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PERFORMANCE EVALUATION OF TCP NEWRENO AND TCP VEGAS TO AVOID CONGESTION OVER LTE

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Abstract

The one of implementation of a wireless network is based on mobile broadband technology Long Term Evolution (LTE), which offered a variety of advantages, especially in terms of accessing speed, capacity, architectural simplicity and ease of implementation, as well as the breadth of choice of the type of user equipment (UE) that could access the LTE. The majority of internet connections in the world occurred by using the TCP (Transmission Control Protocol), this is caused by TCP reliability in transmitting packets in the network. TCP reliability lied in the ability to control congestion. TCP was originally designed for wired media, but LTE connected through a wireless medium that was not stable in comparing to wired media. A wide variety of TCP made to produce a better performance than its predecessor. In this study, the researcher was simulated the performance that provided by the TCP NewReno and TCP Vegas based on the simulation using network simulator version 2 (ns2). TCP performance is analyzed in terms of throughput and end-to-end delay. The simulation result that obtained from throughput TCP NewReno had an average throughput about 1033.34 kbps and received packets of 4594 bits and the average throughput on TCP Vegas was about 912 624 kbps and received packets of 4161 bits, then the results of the simulation delay obtained, the average delay on TCP Vegas was about 4768.66 ms, and the average delay on TCP NewReno was about 4864.21 ms. Both the analysis throughput and end-to-end delay performed by time for 0 to 60 second.

Keywords: Transmission control protocol, TCP Newreno, TCP Vegas, LTE (Long Term Evolution)

1. INTRODUCTION

Long Term Evolution (LTE) is an evolution of mobile network technology in the world that offers a variety of advantages, especially in terms of access speed, which provides a level of capacity of at least 100 Mbps downlink and uplink at least 50 Mbps and RAN round trip of less than 10 ms. In a telecommunication system network, the inevitable exchange of data. Involves data exchange protocols in the OSI (Open Systems Interconnection). One of the layer is the transport layer.

Transmission Control Protocol (TCP) is a protocol that works at the transport layer of the OSI model (layer 4) which serves to transmit data per segment, meaning that data packets in burst in an amount corresponding to the amount of the package then sent one by one to finish. In order for data transmission up well, then at the time of each packet transmission, TCP will include a serial number (sequence number).

LTE supports deployment on different frequency bandwidths. The current specification outlines of the following bandwidth blocks are: 1.4MHz, 3MHz, 5MHz, 10MHz, 15MHz, and 20MHz. 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 (Khan, 2009). On the other hand in the development of data networks, Transmission Control Protocol (TCP) is a protocol that works at the transport layer of the OSI model (layer 4) and pretty much used, as a connection oriented protocol that has congestion avoidance mechanisms to ensure the delivery of data packets are not lost and get to the destination.

The most popular version and most widely used today is TCP NewReno. However, the TCP NewReno considered less effective in terms of media utility when congestion in the network and when should pass loosy-medium such as wireless. Therefore, there are many algorithms have been proposed to improve the performance of TCP, One of them is TCP Vegas.

2. RELATED WORK

In this section, the researchers provide an overview of congestion and TCP congestion control, especially those closely related to our work.

2.1. Congestion

Definition of congestion as follows: A network congestion from a user perspective if the quality of service perceived by the user decreases as the increasing in network load. If the time allocation of minimum threshold for each user has reached, but the load is still increasing, the allocation will be smaller. If this is the case, the allocation will reach a small enough value such that the perceived user can not perform data communication. This condition is called congestion (Welzl, 2005).

2.2. TCP Congestion Control

At today's TCP, the core of congestion control is to adjust the variable congestion window (cwnd), which determines that how many packages are not recognized by sender that 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 (Jacobson, 1988).

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 (RTT) and decreases it by half for every congestion event. One of the good advantage of AIMD algorithm is that it allowed the cwnd of multiple flow through a link to converge to a fair value (Hoe, 1996).

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 to the actual rate and the cwnd is additively increased, kept constant, or additively decreased (Brakmo & Peterson, 1995).

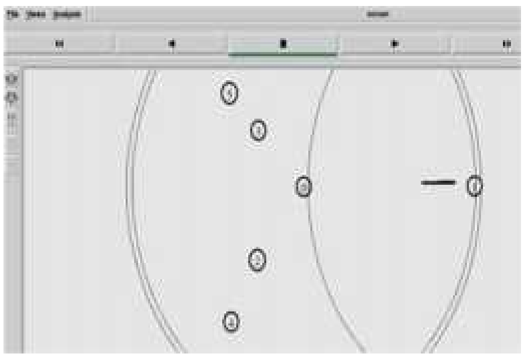

Figure 1 : LTE representation

3. METHODOLOGY

A simple LTE architecture has shown in Figure 1, it consists of one server (node 1) for serving FTP and providing source connection for the TCP link over the topology. In LTE system, the main job of aGW router (node 0) is to control the flow rate of the streaming data from server to user equipment (UE) (node 4,5) called evolved-NodeB (eNB) (node 3,2), where these nodes responsible for buffering the data packets for UE over the network. Each eNB, connected to the corresponding aGW through wireless of 11 MHz bandwidth. The proposed topology has shown 2 UE's are used, and connected to eNB within constant bandwidth 11 MHz.

The researchers evaluated performance of the proposed model by using NS-2 simulator. NS2 network simulator simulates network based on TCP / IP with a wide variety of media, the researchers can simulate network protocols (TCP / UDP / RTP) traffic behavior (FTP, telnet CBR, etc.) Queue Management (RED), FIFO, CBQ) unicast routing algorithm (DSDV, Link State,) multimedia applications such as video layer, quality of service, and audio-video transcoding (Fall & Varadhan, 2007). The parameters of modeling and simulation are presented in the following table (Abed et al, 2010).

Table 1 : Simulation Parameters

Parameter	Value
TCP Protocol	TCP Newreno and TCP Vegas
Bandwidth	11 MHz
Propagation Model	Two Ray Ground
Packet Size	1500 Bytes
Simulationtime	6 second

4. RESULT AND ANALYSIS

The goal of the experiments is to comprehend the performance of TCP Newreno and TCP Vegas over a network topology based on LTE system.

In the simulation model, it shown in Figure 1, the nodes 0 to 5 established with same parameters and behavior, where all of them use either Newreno or Vegas. In figures 3 to 8, represent the comparison of Newreno and Vegas under similar network conditions, which the bandwidth, propagation delay, packet size, window size, and all other link parameters are kept the same.

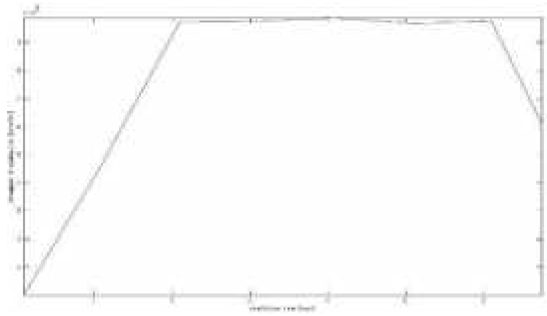

Figure 3: TCP Newreno throughput sending bits

In the chart above, it shows the relationship between packet delivery and increasing throughput occurs at the beginning of sending packet and relatively stable when it is sending packets, but, when the sending packet almost end, throughput will decrease that occurred in TCP Newreno.

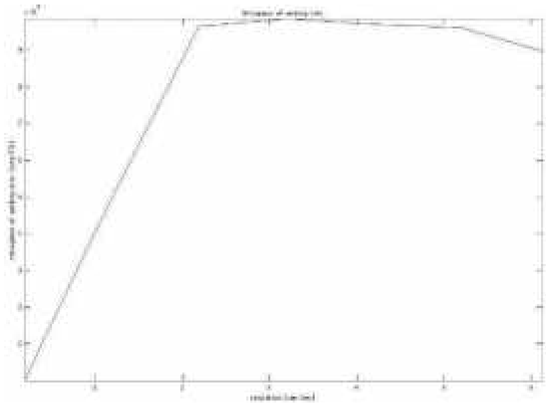

Figure 4: TCP Vegas throughput sending bits

Throughput occurs on the chart above, TCP Vegas, as happened in the TCP Newreno upon delivery of data. However, a little different at the end when sending packets almost finish, a decrease in throughput is not as significant as it did in the TCP Newreno.

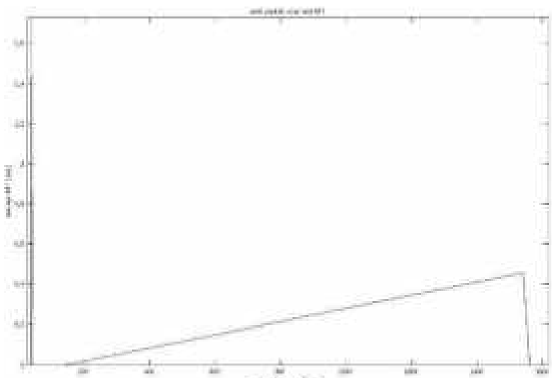

Figure 5: TCP Newreno packet size and RTT

In figure 5, shown a RTT (Round-Trip Time) result on TCP Newreno when sent packet size, the result of the average RTT is less than 0.6 second when packet size sent in TCP Newreno.

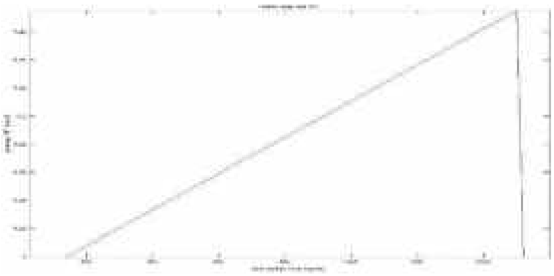

Figure 6: TCP Vegas packet size and RTT

In the TCP Vegas, the results seen RTT on TCP Vegas is less than in the TCP Newreno when packet size sent, the RTT on TCP Vegas is less than 0.2 second.

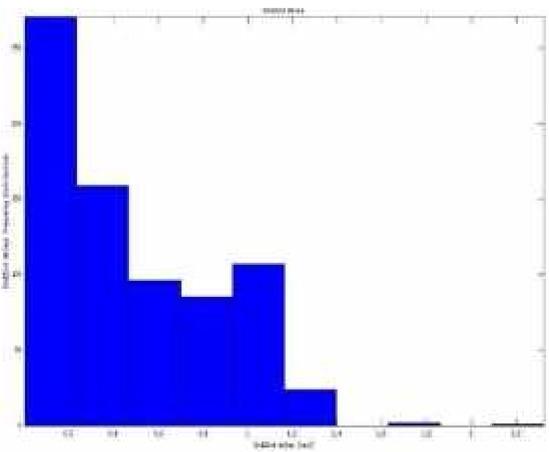

Figure 7: TCP Newreno end-to-end delay

End-to-End delay that occurs in TCP Newreno in the diagram above can be seen in a time of 1 second, the frequency distribution delay which occurs at 100.

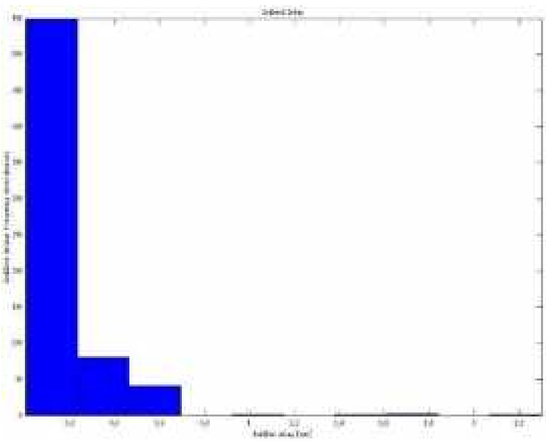

Figure 8: TCP Vegas end-to-end delay

In contrast, end-to-end delay shown on TCP Vegas when compared with TCP Newreno. The diagram above shows end-to-end delay that occurs in a time of 1 second, frequency distribution of end-to-end delay is only shows numbers approximately 0.

5. CONCLUSION

In the conclusion, the researchers have presented the simulation results for evaluating of TCP Newreno and TCP Vegas over LTE network. Several studies establish that TCP Vegas does achieve higher efficiency than Newreno, causes the end-to- end delay in TCP vegas less than Newreno, and is not biased against the connections with longer RTT's. The analysis of TCP performance over LTE ensures that both TCP's have a similar throughput and the best performance return to TCP Vegas than TCP Newreno.

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