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Fast Recovery Approach to Improve TCP Congestion Problem in 4G LTE Network Using NS-3

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ABSTRACT

The Long Term Evolution (LTE) of the UMTS Terrestrial Radio Access and Radio Access Network is another correspondence standard went for business arrangement in 2010. Objectives for LTE incorporate backing for enhanced framework limit and scope, high top information rates, low idleness, lessened working expenses, multi- recieving wire support, adaptable transfer speed operations and consistent mix with existing systems. The point of this proposal is to study the effects on the end-client and framework execution when clients with high bit rates TCP administrations are traveling through the system. These effects influence the decreased end-client or framework throughput, e.g., because of clogging in the vehicle system, prompting poor usage of the vehicle and radio assets accessible. To reach such a point, it has been important (1) to make another test system with the ns-3 and perform reenactment in diverse system settings and (2) define a quick recuperation approach ready to catch the essential flow of the genuine framework and to keep away from blockage circumstance. Conceivable answers for relieve the effects are researched by contrasting the recreations consequences of TCP execution in the radio and transport system and we perform Fast recovery in TCP Stream.
Keywords: LTE, UMTS, TCP, Quality of service, Evolved Packet Core, OFDM, MIMO, RATs

I. INTRODUCTION

The recent increase of mobile data usage and the emergence of new applications, such as Multimedia Online Gaming (MMOG), mobile TV, Web 2.0, streaming contents, have motivated the 3rd Generation Partnership Project (3GPP) to work on the Long Term Evolution (LTE). LTE is the latest standard in the mo- bile network technology tree, which previously implemented the GSM/EDGE and UMTS/HSxPA network technologies now account for over 85% of all mo- bile subscribers. LTE will ensure 3GPP's competitive edge over other cellular technologies.

LTE, whose radio access is called Evolved UMTS Terrestrial Radio Access Network (E-UTRAN), is expected to substantially improve end-user through- puts and sector capacity also to reduce user plane latency, bringing significantly improved user experience with full mobility. With the emergence of Internet Protocol (IP) as the protocol of choice for carrying all types of traffic, LTE is scheduled to provide support for IP-based traffic with end-to-end Quality of service (QoS). Voice traffic will be supported mainly as Voice over IP (VIP) enabling better integration with other multimedia services. Initial deployments of LTE are expected by 2010 and commercial availability on a larger scale 1-2 years later.

Unlike High Speed Packet Access (HSPA), which was accommodated within the Release 99 UMTS architecture, 3GPP is specifying a new Packet Core, the Evolved Packet Core (EPC) network architecture to support the E-UTRAN through a reduction in the number of network elements, simpler functionality, im proved redundancy but most importantly allowing for connections and handover to other fixed line and wireless access technologies, giving the service providers the ability to deliver a seamless mobility experience.

LTE has been set aggressive performance requirements that rely on phys- ical layer technologies, such as: Orthogonal Frequency Division Multiplexing (OFDM), Multiple-Input Multiple-Output (MIMO) systems and Smart Anten- nas to achieve the baseline targets. The main objectives of LTE are to minimize the system and User Equipment (UE) complexities, allow flexible spectrum de- ployment in existing or new frequency spectrum and to enable co-existence with other 3GPP Radio Access Technologies (RATs).

II. METHODS AND MATERIAL

RELATED WORK

The most commonly used transport layer protocols are TCP and udp. udp[12] is a connectionless protocol that does not guarantee delivery of data, it does not make any error check on the payload. It is used in server client type request-reply situations and in applications where prompt delivery of data is more important than accurate delivery, like video streaming.

The Transmission Control Protocol (TCP) is one of the core protocols of the Internet Protocol Suite. TCP is so central that the entire suite is often referred to as "TCP/ IP". Whereas IP handles lower-level transmissions from computer to computer as a message makes its way across the Internet, TCP operates at a higher level, concerned only with the two end systems, for example a Web browser and a Web server. TCP/IP manages the end-to-end reliability in the Transport layer. The topic of this section is the Transmission Control Protocol (TCP) which is the predominant transport protocol of the Internet today carrying more than 80% of the total traffic volume. TCP is used by a range of different applications such as web-traffic (www), file transfer (ftp, ssh), e-mail and even streaming media application with "almost real-time" constraints such as, e.g., Internet radio. Among its management tasks, TCP controls message size, the rate at which messages are exchanged, and network traffic congestion.

2.1 Importance of TCP

TCP provides a communication service at an intermediate level between an application program and the Internet Protocol (IP). That is, when an application program desires to send a large chunk of data across the Internet using IP, instead of breaking the data into IP -sized pieces and issuing a series of IP requests, the software can issue a single request to TCP and let TCP handle the IP details.

IP works by exchanging pieces of information called packets. A packet is a sequence of bytes and consists of a header followed by a body. The header de- scribes the packet's destination and, optionally, the routers to use for forwarding generally in the right direction - until it arrives at its final destination. The body contains data which IP is transmitting. When IP is transmitting data on behalf of TCP, the content of the IP packet body is TCP payload.

Due to network congestion, traffic load balancing, or other unpredictable net- work behavior, IP packets can be lost or delivered out of order. TCP detects these problems, requests retransmission of lost packets, rearranges out-of-order packets, and even helps minimize network congestion to reduce the occurrence of other problems. Once the TCP receiver has finally reassembled a perfect copy of data originally transmitted, it passes that datagram to the application program. Thus, TCP abstracts the application's communication from the underlying networking details.

2.2 TCP key features

TCP[11] is a reliable transport protocol with connection procedure that pro- vides a in-sequence data delivering from a sender to a receiver. Some of the TCP properties may be desired by certain applications.
The TCP reliability is achieved using ARQ mechanism based on positive ac- knowledgments.

TCP protocol provides transparent segmentation and reassembly of user data and handles both data flow and congestion. TCP packets are cumula- tively acknowledged when they arrive in sequence; out of sequence packets cause the generation of duplicate acknowledgments. For each segment transmission, a retransmission timer is started; retransmission timers are continuously updated on a weighted average of previous round trip time (rtt) measurements, i.e. the time it takes from the transmission of a segment until the paired acknowledgment is received. TCP sender detects a loss either when multiple duplicate acknowl- edgments (3 is the default value) arrive, this imply that the packet after the last acknowledged one has been lost, or when a retransmission timeout (RTO) expires. The RTO value is calculated dynamically based on rtt measurements. Its accu- racy is critical, delayed timeouts slow down recovery or redundant retransmissions can occur due to mistakes in the RTO evaluation.

III. RESULTS AND DISCUSSION

LTE is a next generation 4G technology, and still it demands to evaluate and experiments the performance for better and speedily accessing to the back bone internet network.

We proposed evolutionary experiment of LTE technology under various parameters to enhance its performance and optimization. The main propaganda of this synopsis is to provide satisfactory QoS of LTE and TCP performance under various mobility impacts in LTE network.

Following points will be consider in proposed work-

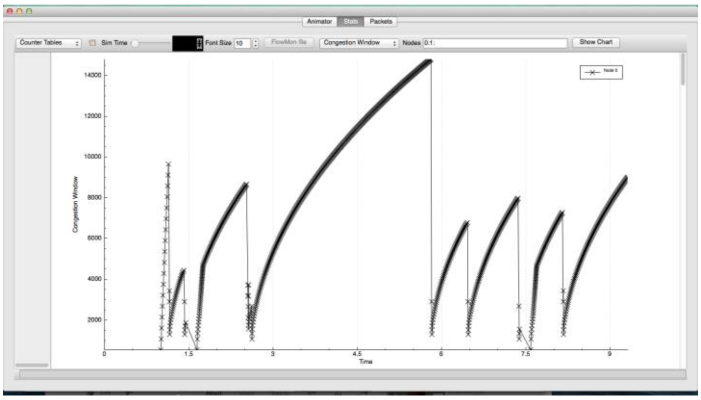
1. Evaluation of LTE network using Simulator
2. TCP and UDP performance in LTE 4g technology.
3. Mobility impact

3.1 Proposed Method for Congestion Control in TCP on LTE

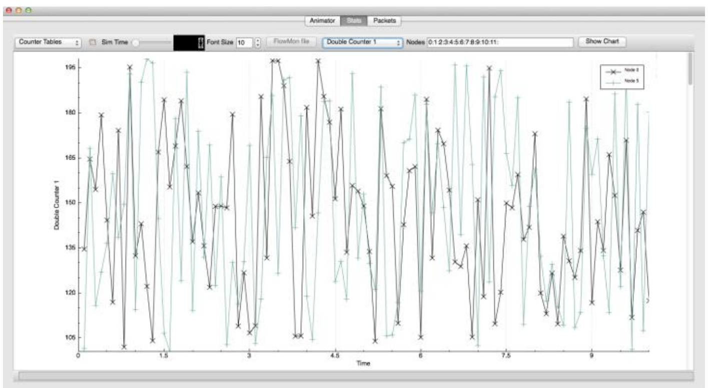
We will adopt following procedure to control the congestion in TCP-

if (ndupacks and CW = 1)
{
ssthresh = cwnd
Retransmit the lost packet
Enter fast recovery ()
}
if (ndupacks and CW = 0)
{
Calculate the delay_Th-Val
Send new packet
Enter fast recovery ()
}

(a)This Graph represent congestion window with respect to time and congestion window peak value in graph 14200 and 5.8 sec represent the TCP congestion and downward line indicate recovery of packets which will improve the throughput of packets between nodes in wireless network**.**



(b)This graph represent scenario of congestion window between node o and node 5



IV. CONCLUSION

In this paper we have examined the effects on the end-client and framework performance when clients with high bit rates TCP administrations are traveling through the system. Specifically we have concentrated on the TCP exhibitions amid the LTE clogging procedure.To reach such a point we proposed a TCP blockage utilizing quick recuperation instrument the approval of this methodology has been finished by recreations performed with ns-3 to assess LTE execution.

The outcome demonstrated a decent assention examination recreation of the genuine conduct of the blockage methodology in LTE.

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