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Queueing Evaluation of Transmission Control Protocol in LTE-Advanced

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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 is targeted to advanced development of the requirements of LTE in terms of throughput and coverage. Then, LTE-Advanced is not new as a radio access technology, but it is an evolution of LTE to enhance the performance. The employment of the existing protocols such as Transmission Control Protocol (TCP) cannot achieve the requirements of LTE-Advanced due to the large-bandwidth and low-latency links used in this network. Therefore, it's necessary to enhance the TCP variants such as improving the congestion control performance over LTE-Advanced to make these variants fulfill the needs of the huge data which transferred over the secured links. This article offered comparative evaluation and estimation of the queue size performed by different TCP source variants over traffic model of LTE-Advanced system using the Network Simulator 2 (NS-2).**

Index Terms— **LTE-Advanced; queue size; TCP; NS-2**

I. Introduction

In September 2009, the partners of 3GPP have prepared the official suggestion to the proposed new ITU (International Telecommunication Union) systems, represented by LTE with Release 10 and beyond to be the appraised and the candidate toward IMT-Advanced (IMT: International Mobile Telecommunications). After attaining the requirements, the main object to bring LTE to the line call of IMT-Advanced is that IMT systems must be candidates for coming novel spectrum bands that are still to be acknowledged [1][2]. LTE-Advanced is applying various bands of the spectrum which are already valid in LTE along with the future of bands of IMT-Advanced. More developments of the spectral efficacy in downlink and uplink are embattled, specifically if users serve as an edge of the cell. In addition, LTE-Advanced aims quicker exchanging between the resource of radio states and between additional enhancements of the figures of latency. All at once, the bit cost must be decreased [3].

IMT-Advanced represents the next generation in systems of wireless communications, which aim to accomplish other main advances of the current third-generation systems, by reaching to uplink (UL) rate of 500 Mbps and to 1Gbps in the downlink (DL) [4]. With LTE-Advanced starts, there are many keys of requests and features that are up coming to the light.

II. Architecture of LTE-Advanced Networks

3GPP identified in Release 8 the requirements and features and requirements of the architecture of Evolved Packet Core (EPC) which that serving as a base for the next-generation systems [5]. This identification specified two main work objects, called LTE and system Architecture Evolution (SAE) that leading to the description of the Evolved Packet Core (EPC), Evolved Universal Terrestrial Radio Access Network (E-UTRAN), and Evolved Universal Terrestrial Radio Access (E-UTRA). Fig. 1 illustrates the architecture of LTE-Advanced networks based on EPC and E-UTRAN. Each of these parts corresponded respectively to the network core, system air interface, and the radio access network. EPC is responsible to provide an IP connection between an external packet data network by using E-UTRAN and the User Equipment (UE).

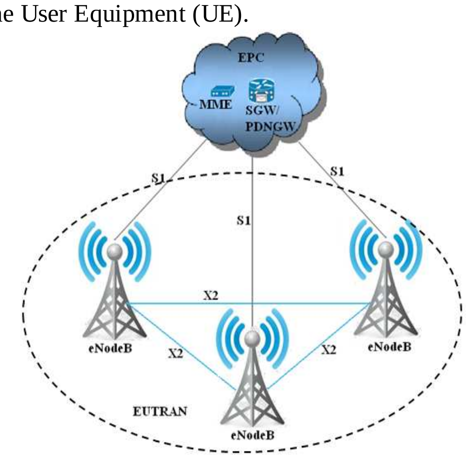

Fig. 1. LTE-Advanced architecture

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In the environment of 4G systems, the radio access network and the air interface are actually improving, while the architecture of the core network (i.e., EPC) is not suffering large modifications from the previous systematized architecture of SAE. The main part in the architecture of E-UTRAN is the improved Node B (eNB or eNodeB), that provide the air interface between the termination of control plane protocol and the user plane towards the user equipment (UE). Both eNodeBs are a logical element that serving one or more E-UTRAN cells and the interfacing between the eNodeBs is termed the X2 interface. The interfaces of network are built on IP protocols. The eNodeBs are connected by an X2 interface and to the MME/GW (Mobility Management Entity/Gateway) object with an S1 interface. The interface S1 supported many relationships between eNodeBs and MME/GW [6]. The two entities of the logical gateway are termed Serving Gateway (S-GW), and the other is Packet Data Network Gateway (P-GW).

The Serving Gateway (S-GW) acts as a limited anchor for the mobility service to receiving and forwarding packet rates from and to the eNodeB to serve the UE, while the P-GW is interfaced with the exterior Packet Data Networks (PDNs) for example, the IMS (Internet Multimedia Server) and the Internet. P-GW provides other IP functions such as packet filtering, routing, policy statement, and address allocation. The MME is an entity to provide signaling only and later the user packets on the IP do not pass over the MME.

The main benefit of separating the network entities is for indicating if the capacity of network for traffic and signaling can independently grow. Actually, the core tasks of MME are too idle mode the reach ability of UE together with controlling the retransmission of paging, roaming, authorization, P-GW/S-GW selection, tracking area list management, bearer management, including dedicated bearer establishment, authentication, security negotiations and signaling of NAS[7]. The eNodeB is implementing the functions of eNodeB along with protocols usually applied in Radio Network Controller (RNC).

The eNodeB functions are ciphering, packet reliable delivery, and header compression. However, in controlling side, eNodeB is incorporating functions such as:

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

Several benefits of using one node in the network accessing are to reduce the latency and the RNC processing distribution load into many eNodeBs.

III. User Plane and Control Plane Protocols

The stack of user plane protocol is shown in Fig. 2. From the Fig. 2, the Radio Link Control (RLC) and the Packet Data Convergence Protocol (PDCP) layers usually concluded in RNC on the network side are now concluded in eNodeB.

The control plane protocol stack is demonstrated in Fig. 3, where the Radio Resource Control (RRC) functional conventionally applied in RNC is integrated into eNodeB [8]. The layers of Medium Access Control (MAC) and Radio Link Control (RLC) are implementing similar roles to the user plane.

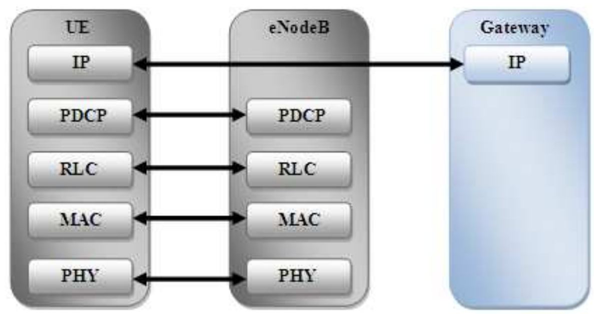

Fig. 2. User plane protocol

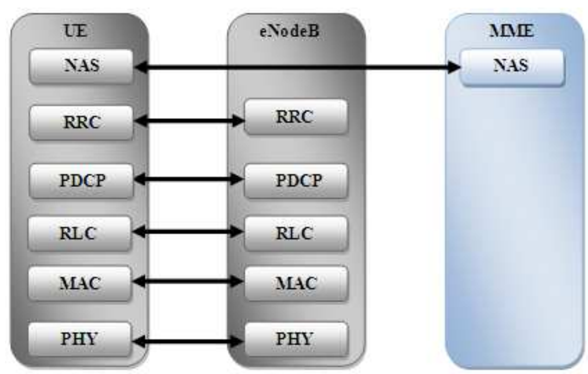

Fig. 3. Control plane protocol architecture

The RRC functions include paging, system information broadcast, radio bearer control, connection management for RRC, measurement reporting to UE, and mobility functions. On the MME network side, the Non-Access Stratum (NAS) protocol is terminated while on the terminal side, the UE executes functions such as Evolved Packet System (EPS), authentication, security control, and bearer management. In Fig. 4 and Fig. 5, the interface protocol stacks S1 and X2 are presented where the protocols that used are similar in the two interfaces. The interface between S-GW and anode are interconnected by the S1 user plane interface (S1-U).

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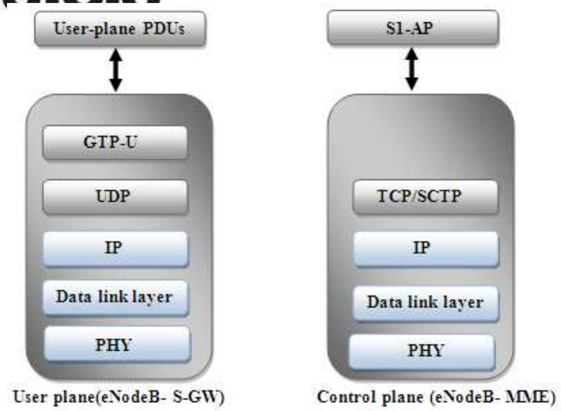

Fig. 4 .S1 Interface user and control planes

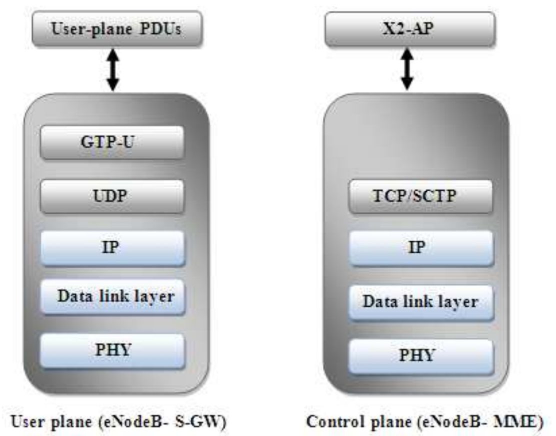

Fig.5. X2 Interface user and control planes

This interfacing is using GPRS Tunneling Protocol-User Data Tunneling (GTP-U) over UDP/IP transport. Moreover, it provided none numerated delivery to the user plane PDUs between S-GW and eNodeB [6]. GTP-U is a comparatively simple IP and is based on tunneling protocol that allows a lot of tunnels between end point's sets. In details, the S1 interfacing is separating the EPC and the E-UTRAN. It is split into two interfaces; the first is S1-U that are transfer's traffic data among S-GW and the eNodeB, and the second is S1-MME that is a signaling the interface between the MME and eNodeB.

X2 is the interfacing between the eNodeBs and also involving two interfaces; the first is X2-C, which is the control plane interface between eNodeBs, and X2-U are the user plane interface between eNodeBs. It is always supposed that there is an X2 interface between eNodeBs which is to provide communicating between each other [9]. S1-MME represents the S1 control plane interfacing between MME and eNodeB. Similarly, the transport network layer and user plane are based on IP transport and in case of a reliable transport to the signaling messages; the Stream Control Transmission Protocol (SCTP) is applied over IP top.

These protocol functions analogously to TCP confirming a reliable, in sequence transmission of all messages with congestion control. SCTP drives analogous to TCP certifying reliable and offer in-sequence transport of messages with congestion control [10]. The application layer signaling protocols is mentioned to S1 application protocol (S1-AP) and X2 application protocol (X2-AP) for S1 and X2 interface control planes respectively. 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.

IV. Traffic Model and Simulation

The next generation and challenging network govern a set many guidelines that state movement, behavior, and the mobility (if mobile nodes supposed to be movable). Network simulators can then, by using this information, create random topologies based on node's position, parameters, and the tasks between the nodes [11]. Currently, there are many network simulators that have different features in different aspects. A short list of the current network simulators includes OPNET, NS-2, NS-3, OMNeT++, REAL, SSFNet, J-Sim, and QualNet. NS-2 is the most popular one in academia because of its open-source and plenty of component's libraries. A lot of non-benefit organizations contribute a lot in the component library, and it has been proven that the development mode of NS2 is very successful [12].

As mentioned before, the modeling in this article based on NS-2 as a modeler NS-2 not just a simulator, but it's a discrete event aimed to support the research and studies that deal with communications and network analysis. In addition, NS-2 provides environments to simulate and modeling multicast protocols; network traffic, handovers, and other network resources and conditions for wireless and wired channels [13]. In our research, we used NS-2 version 2.34, and this version installed over Windows XP or using Cygwin, where Cygwin provides a Linux-like environment under Windows, because NS-2 is supported by Linux operating system only.

The proposed model is shown in Fig. 6. It involves one main server for serving data as FTP and HTTP, also to providing source connection for the TCP. The routers Gateway1 and Gateway2 are connected directly to the Server with a duplex link with bandwidth reach to 1Gbps, and propagation delay of three msec. In fact, the propagation delay for all links over the proposed model kept the same value of three msec, where this value represents the practical latency of the links interfacing and connections on LTE-Advanced networks. The function of Gateway routers is to control the flow rate of the streaming data from the server to the base stations eNodeB1, eNodeB2, and eNodeB3.

Gateway within the wired simplex link with Bandwidth reaches to 10 Mbps. The interface between base stations (X2) is very important in a model setting due to the relation between eNodeBs will detect the handover scenario when the UEs move from one eNodeB to another. The base station nodes are responsible for buffering the data packets to the User Equipment's (UEs).

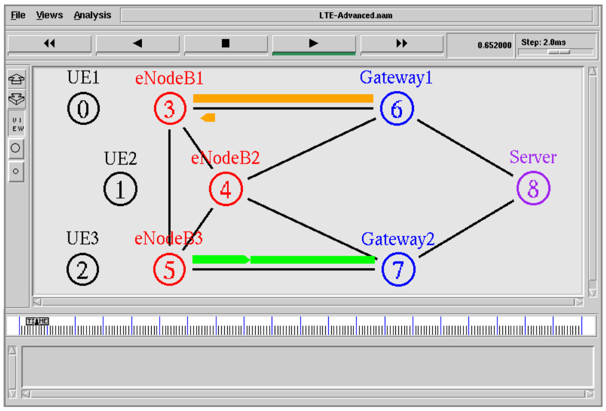

Fig.6. Animation of Proposed LTE-Advanced Model

TABLE I. MODEL PARAMETERS

Parameter	Value
Protocol	TCP-Sack, TCP-Fack,TCP-Reno, TCP-Vegas, TCP-Tahoe, TCP-Newreno
Propagation Delay of all links	3 msec
Bandwidth eNodeB-eNodeB	20 Mbps
Bandwidth eNodeB-Gateway	10 Mbps
Bandwidth Server-Gateway	1 Gbps
Packet Size	1500 Byte
Window size	128 Packets
Simulation Time	10 sec

Each base station (eNodeB) is connected to the corresponding the average bandwidth size of X2 proposed to be 20 Mbps and this represent the estimated and practical range. In the proposed model, three UEs used with wireless features and each UE coupled to eNodeB, these UE nodes don't have full mobility features because avoiding the handover scenario in this model where that represents the next step. The other main parameters of proposed LTE-Advanced model is illustrated in Table 1, and we can note that all link kept for one propagation delay of 3msec, and the maximum packet size used sets to 1500 Byte, with minimum window size of 128 packets.

The final model tested with simple experiments to evaluate the model. This experiment based on use TCP-Reno, TCP-Vegas, TCP-Tahoe, TCP-Newreno, TCP-Sack, and TCP-Fack to estimate and evaluate the queuing performed by each TCP version over the proposed LTE-Advanced model.

The queue size at a bottleneck would affect the performance of TCP protocols, especially with the single TCP flow and large bandwidth path. There are many studies on TCP/RTT determined by evaluating packet traces, but it is not efficient if the maximum RTT is used to estimate the bottleneck queue size. In this study, the maximum values of one-way delays are immediately measured when a packet loss occurs and leads to queue overflow. To estimate the number of queued packets for each RTT interval in the slow-start phase, the following formula in Eq. 1 is used [14].

$$N = \frac{W * RTT(R_i - R_o)}{BW} \qquad (1)$$

Where *N* is the number of queued packets, *W* is the window size in packets, *BW* is the link bandwidth capacity, *Ro* is the bottleneck rate, *Wi* is the initial window size, and *Ri* is the burst rate at which the TCP source generates the traffic.

To estimate the number of queued packets for each RTT interval in the slow-start phase, the following formula in Eq. 2 is used. In Eq. 2, the standard TCP increases the window size by two when an ACK is received so that *RI* is limited to 2*Ro*.

$$W = \frac{W_i * 2 * Time}{RTT} \qquad (2)$$

The queue size at a network bottleneck would affect the performance of TCP protocols, particularly when executed in a single TCP flow in networks [15]. One of the important metrics to validate the performance is to know the queue sizes of network pipeline, because the dynamic behavior of TCP protocol implementations varies according to bottleneck queue size. The next section focuses on evaluating and investigates the performance of TCP source variants according to the queue length by taking all the packet traces at the sender [16].

V. RESULTS AND DISCUSSIONS

This section investigates the performance of TCP variants that are affected by the queue length by taking all the packet traces at the sender, where it is possible to calculate the number of queued packets at the bottleneck by simulating drop-tail token buckets [17]. The estimation of queue size over TCP running is subject to several factors and limitations; one of these limitations is at what average rate TCP expects to transmit, but there is no control of the rate within the slow-start period or discovery of the available bandwidth capacity rapidly.

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Therefore, TCP transmits packets as a burst per RTT intervals at a rate up to the transmitter capability. Because standard TCP's increasing window size almost jumps for each RTT interval, the final growth of window size may save overflowing a bottleneck queue till a packet loss indicates because congestion reaches the sender at most afterward RTT.

The default queue length proposed in LTE-advanced model is set to 12 packets and the queue monitoring focused on the link eNodeB1-Gateway and eNodeB2-Gateway as shown in Fig. 6. The motive of using 2 bottlenecks in the queue monitor instead of many bottlenecks relates to the testing of the queue management of standard TCP (Tahoe) and comparing it with the management by other TCP's due to the main goal here is not to test the model, but to evaluate the TCP versions. That's why we should expect the queue size not to exceed 12 packets, but simultaneously it should be in large level to avoid congestion in bottlenecks as far as possible. The queue size of the six TCP's is shown in the figures 8, 9, 10, 11, and 12. It can be noted the large difference in the queue management performed by the candidate TCP versions in both of steady run and during initialization phase.

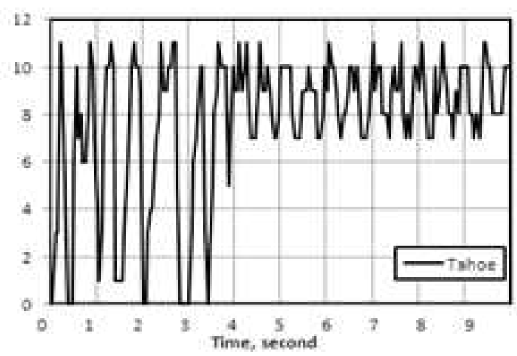

Fig. 7 Queue Size of TCP-Tahoe

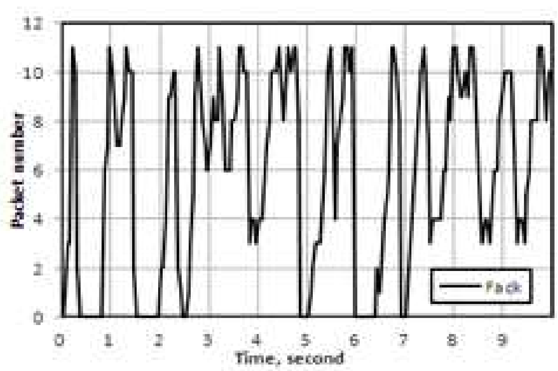

Fig. 8 Queue Size of TCP-Fack

In Fig. 7, Tahoe showed a reasonable queuing control in spite of the poor size in the first 4 seconds but it keeps decent queue level until the simulation end.

On the other hand, Fack not performed well during the simulation period due to the weakness strategies used in the congestion control mechanism adopted by Fack as shown in Fig. 8.

Despite Reno and Newreno are based on similar congestion control mechanism (the difference only happens with multiple packet loss) but they behaved differently. The reason behind this difference that Reno not perform well over multiple packet loss while Newreno can perform better with multi loss therefore Reno showed better queuing comparing with Newreno due the proposed scenario not included multiple loss of the transferred packet and only congestion event added to the scenario.

Nevertheless, Reno and Newreno not able to give high queue management as shown in Fig. 9 and Fig. 10 while the improved Reno (TCP-Sack) showed better queue size as shown in Fig. 11. The congestion control of TCP-Sack based on Reno congestion control with selective Acknowledgment (ACK) and this property assist Sack to save the required time to acknowledging the transferred packet and then keep the queue size in good level compared with the typical Reno.

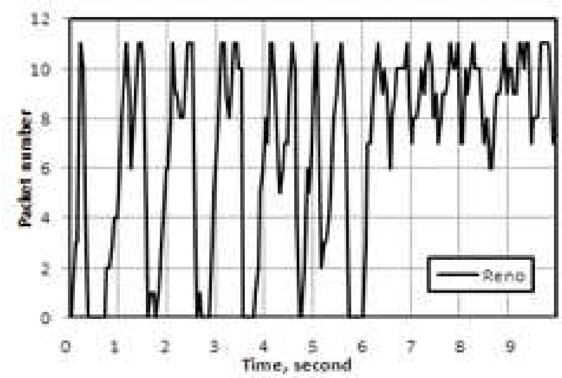

Fig. 9 Queue Size of TCP-Reno

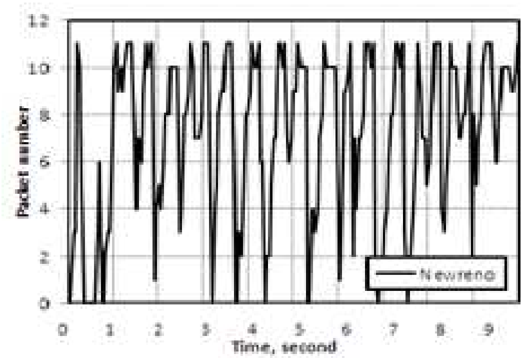

Fig. 10 Queue Size of TCP-Newreno

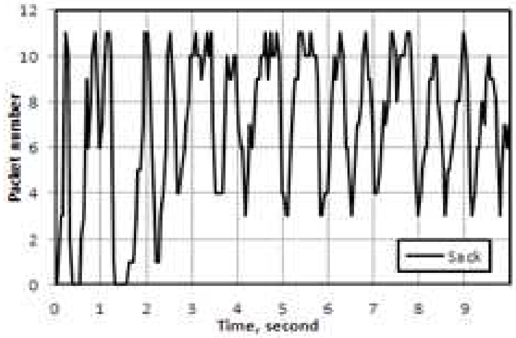

Fig. 11. Queue Size of TCP-Sack

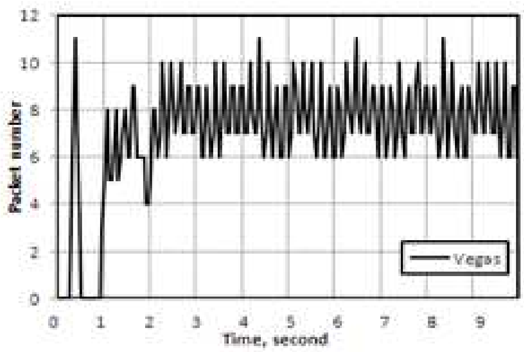

Fig. 12. Queue Size of TCP-Vegas

The all TCP variants used in this study used same slow-start mechanism and share many routines in congestion control except TCP-Vegas, where Vegas uses a different approach to avoid congestion in the pipeline network. Vegas congestion control senses the available bandwidth and compares it with the target bandwidth before increase the congestion window. This facility makes Vegas to avoid the early congestion and give steady throughput with minimum packet loss. On te other hand, even though the throughput is steady but it will not be in high level. Fig. 12 shows the excessive queue size performed by Vegas over LTE-Advanced model where the size oscillated near the maximum size and the size of queue opened earlier than the other TCP variants.

VI. CONCLUSION

This article provides experimented results to estimate the queue size given by six TCP variants, Tahoe, Reno, Newreno, Sack, Fack and Vegas over a proposed traffic model of LTE-Advanced System. Furthermore, this article provides the basic procedures to implement the link interface for LTE-Advanced networks by using network simulator NS-2.

In addition, it offers the main features of the user plane protocol, control plane protocol, and the link interface of the protocol stack and illustrate the parameter values which make the limitation to the behavior of these protocols over LTE-Advanced. The comparative queue size estimation proved that TCP-Vegas performed better than other TCP versions while the improved Reno (Sack) performed better than the typical Reno while TCP-Fack gave poor queue management in spite of the similarities in congestion control of Fack with other TCP source variants.

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