Department of Electronics and Communications Engineering

Study of AQM based congestion control in LTE network

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Letter of Transmittal

To

Anup Kumar Paul

Assistant Professor

Department of Electronics and Communication Engineering

East West University

Subject: Submission of Thesis Report as **(ETE-498)**

Dear Sir,

We are pleased to let you know that we have completed our project on ''Study of AQM based congestion control in LTE network ''. The attachment contains of the thesis that has been prepared for your evaluation and consideration. Working on this thesis has given us some new concepts. By applying those concepts, we have tried to make something innovative by using our theoretical knowledge which we have acquired since last four years from you and the other honorable faculty members of EWU. This project would be a great help for us in future.

We are very grateful to you for your guidance, which help us a lot to complete our project and acquire practical knowledge.

Thanking You.

Yours Sincerely

Abu Kamal Hasan

ID: 2013-1-53-023

Md. Foysal Khan

ID: 2011-1-55-019

Department of ECE

East West University

Acceptance

This Thesis paper is submitted to the Department of Electronics and Communication Engineering; East West University is submitted in partial fulfillment of the requirements for the degree of Bachelor of Science in Electronics & Telecommunications Engineering (ETE) under complete supervision.

…………………………….	………………………………….
DR. Anup Kumar Paul	DR. MofazzalHossain
Assistant Professor&Chairperson	Professor
Dept. of ECE	Dept. of ECE
East West University	East West University

Declaration

This is certified that the thesis is done by us under the course **Thesis (ETE-498**) ''Study of AQM based congestion control in LTE network '' has not submitted elsewhere for the requirement of any degree or any other purpose except for publication.

Abu Kamal Hasan (Badhon)

2013-1-53-023

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2011-1-55-019

Abstract

Wireless communication is the heart of all the modern communication system. This communication is mostly all about to wireless communications. In wireless communication, different modulation methods are playing the vital roles. With the advancement of the Long-Term Evolution (LTE) network and smart-phones, most of today's internet content is delivered via cellular links. Due to the nature of wireless signal propagation, the capacity of the last hop link can vary within a short period of time. In our simulations, we demonstrate that the performance of different schedulers can be enhanced via proper dropping function.

Table of content

CHAPTAR 1

Introduction

Bangladesh is a rapidly growth country in terms of Mobile Subscribers. Till June 2016, the number of mobile subscribers were **131.376** million [1] and the number is rapidly increasing. To increase the subscribers in Bangladesh, Mobile Network Operators are increasing their Coverage Planning for improving network infrastructure. Coverage planning is performed with a planning tool including a digital map with topography and orthography information. The model selection is done according to the planning parameters. The coverage prediction is based on the map and the model and the accuracy is dependent on those as well [2]. The wireless cellular network services are growing fast and are coupled with an ever increasing high bit rates and multimedia applications that inevitably translate into demands for better coverage and capacity of the networks to guarantee Quality of Services (QoS) to end users. The global wireless cellular subscriptions stand at 6.8 billion users in 2013 with global penetration rate of 96%; 128% penetration rate in developed countries and 89% in the developing countries [1]. According to ERRICSSON Mobility Report [2], total mobile subscriptions are expected to grow from 6.8 billion in first quarter, 2014 to 9.2 billion by the end of 2019. Also, in this report, global mobile broadband subscriptions are predicted to reach 7.6 billion by 2019 and will gain an increasing share of the total mobile subscriptions over time, with GSM/EDGE subscriptions-only represent the largest share of the mobile subscriptions.

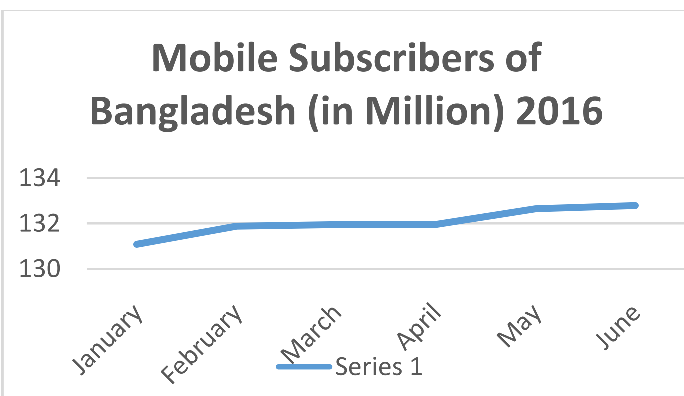

Fig. 1. Mobile Subscribers Growth in Bangladesh

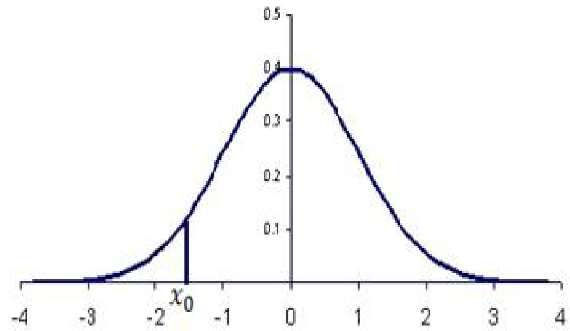

Fig. 2. Lower tail of the normal probability distribution for the location probability calculation

In this work, we have examined the impact of Active Queue Management (AQM) based congestion control furthermore, intra LTE handover on the execution of various Medium Access Control (MAC) schedulers with TCP activity by ns3 simulation.

The key necessities for the cutting edge remote system, cell administrators around the world give consistent Quality of Service (QoS) for joined versatile and mixed media administrations. This is conceivable because of the arrangement of 4G systems in light of the Long-Term Evolution (LTE) standard. The growing demand for network services, such as video streaming, video telephony,
Voice Over Internet Protocol (VOIP) with constraints on delay and bandwidth requirements, posesnew challenges for researchers in the design of the future generation cellular network. LTE and SystemArchitecture Evolution (SAE) were formulated by the 3rd Generation Partnership Project (3GPP) in2004 [2].LTE presents another air interface and radio access network, which gives substantially higher throughput and low inactivity, extraordinarily enhanced framework limit and scope contrasted with those of the Wideband Code Division Multiple Access (WCDMA) frameworks. These enhancements prompt expanded desires on the end-client Quality of Experience (QoE) over LTE when contrasted with existing 2G/3G frameworks [3]. Consequently, both research and mechanical groups are attempting in the investigation of LTE frameworks, proposing new and creative arrangements with a specific end goal to break down and move forward their execution.

To date, there has been much research on LTE, on the physical layer as well as on higher layers, e.g., the Packet Data Convergence Protocol (PDCP) and Radio Link Control (RLC) layer. The primary information cushions of the UI are situated at the RLC layer where information is held before its transmission to the goal User Equipment (UE). As a fast broadband system,[3] LTE guarantees continuous availability to the parcel information arrange. Blockage may occur at the radio interface because of different reasons. One such purpose behind blockage is correspondence quality in the channel that shifts with the separation of the UE from the base station (Evolved Node B, eNB) that leads eNB to choose a low piece rate regulation plan on account of poor correspondence channel quality what's more, may section the bundle. In this way, if the channel condition isn't adequate to transmit RLC Administration Data Unit (SDUs) in time, they will be in cradles of RLC elements sitting tight for the guideline for division or link from the Medium Access Control (MAC) layer.

Additionally, when a client is moving starting with one cell then onto the next cell, handover happens. Amid the handover, the source eNB advances the approaching information and the information that is as of now in the RLC cradle for the client to the objective eNB to ease the administration disturbance. Be that as it may, the sending of the client information may cause expanded deferral of sent information. In addition, there is a period interim quickly after the handover when the information on both the immediate way and the sent way may land in parallel to the target eNB [4]. Subsequently, there is a high likelihood of clog or flood in RLC cushions due to the vast volume of activity in a brief timeframe, prompting a high defer that outcomes in poor end-to-end application execution. To ensure high throughput and low defer when blockage happens, various investigations on Active Queue Management (AQM) based clog control conspires in the organize have been proposed previously, particularly for transport-layer conventions. Among the proposed AQM schemes, the Random Early Detection (RED) calculation is a standout amongst the most well-known calculations.

Many AQM schemes have been designed based on heuristic, control theoretic and optimization Approaches. Examples of heuristic schemes are RED , gentle RED (GRED) , adaptive RED (ARED) , parabola RED , hyperbola RED (HRED) , dynamic RED , flow RED , StabilizedRED, balanced RED , BLUE , Yellow etc. These algorithms aimed to improve some or all of the features, such as fairness, stability, network utilization, packet loss and adaptabilityfor different traffic loads. In GRED, the packet dropping probability from the maximum packetdropping probability (Pmax) to 1 is replaced by a gentle slope. The author suggested that, by doingso, the stability would increase. However, the main problem of GRED is parameter tuning. Differentparameters should be tuned according to different network conditions. ARED adaptively adjusts themaximum packet dropping probability Pmaxusing the multiplicative increase–multiplicative decreaseapproach. However, ARED is also sensitive to different parameter configuration and its performanceis not superior to RED when the network environment is complex. BLUE uses link idle events. And packet loss to model the congestion control scheme in a simple way and its packet loss is low.However, the long delay causes congestion in certain situations.

In the control theoretic approach, the fundamental concentration is to show TCP with RED progression. This approach at that point gives a steady and quicker reacting framework. In this approach, the principle push is to decipher the basic issues to logically decide different RED parameters.

Concerning illustration, a secondary speed broadband network, LTE guarantees an un-. Intruded on connectivity of the bundle information system. Blockage. Might occur on the radio interface because of Different motivations.

One such purpose behind blockage may be correspondence caliber done. Those channel that varies for the separation of the UE from the. Build station (eNB) and likewise different correspondence frameworks. Vary from one another clinched alongside powerful information rate that heads eNB. With select low spot rate regulation plan in the event about poor. Correspondence channel nature Furthermore might fragment those packets. Therefore, if the channel state is not sufficient to trans-. MIT RLC SDUs to time, they will make for buffers from claiming RLC substances. Holding up to those direction book for division or connecting. Starting with the Macintosh layer. Hence, there is a secondary likelihood for. Blockage or flood On RLC buffers because of those substantial volumes. Of movement over An brief time from claiming time, prompting An helter-skelter delay. That brings about poor end-to-end requisition execution [6].

To surety secondary throughput Also low delay The point when clogging. Occurs, various researches looking into animated Queue management. (AQM) built blockage control schemes in the system. Bring been suggested in the past, particularly for transport-layer. Conventions. "around those recommended AQM schemes, the irregular. Initial identification (RED) [5] calculation is a standout amongst those A large portion popular.

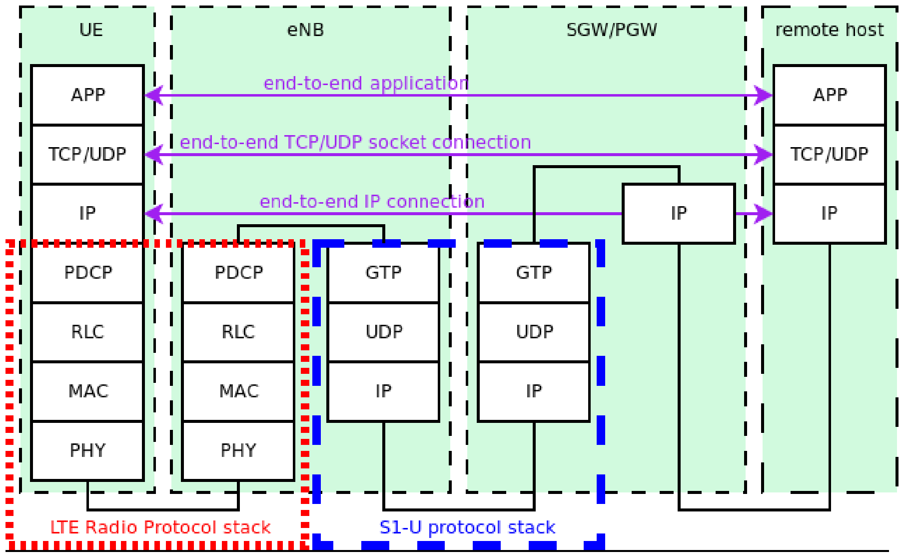

Fig-3. The LTE protocol stack (user plane)

The grade objective from claiming this paper will be will plan an straightforward Furthermore. Useful cushion administration plan for eNB RLC support that is unable on administer the queue measure of the predictable worth. Under an extensive variety of system loads. This paper proposes. An insignificant change with RED, which is known as advanced mobile red. (SmRED), On which those bundles dropping likelihood capacity. May be partitioned under two segments will recognize separate network Loads. The control law of SmRED camwood control those queue span. Of the predictable queue extent that accomplishes higher throughput. During low movement blockage and enhances end-to-end postponements toward. Helterskelter movement clogging without debasing those throughputs significantly.

As a result, SmRED rapidly adjusts the bundle dropping. Likelihood of the level about system load. Furthermore, will. Move keeping starting with red to SmRED over a genuine network, next to no. Worth of effort necessities should a chance to be carried Since just the bundle dropping. Likelihood profile may be balanced.

Chapter 2

Specifications

Now we are going to discuss briefly on the specifications of Mobile Quality of Service (QOS)Quality of Service or QoS in an important factor for Network infrastructure that enhances the mechanisms which include the performance control, stability, reliability, scalability, usability and customer satisfaction. Therefore not only it has to be standardized but also needs to be monitored whether necessary changes in the specifications are required. These specifications include the followings:

- Proper network planning with the provision of an effective further expansion.
- Stability and reliability of the network.
- Periodic monitoring and proper optimization of different network performance parameters.

Addition of different value-added services without hindering the network performanceproblems within the networks. This is done to improve the network performance.

Implementation of recommendation involves tuning the engineering antenna parameters such as the antenna tilt, and parameter auditing.

This followed by carrying out another drive test (i.e. Post swap drive test) exercise to ascertain the effect of the changes in the network parameters. The Post Swap drive test report will include the same reports as the Pre Swap Drive Test.

After each optimization exercise, the Key Performance Indicators (KPIs) obtained are analyze and checked against the desired threshold. See figure 1 for a summary of network optimization process.

In this paper, we specifically engaged a combination of essential KPIS such as dropped call rate, call setup success rate and outage call rate to examine overall QoS and GoS performance of the investigated GSM network. This in turn allowed us to determine the end user satisfaction rate and the general network performance. This exercise took place after some key problems such as poor transmission line performance, reduced power output, cell coverage degradation, coverage hole, over shooting, cell imbalance, poorly connected feeders, Increased interference and among others were identified across the studied GSM cell clusters, the following recommendations were made and implemented on the network to optimise its performances:

• Addition of missing adjacent cells in each cell cluster • antenna azimuth and tilt changes • BTS Equipment/Filter change • Re–tuning of interfered frequencies • Adjustment accessibility parameters • power parameters changes • Increase the reuse distance between the co-frequency and adjacent frequencies

Wireless network optimization

In the first place, Single site verification exercise enables one to verify the status of the base stations within the cluster to be optimized. Here, the individual sites are verified to make sure they are free of critical hardware problems before the optimization process is kick off. Also, in this exercise, some key engineering parameters like antenna height, antenna tilt, transmit power among others, are properly checked in each of the cell sites for errors and inconsistency with that in the site data obtained before the optimization exercise. This is followed by collecting and analyzing data from Pre Swap Drive Test carried out on the selected routes and also data from networks nodes by using customized software.

After collection, the log-file is analyzed to identify possible problems within the networks. This is done to improve the network performance.

Implementation of recommendation involves tuning the engineering antenna parameters such as the antenna tilt, and parameter auditing.

This followed by carrying out another drive test (i.e Post swap drive test) exercise to ascertain the effect of the changes in the network parameters. The Post Swap drive test report will include the same reports as the Pre Swap Drive Test.

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To meet the QoS requirements, one of the important features in the LTE system is multi-user scheduling. Multi-user scheduling is responsible for distributing available resources among active users. Both downlink and uplink packet schedulers are deployed at the eNB, and, since there is no inter-channel interference provided by OFDMA, they work with a granularity of one TTI and one RB in the time and frequency domain, respectively. Resource allocation for each UE is usually done based on the comparison of per-RB metrics. These metrics can provide the status of transmission queues, channel quality, resource allocation history, buffer state and QoS requirements. The main differences among resource allocation strategies exist due to the trade-off between computational complexity and decision optimality.

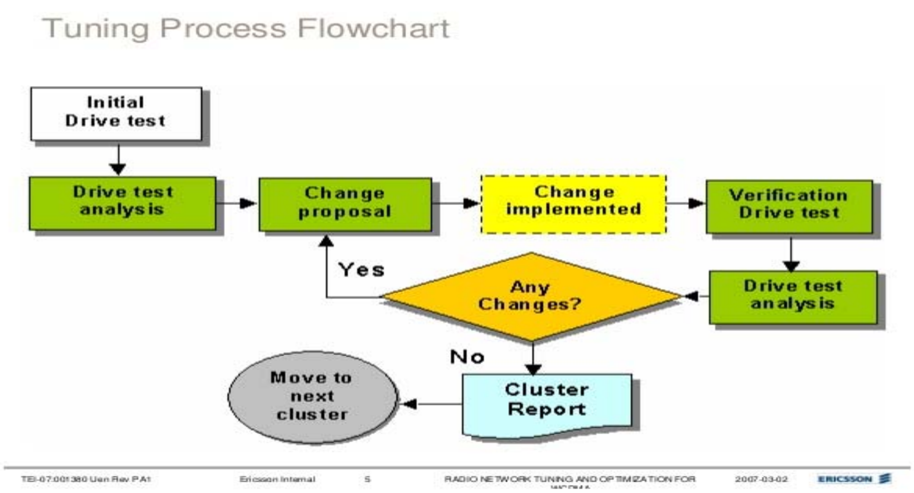

Figure 4: Network Optimization flowchart.

CHAPTAR 3

Long-Term Evolution (LTE)

Long-Term Evolution (LTE) is a standard for high speed wireless communication for cell phones and data terminals which based on the GSM/EDGE and UMTS/HSPA technologies in telecommunication. It increases the speed and capacity using different radio interference together instead of core network improvements.In telecommunication, **Long-Term Evolution** (**LTE**) is a standard for high-speed wireless communication for mobile devices and data terminals, based on the GSM/EDGE and UMTS/HSPA technologies. It increases the capacity and speed using a different radio interface together with core network improvements. The standard is developed by the 3GPP (3rd Generation Partnership Project) and is specified in its Release 8 document series, with minor enhancements described in Release 9. LTE is the upgrade path for carriers with both GSM/UMTS networks and CDMA2000 networks. LTE is the upgrade path for carriers with both GSM networks and CDMA2000 networks. The different LTE frequencies and bands used in different countries. It means that only multi-band phones are able to use LTE in all countries where it is supported.

3GPP engineers named the technology ‘’Long Term Evolution’’, it represents the next step (4G) in a progression from GSM, a 2G standard, to UMTS, the 3G technologies based on GSM.

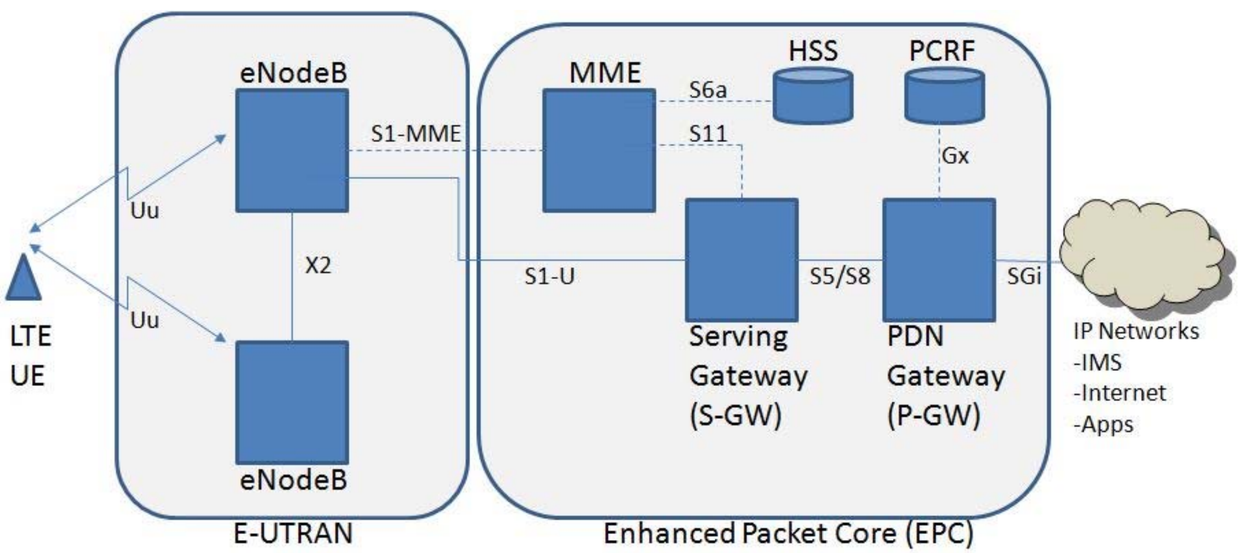

Fig- 05: LTE architecture

Using of LTE:

LTE uses OFDM (Orthogonal Frequency Division Multiplexing). The upper layer of LTE is based upon TCP/IP, which will likely result in an all- IP network similar to the current state of wired communications.

LTE will support mixed data, voice, video and messaging traffic. The higher signal to noise ratio (SNR) at the receiver enabled by MIMO, along with OFDM. It provides improved coverage especially in dense urban areas.

These networks will complete with WiMax for both enterprise and consumer broadband wireless customers. Outside of the US telecommunications market, GSM is the dominant mobile standard, with more than 80% of the world's mobile phone users.

LTE changes the current method of moving data to an internet protocol system. Rather than move small amounts of data, as both CDMA and GSM do, it will move large packets of data and streamline the service. Think of it as taking the speedbumps off the roads in your city so you can zoom around faster.The major benefit to LTE is that in reduces the latency in data transfer. GSM uses technology called time delay duplex (TDD), while CDMA uses code division duplex (CDD). Both are a method of coding information for travel across airwaves. The advent of CDD proved to be faster, but the world operates on GSM technology. As such, GSM was improved to HSPA, or high-speed packet access. Like LTE, it moves larger packets of information at a faster rate.LTE incorporates digital signal processing (DSP) to better adjudicate the data packet transfer. In a nutshell, LTE is a supercharger for your GSM or CDMA car which is zipping around a speedbump-less city.

LTE is an upgrade to the current systems in place, we also know it works for both GSM and CDMA. In that respect, everyone is available to use it, and it is widely believed to be a worldwide standard at some point. As LTE is a system upgrade, it's left up to frequencies and spectrum to carry the day.A true LTE phone will operate on a variety of frequencies. Across the globe, different countries operate on different frequencies. A higher frequency does not denote a better network, either, as a lower frequency is more useful in rural areas. The best way to understand this is to examine countries in Europe. While we here in the US have been fortunate to have exposure to LTE for some time now, Europe is just starting their LTE implementation.LTE typically operates on a frequency spectrum of 700MHz to 2.6GHz. A lower spectrum like the 700MHz will carry a signal over a larger area, thus reaching a larger number of people with less infrastructure change the carrier has to make. Something higher like the 2.6GHz will be much faster, but reach less people.

LTE is where we are moving. LTE is not, however, where we currently reside. Every carrier is moving toward LTE as our data needs increase. Even though Verizon here in the US brags about their LTE network, a closer look will put things into perspective a bit. A favorite source of mine, open signal map, shows all manner of coverage in the world's major markets. A closer look at the LTE situation in Chicago shows the signal to be pretty strong centrally, but non-existent outside the confines of the windy city.

LTE is commonly marketed as 4G LTE, but it does not meet the technical criteria of a 4G wireless service. The most significant advancement is that Wimax, Evolved High Speed Packet Access and LTE bring to the original 3G technologies. To differentiate LTE advancement and WiMAX Advanced from current 4G technologies, ITU has defined them ''true 4G''.

Chapter 4

Congestion control in LTE:

Congestion, in the context of networks, refers to a network state where a node or link carries so much data that it may deteriorate network service quality, resulting in queuing delay, frame or data packet loss and the blocking of new connections. In a congested network, response time slows with reduced network throughput. Congestion occurs when bandwidth is insufficient and network data traffic exceeds capacity. Congestion control adjusts movement passage into a broadcast communications organize keeping in mind the end goal to dodge congestive fall coming about because of oversubscription. This is ordinarily refined by decreasing the rate of bundles and it ought not be mistaken for stream control, which keeps the sender from overpowering the collector.

Congestion Control in the Internet network is divided into Congestion Control at the end system andCongestion Control at the network side [8]. Congestion Control at the end system mainly rely on variousVersions of TCP [2~4] protocol, while Congestion Control at the network side is achieved by carrying onactive queue management [5] in buffer queue of router, dropping packets predictably before the formation of a full queue, avoiding deadlock, full queue and global synchronization caused by packet loss, type of Drop-Tai. Active queue management mechanism can be divided into two categories. One based on real queue and the other based on virtual queue. Active queue management mechanism based on real queue contains REM, RED, PI control. Based on virtual queue and corresponding to real queue, queue management has a rate less than real queue. Packet drop or mark is based on whether the virtual queue is full or not. The congestion control mechanism combined by virtual queue and real queue. Through simulation, there comes the conclusion that dealing with burst data's robustness, queuing delay and jitter, active queue management mechanism based on virtual queue has the original active queue management mechanism.

Classification:

Open-loop control
· Mainly used in circuit
switched network (GMPLS)

Closed-loop control
· Mainly used in packet switched network
· Use feedback information: global & local

Implicit feedback control
· End-to-end congestion control
· Examples:
TCP Tahoe, TCP Reno, TCP Vegas, etc.
Explicit feedback control
· Network-assisted congestion control
· Examples:
IBM SNA, DECbit, ATM ABR, ICMP
source quench, RED, ECN

Open Loop Congestion Control

• In this method, policies are used to prevent the congestion before it happens.

• Congestion control is handled either by the source or by the destination.

Closed Loop Congestion Control

• Closed loop congestion control mechanisms try to remove the congestion after it happens.

LTE Protocol Structure (User Plane)

The user plane protocol structure of LTE is shown in Fig. 1. The RLC entity is specified in the 3GPP technical specification [4], and comprises three different types of RLC modes: TM, UM, and AM. The RLC entities provide the RLC service interface to the upper PDCP layer and the MAC service interface to the lower MAC layer. Among the three modes, TM RLC is configured to the Radio Resource Control (RRC) messages that do not need RLC configuration. Error-sensitive and delay-tolerant non real-time applications are provided by

AM RLC. On the other hand, UM RLC is for delay-sensitive and error-tolerant real-time applications such as VoIP

The AM RLC entity manages 3 buffers:

• Transmission Buffer: it is the RLC SDU queue. When the AM RLC entity receives a SDU from the upper PDCP entity, it enqueues it in the transmission buffer.

• Transmitted Protocol Data Units (PDUs) Buffer: it isthe queue of transmitted RLC PDUs for which an ACK/NACK has not been received yet. When the AM RLC entity sends a PDU to the MAC entity, it alsoputs a copy of the transmitted PDU in the transmitted PDUs buffer.

• Retransmission Buffer: it is the queue of RLC PDUs that are considered for retransmission (i.e., they have been NACKed). The AM RLC entity moves this PDU to the retransmission buffer, when it retransmits a PDU from the transmission buffer.

Chapter 5

Active Queue Management (AQM)

Active queue management (**AQM**) is the intelligent drop of network packets inside a buffer associated with a network inference controller (NIC), when that buffer becomes full or gets close to becoming full, often with the larger goal of reducing network congestion. This task is performed by the network scheduler, which for this purpose uses various algorithms such as random early detection (RED), explicit congestion notification (ECN), or controlled delay .RFC 7567 recommends active queue management as a best practice.

The benefits of AQM is drop-tail queues have a tendency to penalize bursty flows. By dropping packets probabilistically, AQM disciplines typically avoid both of these issues. By providing endpoints with congestion indication before the queue is full, AQM disciplines are able to maintain a shorter queue length than drop-tail queues, which combats bufferbloat and reduces network latency. A congestion control plot in light of AQM has progressed toward becoming an exploration problem area in the business, and the AQM system is prescribed on Internet switches to accomplish the accompanying objectives: overseeing line lengths to assimilate here and now clog (e.g., blasts), giving a lower intelligent deferral, and keeping away from worldwide synchronization [6]. Among the AQM instruments, the most run of the mill conspire is RED proposed by Floyd and Jacobson. The RED calculation works by distinguishing beginning congestion and informing the Transmission Control Protocol (TCP) by probabilistically dropping bundles previously the line tops off. Quickly, the calculation works by keeping up a normal line estimate. As the normal line estimate fluctuates between the base and most extreme limits, the bundle dropping likelihood straightly changes in the vicinity of zero and maximum drop probability Pmax. Thus, the packet dropping probability function is linear to the change of the average queue size. If the average queue size exceeds the maximum threshold, all arriving packets are dropped. Since the packet dropping mechanism is based on the moving average algorithm, RED can control the transient congestion by absorbing arrival rate fluctuations. Although RED is a significant improvement over simpleDrop tail that simply drops all incoming packets when a queue is full, RED is particularly sensitive to the traffic load and the parameters of the scheme itself.

It has been demonstrated that the execution of the RED calculation on evasion clog is incredible . Be that as it may, the RED calculation is utilized in TCP/IP layers, which is altogether different from the RLC layer. RED is utilized with TCP window control. Be that as it may, delay-touchy information streams transmit by the User Datagram Protocol (UDP), which does not have the window control system. Hence, for delay-delicate information streams, a cross layer approach is important to control the blockage in RLC in view of AQM systems.

Congestion control schema

The objective of an appropriate AQM plot is to keep up the normal line estimate between least edge and most extreme limit of the line at low motions and thusly help maintain a strategic distance from constrained drops. It is unseemly that the first RED parcel dropping likelihood and the normal line measure are straightly related. It has been found in that the connection transfer speed is not completely used with a little normal deferral in the low-stack situation; in this way, a littler parcel dropping likelihood ought to be utilized as a part of request to enhance the connection use. The linkbandwidth is fully utilized with a large average delay in the high-load scenario; thus, a larger packet dropping probability should be used in order to reduce the average delay.

When the average queue length is close to the minimum threshold, which indicates that the network congestion is not very serious, the packet dropping probability should be smaller than RED so that there will be fewer numbers of packets to be dropped to improve the utilization of the network. In the
same way, the throughput could be enhanced. Meanwhile, the smaller the average queue length, the smaller the change rate of the packet dropping probability should be to ensure all the packet dropping probability is lower than RED in this stage. In other words, the slope of the packet drops probability curveincreases slowly with the average queue length in this stage.[7}

Oppositely, when the average line length is near the maximum threshold, which shows that the system
blockage is intense, the parcel dropping likelihood ought to be bigger than RED to guarantee that the line does not flood. Therefore, less bundles will confront constrained drops what's more, retransmissions, and this is the manner by which the normal deferral can be diminished, that progressions with the normal line length. Additionally, the larger the average queue length, the greater
the change rate of the packet dropping probability should be in this stage to avoid serious congestion earlier. In other words, the slope of the packet dropping probability curve increases moderately with the average queue length in this stage.

RED’s original packet dropping probability can be defined as
Pd= Pmax× avg− Minth
Maxth– Minth(1)

Where,
*Pd*is the packet dropping probability
*Pmax*is the maximum packet dropping probability
*avg*is the average queue length
Minth is the minimum threshold

We first decided one target value below which we treat the traffic volume as low and above the target the traffic volume is high. We define the target value as
Target = *Minth*+ *Maxth*– *Minth* / 2 (2)

If the *avg*is below the target value, we set the packet dropping probability as
Pd= Pmax× (avg– Minth /Maxth–Minth)^2(3)

On the other hand, when the traffic volume starts getting high, i.e., if the *avg*is between *Target* and *Maxth*, then the packet dropping probability function is defined as

$Pd = Pmax \times \sqrt{}$(*avg– Minth / Maxth– Minth)* *(4)*

Network Topology and System Parameters:

In this section, we will describe the network model that we used for simulation in ns3, system parameters and provide a brief description of different scheduling algorithms. In order to systematically explore the interactions between the AQM-based buffer management and scheduling algorithm, we have to delve deeper to consider mobile network elements and protocols. We implemented our proposed strategy using the LTE module of the ns3 [9] simulator configured with the topology depicted in Figure 6 for single-cell and Figure 7 for the handover scenario in multi-cell topology. We used the LTE/EPC Network Simulator and Analysis (LENA) module to create an end-to-end LTE network. The LENA module has all the major elements of a real LTE system including the Evolved Packet Core (EPC) and air interface Evolved UMTS Terrestrial Radio Access (E-UTRA). The LTE model in ns3 provides a detailed implementation of various aspects of the LTE standard such as adaptive modulation and coding, Orthogonal Frequency Division Multiple Access (OFDMA), hybrid Automatic Repeat Request (ARQ) etc. The ns3 implementation follows the detailedspecification of 3GPP LTE and various versions of TCP. Hence, the results obtained in the simulationcan be representative of what happens in a real system.

A good AQM algorithm should require very low computational cost and be easily implemented in a real network. SmRED-4, a smart congestion control mechanism based on RED, is aimed at solving link under-utilization and large delay problems in low and high traffic-load scenarios for eNB RLC in the LTE network. Furthermore, the migration from RED to SmRED-4 in a real network needs very little work because of its simplicity (only the packet dropping profile is modified). To achieve gains in user application performance, an optimal RLC buffer occupancy at eNB is expected. SmRED-4 effectively improves the drawbacks of RED, and achieves a better balance between higher throughput and lower delay.

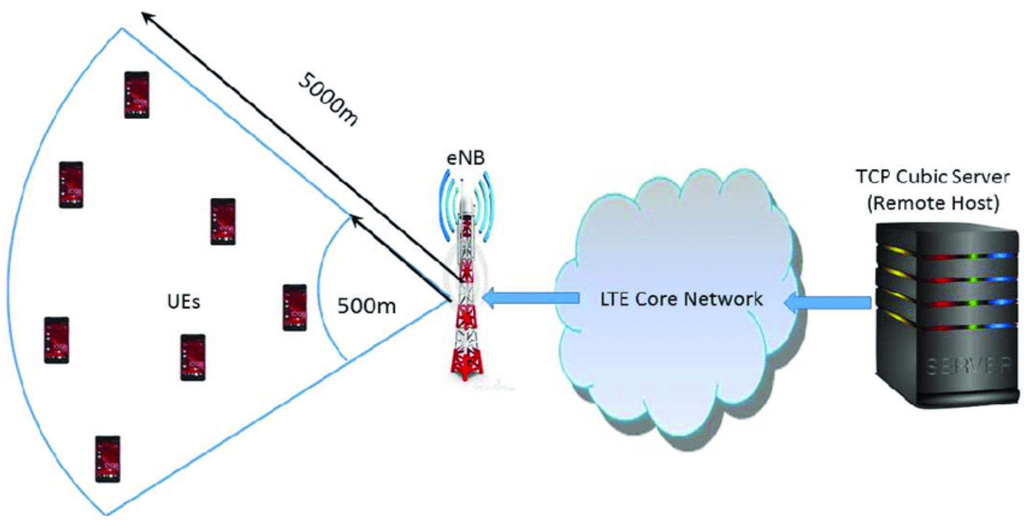

Fig-6 :Single-cell LTE network topology

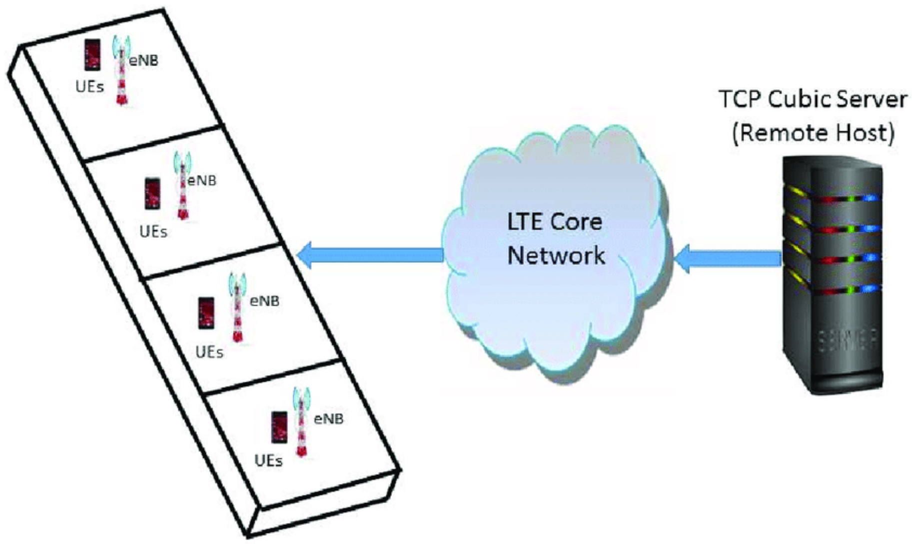

Fig- 7: Multi-cell LTE network topology

The guaranteed bit rate video traffic is simulated, the remote host on the right side acts as sources,and the UEs on the left side act as sinks. TCP data senders are of the TCP-Cubic type. We usedTCP-Cubic as it is the default TCP congestion control algorithm in Linux OS in real networks. The TCPpacket sizes are 1000 bytes. We also used the Single Input Single Output (SISO) transmission mode forboth UE and the eNB. The remote host is connected to the LTE core network via a wired link with thelink capacity of 10 Mbps (1250 packets per second) and the propagation delay on this link is 50 ms.the application data rate is 100 Mbps.The thresholds for the packet dropping function are set asMinth = 20 packets and Maxth = 3_ Minth packets and the maximum packet dropping probability isset as 0.1. The eNB RLC buffer size is 100 packets. The total simulation time is 75 s. Table 1shows the details of some of the important simulation parameters and their values.

Simulation Parameter	Value
Radio Link Control (RLC) Buffer Size	100-200
Number of Traffic Source	2-10
Pkt Size	1000 Bytes
Number of Resource Block	15
Application Data Rate	100 Mbps
Wired Link Capacity	10 Mbps
Wired Link Delay	50 ms
eNB Transmission Mode	SISO

Transmission Control Protocol (TCP) Traffic Type	Cubic
Handover Algorithm	A3Rsrp
Mobility	Random Walk 2D
Node Movement Speed	20 m/s
Total Simulation Time	100 s
Application Start Time	From 0.1 s
Application Stop Time	100 s
Simulation Area	Single and multi cell with 5 Km Radius

Chapter 6

SIMULATION RESULT AND PERFORMANCE EVALUATION:

In order to systematically explore the interactions between the mobile network architecture and TCP, we have to delve deeper to consider mobile network elements and protocols. We implemented our proposed strategy using the LTE module of the ns3 [9] simulator configured with topology depicted in Fig. 3. We used the LENA module [10] to create an end-to-end LTE network which has all the major elements of a real LTE system including the air interface Evolved UMTS Terrestrial Radio Access (E-UTRA) and Evolved Packet Core (EPC). The LTE model in ns3 gives an in depth implementation of various factors of the LTE preferred which include OFDMA, hybrid ARQ, and adaptive modulation and coding. The ns3 implementation follows detailed specification of TCP and 3GPP LTE. Hence, the results provided should be representative of what happens in a real system.

With the topology in Fig. 3, assured Bit price (GBR) video traffic are simulated, the far flung host at the right side acts as assets, the UEs on the left side act as sinks. TCP data senders are of the TCP-Cubic type. We used TCP-Cubic as it is the default TCP congestion control algorithm in Linux OS in real networks. The TCP packet sizes are 1000 bytes. We used the SISO transmission mode for both UE and the eNB. The remote host is connected to the LTE core network via wired link with the link capacity of 10 Mbps (1250 packets per second) and the propagation delay on this link is 50 ms. The application data rate is 100 Mbps.The thresholds for the packet dropping function are set as Min = 20 packets and Max =3× Min packets and the maximum packet dropping probability is set as 0.1. The eNB RLC buffer size is 100 packets. The total simulation time is 100 seconds.

To validate the improvement achieved by SmRED, we compared the performance of SmRED with Droptail and RED with the topology as shown in Fig. 3. We varied the number of UEs from 2 to 10 to simulate different traffic load conditions. The UEs are distributed randomly in the cell within a distance of 500 m to5000 m from the eNB so that the UEs face different channel conditions. All UEs downloaded GBR Video traffic from a single server. We compared the download performance in terms of end-to-end average throughput and end-to-end average delay. The results are shown in Fig. 4 and Fig. 5. . Thefiguresshowthattheend-to-endaveragethroughput decreases with the increase in traffic load. With low traffic load conditions, such as with 2 UEs, Droptail outperforms RED and SmRED in terms of throughput but at the cost of a significantly higher end-to-end average delay. Because Droptail has no mechanism to inform the TCP source about the possible congestion, the TCP window size grows significantly until the queue in the eNB RLC becomes full and continuous drop occurs. Therefore, it takes more time to resend those lost packets that incurs larger end-to-end average delay. SmRED outperforms RED in terms of throughput and delay.

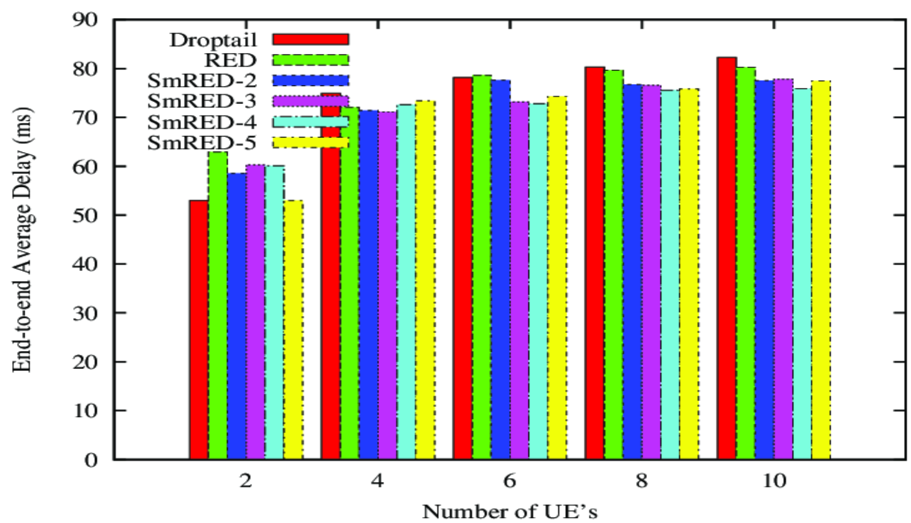

Fig. 5. End-to-end average throughput with increasing traffic load.

When the traffic load is low, SmRED's throughput is higher than RED's throughput but as the traffic load increases, the end-to-end throughput of SmRED becomes almost the same as that of RED. This is because, when the traffic load is low, SmRED's dropping probability is low and thus fewer packets are dropped. So, the TCP window size does not shrink too frequently which helps to achieve higher throughput than that of RED. On the other hand, the end-to-end delay is almost the same as that of RED's end-to-end average delay when the traffic load is low, but as the traffic load increases, SmRED outperforms RED. This is because SmRED has a higher dropping probability than RED when the traffic load is high, which helps to inform the source earlier in advance than RED about the possible congestion. Thus, the TCP source reduces the window size in advance to avoid resending lost packets, which helps to achieve lower end-to-end average delay at the cost of lower throughput. Figure 6 shows the performance of SmRED and RED in terms of forced drops i.e., packets drop due to full queue. The figure shows that, when the traffic load is low, RED's forced drop is lower than SmRED but as the traffic load increases, SmRED significantly reduces the forced drop compared to that of RED, which helps to achieve lower end-to-end average delay.

Lower end-to-end delay is the most important criterion for delay-sensitive streams. But as we know, AQM algorithms are used to cooperate with the TCP source to manage congestion. However, delay-sensitive streams are always with the UDP protocol that does not cooperate with the TCP window control mechanism. Therefore, to effectively utilize our idea with delay-sensitive streams, cross layer information is necessary to control the congestion in the eNB RLC buffer. The LTE protocol stack in the PDCP layer is upon the RLC layer. The PDCP layer utilizes a discard timer todiscard packets [16]. When a packet enters from the higher layer, a maximum limit is imposed on the waiting time of a packet inside the queue. Packets are time stamped upon their arrival to the PDCP layer. When the discard timer expires, the packet is discarded. Thus when the RLC entity detects the congestion using SmRED, it will inform the PDCP layer to decrease the value of the discard timer and vice versa when the traffic load is low. But, based on congestion notifications from the RLC layer, how much the discard timer should be decreased depends on the QoS/QoE of particular applications and is a matter for further research. Due to the lack of space, we will address this topic in our future studies.

CHAPTAR 7

Conclusion:

This paper presented a performance evaluation of different scheduling algorithms in the presenceof AQM-based congestion control with handover. From a maximum utilization of the channel capacitypoint of view, the best solution is to allocate the RB to those users that experience good channelcondition. However, doing so will lead to unfair resource allocation to other users. Thus, providingequal service to all users, i.e., fairness, QoS provisioning, computational complexity, and energy savings,can be achieved at the cost of lower cell capacity. According to this requirement, the implementationdesign of a RB allocation strategy is a trade-off among the goals that the network operator wants toachieve and the spectral efficiency.

On the other hand, a good AQM algorithm should require very low computational cost and be easily implemented in a real network. SmRED-4, a smart congestion control mechanism based onRED, is aimed at solving link under-utilization and large delay problems in low and high traffic-loadscenarios for eNB RLC in the LTE network. Furthermore, the migration from RED to SmRED-4 ina real network needs very little work because of its simplicity (only the packet dropping profile ismodified). To achieve gains in user application performance, an optimal RLC buffer occupancy ateNB is expected. SmRED-4 effectively improves the drawbacks of RED, and achieves a better balancebetween higher throughput and lower delay. Thus, a good combination of the AQM algorithm withthe scheduling techniques can provide optimal performance to network operators.

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