packet loss and end-to-delay that occurs in TCP NewReno is higher than TCP Vegas, meaning that TCP Vegas is better than TCP NewReno in the process of data delivery in terms of packet loss and end-to-end delay.

Table 2
Packet loss and End-to-End Delay

TCP	Packet loss	End-to-End Delay (Average)
TCP NewReno	10.07 %	0.46724
TCP Vegas	8.75 %	0.13720

IV. CONCLUSION

As a conclusion, it is shown that the performance of TCP Vegas is better than TCP NewReno in sending FTP data in terms of Throughput, RTT, Packet loss and End-to-end delay over LTE network. TCP Vegas gives better packet loss, RTT and end-to-end delay than TCP NewReno, even though TCP NewReno has slightly higher throughput than TCP Vegas. The result conforms to the studies conducted by other researchers [13, 14] regarding the performance of TCP NewReno and TCP Vegas over LTE network for the FTP data delivery. For this reason, TCP Vegas is recommended to be used as a transport protocol over LTE network.

ACKNOWLEDGMENT

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REFERENCES

[1] LTE, retrieved from http://www.3gpp.org/technologies/keywords-acronyms/98-lte, 2017.

[2] LTE, retrieved from http://www.extremetech.com/mobile/110711-what-is-lte, 2017.

[3] F. Khan, LTE for 4G Mobile Broadband: Air Interface Technologies and Performance, Cambridge University, 2009

[4] S. Parkvall, E. Dahlman, J. Sköld, and P. Beming, "3G Evolution: HSPA and LTE for Mobile Broadband," Elsevier Publishers, Second Edition, 2008.

[5] Peterson, Larry L. & Bruce S. Davie; "Computer Networks: A systems approach 3rd ed.", Morgan Kaufmann, San Francisco, Calif.; London 2003.
[6] D.E. Commer, Internetworking TCP/IP: Principles, Protocols, and Architecture, Upper Saddle, New Jersey: Prentice Hall, 2006.
[7] R. Braden. "A requirement for Internet hosts -communication layers". IETF RFC 1122, 1989.
[8] J. Postel. "Transmission control protocol". RFC 793 Standard, 1981.
[9] TCP Three-way handshake, retrieved from http://www.networkuptime.com/nmap/images/tcp_handshake.gif
[10] M. Welzl, Network congestion control: managing Internet traffic, John Wiley & Sons, 2005.
[11] V. Jacobson, "Congestion avoidance and control". In ACM SIGCOMM Computer Communication Review. 18(4), 314-329, 1988.
[12] J.C. Hoe, "Improving the start-up behavior of a congestion control scheme for TCP". In ACM SIGCOMM Computer Communication Review. 26(4), 270-280, 1996.
[13] L.S. Brakmo, and L.L. Peterson, "TCP Vegas: End to end congestion avoidance on a global Internet". Selected Areas in Communications, IEEE Journal. 13(8), 1465-1480, 1995.
[14] G.A. Abed, M. Ismail, and K. Jumari, "Behaviour of cwnd for Different TCP source Variants over Parameters of (LTE) Networks". Information Technology Journal, 2010.
[15] K. Fall, and K. Varadhan, The network simulator–NS-2. 2007.
[16] LTE Architecture, retrieved from http://www.cisco.com/c/dam/en/us/solutions/collateral/service-provider/, 2012.