



Improvement of TCP Congestion Window over LTE-Advanced Networks

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ABSTRACT—*Long Term Evolution Advanced (LTE-Advanced) network is the continuation of 3GPP-LTE (3GPP: 3rd Generation Partnership Project) and it targets to advanced develop of the requirements of LTE in terms of throughput and coverage. Then, LTE-Advanced is not new as a radio access technology, but it's an evolution of LTE to enhance the performance. The direct employment of the existing protocols cannot achieve the requirements of LTE-Advanced due to the large-bandwidth and low-latency links used in this network. Therefore, an enhanced congestion control mechanism is proposed in this article to improve the performance of Transmission Control Protocol (TCP) over LTE-Advanced. The proposed algorithm used a new mechanism based on the available bandwidth of the connection to detect the capacity of network path in order to improve the congestion avoidance mechanism. The work in this paper based on using Network Simulator 2 (NS-2) to model the traffics of LTE-Advanced and tests the proposed mechanism.*

Keywords—TCP; congestion avoidance, congestion control; LTE; LTE-Advanced; NS-2

I. INTRODUCTION

LTE-Advanced must be compatible backwards in the sense that it should be possible to deploy LTE-Advanced spectrum that occupied LTE with no effect on existing plants of LTE, to include the possibility for data rates [1]. However, more important than the peak data rates is the possibility of providing high data rates over the greater part of the cell. While LTE Rel.8 supports peak data rates exceeding 300 Mbps in the downlink (DL) and 75Mbps in the uplink (UL), LTE is expected to provide in Rel.10 up to 1Gbps in DL and 500 in UL [2]. Also, it will be extended carrier bandwidths up to 100 MHz for LTE-Advanced, while the limited bandwidth in the Far Rel.8 (LTE) is only 20 MHz. Furthermore, the increased spectral efficiency even in 30 bps/Hz in DL and 15-bit / Hz in the UL side by side with the ability to improve the edge of the cell, the user and control plane disappearances, and urged on the user experience more homogeneous over the area of the cell. The specification of the Evolved Universal Terrestrial Radio Access (E-UTRA) is on-going within 3GPP. 3GPP has started LTE of 3G since 2004, whose objectives include reduced latency, higher user data rates, improved system capacity and coverage, and reduced overall cost for the operator [3]. LTE-Advanced systems under developments now, and TCP protocol is the most widely used protocol for wired and wireless systems, although TCP was not originally designed for real time applications and not for wireless networks. Then we need to develop new TCP versions, or at least choose a suitable TCP variant for each

new network to be more efficient and more reliable with this network. For the first time to the congestion control mechanisms in early 1980 and was designed primarily to stop the collapse with traffic congestion typical of that era. In recent years, increased volume of traffic generated by real-time applications, a high proportion of large transport package and congestion control in more often than not opt for these types of applications. By this reason, the systems can fit into the new system is uncontrolled, so much the overall productivity of the oscillating flow and one which, in turn, can result in poor application performance. Apart from concerns about the network level, and those types of applications a lot of care, end-to-end delay in the speed and smooth congestion control, this does not fit the traditional schemes. In this research, we will investigate improving the performance of the new congestion control algorithm, developed to be able to transfer high rate of packet over large bandwidth low-latency platform, such as Advanced LTE. When using TCP over the cellular infrastructure, and the result is that both times of end-to-end production and use of radio link is very weak. This is because the dynamic characteristics of TCP and wireless connections do not fit well together [4]. TCP limits transmission rate by controlling the send congestion window size, which is the number of packets that can be transmitted in the flow. Generally, the time between the submission and receipt of ACK packet is Round-Trip Time (RTT). TCP sender can send up to the congestion window size of data packets during one RTT. Once send the TCP window size of data packets, it can send the new data packet only after it is

reached to some of the ACKs to the sender. Therefore, the average rate of more than TCP RTT is almost the size of the window divided by the RTT [5]. The most important concept in TCP congestion control is the congestion window, where the window is the amount of data that have been sent, but which have not yet received any confession. Congestion window constant means that the broadcast packet and a new one for each ACK that is received while the control is in the rate of transmission indirectly by adjusting the size of the congestion. Been documented standard way to do so in RFC2581 [6], usually referred to as TCP Reno. The other common TCP versions are Newreno [7] and TCP with selective ACK (Sack) [8,9].

In order to improve TCP performance, many TCP variants, which mainly differ with each other on the functions of congestion control, have been proposed. Currently, there are six TCP source variants; Tahoe, Reno, Newreno, Fack, Sack, and Vegas, adding many other extended variants. Each variant has been a private congestion control algorithm or developed a previous algorithm, to be efficient and reliable with a new end-to-end application.

II. ARCHITECTURE OF LTE-ADVANCED NETWORK

Developed by 3GPP, LTE is the leading Orthogonal Frequency-Division Multiple Access (OFDMA) wireless mobile broadband technology. LTE offers high spectral efficiency, low latency and high peak data rates. LTE/LTE-Advanced leverages the economies of scale of 3G, as well as the ecosystem of infrastructure and devices vendors to provide the highest performance in a cost effective manner [10]. The notion of LTE technology depends on the understood that it improves system performance in terms of data rate, throughput, latency, coverage and cost. LTE offers a pure packets assigned architecture, with viability of the movement management. The latest step being studied and developed in 3GPP is an evolution of 3G into an evolved radio access referred to as the LTE and an evolved packet access core network in the System Architecture Evolution (SAE) [11].

In this section, architecture for LTE/LTE-Advanced introduced. So, the architecture which is considered for 3GPP is shown in Fig. 1. E-UTRAN architecture based on, eNodeB which acts as a base station, and called E-UTRAN Node B. MME represents the Mobility Management Entity. S-GW is a serving gateway, and last, P-GW is a PDN (Packet Data Network) Gateway.

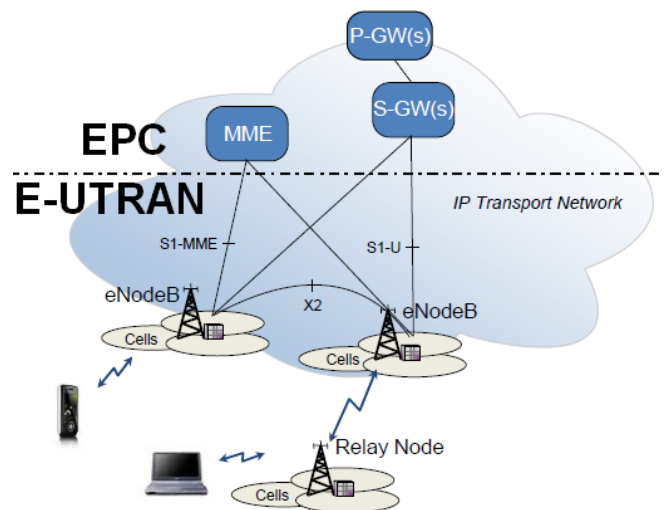


Fig.1 LTE-Advanced E-UTRAN and EPC architecture

Each eNodeB are connected to the MME/SAE Gateway via S1 interface where X2 interface is interconnecting the eNodeBs (or eNBs). The X2 interface is used also on U-plane for temporary user downlink data. The main functions of the eNodeB are:

- Radio resource management (radio bearer control, radio admission and connection mobility control, dynamic scheduling).
- Routing user plane data towards SAE Gateway.

The function of MME/SAE Gateway is to distribute the migration of messages to eNBs; security control; encryption of user data, switching of U-plane to support of UE mobility; idle mode mobility handling [12]. Fig. 2 gives a graphical overview of both protocol stacks. In the user plane, the protocols that are included are the Packet Data Convergence Protocol (PDCP), the Radio Link Control (RLC), Medium Access Control (MAC), and Physical Layer (PHY) protocols.

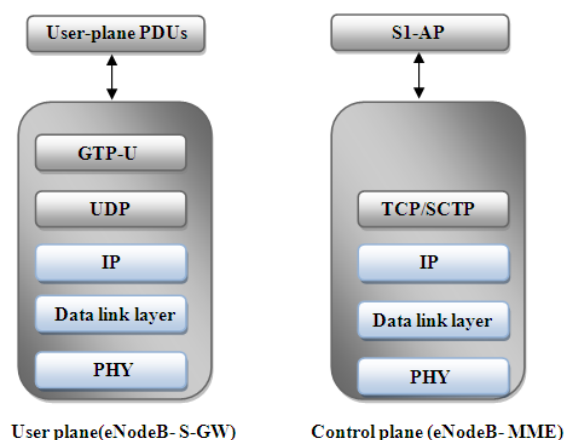




Fig.2 Protocol stack

The control plane stack additionally includes the Radio Resource Control (RRC) protocols. The main functionalities carried out in each layer are summarized in the following [13, 14]:

- NAS (Non-Access Stratum) – Connection/session management between UE and the core network.
- RRC (Radio Resource Control) – Broadcast system information related to Non-Access Stratum (NAS) and Access Stratum (AS).
- PDCP (Packet Data Convergence Protocol) – Header compression and Duplicate detection.
- RLC (Radio Link Control) – Error correction through Automatic Repeat request (ARQ).
- MAC (Medium Access Control) – Multiplexing/ De-multiplexing of RLC Packet Data Units (PDUs).

LTE, 3GPP is also defining IP-based, flat network architecture. This architecture is defined as part of the (SAE) effort. The LTE/SAE architecture and concepts have been designed for efficient support of mass-market usage of any IP-based service. An example of such basic application data which can be configured automatically via the S1 SETUP procedure is the tracking area identities. These identities are very important for the system operation because the tracking areas correspond to the zones, in which UEs are paged, and their mapping to eNBs must remain consistent between the E-UTRAN and the EPC. Thus, once all the tracking area identities which are to be broadcast over the radio interface [15].

III. PROPOSED CONGESTION AVOIDANCE

Primary role, to control congestion, is adjust the window of data transmission at sender side in such a way that is preventing buffer overflow in the recipient, but also in the intermediate routers. To achieve this, TCP used another variable to control the congestion window. Congestion control represents a number of segments of appreciation that can be injected in the network without causing congestion. The challenge is to take advantage of the available space in the store network routers. Routers do not participate in the TCP layer and the chip cannot be used to adjust the TCP ACK frame. To resolve this problem, TCP assumes network congestion as the retransmission timer expires, and that it interacts with the network congestion by adjusting the congestion window using two algorithms, slow start and congestion avoidance, as shown in the Fig. 3. In the slow start phase, and when the connection is established, is first set the value of $cwnd$ to one and then each received ACK value is updated to: $cwnd = cwnd + 1$ which means doubling the $cwnd$ per RTT.

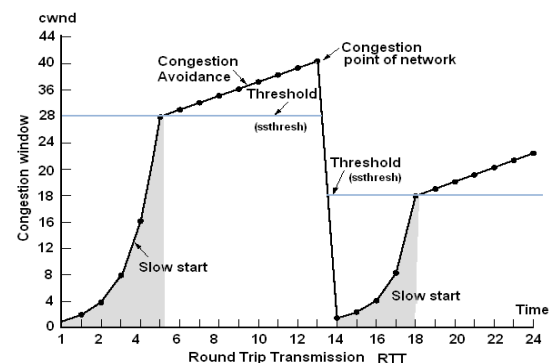


Fig.3 TCP Slow-Start and Congestion Avoidance

The rapid growth of $cwnd$ continues until the packet loss was observed, causing the value of slow-start threshold ($ssthresh$) is updated to: $ssthresh = cwnd/2$. After losing the packet, the connection starts from slow start again by set $cwnd=1$, and is increase exponentially until the window is equal to $ssthresh$, the estimate of available bandwidth in the network. At this point, it goes to the congestion avoidance phase, where the value of $cwnd$ is less aggressive with the pattern: $cwnd = cwnd + 1/cwnd$, which implies a linear rather than exponential growth. And will continue to increase until the written disclosure of packet loss. The new mechanism proposed in this article introduces new congestion avoidance algorithm by estimate the predictable throughput with the prospect of higher productivity, regardless of the level of network congestion. The new algorithm depends on using the available capacity on the network links to detect the increasing or decreasing the size of the congestion window to obtain an adaptive congestion avoidance mechanism. The evaluation and representation of the new algorithm performed using NS-2 to analyse the performance of the proposed mechanism over many experiments. Certainly, the proposed algorithm provided an increment in network path about 20-30%, and that allows to growth the bottleneck capacity too, even the network suffer from congestion. In proposed mechanism, we used classic exponential increment to in slow-start phase. Where congestion window $cwnd$ less than slow-start threshold $ssthresh$ and the window size increases by one, as explained in Eq. (1):

$$\text{if } (cwnd < ssthresh) \\ \text{Then } cwnd = cwnd + 1 \quad (1)$$

TCP sender updates congestion window size $cwnd$ in the congestion avoidance phase according to the following equation when it receives an ACK packet from receiver TCP as shown in Eq. (2):

$$cwnd = cwnd + (f/cwnd) \quad (2)$$



Where f is control parameter. From this equation, we expect that the congestion window size increases by f segments in every RTT. The main function of the proposed mechanism is to regulate f dynamically and adaptively, while the original TCP Reno uses fixed value of $f=1$. In the rest of this subsection, we explain how to change f to the network condition and the current throughput of the TCP connection. The classic algorithm's code is as follows:

$$cwnd_ = cwnd_ + (f / cwnd_)$$

where:

$cwnd_$: is the real $cwnd$ variable in NS-2.

$ssthresh_$: is the real $ssthresh$ variable in NS-2.

f : control parameter, and $f=1$ in Reno.

In proposed mechanism, we need to update f for equation (2) when the sender of TCP receives a new ACK. By this ACK-based mechanism, the proposed mechanism can accommodate the fluctuation of RTTs of the network path. In fact we depend on four main parameters to updates f values every RTT. These parameters illustrated below:

- $ssthresh$: slow-start threshold of network path.
- $cwnd$: the last value of $cwnd$.
- wnd_const : packets per RTT.
- $K_parameter$: k parameter in binomial controls.

By using these parameters for determining f , the degree of increase of congestion window size becomes too large when the current throughput of a TCP connection is far below the target throughput. The large values of f will cause bursty packet losses in the network and resulting in performance degradation due to retransmission timeouts. On the other hand, when the network has sufficient residual bandwidth, the degree of increase of the congestion window size becomes smaller than one. Therefore, we limit the maximum and minimum values of f . For that, every RTT, the new value of f becomes:

$$f = ssthresh * wnd_const(3)$$

In Eq. (3), f will equal the available bandwidth; this will adjust the sending interval of the data packets according to the available bandwidth of the network path. Next, when the proposed formula in Eq. (3) has obtained a larger throughput as explained, we need to minimize the value of f to avoiding packets losses. The control of this problem by divide the last value of f obtained from Eq. (3) by the

previous $cwnd$ value multiplied by the exponent $cwnd$ to the k parameter in binominal control as shown in Eq. (4):

$$f = f / (cwnd * pow(cwnd, k_parameter)) \quad (4)$$

This means, the proposed mechanism steals bandwidth from competing flows in the network in order to achieve the bandwidth required by the upper-layer application. In summary, the proposed mechanism updates f using the following formula when TCP sender receives a new ACK:

$$f = wnd_const * ssthresh_ / cwnd_ * pow(cwnd_ , k_parameter_);$$

$$cwnd_ = cwnd_ + (f / cwnd_);$$

IV. MODELLING AND SIMULATION OF LTE-ADVANCED

The traffic model designed for an LTE/LTE-Advanced system is generally more complex in nature than the traffic model used for a GSM or 3G system. In an LTE-Advanced system, the Mobility Management Entity (MME) and the System Architecture Evolution (SAE) gateway communicate with the eNB by way of an S1 interface. For traffic transport, the S1 interface employs the hub-and-spoke model, which is similar to the traffic model of Iub interface in a 3G system. An X2 interface is used between eNBs in the LTE system to ease the SAE gateway traffic load caused by frequent handover. However, using this type of interface greatly increases the complexity of the traffic model for backhaul networks, as shown in Fig. 1. Theoretically, each eNB must have a direct physical or logical link with its neighbouring eNBs in order to support X2 interfaces, if connection-oriented technology is used. Before describing the model requirements and configuration, and because we used the network simulator, NS-2, we must explain, NS-2 is a discrete event simulator targeted at networking research. NS-2 provides substantial support for simulation of TCP, routing, and multicast protocols over wired and wireless (local and satellite) networks [16]. In NS-2 simulation, all the data in the network is available, thus the performance of the network can be easily analysed. NS-2 is free and open source code and suitable to build system level simulation, so it is deployed to simulate LTE/LTE-Advanced, or any other network. In our research, we used NS-2 version 2.34, and this version, can installed over Windows XP with using Cygwin, where Cygwin provides a Linux-like environment under Windows. Fig. 4, shows the proposed model, for this research, it consist from one server for serving FTP and HTTP, and to provide a source connection for the TCP link over the topology.

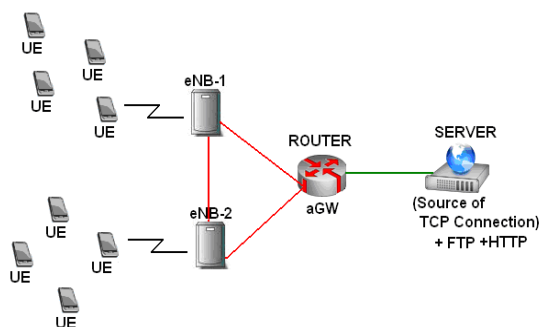


Fig.4 Proposed Topology

The router aGW, connected to the server with duplex link (two ways) and Drop Tail, with bandwidth of 1Gbps, and propagation delay of 5 msec. The main job of aGW router is to control the flow rate of the streaming data from server to eNB-1, and eNB-2. These two nodes are responsible for buffering the data packets to User Equipment UEs. Each eNB, connected to the corresponding aGW through wired simplex link (one way) of 10 Mbps bandwidth and 5 msec delay [17,18]. The other main parameters of proposed LTE-Advanced topology illustrated in Table I, and we can note that all link kept for one propagation delay of 5 msec, and the maximum packet size of TCP was set to 1500 Byte, with minimum window size of 48 Kbytes. Fig. 5 represents the real topology animation created by NS-2 network simulator. The wireless nodes (UEs) linked to the corresponding eNB, through wireless link as a mobile nodes, where the nodes 4,5,6, and 7 connected to eNB-1, and the nodes 8,9,10, and 11 connected to eNB-2. Really, the UE nodes not have full mobility features, because they not move yet, and if we support the movement of these nodes we must add a handover scenario to the topology, and this is not our goal in this research.

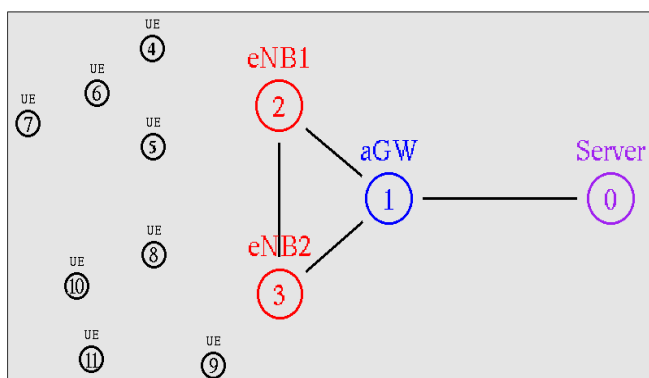


Fig.5 Screenshot of Proposed Network Model

TABLE I
SIMULATION PARAMETERS

Parameter	Value
Propagation Delay of all links	5 msec
Bandwidth eNB-eNB	10 Mbps
Bandwidth eNB-aGW	10 Mbps
Bandwidth aGW-Server	1 Gbps
Bandwidth UE-eNB	2 Mbps
Packet Size	1500 Byte
Window size	48 Kbytes
Simulation Time	30 sec

The evaluation of new congestion control mechanism, already added to a new TCP. This new TCP have the same parameters and specification of other TCP source variants; only the congestion control algorithm changed. In first time, when this new TCP developed, and before supported by new congestion control mechanism, it was tested with constant function of congestion window; $cwnd=h$, where his a constant value such as 40 (packets). This permit to developers to testing the other main functions of the new TCP before adding new congestion control mechanism, and evaluation of the behaviour of congestion window when it have a constant value.

V. RESULTS DISCUSSION AND PERFORMANCE EVALUATION

Firstly, we check whether new mechanism can be effective to the bandwidth available in network path. In these experiments, we performed the data transfer using the new TCP by change the bottleneck bandwidth of the network and keep all other parameters. As shown in Fig. 6, we used a bandwidth of 100 Mbps and test the behaviour of $cwnd$ of new TCP and compared it with TCP Reno.

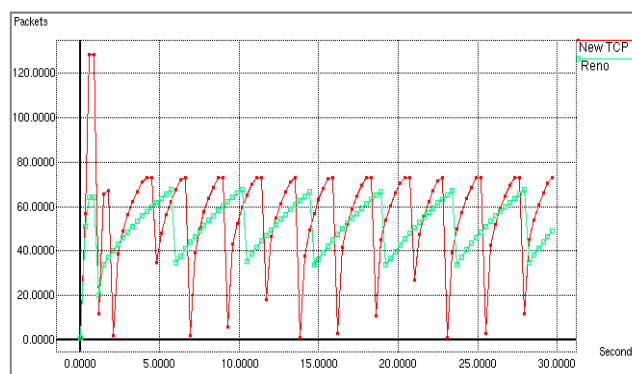


Fig.6 cwnd Behaviour of New Mechanism and Reno (BW=100Mbps)

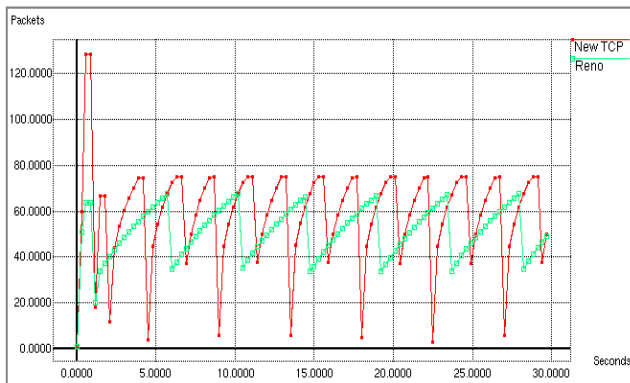


Fig.7 cwnd Behaviour of New Mechanism and Reno (BW=200 Mbps)

The measurement results of cwnd during the experiment explain that the congestion window of new TCP performed well comparing with Reno congestion window, so the clocking of new TCP is also better than Reno clocking; in addition we have a smooth slow-start phase like Reno.

In Fig. 7, we can notes the large increment in *cwnd* between new TCP and Reno, and this difference because that we increased the bandwidth from 100Mbps in first experiment (Fig. 6) to 200 Mbps. This increment permit to the new mechanism to open *cwnd* reaches to 75 packets while Reno kept the same packet level in two experiments of less than 65 packets. The reason beyond to the difference in mechanism used in Reno and new TCP, as we explained that because Reno used $f=1$, while new TCP estimates the available capacity of network path.

The performance variation of two mechanisms appear clearly in next test, when we used a bandwidth of 500 Mbps as shown in Fig. 9. This leads to get a *cwnd* exceeded 80 packets provided by new TCP, with constant window size in Reno. Furthermore, Fig. 8 shows that the available bandwidth will gives a throughput of cross traffic. That is, TCP Reno cannot be used for background data transfer. This means that the new TCP will increase the *cwnd* in same time that it will not decrease the throughput of the co-existing foreground traffic. Additionally, Figures, 6,7, and 8, shows that the *cwnd* of new TCP is closest to the available bandwidth. Therefore, the new mechanism can utilize the available bandwidth better than Reno. These results clearly show the proposed mechanism which utilize the available bandwidth information obtained by the new formula, performs well in the experimental network environment. These three experiments shows that the new mechanism really related with the available bandwidth and this concept will permit to use it in a full traffic model of LTE-Advanced system where the bottleneck of network reaches to 1Gbps.

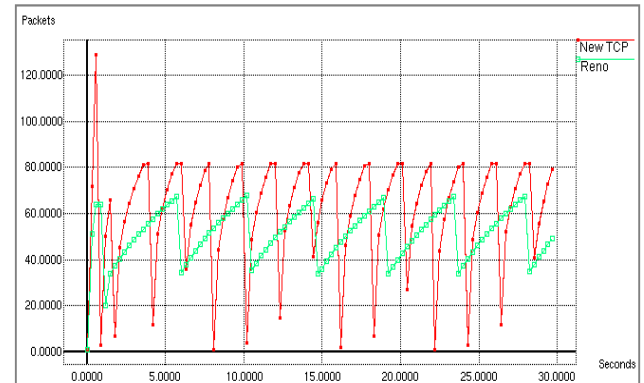


Fig.8 cwnd Behaviour of New Mechanism and Reno (BW=500 Mbps)

For this reason and from previous experiments, it is not efficient to use Reno or any other classic TCP variants over high bandwidth networks like LTE and LTE-Advanced because of the limited performance of congestion control for these variants and that is the main reason to develop new mechanisms has more efficiency. Next, and after we ensured that new mechanism be ready to work over high bandwidth-low latency links, we demonstrate the performance of the proposed mechanism over model described in Section 4. In the simulation, we establish multiple wireless nodes connected to base stations eNB1 and eNB2. This setting means that the new TCP will transfer data from USs to Server through network path with network bottleneck of 1Gbps as a bandwidth and 5 msec as a propagation delay. Here, the new TCP will detect the current information on the available bandwidth and the physical capacity of the network path. This assumption is necessary in order to focus on evaluating the algorithm described before. Fig. 9 shows the cwnd behaviour in which the proposed mechanism can obtained and shows that the original TCP Reno have about of a half *cwnd* compared with *cwnd* obtained from new mechanism.

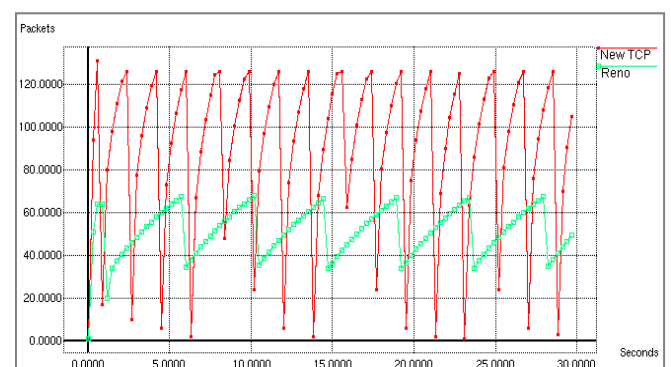


Fig.9 cwnd Behaviour of New Mechanism and Reno(1Gbps Bandwidth)

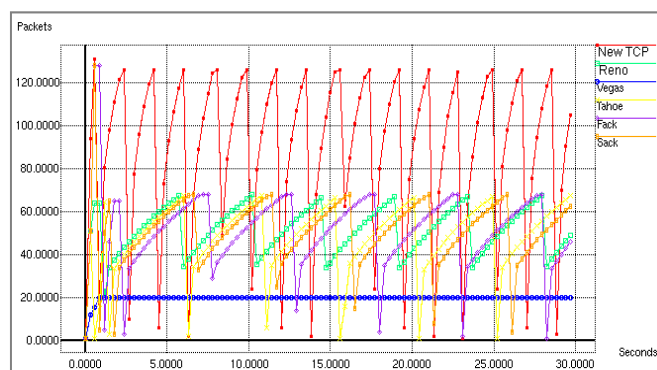


Fig.10cwnd Behaviour of New Mechanism and Six TCP Variants (BW=1Gbps)

The new mechanism gave about 125 packets as a maximum congestion point, but Reno kept the same previous value of 65 packets. On the other hand, the new TCP gave a performance speed more than twice that obtained from Reno, where everyone complete cycle clock of cwnd in Reno, we obtained more than two cycles in new TCP. That mean we will send a more than double amount of packets in new mechanism. Finally, and for more accuracy, we performed more experiment with other five TCP variants, Newreno, Tahoe, Sack, Fack, and Vegas to achieving the full requirement to depend the new TCP as a new mechanism that support data transfer in LTE-Advanced systems with high performance and with more efficiency and reliability form other classic TCP variants, as shown in Fig. 10.

VI. CONCLUSION

In this paper, the authors focused on developing enhanced TCP congestion control with new methodology to work over LTE and LTE-Advanced systems. The new proposed mechanism modifies the degree of increase of congestion window size of a TCP connection in the congestion avoidance phase by using the information of the available bandwidth of the network path. Through simulation experiments, we demonstrated that proposed mechanism over traffic model of LTE-Advanced system and from results obtained that new mechanism achieve the required cwnd with high probability, even when there is no residual bandwidth on network path. We also reported that the extended mechanism performs efficiently to provide the required widow size for Large-Bandwidth and Low-Latency networks. In addition, we implemented the proposed mechanism on a NS-2 simulator and support it by new TCP and confirmed from the implementation that the new TCP

works well in actual LTE-Advanced networks. The performance of the proposed mechanism is highly dependent on the estimation results of the available bandwidth given by the last status of network path.

VII. FUTURE WORK

We will evaluate the performance of the proposed mechanism in other actual high speed network environments. In addition, we will test the behaviour of the congestion window when the UEs nodes have a full mobility features and evaluate the performance with some simple handover scenarios when these nodes move from one eNB to another.

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Biography



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