



## **Study on AQM Based Buffer Management in LTE Network**

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Dept. of Electronics & Communications Engineering, East West University  
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Engineering**

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# DECLARATION OF AUTHORSHIP

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## ABSTRACT

This study is emerged on AQM based buffer management of LTE system where we have observed the performance of multiple Medium Access Control (MAC) schedulers with TCP traffic with the help of ns3 simulation. We tried to carry out the knowledge of the effect of Active Queue Management (AQM) based congestion control and intra LTE handover. The extended idea of Smart RED (SmRED) to SmRED-i, where packet dropping probability function is different in accordance with the value of  $i = 2, 3, 4 \dots$  which is proposed by Effect of AQM-Based RLC Buffer Management on the eNB Scheduling Algorithm in LTE Network has been discussed on this paper.

The influence of this method has been studied previously. Herein, a more efficient manner to manage interference has been represented. Based on the results obtained in this report through different ns3 simulations, the options presented in this performance of different schedulers can be enhanced via proper dropping function which can be seen as positive solutions.

## ACKNOWLEDGEMENTS

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# ABBREVIATIONS

- LTE** – Long Term Evolution
- TCP** – Transmission Control Protocol
- QoS** – Quality of Service
- QoE**- Quality of Experience
- AQM** – Active Queue Management
- MAC** – Medium Access Control
- RLC** – Radio Link Control
- eNB** - E-UTRAN Node B
- RED** – Random Early Detection
- SmRED**- Smart Random Early Detection
- VOIP** – Voice Over Internet Protocol
- SAE** – System Architecture Evolution
- 3GPP**- 3rd Generation Partnership Project
- UMT** - Universal Mobile Telecommunications Service
- PDCP** – Packet Data Convergence Protocol
- WCDMA** - Wideband Code Division Multiple Access
- UE** – User Equipment
- SDU** – Service Data Unit
- PI** – Proportional Integral
- PD** – Proportional Derivative
- IMS**- IP Multimedia Subsystem
- HSS** – Home Subscribe Server
- EPC** -Evolved Packet Core
- MME**- Mobility Management Entity

**SGW** – Servicing Gateway

**PWG** – Packet Data Network Gateway

**RRM**- Radio Resource Management

**OFDM** - Orthogonal frequency-division multiplexing

**OFDMA**- Orthogonal frequency-division multiple Access

**TTI** – Transmission Time Interval

**ARQ**- Automatic Repeat Request

**MT**- Maximum Throughput

**BET**- Blind Equal Throughput

**PF**- Proportional Fair

**TTA**- Throughput to Average

**RR**- Round Robin

**PSS**- Priority Set Scheduler

**TBFQ**- Token Band Fair Queue

**GBR** – Guaranteed Bit Rate

**HO** – Hand Over

**RSRP** -Reference Signal Received Power



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# CHAPTER 1: INTRODUCTION

## 1 Introduction

This chapter will represent a brief introduction to the research area along with an explanation of the problems addressed in this thesis.

### 1.1 Motivation of the Study

As the number of mobile broadband subscriptions grow at a massive rate since the last two decades. With the evolution of new technologies devices, smartphones are now capable of displaying high quality videos or real time video, which will definitely put high efforts on cellular networks capacity. To meet the mobile traffic requirement, mobile operators are setting up a new generation mobile network. In recent years, the Third Generation Partnership Project (3GPP) was established a long time ago to establish Long Term Evolution (LTE). However, even with the increase in LTE growth, it is not able to manage the growing needs for increasing power and increasing data throughput. This phenomenon is increasing in densely populated areas where high volume users want to connect to the base station at the same time and the system may exceed the power limit. The traditional architecture of the macro cell network is not sufficient for that environment.

Further, to meet the requirements of the next generation of wireless networks, global cellular operators offer flat-quality (QoS) services to combine mobile and multimedia services. This is possible due to the establishment of a 4G network based on the quality of long-term evolution (LTE). With the blooming of increasing demand, delays and bandwidth requirements in network services such as video transmission, video telephony, and voice over Internet protocol (VoIP), researchers create new challenges in the designs of cellular networks of the future generation. LTE and System Architecture Evolution were prepared in 2004 by the Third Generation Partnership Project (3GPP).

LTE established a new air interface and a radio access network that provides much higher performance and less trending than the broadband code division

system (WCDMA), which boost the power and coverage of the system. These improvements lead to higher expectations in the LTE end-user quality (QoE) experience compared to existing 2G / 3G systems. Due to this, both research and industry propose a significant effort in the study of LTE systems, offering new and innovative solutions to analyze and enhance their performance.

## 1.2 Problem Statement

With the development of long-term evolution (LTE) networks and smartphones, most of the current Internet content is distributed through cellular links. Due to the nature of the radio signal transmission, the capacity of the last hop link can vary in a short time. Unfortunately, the Transmission Control Protocol (TCP) does not work well in such situations, it may be possible to have low quality services (QoS) for the end user (for example, entry and delay at the end). In this paper, we have studied the impact of congestion control based on active queue management (AQM) and the transfer of LTE interterm related to the performance of several medium access control (MAC) schedules, including TCP traffic by NC simulation 3. The EnB Radio Link Control (RLC) buffer in the LTE network avoids linking an appropriate AQUIM design with strong links to forced drops and link layers as well as network traffic loads. First we show that the original random initial detection (red) linear fall function cannot handle several traffic load situations well. Then, we establish a heuristic method with a parameter to define different non-linear functions. In our simulations, we exhibit that the performance of different schedules can be improved with the appropriate elimination function.

The following activities are identified as the main points covered by this investigation:

we have studied the extended idea Smart RED (SmRED) to SmRED-i, where packet dropping probability function is different in accordance with the value of  $i = 2, 3, 4 \dots$  which is proposed in [1]

We tried to overview the effect of AQM-based buffer management on the performance of different scheduling algorithms with and without handover. The performance we measured in terms of end-to-end average Transmission Control Protocol (TCP) throughput and delay. While there is significant work on the performance of different schedulers in the absence of AQM-based buffer

management and handover, it is very hard to capture the effect of both on the scheduling algorithms using analytical models.

We worked on particular and extensive ns3 realistic simulation to simulate the effect of AQM-based buffer management on the scheduling algorithms in the single cell network topology without handover and in the multi-cell network topology in the presence of handover in LTE networks across a wide range of traffic-loads.

## CHAPTER 2: LTE OVERVIEW

# 2 An Overview of LTE

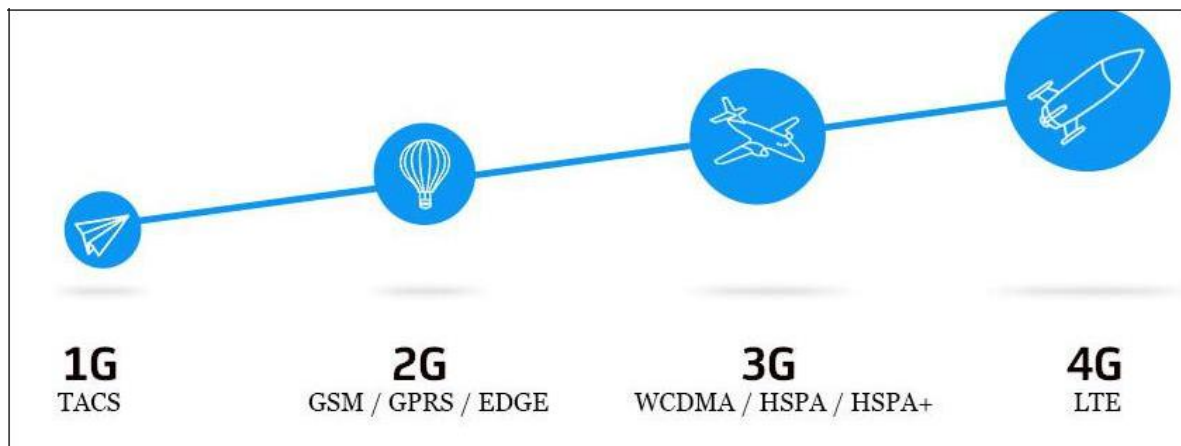
This chapter will briefly introduce the theoretical and background knowledge about LTE network.

LTE's vision of wireless access will result in a comprehensive transition towards a packet-switched-only system which makes wide use of Internet Engineering Task Force (IETF) protocols and practices. LTE is further designed to be practical with legacy UMTS systems and offer support for seamless mobility through non-3GPP wireless accesses including, but not limited to, WiMAX, 1x-EVDO, and Wi-Fi.

The LTE access network incorporates state-of-the-art air interface technologies including OFDMA (Orthogonal Frequency Division Multiple Access) and advanced antenna techniques to maximize the efficient use of RF spectrum. It also accommodates several options for frequency bands, carrier bandwidths, and duplexing techniques to effectively utilize the different portions of unused spectrum in different countries and geographies.

Most significantly, the LTE network architecture's evolution to an all-IP architecture enables seamless delivery of applications and services over what were previously two separate and distinct networks. In addition to reducing deployment and operational costs and complexity, the transition to IP enables LTE to support Quality of Service (QoS) for real time packet data services like VoIP and live video streaming. [2]

3<sup>rd</sup> generation of standards and technologies for mobile communication is based on the standards of International Mobile Telecommunications-2000 (IMT-2000) specified by the International Telecommunication Union (ITU). The WCDMA is specified in 3GPP which promises 2 Mbps peak data rate for downlink transmission and 384 kbps for uplink. Next phase of WCDMA is High-Speed Downlink Packet Access (HSDPA), which offers up to 14.4 Mbps for downlink. The 3G evolution is presented in Figure 2.1



*Figure 2:1 4G-Evolution*

3GPP also specifies a new core network evolution, which is called System Architecture evolution (SAE). The main purpose of LTE and SAE is to improve the UMTS system for the future. The aim of LTE/SAE is to ensure mobile networks with higher data rate, spectrum efficiency, lower latency, etc. In order to fulfill these requirements, some new techniques are introduced in LTE [3]

## 2.1 LTE design requirements

### 2.1.1 Capabilities-related requirements

- The instantaneous downlink peak data rate should be at least 100 Mbps within 20 MHz spectrum allocation and with two receiver antennas at the UE.
- The instantaneous uplink peak data rate should be at least 50 Mbps within 20 MHz spectrum allocation and with one transmit antenna at the UE
- The control-plane latency requirement has two measures, one is the transition time from a camped-state to active state, where the requirement is 100 ms; the other measure is the transition time from a dormant state to active state, where the requirement is 50 ms.

### **2.1.2 Deployment-related requirements**

- LTE system can be deployed both standalone and integrating with an existing WCDMA/HSPA and/or Global System for Mobile communications (GSM) network.
- LTE-based radio access can be deployed in both paired and unpaired spectrum allocation. Thus, LTE should support both Frequency Division Duplex (FDD) and Time Division Duplex (TDD) [4]

### **2.1.3 Requirements for architecture and migration to LTE**

- LTE architecture should simplify and minimize the introduced number of interfaces; thus, a single E-UTRAN architecture is required.
- LTE should be designed to minimize the delay variation for e.g. TCP/IP packet communication.

### **2.1.4 Radio Resource Management requirements**

- LTE should provide Enhanced support for end-to-end Quality of Service (QoS). This requires improved supporting for various applications, services and protocols etc. And should provide mechanisms to support operation and transmission of higher layer protocols over radio interface e.g. IP header compression.

Overall, the LTE system requirements are based on System Capability (Peak Data Rates and Latency); System Performance (Throughput, Spectrum Efficiency, Mobility, Coverage and Enhanced Multimedia Broadcast/Multicast Services, eM-BMS); System Spectrum Allocation; System Architecture and Cost Reductions. [5]



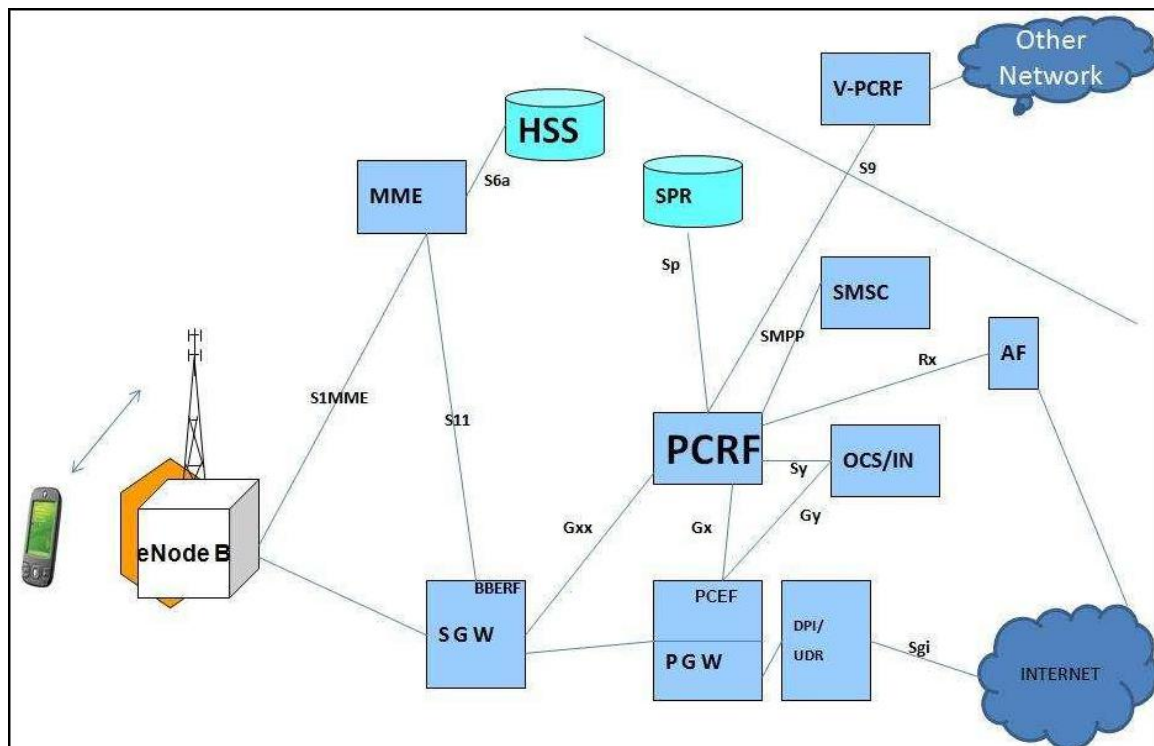
## 2.2 Overall LTE Architectural Overview

The radio access basically is the evolution of the LTE Physical Layer, while the non-radio access grouped under the System Architecture Evolution (SAE), is the evolution of the network architecture of the LTE. The major components of the LTE System

Architectures are:

1. User Equipment (UE)
2. Radio Access Network (RAN)
3. Evolved Packet Core (EPC)

The Evolved Packet System (EPS) is comprised of the LTE Radio Access Network and Evolved Packet Core (EPC⇒RAN + EPS). At the high level, the LTE network is composed of the Core Network (CN), also called the EPC while there is also the Access Network, which is referred to as E-UTRAN.



*Figure 2:2 The EPS showing Network Elements and Standardized Interfaces*

## 2.3 Core Network

The Core Network (also known as EPC) does the overall control of the UE and establishes the bearers. The Core Network has a number of different logic nodes, some of which are:

**MME:** The MME is the main control node in the EPC. The control plane information coming from the eNodeB is mainly routed to the MME. One of the most essential functions of the MME is that it handles the signaling between the UE and the CN. Also, it handles the issue of security and authentication for keys offering; in addition to mobility management - where the MME does management functions by making request setup and release of appropriate resources in eNodeB and the S-GW; the MME also manages the subscription profile and service connectivity. The responsible protocols between the UE and the CN are the Non-Access Stratum (NAS) protocols.

**P-GW:** The P-GW serves as the end point intermediary router between the EPS and external networks. It mainly provides IP connection at its active point; and is refer to as the highest-level mobility or final anchor in the system. Also, it does IP addressing to UEs, performs traffic gating and filtering duties when needed.

**S-GW:** The S-GW is responsible for the U-plane tunnel management and switching; it acts as the mobile anchor between EPC and the LTE RAN. All the users' packets are routed through the S-GW. Although, the S-GW has a role in control functions, it is very important in terms of inter-connectivity to other 3GPP technologies like GPRS/GSM and UMTS. Also, when the UEs' bearers are setup, cleared or undergo modification, the S-GW make resource allocation depending on the various requests from the MME, P-GW and/orPCRF.

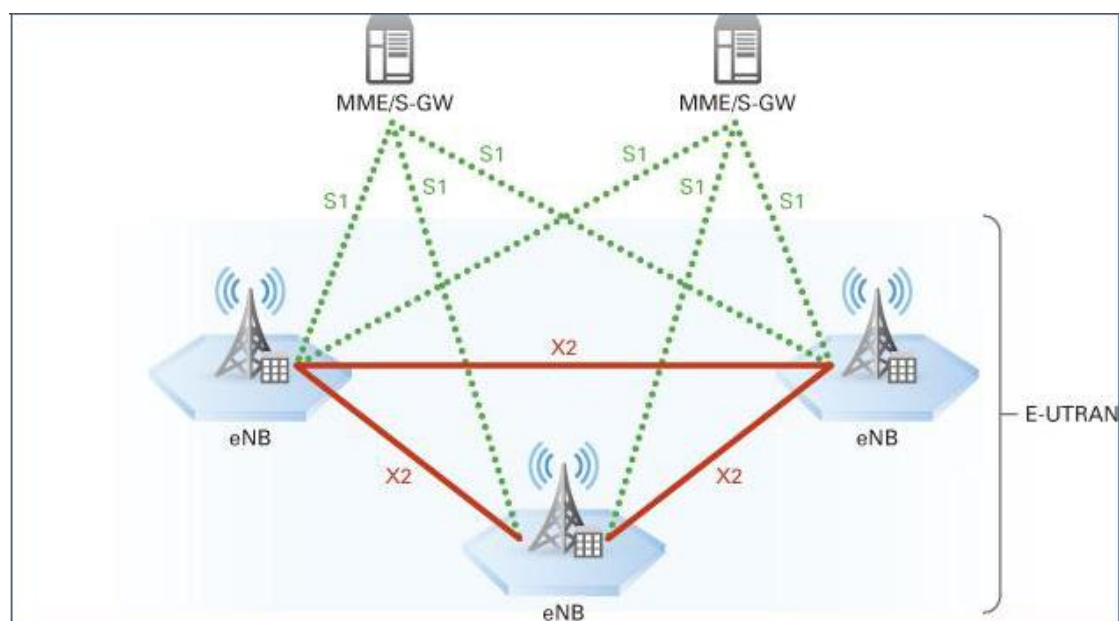
**E-SMLC:** The E-SMLC is responsible for the management of all the coordination and resource scheduling needed for UEs' locations in connection with the E-UTRAN. The final location is estimated based on calculated values from the E-SMLC and it does the estimation for the UE speed and its accuracy level.

**PCRF:** The PCRF is responsible for the QoS as well as the policy control decision making. Also, it controls the flow-based charging for functions within the Policy Control Enforcement Function (PCEF) which is part of the P-GW. In other words, it does the Policy and Charging Control (PCC) functions.

**HSS:** The HSS is a database that contains all users' subscription details. It contains the information about the PDN every user is connected to or can connect to. Essentially, it holds all permanent subscribers' data. As part of its side functions, the HSS can also integrate the Authentication Centre (AuC). [6]

## 2.4 The E-UTRAN (The access network)

The architecture of evolved UMTS Terrestrial Radio Access Network (E-UTRAN) has been illustrated below.



*Figure 2:3 E-UTRAN*

The E-UTRAN handles the radio communications between the mobile and the evolved packet core and just has one component, the evolved base stations, called eNodeB or eNB. Each eNB is a base station that controls the mobiles in one or more cells. The base station that is communicating with a mobile is known as its serving eNB.

LTE Mobile communicates with just one base station and one cell at a time and there are following two main functions supported by eNB:

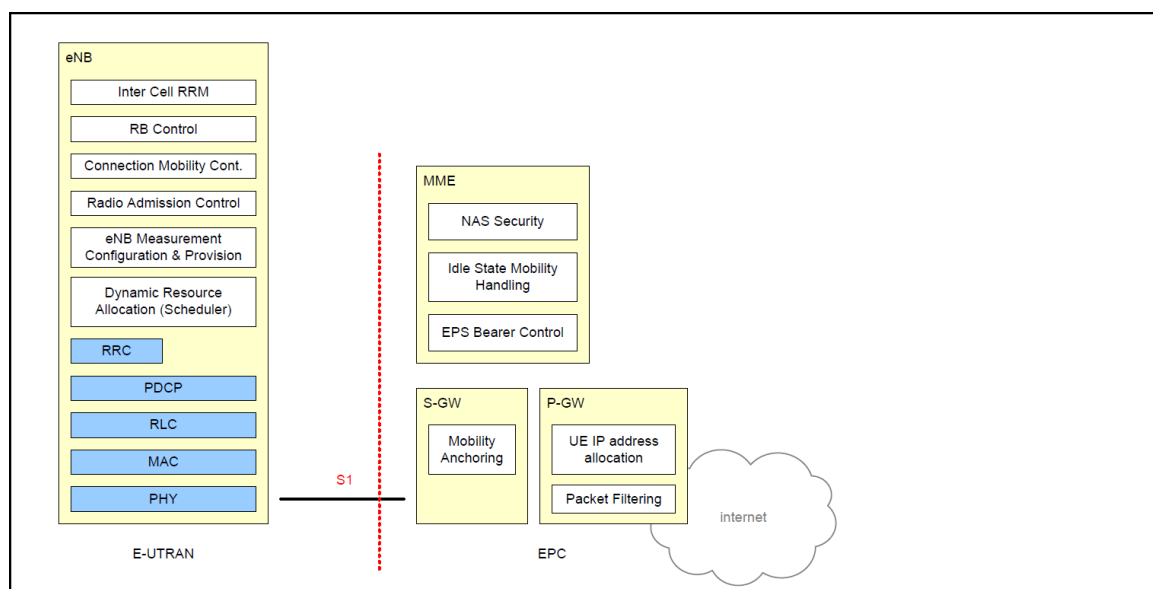
- The eNB sends and receives radio transmissions to all the mobiles using the analogue and digital signal processing functions of the LTE air interface.
- The eNB controls the low-level operation of all its mobiles, by sending them signaling messages such as handover commands.

Each eNB connects with the EPC by means of the S1 interface and it can also be connected to nearby base stations by the X2 interface, which is mainly used for signaling and packet forwarding during handover.

A home eNB (HeNB) is a base station that has been purchased by a user to provide femtocell coverage within the home. A home eNB belongs to a closed subscriber group (CSG) and can only be accessed by mobiles with a USIM that also belongs to the closed subscriber group. [7]

## 2.5 Functional split between the E-UTRAN and the EPC

Following diagram shows the functional split between the E-UTRAN and the EPC for an LTE network:



*Figure 2:4 Functional split between the E-UTRAN and the EPC*

## 2.6 Radio Resource Management

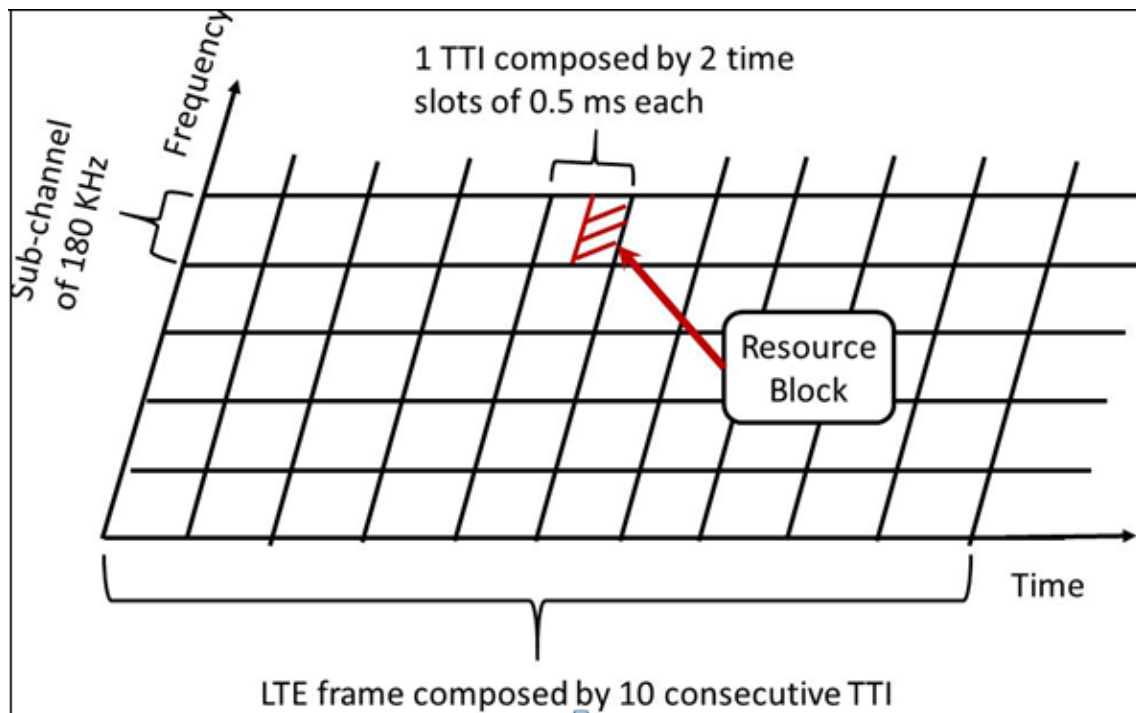
Radio Resource Management (RRM) is used in LTE-Advanced to secure the radio resources which available and utilized fairly. In order to do that, it comprise all strategies for controlling parameters includes transmit power, handover measures, modulation scheme, error coding scheme and channel allocation. It improved the speed and data rate. OFDM is

Used by LTE for downlink transmission where OFDM divide the bandwidth into multiple narrower sub-carries and data is transmitted on these carries in parallel streams. In OFDM the subcarrier is modulated with different modulation method like QPSK, QAM, 64QAM and the use of higher order modulation such like 16QAM and 64QAM provides the higher bandwidth utilization and high data rate, within particular bandwidth an OFDM symbol is obtained by adding different modulated

Subcarrier signals in downlink of an OFDMA the resources are divided in the Frequency and time domains. In frequency domain the resources are divided into Traffic channels which are a cluster of OFDM subcarriers. Whereas in the time domain

The resources are divided into slots called frames and super frames. OFDM is used in other of systems like WLAN, WIMAX to broadcast technologies.

Radio resources are allocated into the time/frequency domain (in Figure 2.5). In the frequency domain, as the total bandwidth is divided into sub channels of 180 kHz, where each one with 12 consecutive and equally spaced OFDM sub-carriers. Where In the time domain, they are divided in every Transmission Time Interval (TTI), each one lasting period 1 ms. The time is cleave into frames, each one comprise of 10 consecutive TTIs. However , each TTI is has two time slots with length 0.5 ms, corresponding to 7 Orthogonal Frequency Division Multiplexing (OFDM) symbols in the default configuration with a short cyclic prefix. A time/frequency radio resource spanning over two time slots in the time domain and over one sub-channel in the frequency domain is called Resource Block (RB) and corresponds to the smallest radio resource unit that can be assigned to an UE for data transmission. The number of RBs varies according to the system bandwidth configuration because of sub-channel size is fixed [8]

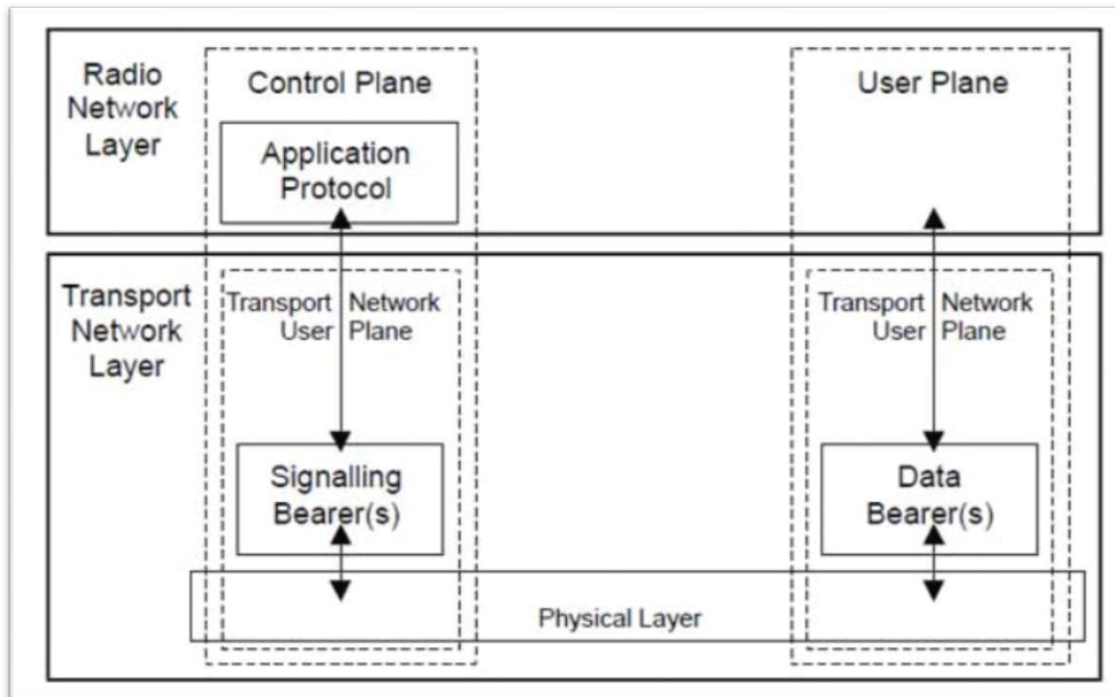


*Figure 2:5 Time-Frequency radio resource grid*

In LTE-Advanced, a dynamic RRM is considered, meaning that the radio network parameters are adaptively adjusted to the identify load, user positions, QoS requirements, etc [9]. Link Adaptation (LA) and other objects like the Packet Scheduling (PS) or Hybrid Automatic Repeat Request (HARQ) play such a finest role.

## 2.7 LTE protocol structure

The radio protocol architecture for LTE can be separated into control plane architecture and user plane architecture as shown below in Figure 2.6



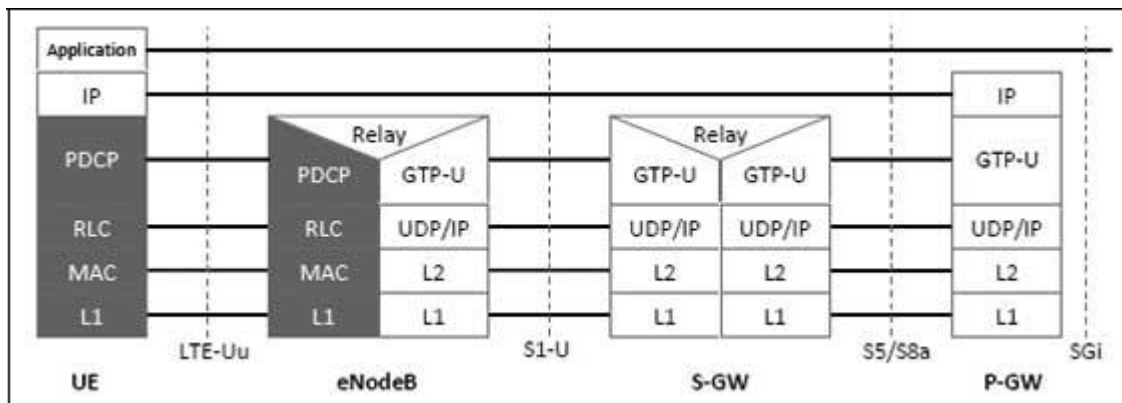
*Figure 2:6 Control plane architecture and user plane architecture*

At user plane side, the application creates data packets that are processed by protocols such as TCP, UDP and IP, while in the control plane, the radio resource control (RRC) protocol writes the signaling messages that are exchanged between the base station and the mobile. In both cases, the information is processed by the packet data convergence protocol (PDCP), the radio link control (RLC) protocol and the medium access control (MAC) protocol, before being passed to the physical layer for transmission.

The user plane protocol stack between the e-Node B and UE consists of the following sub-layers:

1. PDCP (Packet Data Convergence Protocol)
2. RLC (radio Link Control)
3. Medium Access Control (MAC)

On the user plane, packets in the core network (EPC) are encapsulated in a specific EPC protocol and tunneled between the P-GW and the eNB. Different tunneling protocols are used depending on the interface. GPRS Tunneling Protocol (GTP) is used on the S1 interface between the eNB and S-GW and on the S5/S8 interface between the S-GW and P-GW.



*Figure 2:7 User Plane*

Packets received by a layer are called Service Data Unit (SDU) while the packet output of a layer is referred to by Protocol Data Unit (PDU) and IP packets at user plane flow from top to bottom layers. [10]

The RLC entity is specified in the 3GPP technical specification, 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 [10].

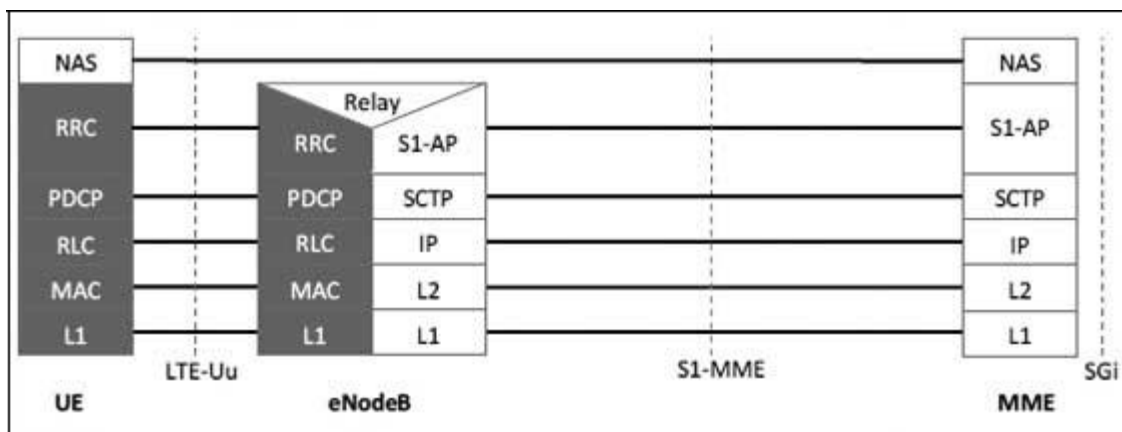
The AM RLC entity manages 3 buffers:

1. **Transmission Buffer:** It is the RLC SDU queue. When the AM RLC entity receives a SDU from the upper PDCP entity, which enquires it in the transmission buffer.
2. **Transmitted Protocol Data Units (PDUs) Buffer:** It is the 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 also puts a copy of the transmitted PDU in the transmitted PDUs buffer.
3. **Retransmission Buffer:** It is basically refers the queue of RLC PDUs that are considered for retransmission. The AM RLC entity moves this PDU to the retransmission buffer, when it retransmits a PDU from the transmission buffer.



The technique of segmentation and concatenation for the RLC SDU of the AM RLC entity follow the same technique which UM RLC entity follow but there are new state variables only present in the AM RLC entity. Depending on the channel condition or the distance of the UE from the eNB, the MAC entity can give instruction to the RLC entity to segment or concatenate the RLC SDU. For example, if the channel condition is not good then the RLC entity has to segment the RLC SDU to fit the length decided by the MAC layer to avoid large bit errors. Thus, a long queue in the RLC buffer or an overflow is likely to occur. Either of these will cause a long delay and lower end-to-end throughput, which are undesirable for a delay-sensitive stream service and non-real-time application services, respectively. [11]

The protocol stack for the control plane between the UE and MME is shown Figure 2.8 The grey region of the stack indicates the access stratum (AS) protocols. The lower layers perform the same functions as for the user plane with the exception that there is no header compression function for the control plane.



*Figure 2:8 Control Plane*

The Control Plane handles radio-specific functionality which depends on the state of the user equipment which includes two states: idle or connected.

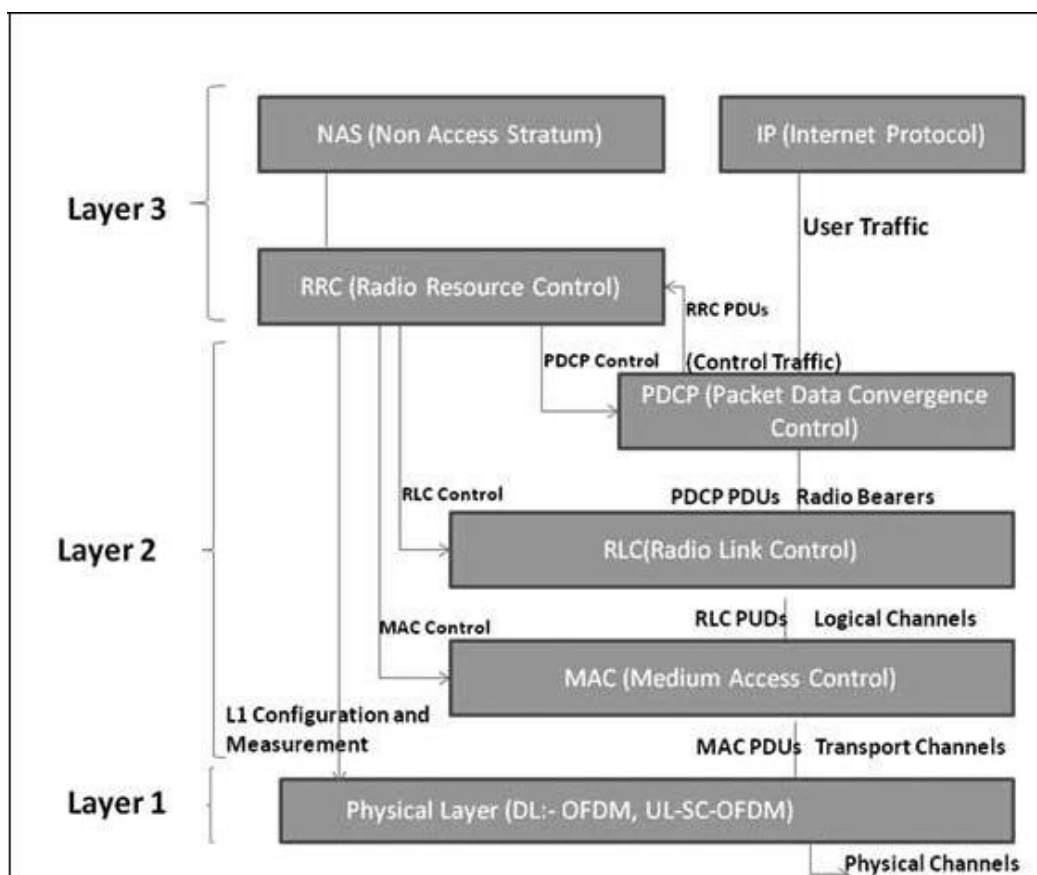
### Idle-

The user equipment camp on a cell selection or reselection process where factors like radio link quality, cells status and radio access technology are considered. The UE also monitors a paging channel to detect incoming calls and acquire system information. In this mode, control plane protocols include cell selection and reselection procedures. [12]

### Connected-

The UE supplies the E-UTRAN with downlink channel quality and neighbor cell information to enable the E-UTRAN to select the most suitable cell for the UE. In this case, control plane protocol includes the Radio Link Control (RLC) protocol. [13]

In LTE Protocol Stack Layers all the layers available in E-UTRAN Protocol Stack which we have seen previously in figure 2.9 we can see elaborated diagram of E-UTRAN Protocol Stack:



*Figure 2:9 E-UTRAN Protocol Stack Layer*

Physical Layer (Layer 1)-Physical Layer carries all information from the MAC transport channels over the air interface. Takes care of the link adaptation (AMC), power control, cell search (for initial synchronization and handover purposes) and other measurements (inside the LTE system and between systems) for the RRC layer.

Medium Access Layer (MAC)-MAC layer is responsible for Mapping between logical channels and transport channels, Multiplexing of MAC SDUs from one or different logical channels onto transport blocks (TB) to be delivered to the physical layer on transport channels, de multiplexing of MAC SDUs from one or different logical channels from transport blocks (TB) delivered from the physical layer on transport channels, Scheduling information reporting, Error correction through HARQ, Priority handling between UEs by means of dynamic scheduling, Priority handling between logical channels of one UE, Logical Channel prioritization.

Radio Link Control (RLC)-RLC operates in 3 modes of operation: Transparent Mode (TM), Unacknowledged Mode (UM), and Acknowledged Mode (AM). RLC Layer is responsible for transfer of upper layer PDUs, error correction through ARQ (Only for AM data transfer), Concatenation, segmentation and reassembly of RLC SDUs (Only for UM and AM data transfer).

RLC is also responsible for re-segmentation of RLC data PDUs (Only for AM data transfer), reordering of RLC data PDUs (Only for UM and AM data transfer), duplicate detection (Only for UM and AM data transfer), RLC SDU discard (Only for UM and AM data transfer), RLC re-establishment, and protocol error detection (Only for AM data transfer).

Radio Resource Control (RRC)-The main services and functions of the RRC sublayer include broadcast of System Information related to the non-access stratum (NAS), broadcast of System Information related to the access stratum (AS), Paging, establishment, maintenance and release of an RRC connection between the UE and E-UTRAN, Security functions including key management, establishment, configuration, maintenance and release of point to point Radio Bearers.

Packet Data Convergence Control (PDCP)-PDCP Layer is responsible for Header compression and decompression of IP data, Transfer of data (user plane or control plane), Maintenance of PDCP Sequence Numbers (SNs), In-sequence delivery of upper layer PDUs at re-establishment of lower layers, Duplicate

elimination of lower layer SDUs at re-establishment of lower layers for radio bearers mapped on RLC AM, Ciphering and deciphering of user plane data and control plane data, Integrity protection and integrity verification of control plane data, Timer based discard, duplicate discarding, PDCP is used for SRBs and DRBs mapped on DCCH and DTCH type of logical channels.

Non-Access Stratum (NAS) Protocols-The non-access stratum (NAS) protocols which is actually form the highest stratum of the control plane between the user equipment (UE) and MME.NAS protocols support the mobility of the UE and the session management procedures to establish and maintain IP connectivity between the UE and a PDN GW. [13]

## CHAPTER 3: ACTIVE QUEUE MANAGEMENT (AQM)

# 3 Introduction to Active Queue Management (AQM)

Active Queue Management (AQM) is a mechanism designed to improve the TCP performance. When the amount of queuing data in the buffer exceeds the physical limit, some packets have to be dropped. The most straightforward method of doing this is drop-tail, which means all the incoming packets are discarded because there is no “room” for them. This method is intuitive and easy to implement; however, it suffers performance degradation in some situations.

One possible problem is known as unfair sharing. Unfair sharing happens when one or several TCP flows occupy the whole buffer so that new incoming flows are locked out. To avoid unfair sharing, new methods called drop-from-front and random drop are proposed. The differences of these methods relate to say how the packets to be discarded are picked up when the buffer is full. Instead of dropping incoming packets, drop-from-front drops the packets which have stayed the longest time in the queue, while the random drop picks the packets to be discarded randomly. These methods indeed improve the TCP performance as compared to drop-tail. However, they still suffer from fact that they are passive in responding to congestion, which means that only when the buffer is full, something is done. Because passive queue management allows queues to be full or almost full, one obviously problematic situation occurs when a burst of packets arrives when the buffer is almost full. Then multiple packets have to be discarded. If those packets belong to different TCP flows, all those flows might reduce the sending rate at the same time or even trigger the slow-start phase. This is known as global synchronization. Global synchronization can decrease the throughput significantly; drawbacks are even more severe in radio networks because of the variation of data rates and thus a higher probability to packets dropping. One straightforward way to avoid the global synchronization is increasing the physical buffer; nevertheless, implementing large buffers do not only increase the system cost but also causes other problems, e.g. significant delay, timeouts

inflation, viscount web surfing etc. Thus, more sophisticated mechanisms are needed to control the queue [14]

Active Queue Management (AQM) can somehow overcome the drawbacks of passive queue management by maintaining the queue size, queuing time or both of them within a reasonable value. Considerable work has been done in this field and several algorithms have been proposed. RED and its variations are the best-known algorithms of AQM. Although RED is a significant improvement over simple Drop 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. [13]

In next subsections we will introduce the AQM algorithms in more detail.

### 3.1 RED

The RED algorithm is widely accepted and implemented AQM method in current Internet routers. It is a queue-size-based algorithm which have two threshold called  $min_{th}$  and  $max_{th}$ . When the queue size is less than  $min_{th}$ , every coming packet is taken into the queue, and when the queue size is greater than  $min_{th}$  but less than  $max_{th}$ , every incoming packet have a probability  $p_{\alpha}$  to be dropped, where  $p_{\alpha}$  is a value with range from 0 to  $max_p$ . If the queue size is greater than  $max_{th}$ , then every coming packet will be dropped. In order to avoid unnecessary dropping when packets arrive in a burst, an average queue size  $Avg$ , is used instead of the current queue size.

The general RED algorithm is give below:

#### Algorithm

*for each packet arrival*  
*calculate the average queue size Avg*  
*if  $min_{th} \leq Avg \leq max_{th}$*   
*calculate drop probability  $p_{\alpha}$*   
*with probability  $p_{\alpha}$*   
*mark (or drop) the arriving packet*  
*else if  $Avg \geq max_{th}$*   
*mark (or drop) the arriving packet*

The RED algorithm basically has two separate procedures: one is determining the average queue size to compute the degree of burst; the other one is computing the probability to drop the incoming packet, which totally depends on the previously calculated average queue size.

The Avg is calculated using the equation:

$$Avg = (1 - wq) * previous\_Avg + wq * current\_size$$

Here wq is a queue weight having a value between 0 and 1. The smaller the value of wq, the smoother averaging of Avg, meaning that incoming packets can be maintained in a burst without significant increase in the dropping probability.

$p_a$ , on the other hand, is more complicated to calculate. To calculate  $p_a$ , another related parameter  $p_b$  is calculated depending on  $max_p$ ,  $min_{th}$  and  $max_{th}$ :

$$p_b \leftarrow Max_p (avg - Min_{th}) / (Max_{th} - Min_{th})$$

the drop probability  $p_a$  is calculated as

$$p_a \leftarrow p_b / (1 - count \times p_b)$$

where count is the number of packets since last drop

## 3.2 A delay-based AQM

The algorithm of delay-based, which means that it decides the packets to be dropped based on the time they have been in the queue. There are four parameters in this algorithm: *minAgeThreshold*, *maxAgeThreshold*, *lowerDropThreshold* and *minInterDropTime* [15]

In delay-based AQM, a parameter called *minAgeThreshold* is used determined by the pipe capacity of the link the *minAgeThreshold* is determined by end-to-end Round-Trip Time (RTT<sub>e2e</sub>) of the link. The RTT<sub>e2e</sub> is less dependent on the instantaneous bandwidth than Pipe Capacity. As a consequence, the *minAgeThreshold* is less dependent on bandwidth than *Tmin*; therefore, the delay-based AQM usually has better throughput in circumstance of large varying bandwidth [16].

The *minInterDrop Time* is defined to prevent the dropping multiple packets consecutively

The *lowerDrop Threshold* is used to make sure that the algorithm does not drop from an almost drained queue, no matter how long time the packets have been in the buffer. Another reason for this parameter is ensuring that there is sufficient number of packets left in the queue to trigger duplicate ACKs after the drop and thereby avoid RTO, if the data belongs to some TCP flow. The *maxAge Threshold* defines the algorithm drops the packets that have stayed in the queue longer than the parameter, regardless of the *minInter DropTime*. To this end, the AQM algorithm can be defined as in Algorithm

### Algorithm

```

for each outgoing packet
  if(size ≤ lowerDropThreshold) transmit packet
  else
    if((delay > minAgeThreshold AND now - previousDropTime > minInterDropTime))
      OR (delay > maxAgeThreshold ) )
      discard packet
      previousDropTime = now
    else
      transmit packet

```



### 3.3 SmRED

RED algorithm discussed in previous chapter which is proposed by Floyd and Jacobson, is the prominent solution for global synchronization and this opens a vast research area in AQM.

As we have been discussed that the goal of a proper AQM scheme is to maintain the average queue size between  $Min_{th}$  and  $Max_{th}$  of the queue at low oscillations and in turn help avoid forced drops of packets. But It is inappropriate for the average queue size and the original RED packet dropping probability to be linearly related[17] It has been found in many research that with a small average delay in the low-load scenario of packets, the link bandwidth is not fully utilized; thus, in order to ameliorate the link utilization, a smaller packet dropping probability should be introduced. The link bandwidth is fully utilized with a large average delay in the high-load scenario of packets; thus, a larger packet dropping probability should be introduced in order to minimize the average delay. Considering the above requirements, we remodeled the packet dropping probability of RED according to the traffic-loads which we call Smart RED (SmRED), in which we divided the packet dropping probability function into two regions according to the setting of the  $Min_{th}$  and  $Max_{th}$  to discriminate between low and high traffic-load conditions. By doing this, we want to achieve a trade-off between the throughput and the delay.

RED's original packet dropping probability can be defined as

$$p_d = p_{max} \times \left( \frac{avg - Min_{th}}{Max_{th} - Min_{th}} \right) \dots \dots \dots (1)$$

As previously mentioned that where avg is the average queue length,  $Min_{th}$  is the minimum threshold that the average queue length must exceed before any packet marking or dropping is done, and  $Max_{th}$  is the maximum threshold that the average queue length must exceed before all packets are marked and dropped.

We have to first decide one target value of the eNB buffer size below which we treat the traffic volume as low and above which we treat the traffic volume is high. We can define the target value as the middle of the minimum threshold and the maximum threshold and it can be defined mathematically as:

$$Target = Min_{th} + \frac{Max_{th} - Min_{th}}{2} \dots\dots\dots (2)$$

If the avg is below the target value, we set the packet dropping probability as

$$p_d = p_{max} \times \left( \frac{avg - Min_{th}}{Max_{th} - Min_{th}} \right)^i \dots\dots\dots (3)$$

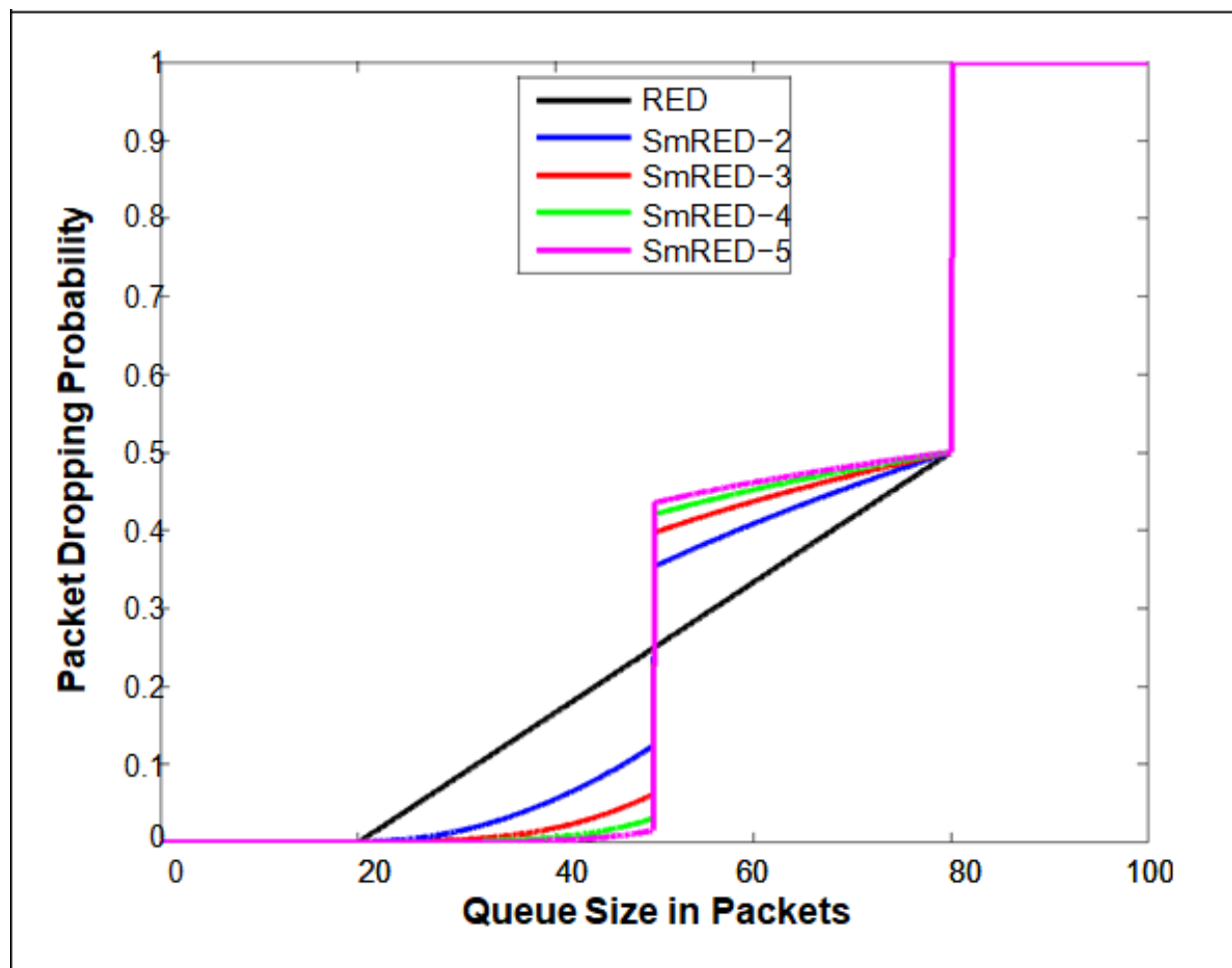
On the other hand, when the traffic volume starts to become high, i.e., if the avg is between Target and Max<sub>th</sub>, then the packet dropping probability function is defined as:

$$p_d = p_{max} \times \left( \frac{avg - Min_{th}}{Max_{th} - Min_{th}} \right)^{1/i} \dots\dots\dots (4)$$

where,  $i = 2, 3, 4, 5, \dots$ . Increasing the value of  $i$  will lead to lower dropping probability during low traffic-load condition and high dropping probability during high traffic-load condition. We can tune the parameter  $i$  to obtain different dropping functions as shown in Figure, where  $Min_{th} = 20$ ,  $Max_{th} = 80$ , and  $P_{max} = 0.5$ . Depending on the value of the parameter  $i$ , we define SmRED's different versions as SmRED- $i$ . In summary, the expressions of SmRED- $i$ 's packet dropping probability, being the function of the average queue length, are shown in Equation (5).

$$P_d = \begin{cases} 0, avg \in [0, Min_{th}) \\ P_{max} \times \left( \frac{avg - Min_{th}}{Max_{th} - Min_{th}} \right)^i, avg \in [Min_{th}, Target) \\ P_{max} \times \left( \frac{avg - Min_{th}}{Max_{th} - Min_{th}} \right)^{1/i}, avg \in [Target, Max_{th}) \\ 1, avg \in [Max_{th}, +\infty) \end{cases} \quad (5)$$

The detailed discard strategy of the algorithm can be described as shown in Figure 3.1 Here one cannot concurrently have a low queuing delays with high link utilization. Therefore, a reasonable trade-off is required between these two performance measures. Tuning the value of the parameter  $i$  gives a trade-off between high throughput and low delay among different versions of SmRED.



*Figure 3.1 The packet dropping probability function.*

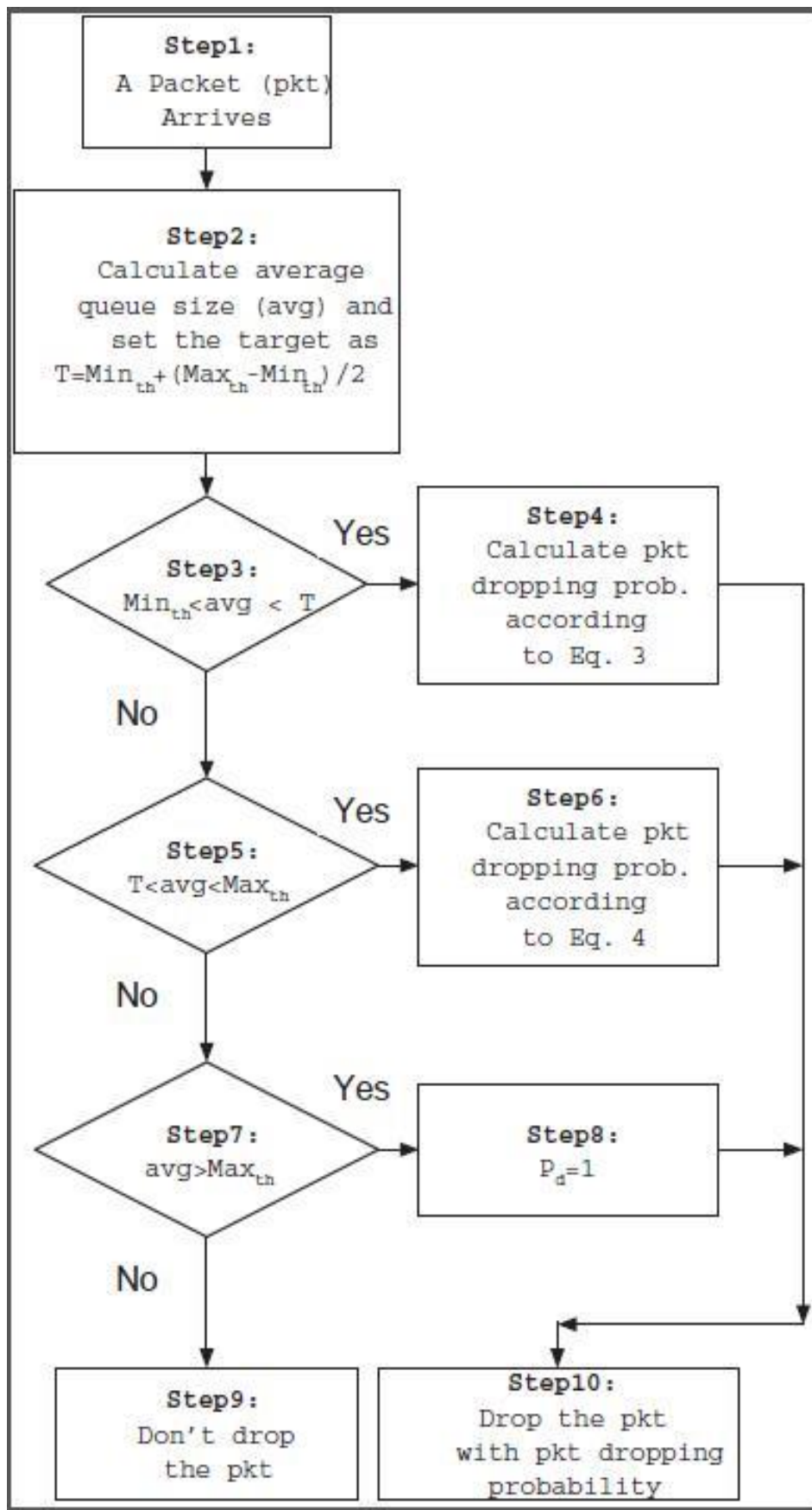


Figure 3:2 The detailed packet discard strategy of SmRED Scheme

# CHAPTER 4: NETWORK MODEL & SIMULATION PARAMETERS

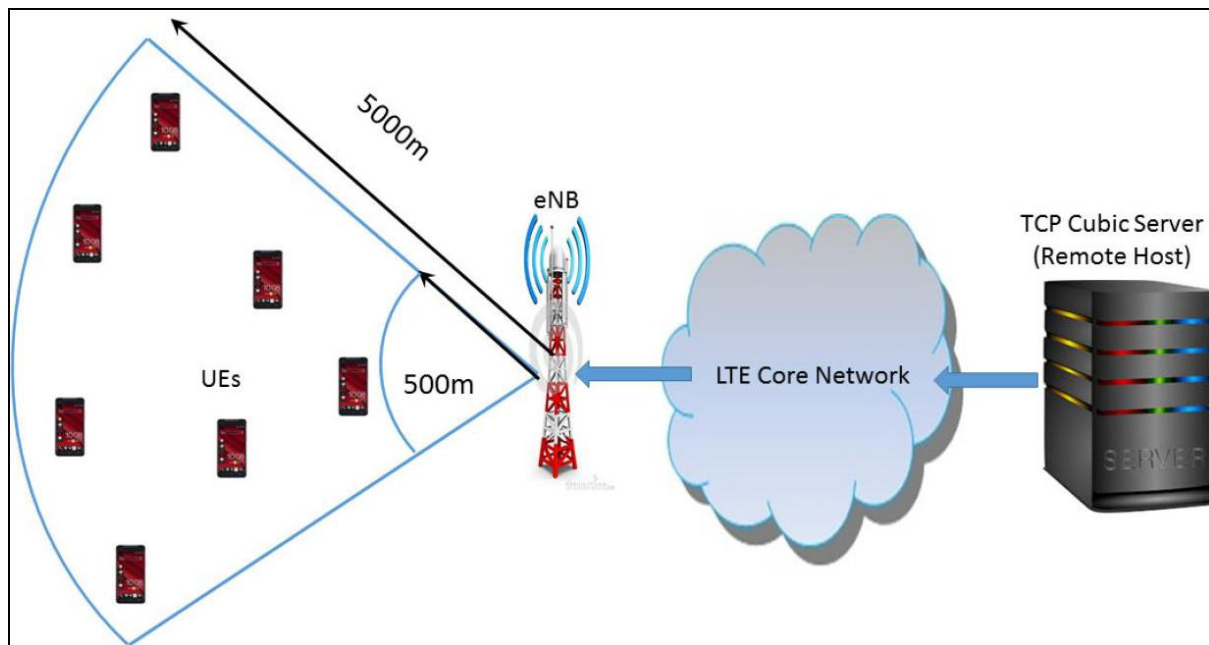
## 4 Network Model

In this chapter, we will describe about the network model that we used for simulation in ns3, system parameters and impart a brief description of different scheduling algorithms which we have been used.

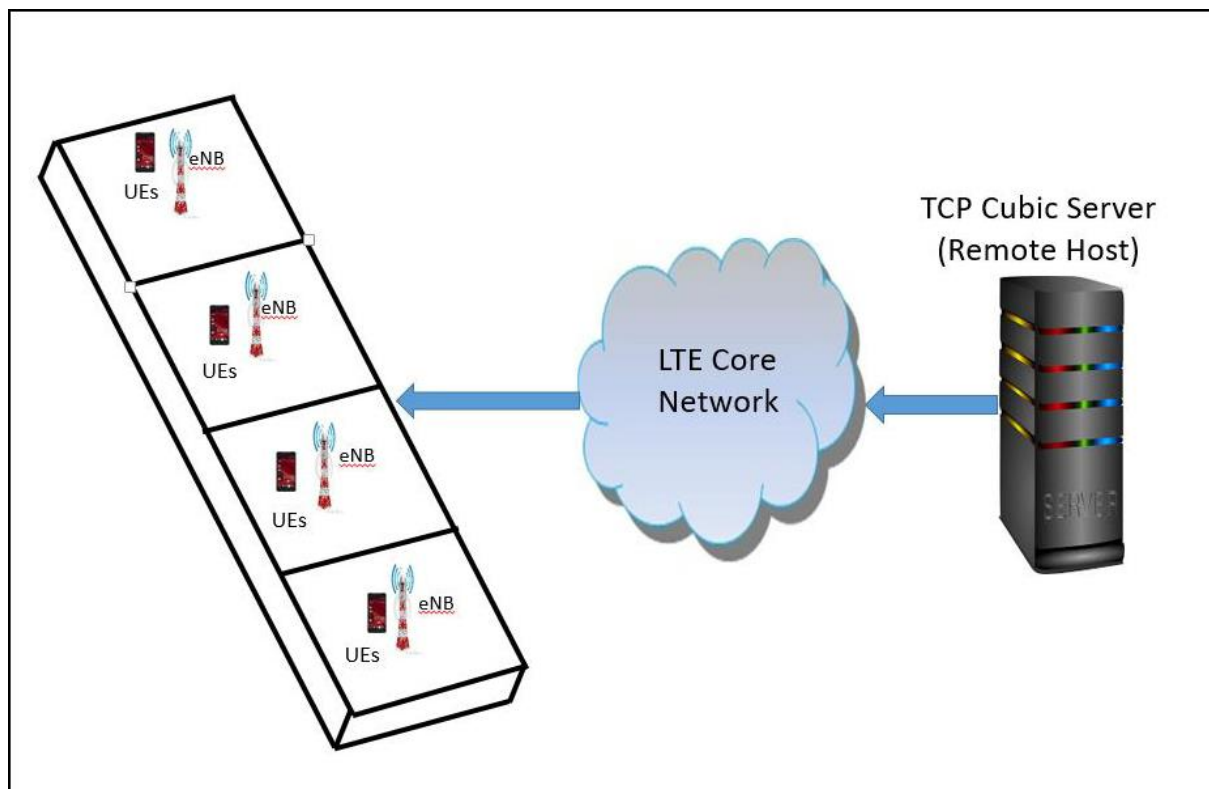
### 4.1 Network Topology and System Parameters

In order to observe the interactions between the AQM-based buffer management and scheduling algorithm, we have to consider mobile network elements and protocols so that we can achieve better idea. We implemented our proposed strategy using the LTE module of the ns3 simulator configured with the topology depicted in Figure 4.1 for single-cell and Figure 4.2 for the handover scenario in multi-cell topology. We used the LTE/EPC Network Simulator and Analysis (LENA) module [17] 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). LENA is an open source product-oriented LTE/EPC Network Simulator that allows LTE small/macro cell vendors to design and test Self Organized Network (SON) algorithms and solutions. Target applications for LENA include the design and performance evaluation of DL & UL Schedulers, Radio Resource Management Algorithms, Inter-cell Interference Coordination solutions, Load Balancing and Mobility Management, Heterogeneous Network (HetNets) solutions, End-to-end QoE provisioning, Multi-RAT network solutions and Cognitive LTE systems. LENA is based on the popular ns-3 network simulator for internet systems. The development of LENA is open to the community in order to foster early adoption and contributions by industrial and academic partners.[19] The LTE model in ns3 gives 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 detailed specification of 3GPP LTE and various versions of TCP. Hence, the results obtained in the simulation can be representative of what happens in a real system.



*Figure 4:1 Single-cell LTE network topology.*



*Figure 4:2 Multi-cell LTE network topology.*

The guaranteed bit rate of 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 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 also used the Single Input Single Output (SISO) transmission mode for both UE and the eNB, where SISO Easiest Antenna Technology In some environments. In a digital communication system, it can reduce the speed of data and increase the number of errors. The remote host is connected to the LTE core network via a 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  $\text{Min}_{\text{th}} = 20$  packets and  $\text{Max}_{\text{th}} = 3 \text{ Min}_{\text{th}}$  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 75 s. Table 4.1 shows the details of some of the important simulation parameters and their values.

The Table is Given Below:

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 (i.e., 1250 Packets/s)
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

*Table 4:1 Different Simulation Parameters in ns3.*

## 4.2 Scheduling Algorithm

In general terms scheduling is basically the process of making decisions by scheduler regarding the assignment of various resources (time and frequency) in telecommunications system between users. In LTE the scheduling is carried out a eNodeB by dynamic packet scheduler (PS) which decides upon allotment of resources to various users under its coverage as well as transmission parameters including modulation and coding scheme (MCS)



To fulfill QoS requirements, one of the key features of the LTE system is multi-user programming. Scheduling determines how the available radio resources in the mobile communication system are shared between the different UEs. In LTE, the basic strategy of scheduling is so-called dynamic scheduling. Multi-user programming is responsible for delivering resources available to active users. It is applied as both the downlink and any interference between the uplink channels provided by the eNB and OFDMA, while operating with a granularity of time domain and frequency and a TTI of frequency. Assignment of resources for each UE is generally made based on comparison of metric by RB. These metrics can provide status of transmission rows, channel quality, asset allocation history, buffer status, and QoS requirements. Due to the computational complexity and optimization of decision-making, the major differences between asset allocation strategies exist. In order to determine the allocation policy for an asset for LTE systems, the main design that should always be taken into consideration is the spectral skills, performance expansion and complexity, integrity and provisioning in QoS. For example, the maximum throttle (MT) blind performance equals (bet), different allocation strategies, a good overview fair proportional (PF), eld average (TTA), round robin (RR) scheduler preference (PSS) and Token Bank Fair Row ( TBFQ) etc. has been introduced for the LTE system, which can highlight the professionals and symptoms related to each solution[20]. This section will describe both downlink and uplink scheduling respectively.

### **4.2.1 Downlink Scheduling**

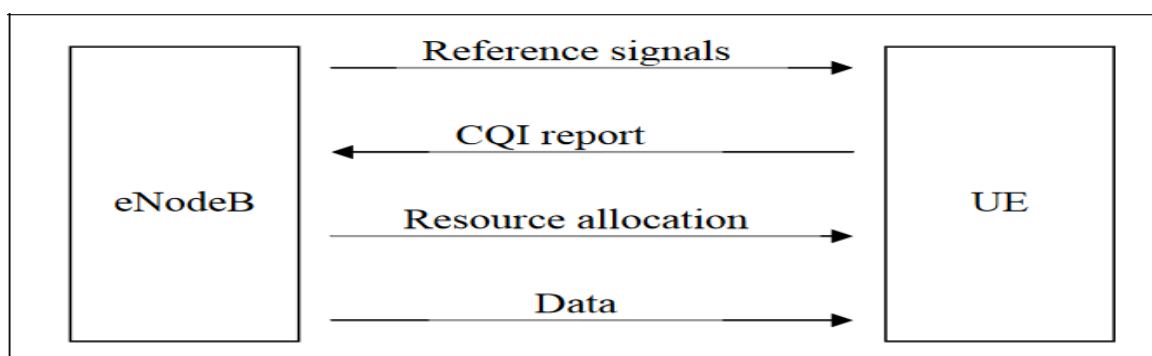
Downlink scheduling is manages in controlling for which UE(s) the eNodeB transmits and with which resource blocks. In addition, the downlink scheduling is organized for the selection of the transport-format e.g. transport-block size, modulation scheme, antenna mapping, etc. and logical channel multiplexing for downlink transmission

One of the remarkable characteristic of mobile communication is the rapid and important variation of the wireless link environment, both on time domain and frequency domain.

Therefore, in order to utilize the scheduling decision, instantaneous channel quality is taken into account; this strategy is called channel-dependent scheduling. To obtain the information of instantaneous channel quality,

Channel Quality Indicator (CQI) is created by the UE and sent to the eNodeB to indicate the channel quality in both time and frequency domain. Basically the CQI is based on the measurement of downlink reference signals.

The all in all target of channel-dependent scheduling is to take advantage of the channel condition variation between UEs and preferably schedule resource to UEs with best channel condition. Figure 4.3 shows the general procedure of downlink scheduling. In addition to supply efficient radio resource utilization, scheduling is also responsible for handling priority issues, due to different requirements of QoS. For example, if the data from VoIP, which requires small end-to-end packet delay, is supposed to have higher priority than the data from a file transferring application e.g. FTP. To support the priority issue, a set of priority queues can be defined. The data is classified to different queues according to their priority level. [21]

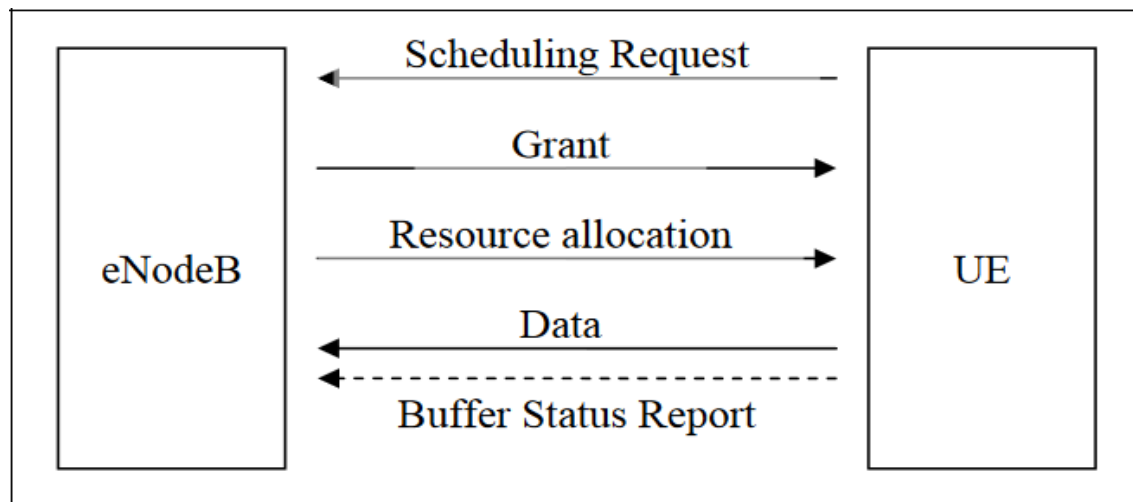


*Figure 4:3 General procedure of downlink scheduling*

## 4.2.2 Uplink Scheduling

It is quite similar to downlink scheduling, where uplink scheduling determines which UE(s) is allowed to transmit and with which resource blocks during a given time interval. However, downlink scheduling; uplink scheduling cannot automatically determine the transmission demand from UE. Thus, before the UE can transmit data to the eNodeB, it first sends a Scheduling Request (SR) to request the transmission permit. When the scheduler in eNodeB receives the SR permit ion, it replies with a scheduling grant for the request. In addition, the scheduler determines the time/frequency resource which the UE should use as well as transport format. After the UE has received the information of assignment, i.e. The UL grant, it can transmit data with required parameters over

a sub-frame when the allow is valid. Additionally, an entity called Buffer Status Report (BSR) can be transmitted together with the data specify the queue length of the UE. The uplink scheduling of general procedure is illustrated in Figure 4.4



*Figure 4:4 General procedure of uplink scheduling*

There are few points trigger the BSR transmission:

- When new data arrives to the UE transmission buffer in that situation the data is with higher priority than the data already existing in the buffer.
- The size of padding bits is larger than the size of BSR and Uplink resource has been allocated. The padding bits defines to the bits in the transport format that are over the actual is need
- The UE moves from one cell to another.
- The Periodic BSR timer is used so that it can trigger Periodic BSR, expires.

A triggered BSR can also be cancelled when the allocated resource is large enough to indulge all the data in buffer which is excluding the size of BSR.

### **4.2.3 Selected Scheduling Algorithms**

In next subsection we will represent scheduling algorithms which have been implemented in the software environment.

#### **4.2.3.1 Round Robin Scheduling**

This scheduling method is based on the idea of being fair in the long term by assigning equal no. of Physical Resource Blocks (PRBs) to all active UEs. It operates by assigning the PRBs to UEs in turn i.e one after another without taking into account their CQI. Hence the users are equally scheduled. [22]

#### **4.2.3.2 Proportional Fair (PF) Scheduling Algorithm**

This algorithm assigns the PRBs to the UE with the best relative channel qualities which is a combination of CQI & level of fairness desired. There are various versions of PF algorithm based on values it takes into account. Main goal of this algorithm is to achieve a balance between Maximizing the cell throughput and fairness, by letting all users to achieve a minimum QoS (Quality of Service).[22]

#### **4.2.3.3 Blind Equal Throughput**

The Blind equal throughput (BET) is a channel unaware strategy that aims at providing throughput fairness among all the users. To counteract the unfair sharing of the channel capacity, the BET scheduler uses a priority metric which considers past average user throughput. BET scheduler prioritizes users with lower average throughput in the past. This implies that users with bad channel conditions are allocated more resources compared to the users with good channel conditions. Thus, throughput Fairness is achieved but at the cost of spectral efficiency [22]

#### **4.2.3.4 Priority set scheduling**

This algorithm aims at providing a defined Target Bit Rate (TBR) to all users. Specifically, in TD scheduler part, the scheduler first sets all available users into two sets based on the target bit rate (TBR)[23]

#### **4.2.3.5 Maximum throughput scheduling**

Maximum throughput scheduling is a procedure for scheduling data packets in a packet-switched best-effort communications network, typically a wireless network, in view to maximize the total throughput of the network, or the system spectral efficiency in a wireless network. This is achieved by giving scheduling priority to the least "expensive" data flows in terms of consumed network resources per transferred amount of information. [24]

#### **4.2.3.6 The token bank fair queuing algorithm**

This algorithm follows the mechanism of the leaky bucket structure with priority handling to address the problem of providing quality-of-service (QoS) guarantees to heterogeneous

Applications in the next generation packet-switched wireless networks. [25]

#### **4.2.3.7 Throughput to Average**

Basically, TTA is a channel-aware/QoS-unaware scheduler. TTA can be considered as a combination of MT and PF. It also ensures that the best RBs are allocated to each user.

# CHAPTER 5: SIMULATION RESULTS & PERFORMANCE ANALYSIS

## 5 Simulations & Performance

In this chapter , we have simulated in ns3 simulation to extend our idea of Smart RED (SmRED) [to SmRED-i, where packet dropping probability function is different in accordance with the value of  $i = 2, 3, 4 \dots$  then the effect of AQM-based buffer management on the scheduling algorithms in the single cell network topology without handover and in the multi-cell network topology in LTE networks across a wide range of traffic-loads and measured the performance in terms of end-to-end average Transmission Control Protocol (TCP) throughput and delay.

### 5.1 Simulation Scenarios

#### 5.1.1 Scenario 1

Implementation of extended idea about Smart RED (SmRED) to SmRED-i, where packet dropping probability function is different in accordance with the value of  $i = 2, 3, 4 \dots$ . In this section, we first exhibit the usefulness of different versions of SmRED as compared to the original RED algorithm. Here, we have simulated some different versions of SmRED where the Guaranteed Bit Rate (GBR) video traffic is downloaded from the source to the target UEs. The number of UEs is varied from 2 to 10 to demonstrate different traffic-loads. We set 15 resource blocks in eNB to serve the UEs. This means that the channel bandwidth is 3 MHz (Other values for resource blocks are 6, 25, 50, 75, 100. We set the value 15 in order to create a bottleneck in the LTE wireless access part. So, even with 10 parallel TCP connections, i.e., 10 UE, the traffic load is considered high when the application is sending packets at a rate of 100 Mbps). The results are shown in Figures 5.1 which illustrates the End-to-end average throughput of

different versions of Smart RED (SmRED) and 5.2 presenting End-to-end average delay of different versions of SmRED.

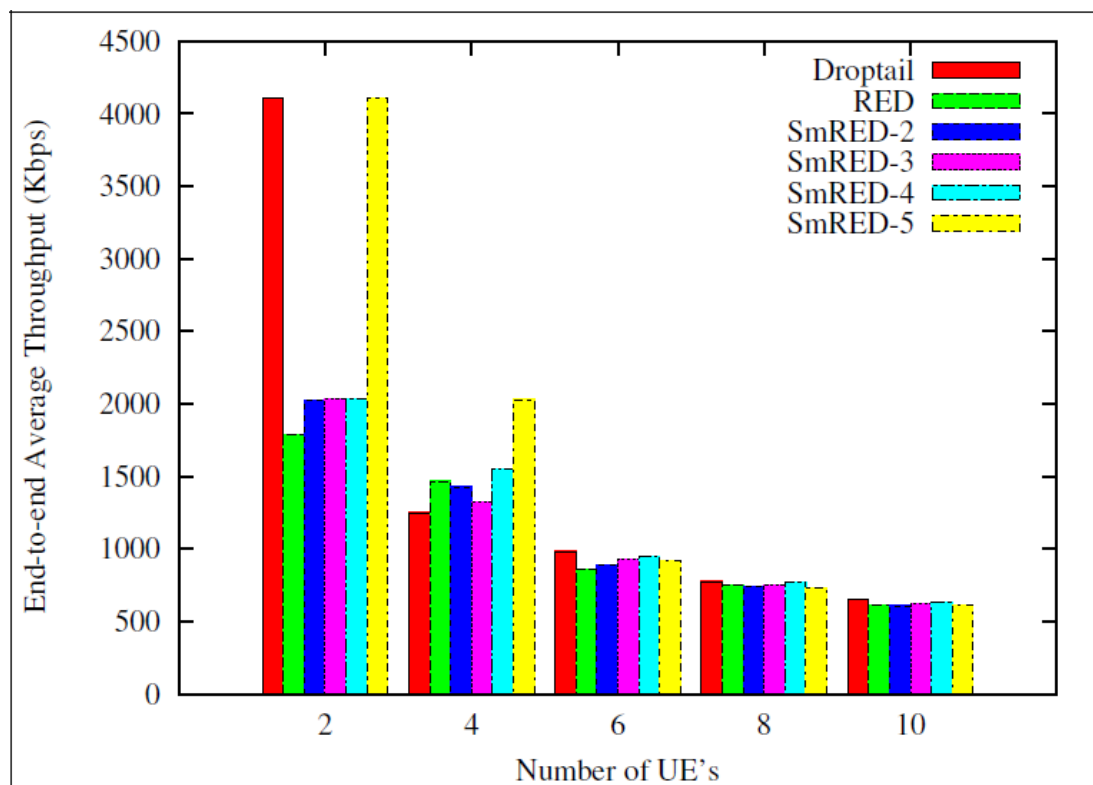


Figure 5:1 End-to-end average throughput of different versions of Smart RED (SmRED).

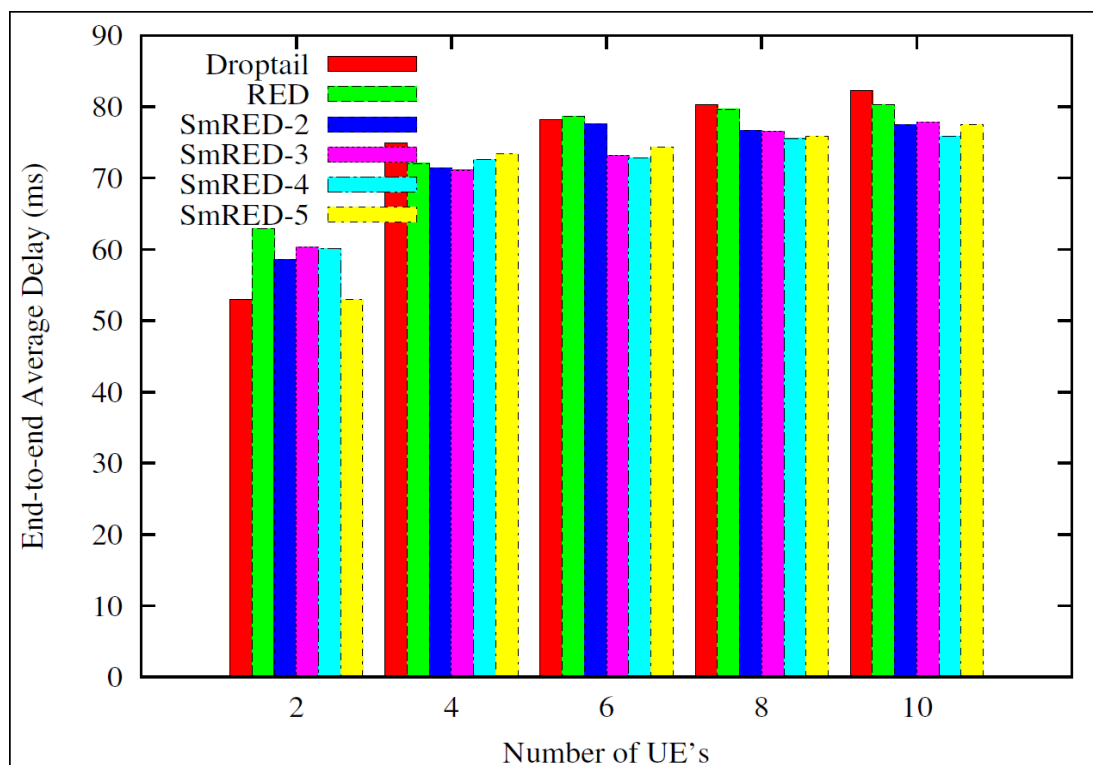


Figure 5:2 End-to-end average delay of different versions of SmRED.

From the simulation results, we can figure out that, in a low traffic-load condition, a higher value of  $i$  gives higher throughput and lower delay. In fact,  $i = 5$  behaves like Droptail. This happens, when the traffic-load is low, higher values of  $i$  result in lower dropping probability and even no dropping occurs for some specific cases. However, when the traffic-load is increasing, then higher values of  $i$  such as  $i = 5$  failed to provide the optimal performance. This is because, for the high traffic-load condition, if the dropping probability is set too low (before the Target), then the AQM algorithm failed to inform the TCP source about the possible congestion more in advance. As a result, when the queue becomes full and packet dropping as well as TCP timeout occurs and the TCP window size enters into the slow start phase, which results in lower throughput and higher delay. On the other hand, the dropping probability should not be set too high after reaching the Target, so that we can avoid unnecessary dropping. From the result, we can see that SmRED with  $i = 4$  achieves a better balance between the throughput and the delay as compared to other versions.

### 5.1.2 Scenario 2

Here, We overview the effect of AQM-based buffer management on the performance of different scheduling algorithms with and without handover, Then We measured the performance in terms of end-to-end average Transmission Control Protocol (TCP) throughput and delay.

To better understand the performance of RED as compared to SmRED-4, we varied the RLC buffer size as the input parameter and calculated the average packet loss rate in percentage with the high traffic load scenario (10 parallel TCP connections) and moderate traffic load scenario (4 parallel TCP connections). The buffer size was varied from 100 packets to 200 packets with a granularity of 25 packets. The simulation results are shown in Tables 5:1 and 5:2 for high load and moderate load respectively. As can be seen from the table, the packet loss rate of SmRED-4 is lower than that of RED with buffer size changed throughout the whole range. With the high traffic load scenario, the reduction rates of the packet loss rates of SmRED-4 compared with RED increased from 3.05 to 14.05. On the other hand, when the traffic load is moderate, the reduction rate is increased from 17.99 to 34.19. This is because, when the traffic load is low, the dropping probability of SmRED-4 is low. So, unnecessary packet dropping is avoided. Thus, the gain in the reduction rate of the packet loss rate of SmRED as compared to RED is higher.



<b>RLC Buffer Size</b>	<b>100</b>	<b>125</b>	<b>150</b>	<b>175</b>	<b>200</b>
Pkt Loss Rate (RED) in Percentage	8.98	7.01	6.18	4.48	4.29
Pkt Loss Rate (SmRED) in Percentage	8.31	6.80	5.31	4.27	3.98
Reduction Rate in Percentage	7.50	3.05	14.05	4.68	7.22

*Table 5:1 Packet Loss Rate in High Traffic Load Scenario*

<b>RLC Buffer Size</b>	<b>100</b>	<b>125</b>	<b>150</b>	<b>175</b>	<b>200</b>
Pkt Loss Rate (RED) in Percentage	5.16	4.28	4.06	3.18	3.10
Pkt Loss Rate (SmRED) in Percentage	3.70	3.51	2.69	2.44	2.04
Reduction Rate in Percentage	28.29	17.99	33.74	23.27	34.19

*Table 5:2 Packet Loss Rate in Moderate Traffic Load Scenario*

To the improved performance achieved by different scheduling algorithms along with SmRED-4, we compared the performance of different scheduling algorithms with SmRED-4 and RED in a single-cell without handover (HO) and in multiple cells with HO. We kept the number of UEs at 10 to simulate high traffic-load conditions. In order to experience different channel conditions, the UEs were placed randomly in the single-cell within a distance of 500 m to 5000 m from the eNB. All UEs downloaded GBR video traffic from a single server. For multi-cell, all UEs were initially located in the first cell, moving towards other cells at a specific speed of 20 m/s. We used a 3GPP-specified A3-RSRP handover algorithm that utilizes Reference Signal Received Power (RSRP) measurements and event A3, as designed. Event A3 is defined as a reporting triggering event which is activated when the neighboring cell's measured RSRP is better than that of the serving cell by a certain offset.

We compared the download performance in terms of end-to-end average throughput and end-to-end average delay. The results are shown in Figures 5:3 and 5:4 for both single-cell and multi-cell. In the multi-cell network scenario, we explored the effect of handover on the TCP performance. Due to the handover, TCP throughput suffers from the packet duplicates and extra latency caused by buffering in the eNB. Thus, in the multi-cell network, the reduction in the total TCP throughput as compared to the single-cell network can be attributed to TCP.

### 5.1.3 Scenario 3

In this part we implemented the idea of AQM-based buffer management on the scheduling algorithms in the single cell network topology without handover and in the multi-cell network topology in the presence of handover in LTE networks.

When the traffic-load is high ( $UE = 10$ ), different scheduling algorithms with SmRED-4 outperform RED. This is because SmRED-4 has a higher dropping probability than RED when the traffic-load is high, which helps to inform the source more 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 (Figure 5:4) without sacrificing the throughput much. An interesting observation is that, in a single-cell topology without handover, MT achieves the highest throughput as expected but in multi-cell topology with handover, its performance degradation is the highest among all the schedulers. This is because MT assigns RB to those users which have good signal quality in order to utilize the full bandwidth. However, as the UEs are moving and handover occurs, different UE's face different signal conditions and MT assigns RB to one UE for a short period of time and then to others. Thus, the overall throughput degraded much but the fairness is increased as expected and can be seen from Figure 5:3

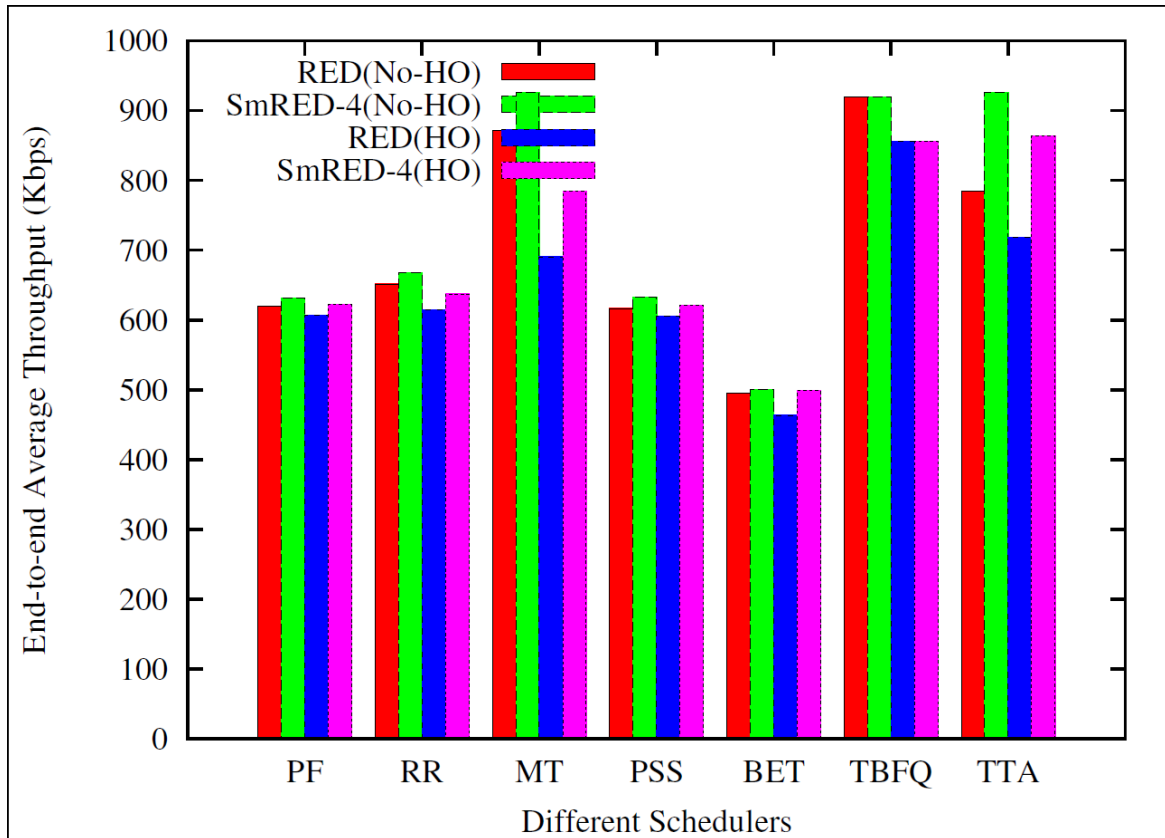


Figure 5:3 End-to-end average throughput of different schedulers.

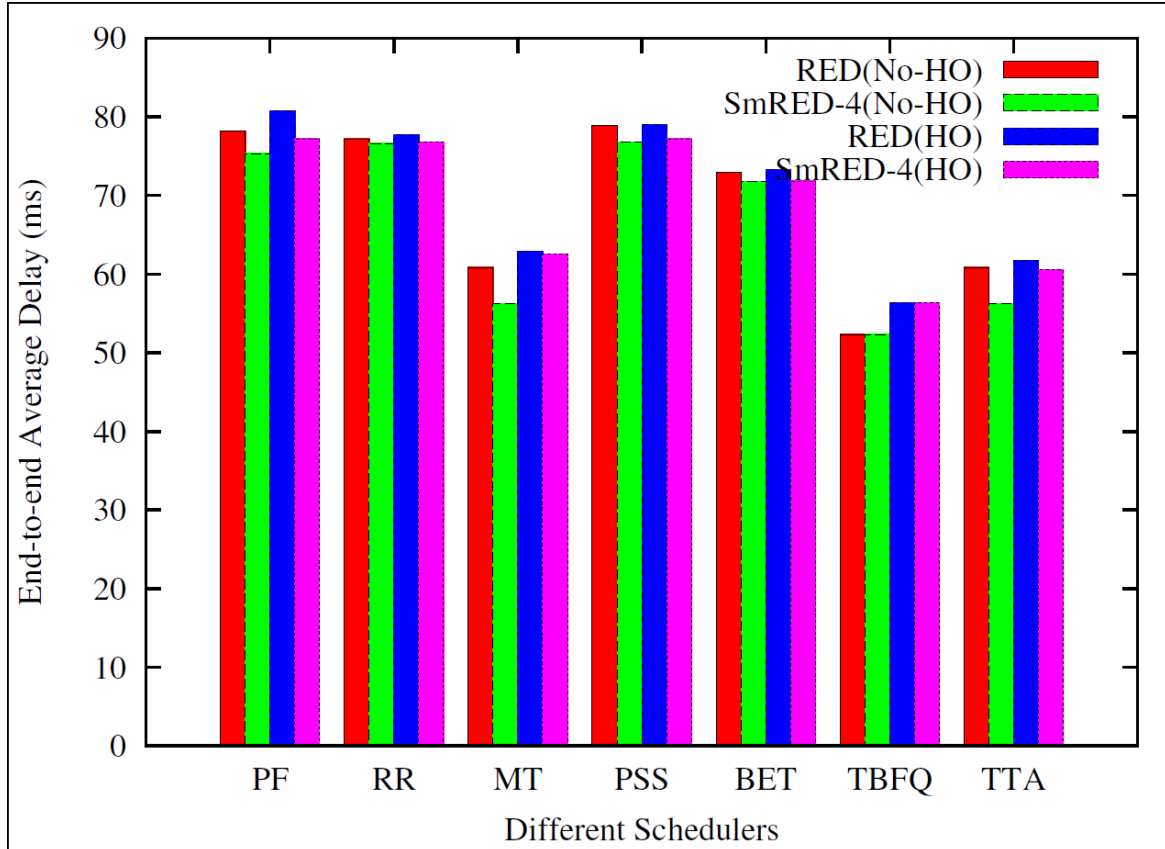


Figure 5:4 End-to-end average delay of different schedulers.

Figure 5:5 shows the fairness of different schedulers Jain's fairness Index [26]. From the performance of MT, TBFQ and TTA, we can see that they blindly enhance the overall cell throughput by utilizing the effective channel in terms of spectral efficiency, but these approaches provide unfair resource sharing among users. Thus, in order to ensure minimum performance even to cell-edge users or in general to those users suffering bad channel conditions, fairness is therefore a criterion of the utmost importance that should be taken into consideration by a good scheduling algorithm. Fairness can be ensured by considering the past service level experienced by each user as is taken into account by many schedulers such as PF, RR, PSS and BET.

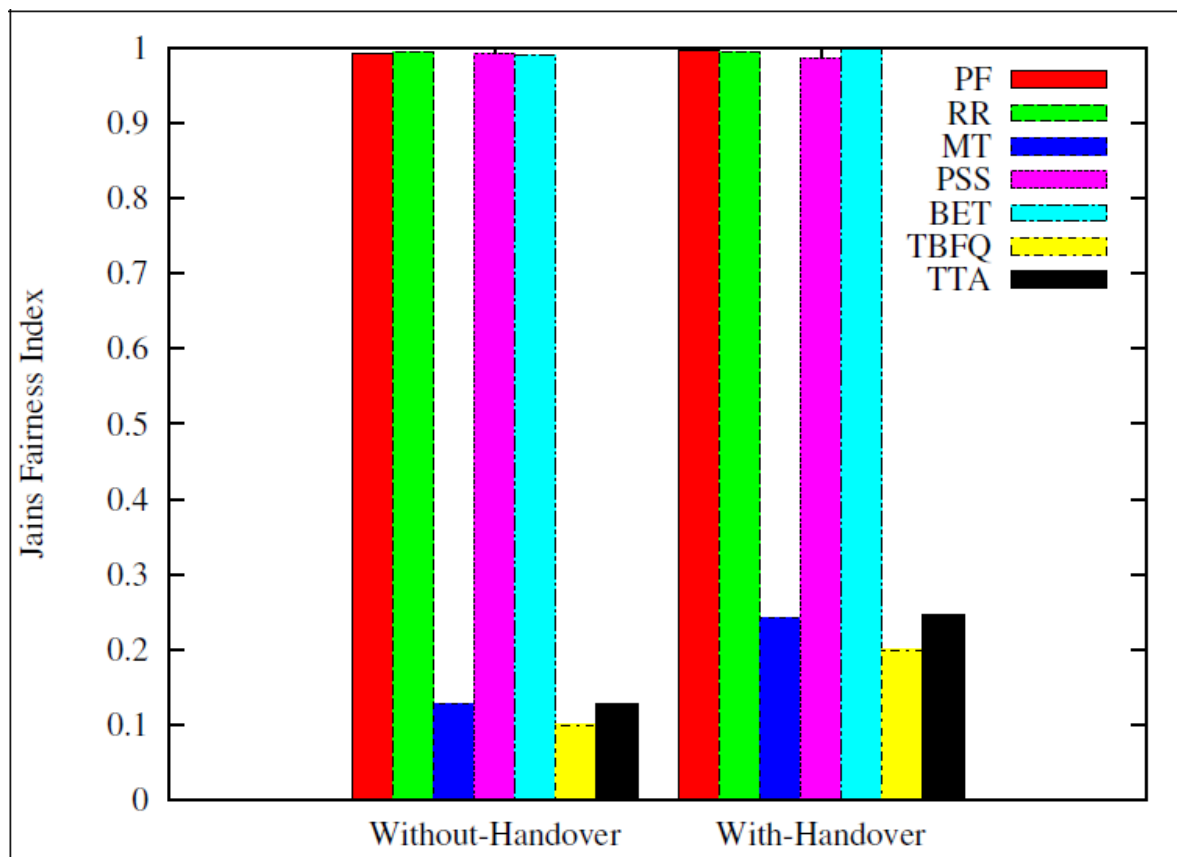


Figure 5:5 Jain's fairness index of different schedulers. Number of UE=10.

## CONCLUSION

The increasing demand for mobile data traffic comes with new challenges in setting up cellular networks.

In this thesis paper we tried to evaluate a performance with help of using various scheduling algorithm in the presence of AQM-based congestion control. A buffer management scheme needs to be kept under RLC buffer at a minimum level, without significant deterioration in user performance. It should be a challenging issue for mobile network operators and thoroughly investigated. Out of the maximum use of the channel's capacity point of view, the best solution is to allocate RB carefully to those users that experience good channel condition. However, doing so will lead to distorted resource allocation to other users. Therefore, equal services can be achieved for all users, such as equity, QoS provisioning, computational complexity and energy savings, low cost costs of call. According to this requirement, the design of implementing the RB allocation strategy is a compromise between the desire to achieve network operators and spectrum functions. On the other hand, a good AQM algorithm needs very less computational cost and should be easily implemented in the real network. SmRED-4 smart process of congestion control based on RED, fixing the implications of connections between the low and high traffic load-load conditions for eNB RLC in the LTE network and the delay of large delays. In addition, due to simplification of SmRED-4 migration from RED to a real network, need very less effort to implement .In order to achieve user application performance SmRED-4 is effectively improves RED disadvantages and achieves a better balance between higher performance and lower latency. Therefore, a good combination of AQM algorithm with scheduling techniques can provide the best performance of network operators.

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