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Comparative Performance Investigation of TCP and SCTP Protocols over LTE/LTE-Advanced Systems

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ABSTRACT—***Fourth Generation (4G) of mobile systems is now replacing the 3G and 2G families of standards.Long Term Evolution (LTE) and LTE-Advanced network is the continuation of 3GPP-LTE (3GPP: 3rd Generation Partnership Project) and achieved the next step towards high data rate transfers and support of all multimedia functions and requiremensts. One of the main solutions to improve the performance of the new wireless communication systems is by improving the protocols used over these networks such as Transmission Control Protocol (TCP) and Stream Control Transmission Protocol (SCTP). This paper investigates the comparative performance of TCP and SCTP protocols over LTE and LTE-Advanced systems.***

Keywords—**LTE; LTE-Advanced; TCP; SCTP**

I. INTRODUCTION

One of the main requirements of 4G systems is that the users should not feel any difference between a wired and a wireless network, and they should have multiple options for connectivity over heterogeneous networks. In other words, it needs a new adaptive, reliable, and efficient TCP that is able to fulfil these requirements. The direct approach to enhance the performance of TCP for the new generation networks is to modify the TCP itself because TCP still represents one of the major causes of poor performance in a wireless environment [1].

Meanwhile, the LTE/LTE-Advanced networks plan to support mixed voice, video and other messaging traffic data. Unfortunately, wired and wireless networks have different main features, such as bandwidth capacity, propagation delay, and reliability in links. The main reason beyond these differences is that packet losses are not caused by network congestion, as it is in the wired networks, but by many other factors of wireless specific reasons and features [2]. In the Wireless Local Area Networks (WLANs) or cellular systems, most lost packets are fundamentally due to the large amount of bit error rates in the wireless networks and the possibility of the handovers between two cells and base stations. In the mobile systems, however, most lost packets accrue are due to medium congestion and route breakages [3][4]. For these reasons, TCP performs well over wired links and networks, but it suffers serious problems and performance degradation over the wireless networks. This will certainly happen when the networks transfer high data rate when the bandwidth reaches 1 Gbps, such as the LTE-Advanced system [5]. The characteristics of the wired channels expect a very small amount of lost packet probabilities and randomly spaced errors, while wireless channels are characterized by time varying packet loss probabilities and are typically very large as compared with those of the wired channels [6]. It is also true that the standard protocols, such as TCP, perform poorly over the wireless links, which is due to lost or delayed packets [7]. In addition, the TCP can deal with these problems and resolve some types of congestion in the network paths, but this may cause reducing in the packet flows. In some cases, such as when the TCP suffers unpredictable losses or packet delay (occurring in wireless channels), the TCP reduces packet rate even when the wireless link bandwidth is still with full capacity [8].

Many researchers have found that TCP supports wireless access inefficiently [9] and many of these researchers found that the key problem is that wireless channel errors would lead to frequent failures of the TCP retransmission time. This will cause an extra congestion by the TCP added to the expected congestion of the network path [10]. Mostly, the networks are built on top of the TCP for data applications because it provides a reliable end-to-end retransmission scenario when the IP packets are dropped [11]. Considering the fact that TCP was originally designed for wired networks, whose packet losses were due to network congestion, the window size of the TCP would be adjusted upon detection of packet loss.The LTE networks were designed to be packet-based and the architecture was developed to contain less network nodes and

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minimum links routing [12]. This would reduce the protocol processing overhead and lead to a reduction of latency in the network queues. In addition, the evolved architecture of the LTE networks focuses on the use of the TCP/IP protocol as a packet data communications protocol, whilst others focus on Stream Control Transmission Protocol (SCTP) [13].

The basic architecture of the LTE and LTE-advanced systems is dependent on the Evolved Packet Core (EPC) network element. Like all IP systems, in the EPC of LTE, the IP cannot provide a reliable service which degrades the packet transmissions in term of packet loss, packet duplication, or when the packet is delivered out of the sequence [14]. However, TCP can provide the required delivery service on the top of IP and can give a more reliable data link. TCP on the networks is responsible for controlling packet flow and resolving problems that are resulting from congestion by using congestion control mechanisms, in addition to providing connection management for all network traffic. Nonetheless, LTE and LTE-Advanced networks support a high throughput by an IP based transport network link. Then, the TCP is expected to play a hazardous role in the performance of LTE systems [15].

Due to the high error rates and because it is inappropriate to directly use the TCP on the LTE system, the LTE employs an error recovery technique at the link layer to overlap the error recovery mechanism which is provided by the TCP in the transport layer [16]. There are many end-to-end solutions to improve TCP over LTE/LTE-Advanced systems during the handover period, or to increase the size of the TCP window. The problem lies in the required modification of the current TCP variants [17]. The TCP actually provides a very good service, where reliability is paramount; however, it is not generally suitable when delays are critical or when the network has a low propagation delay. In fact, this is expected with the LTE-Advanced networks, which support many applications, such as voice and video services. Added to that discrete loss in the data transfer is less important than its time delivery by TCP over the classic mobile systems. Therefore, more research is still required to address these particular transmission characteristics of the new generation systems.

When the TCP is used over the LTE/LTE-Advanced systems, some packets are surely neglected and the TCP sender will be aware of this loss through the acknowledging transmission mechanism. This means that when the TCP sender sends packets and does not receive an ACK after a while, a loss has occurred and the sender must retransmit these packets. In the LTE and LTE-Advanced systems, a high bit rate, which is similar or even larger than that offered in the wired networks, is offered. This will certainly cause a lot of pressure to the performance of the TCP; which is more than that of the traditional wireless networks [16].

Standard TCP versions assume that most lost packets (about 99%) in wired channels are due to the congestion in the network traffics and the remaining (about 1% only) are due to damages [18]. Therefore, it is important to take other actions to avoid any possible congestion in the network, where the congestion is controlled by the TCP to prevent the sender from overflowing the network capacity. This mechanism is known as the TCP congestion control. This TCP congestion control is certainly not a new technique to improve the TCP performance, but many algorithms, which are implemented for different TCP variants, have been developed to deal with network congestion [19].

Many research and studies have suggested improving the ordinary TCP versions in order to obtain a better performance on 3G and beyond 3G networks, but many of these trials have proposed to solve either the handover or the bit error rate problems, such as I-TCP, M-TCP, Freeze TCP, and Snoop TCP. In the incoming 4G mobile systems, a high data rate is expected and the expected services provided by this generation should be carrying because it needs a robust TCP protocol to overcome the new challenges like handover, bit error rate, and asymmetry between the base station and mobile host.

II. TRANSMISSION CONTROL PROTOCOL

TCP is a basic communication language, and a connection-oriented protocol tied with the transport layer. TCP comprises a collection of rules and procedures to control communication over links. Many TCP variants have been modified and developed to achieve the requirements of communications. Most of the current versions of TCP include a set of algorithms to control the congestion for critical links in network to improve the network throughput [20]. Of late, the TCP has grown rapidly to meet the increasing demand to transfer the media over high-speed links, which run on top of TCP. Furthermore, the TCP designed for wired networks can also be used in wireless networks due to the significant features such as, flow control, reliability, congestion control, and connection management for the network connections [21]. The TCP is capable of adjusting its congestion window size, to improve the performance, but that may cause further performance degradation. This represents serious problem in mobile networks, which have rapid topological changes [22]. Besides that, the TCP divides the sequenced data stream into packets, and confirms the packets delivery with the possibility of losing IP layer, retransmit, reorders, or packets duplication, and monitoring the network band capacity to avoid congestions.

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Each TCP sender can regulate the size of the congestion window using the congestion control mechanism and the TCP can dynamically regulate the window size, depending on the packets acknowledgment (ACK) or by the occurrences of packets losses. If the congestion window is constant, the ACK timing, of the sent packets depends on the ACK of the first set of packets (early packets). Furthermore, the TCP window also depends on ACK clock and it calculates the sender flow rate, and when the required time to send packet and receive ACK – Round Trip Time (RTT) - changes with different values, the TCP sliding window determines the mean of the transmission rate of completed window per average RTT. Fig. 1 illustrates an example of three packets transmitted using TCP sender and acknowledged with separated ACK by TCP receiver.

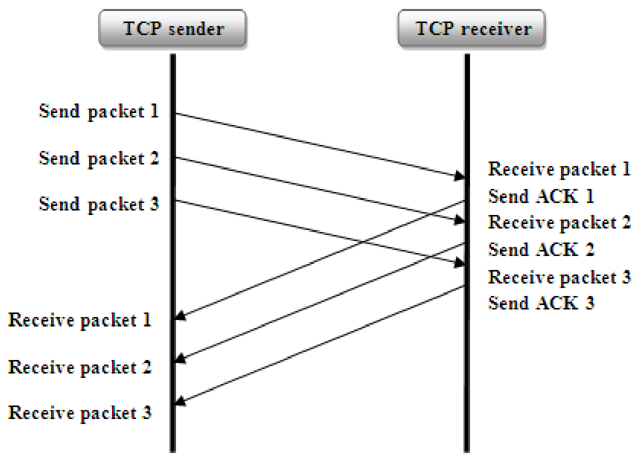

Fig. 1Three Packets Transmitting and Acknowledging using TCP

The transmission window dependently controls the received ACKs for each RTT and the parameters and the control mechanism indicates the general differences between TCP versions. This controlling is used to obtain high packets rate with minimum losses by avoiding the network overloading and at the same time, it provides optimum sharing to the network bandwidth among connections [22].

As discussed earlier, many TCP variants have been developed for different networks and applications. The TCP Tahoe and TCP Reno are mostly applied over many wired and wireless applications, because they include effective congestion control mechanisms. These mechanisms provide different size of congestion windows depending on ACK status. Thus, when packets are acknowledged the window size increases, or it will decrease when detected packets are lost.

In TCP Tahoe, Reno, and Vegas (Vegas one of earlier TCP variants), the congestion control algorithm permits the window size to increase by one segment in every RTT. This increment stops when the window size reaches the congestion point.

Basically, the TCP seeks to provide reliability to the data transmitted between two hosts and it tries to apply set of rules to handle loss of packets, which results from physical errors in transmission or due to the congestion in cross traffics in internet or cellular systems. The provided reliability is hardly achieved by TCP because it is not easy to determine the available bandwidth for TCP packets flow, due to the complexity to indicate the effects of the congestion control of TCP and the network dynamically. In addition, understanding the behaviour and the approaches of TCP is essential to enhance its performance in wireless channels. Consequently, many researches have been geared up in this regard, however some of the problems have already been solved, but many other issues remain unaddressed [23].

III. TCP VERSUS SCTP

Both LTE and LTE-Advanced use either TCP or SCTP, to send messages between two peers. Since both protocols are connection-oriented, the connection between two peers has to be established before sending any command [24]. The SCTP is a transport protocol identified in RFC 2960 [25], which works at an equivalent level in the stack as TCP and UDP. Compared to TCP and UDP, the SCTP is superior in functionality and more robust against the failures in the network connections. The purpose in employing SCTP is to provide an efficient and reliable signalling bearer. To achieve this, the SCTP provides appropriate congestion control techniques with fast retransmission in the case of packet loss and enhanced reliability. Furthermore, it provides extra security against blind attacks and enhances security feature, when connected on top of UMTS and other 3G systems with different operators.

When the functionalities of TCP and SCTP are compared, it is evident that the later provides two key features, multi-streaming and multi-homing, which lacks in TCP. In the SCTP domain, a stream is a unidirectional sequence of user packets to be distributed to upper layers. Consequently, bi-directional communication between two entities includes at least a pair of streams, one in each direction. The multi-streaming is the feature, from which the name of STCP is actually derived. It permits setting up several independent streams between two peers. In such a case, when a transmission error happens on one stream, it does not affect the transmission on the other streams.

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In contrast, TCP only provides one stream for a given connection between IP peers, which may cause additional data transmission delay when packets dropped. When a transmission loss happens on a TCP connection, the packet delivery is suspended until the missing parts are restored. SCTP provides new services and features for IP communication. However, the TCP provides reliable communication service and the UDP provides unreliable service but neither TCP nor UDP can handle multi-homing or have the ability to send information to an alternate address, if the primary becomes unreachable. Many of the features found in TCP and UDP can be found in SCTP. Comparative results between SCTP, TCP and UDP are provided in Table 1.

TABLE 1
SIMULATION PARAMETERS

Feature	**TCP**	**UDP**	**SCTP**
Connection oriented	Y	N	Y
Reliable transport	Y	N	Y
Preserve message boundary	N	Y	Y
In-order delivery	Y	N	Y
Un-order deliver	N	Y	Y
Data checksum	Y(16-bit)	Y(16-bit)	Y(32-bit)
Flow and congestion control	Y	N	Y
Multiple streams within a session	N	N	Y
Multi-homing support	N	N	Y

Y:YES N:NO

Based on the Table 1, it can understand that, the UDP is quickly ruled out as unfit as it is not reliable. From a high-level perspective, the SCTP and TCP are quite close to each other, as they both, support reliable and ordered data delivery, and they have congestion control, to regulate network data flow. Nevertheless, the SCTP is message-oriented and frames individual messages as against the TCP. In SCTP, the messages are transmitted as a whole set of bytes (provided the maximum length is not reached), which helps to improve the transmission efficiency.

The SCTP was first developed to professionally transfer the telephony signalling data across the Internet, but its features make it more attractive for other applications too.The SCTP is well known for its advanced features inherited from TCP that ensure the required reliable delivery of the signalling messages. Furthermore, the enhanced features such as, the handling of multi-streams to easily implement transport network redundancy add more value to the SCTP. The 3GPP working groups debate on the suitable transport protocol for the controlling plane in LTE/LTE-Advanced, to support the signalling message exchange between network entities.

The SCTP can handle congestion and packet drop, better than the standard protocols. In fact, the SCTP implements congestion and flow control such as, TCP. In addition, the SCTP association is distributed into a number of streams, where the SCTP associations are similar to the TCP connection, except that, it is able to provide multiple IP addresses for both ends. The SCTP is similar to TCP in using message acknowledgement and retransmission mechanism, to ensure the segments delivery to the corresponding hosts. Moreover, as in TCP, the SCTP has three main phases in congestion control such as: slow-start, congestion avoidance, and fast retransmit.

As mentioned above the SCTP uses multi-streaming, therefore if the transmission to the primary path is carried out in the congestion avoidance mode, the implementation may still use slow-start for the other paths. The congestion control for TCP and SCTP shares many features in slow-start and congestion avoidance phases. Therefore, when using either TCP or SCTP, the congestion control is affected by the parameters and the limitations of the LTE/LTE-Advanced network. The significant objective for LTE and LTE-Advanced is to decrease the network latency. The network latency is the time taken by a packet to travel from a client to a server and back again and that represents a direct limiter on performance of TCP. The congestion control mechanism, especially the slow-start, is sensitive to the link latency in the network and it is expected to delay the data transmissions. Therefore, the protocol used should guarantee to prevent packet from going beyond the network capacity, even when the number of transferred packets is huge.

The congestion control of SCTP follows the same congestion window reduction mechanism used by TCP. In addition, the SCTP might behave similar to TCP in the event of multiple packet losses. The SCTP applies window-based congestion control similar to TCP, and uses AIMD technique to control the window size according to the network conditions.

IV. CONCLUSION

This paper provides an analysis and investigation to the behaviour and performance of TCP and SCTP protocols over LTE and LTE-Advanced systems. The comparison based on different features such as congestion control, multi-streaming, and data transmission reliability. Moreover, the possibility to improve the performance of TCP over wireless and 4G systems were reviewed too. Based on previous researches, the employment of TCP and SCTP protocols over LTE-Advanced network was discussed with the explanations to the downside and the limitation in using TCP protocol over large-bandwidth low-latency networks. Additionally, the comparisons between TCP and SCTP protocols were presented in this paper investigated the distinctive employment of TCP and SCTP over LTE/LTE-Advanced networks with the key features of SCTP protocol.

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Biography

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