2016

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http://hdl.handle.net/10026.1/6527

http://dx.doi.org/10.24382/1446

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Ph.D.

May 2016

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A thesis submitted to Plymouth University in partial fulfilment for the degree of

DOCTOR OF PHILOSOPHY

School of Computing, Electronics and Mathematics

Faculty of Science and Engineering

May 2016

Acknowledgements

First of all, I would like to express my warm thanks to my Director of Studies Dr Lingfen Sun for her patience, professional guidance and encouragement throughout this project and for the tremendous amount of time and effort she spent to ensure the high quality of my papers and this thesis. I would also like to thank Professor Emmanuel Ifeachor for his guidance, critical comments, and invaluable advices during this project. I would also like to thank the members of Signal Processing Multimedia Communications (SPMC) Research Group: Dr Emmanuel Jammeh and Dr Is-Haka Mkwawa for all the help, support and constructive discussions for this project.

Significantly, my grateful thanks are extended to my wife, Alzahraa. Her support, encouragement, quiet patience and unwavering love were undeniably the bedrock upon which the past 10 years of my life have been built. Warm thanks for my mother, brother and sisters for their unwavering supports.

Finally, this thesis is dedicated to our beautiful two children Hussain and Ruqayah for their endless love.

Declaration

At no time during the registration for the degree of Doctor of Philosophy has the author been registered for any other University award without prior agreement of the Graduate Sub-Committee.

Work submitted for this research degree at the Plymouth University has not formed part of any other degree either at Plymouth University or at another establishment. This study was financed from the sponsor Public Authority for Applied Education and Training, Kuwait. A programme of advanced study was undertaken, which included the extensive reading of literature relevant of research project, 1st Qualinet Summer School on Quality Assessment and research development seminars.

Publications:

The original work presented in this thesis has been published in International Conferences and E-letter:

- Ali Alfayly, Is-Haka Mkwawa, Lingfen Sun and Emmanuel Ifeachor, "QoE-based Performance Evaluation of Scheduling Algorithms over LTE", IEEE Globecom 2012, Workshop: Quality of Experience for Multimedia Communications (GC'12 Workshop - QoEMC), 03-07 December 2012, Anaheim, CA, USA.
- Ali Alfayly, I.-H. Mkwawa, L. Sun and E. Ifeachor, “QoE-driven LTE downlink scheduling for VoIP application,” in the 12th Annual IEEE Consumer Communications and Networking Conference (CCNC), Las Vegas, USA, January 2015.

- L. Sun and A. Alfayly, “QoE-driven Management Schemes for Multimedia Services,” *IEEE COMSOC MMTC E-Letter*, vol. 10, no. 3, pp. 14-17, May 2015.

Word count of main body of thesis: 22875 words.

Signed:____

Date:____

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Ali Alfayly

Abstract

The significant growth in multimedia services and traffic (e.g. VoIP, video streaming and video gaming) in current and emerging mobile networks including the latest 4G Long-Term Evolution (LTE) networks and the rising user expectation for high Quality of Experience (QoE) for these services have posed real challenges to network operators and service providers. One of the key challenges is how to bring multimedia services to the end-user over resource-constrained mobile networks with a satisfactory QoE. Cost-effective solutions are needed for network operators to improve the bandwidth usage of these mobile networks. Therefore, scheduling schemes are of extreme importance in LTE, where scheduling algorithms are responsible for the overall efficiency of resource allocation in an LTE system.

The aim of the project is to develop novel QoE-driven scheduling algorithms for improving system capacity in delivering multimedia services over downlink 3GPP LTE. This is to move away from traditional QoS-driven scheduling schemes to a QoE-driven scheme which guarantee end-user satisfaction in resource allocation. The main contributions of the thesis are threefold:

1. Performance of several existing scheduling algorithms for VoIP applications was evaluated thoroughly in terms of QoE metric (i.e. MOS), instead of QoS metrics (e.g. packet loss and delay). Using QoE metrics instead of QoS ones will facilitate the development of QoE-driven scheduling schemes in order to achieve optimised end-user experiences or optimised mobile system capacity.

2. A novel QoE-driven LTE downlink scheduling scheme for VoIP application was developed to maximize the number of users per cell at an acceptable MOS score. The proposed scheme achieved significant improvement in cell capacity at an acceptable quality (75% compared to MLWDF, and 250% compared to PF and EXP-PF in all three lower speed scenarios considered).
3. A QoE-driven LTE downlink scheduling scheme for multiservice multimedia applications was developed to improve the cell capacity with satisfactory QoE for both VoIP and video streaming services. The proposed algorithm performed well in a pedestrian scenario increasing cell capacity to double for video stream with 'Rapid Movement' (RM) content. For 'Medium Movement' (MM) video content, the capacity was increased about 20% compared to MLWDF and by 40% compared to EXP-PF. In a vehicular scenario, the proposed scheme managed to enhance the cell capacity for MM video stream case.

The project has led to three publications (IEEE Globecom'12 – QoEMC Workshop, IEEE CCNC'15 and IEEE MMTC E-letter/May-2015). A journal paper is in preparation.

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List of Abbreviations and Acronyms

1G	**First Generation**
2G	**Second Generation**
3G	**Third Generation**
4G	**Fourth Generation**
3GPP	**Third Generation Partnership Project**
ACK	**Acknowledgement**
AM	**Acknowledged Mode**
ARQ	**Automatic Repeat Request**
CDMA	**Code Division Multiple Access**
CQI	**Channel Quality Indicator**
CRC	**Cyclic Redundancy Check**
CSI	**Channel State Information**
DL	**Downlink**
DRB	**Data Radio Bearers**

ECM	**EPS Connection Management**
eNodeB	**Evolved Node Base station**
EPC	**Evolved Packet Core**
EPS	**Evolved Packet System**
EXP-PF	**Exponential Proportional Fairness**
E-UTRAN	**Evolved UMTS Terrestrial Radio Access Network**
FDD	**Frequency Division Duplex**
GBR	**Guaranteed Bit Rate**
GSM	**Global System for Mobile communication**
HARQ	**Hybrid Automatic Repeat Request**
HOL	**Head-Of-Line**
HSDPA	**High Speed Downlink Packet Access**
ITU	**International Telecommunication Union**
LTE	**Long Term Evolution**
MAC	**Medium Access Control**

MBMS	**Multimedia Broadcast Multicast Channel**
MOS	**Mean Opinion Score**
MLWDF	**Modified Largest Weighted Delay First**
MME	**Mobility Management Entity**
MMQ	**Modelling Media Quality**
NACK	**Negative Acknowledgement**
OFDMA	**Orthogonal Frequency Division Multiple Access**
PDCP	**Packet Data Convergence Protocol**
PDU	**Protocol Data Unit**
PF	**Proportional Fair**
PHY	**Physical Layer**
PSNR	**Peak Signal to Noise Ratio**
QCI	**QoS Class Identifier**
QoE	**Quality of Experience**
QoE-MS	**QoE scheduling for multi-service**
QoS	**Quality of Service**

RAN	**Radio Access Network**
RB	**Resource Block**
RLC	**Radio Link Control**
RR	**Round Robin**
RRM	**Radio Resource Management**
SBRs	**Signalling Radio Bearers**
SC-FDMA	**Single Carrier Frequency Division Multiple Output**
SDU	**Service Data Units**
SSIM	**Structural Similarity Index**
SSQ	**Surveying Subjective QoE**
S-GW	**Serving Gateway**
SINR	**Signal to Interference plus Noise Ratio**
TDD	**Time Division Duplex**
TDMA	**Time Division Multiple Access**
TM	**Transparent Mode**

TTI	**Transmission Time Interval**
TUQ	**Testing User-perceived**
UE	**User Equipment**
UL	**Uplink**
UM	**Unacknowledged Mode**
UMTS	**Universal Mobile Telecommunication System**

Chapter 1 INTRODUCTION

This chapter presents the motivations behind the project, the fundamental research questions, and the aims and objectives of the project. Furthermore, the chapter highlights the main contributions of this thesis.

The chapter is arranged as follows. Section 1.1 presents the motivations behind the project. The research questions are given in Section 1.2. Section 1.3 presents the project aims and objectives. The major contributions are summarized in Section 1.4. A brief overview and the organisation of the thesis are given in Section 1.5.

1.1 Motivations

The massive growth of mobile devices that are connected to mobile networks worldwide is a major contributor to global mobile traffic growth. According to Cisco's Forecast [1], almost a half billion (497 million) mobile devices and connections were added in 2015 [1] to bring the total active connections to 6.4 billion where smartphones contributed the most of that growth. This is predicted to increase to 7.4 billion mobile devices and connections via the mobile access network by 2019 [1]. The smartphones can act as handheld computers, which offer high quality multimedia applications and meet end-user expectations to use anywhere and anytime. Therefore, it is predicted that mobile multimedia traffic will increase further and mobile video traffic will account for around 72% of the world's mobile data traffic by 2019 [1].

This rapid growth of mobile services and traffic especially mobile video data will present a great challenge to mobile network operators to provide the best service for mobile users. Also, this trend indicates that mobile multimedia has become very popular for both voice and video services. For example, FaceTime and Skype are used for VoIP calling, while Netflix and YouTube are used for video streaming services. These services remain the most popular multimedia services for people in different generations.

The 3GPP is adopting Long-Term Evolution (LTE) as 4G for mobile network to meet requirement of mobile services, which provides higher capacity and high data speed. However the shortage of radio resources and a nature of the mobile network medium have pose great challenges to improve of Quality of Service (QoS) while utilize the radio resources efficiently [2]. One of crucial challenges is to develop downlink packet scheduler that will assign radio resources among users efficiently. The 3GPP standard leaves open the scheme and implementation of radio resource scheduling. Therefore, numerous attempts have been made to design the radio resource scheduler by considering different requirements of the network. For example, Opportunistic contention-based feedback protocol for downlink LTE systems with mixed traffic (i.e. real-time and non-real time) was proposed to improve the achievable throughput for real time traffic by allocating dedicated resource to feedback from all real-time user [3]. Proportional fair scheduling algorithm in OFDMA-based wireless systems with QoS constraints was developed that offers enhanced QoS to voice and video traffic while providing fair data rates to users [4]. The utility-based joint uplink/downlink scheduling algorithm suitable for wireless services was developed to increase the packet loss and improve network throughput [5]. The energy aware packet scheduling was developed and compared against the traditional throughput based proportional fair scheduler [6].

Many studies have been published on LTE scheduling which was primarily focused on the QoS metrics (i.e., throughout, packet loss and delay, etc.), such as the ones proposed in [7], [8] , and [9]. In this context, it will be challenging to pre-define an optimal delay or packet loss ratio values, because these will vary for each service. Further, it is still unknown or unclear how an end user will perceive a delivered multimedia service (e.g. VoIP or video streaming service) given a set of QoS parameters (e.g. 5% packet loss rate and/or 200ms of delay). Therefore, the operators will need to focus progressively on providing superior user perceived service experience, independently of technical requirements.

This leads to utilise the concept of Quality of Experience (QoE), which is defined in [10] as the overall acceptability of a service, as perceived subjectively by the end user, to LTE scheduling. Mean Opinion Score (MOS), ranging from 1 (Bad Quality) to 5 (Excellent Quality) [11], the most commonly used QoE metric is adopted in the thesis to develop QoE-aware LTE scheduling schemes

Due to the fact that the operation cost should be minimized and the utilization of the available radio resources should be as efficient as possible, Radio Resource Management (RRM) is considered the key tool to be focused on for improvement on resource allocation and utilisation when delivering multi-user multimedia services. It has the functions that can be configured to improve the current mobile telecommunication networks. Due to limited radio-frequency spectrum resources of mobile networks, it is always a challenge to design and develop efficient resource allocation algorithms which are able to provide satisfactory QoE for as many as possible end users in delivering multimedia services. This forms up the main goal of the thesis is to develop a novel QoE-driven scheduling schemes for improving system capacity in delivering multimedia services over downlink 3GPP LTE.

1.2 Research questions

This thesis seeks to address the following three key research questions:

Q1) What is the QoE performance of existing typical LTE downlink scheduling algorithms for multimedia applications?

This question led to a significant and thorough investigation of LTE downlink scheduling algorithms and available simulation tools, which could provide a similar test environment for analysing and evaluating the existing LTE downlink scheduling algorithms. The VoIP application was selected as a case study to observe the QoE-based performance of several existing scheduling algorithms (e.g. proportional fairness (PF), exponential proportional fairness (EXP-PF), and modified largest weight delay first (M-LWDF)) in terms of QoE metric (i.e. MOS score) in addition to the normal QoS metrics (e.g. packet loss and delay). The capacity of existing LTE scheduling algorithms in terms of the number of users meeting satisfactory QoE was assessed and limitations of these algorithms were analysed and identified in detail. This work will be discussed in Chapter 4.

Q2) How should the LTE downlink scheduling algorithms be optimized to be QoE-driven for VoIP application?

Recently, many LTE downlink scheduling algorithms were proposed to improve the perceived quality for multimedia applications. For example, in 2012, Fei Liu [12] proposed scheduling targets to maximize the user QoE. This concept was based on packet delay as a main factor affecting QoE as well as the data rate. Also, in 2013, Nabeel Khan [13] proposed a downlink scheduling scheme for scalable video over LTE, which improved the perceived video quality for all users by considering the video frame importance and bit throughput.

The maximization of QoE for multimedia application over LTE networks is a hot research topic today. However, the existing research is only focused on maximising user QoE for a limited number of users. Little efforts have been put on investigating how to utilise mobile resources to increase the number of mobile users with satisfactory QoE. This work is particularly important in mobile resources utilisation when system capacity is the key. Increasing capacity will obviously bring more profit to mobile operators and service providers.

To address this question, a QoE-driven LTE downlink scheduling algorithm for VoIP application was proposed and developed to maximize the number of users per cell at an acceptable MOS score. This work will be discussed in Chapter 5.

Q3) How should the LTE downlink scheduling schemes be optimized the QoE in a general multi-service, multi-user, and multi-cell scenario?

This question led to the proposal of the QoE-aware Multi-Service (QoE-MS) LTE Downlink Scheduling schemes for multiservice multimedia applications, thus answering the two fundamental research questions in this area: What is the performance of LTE scheduling with different real-time multimedia applications in terms of QoE?

The first question leads to another question which is:

Does video content type have an impact on QoE optimization under LTE scheduling in video streaming applications?

This work will be discussed in Chapter 7.

1.3 Project aims and objectives

The main aim of the project is to develop and evaluate a novel QoE-driven downlink scheduling schemes to improve LTE system capacity for multimedia services.

The specific objectives of the research are to:

- Investigate and evaluate the performance of existing LTE scheduling algorithms in terms of QoE (i.e. MOS) instead of the current QoS parameters (i.e. delay and packet loss)
- Develop a novel QoE-driven scheduling scheme for VoIP applications to improve the LTE system capacity with satisfied QoE for all users.
- Evaluate the performance of a novel QoE-driven scheduling scheme for VoIP applications over LTE networks.
- Develop a novel QoE-driven Multi-Service (QoE-MS) scheduling scheme to enhanced the LTE system capacity with satisfactory QoE for VoIP and video streaming services
- Evaluate the performance of the proposed QoE-MS LTE scheduling schemes with different video contents for video streaming services.

1.4 Contributions

The main contributions of the thesis are threefold:

1- The performance of several existing scheduling algorithms (e.g. PF, EXP-PF and MLWDF) for VoIP applications was evaluated in terms of QoE metric (i.e. MOS), instead of normal QoS metrics (e.g. packet loss and delay). These results will assist the development of QoE-driven scheduling schemes in order to achieve optimised end-user experiences or optimised mobile system capacity.

(The associated publication is [14])

2- A QoE-driven LTE downlink scheduling algorithm for VoIP application was proposed and developed to maximize the number of users per cell at an acceptable MOS score. The proposed algorithm achieved remarkable improvement in cell capacity by 75% at an acceptable MOS score compared to MLWDF, 250% compared to PF and EXP-PF in all three lower-speed scenarios considered in the simulated system.

(The associated publication is [15])

3- A novel QoE-driven LTE Downlink Scheduling scheme for multiservice multimedia application was proposed and developed to improve the cell capacity with satisfactory QoE for both VoIP and video streaming services. The proposed algorithm performed well in pedestrian scenarios by doubling cell capacity t for video stream with Rapid Movement (RM) content. For Medium Movement (MM) video content, the capacity increased about 20% when compared to MLWDF and 60% when compared to EXP-PF. Within the vehicular scenario, the proposed system has managed to enhance the cell capacity for the MM video stream case.

(The associated publication is [16])

1.5 Thesis outline

The outline of the thesis is shown in Fig. 1.1 and described as follows:

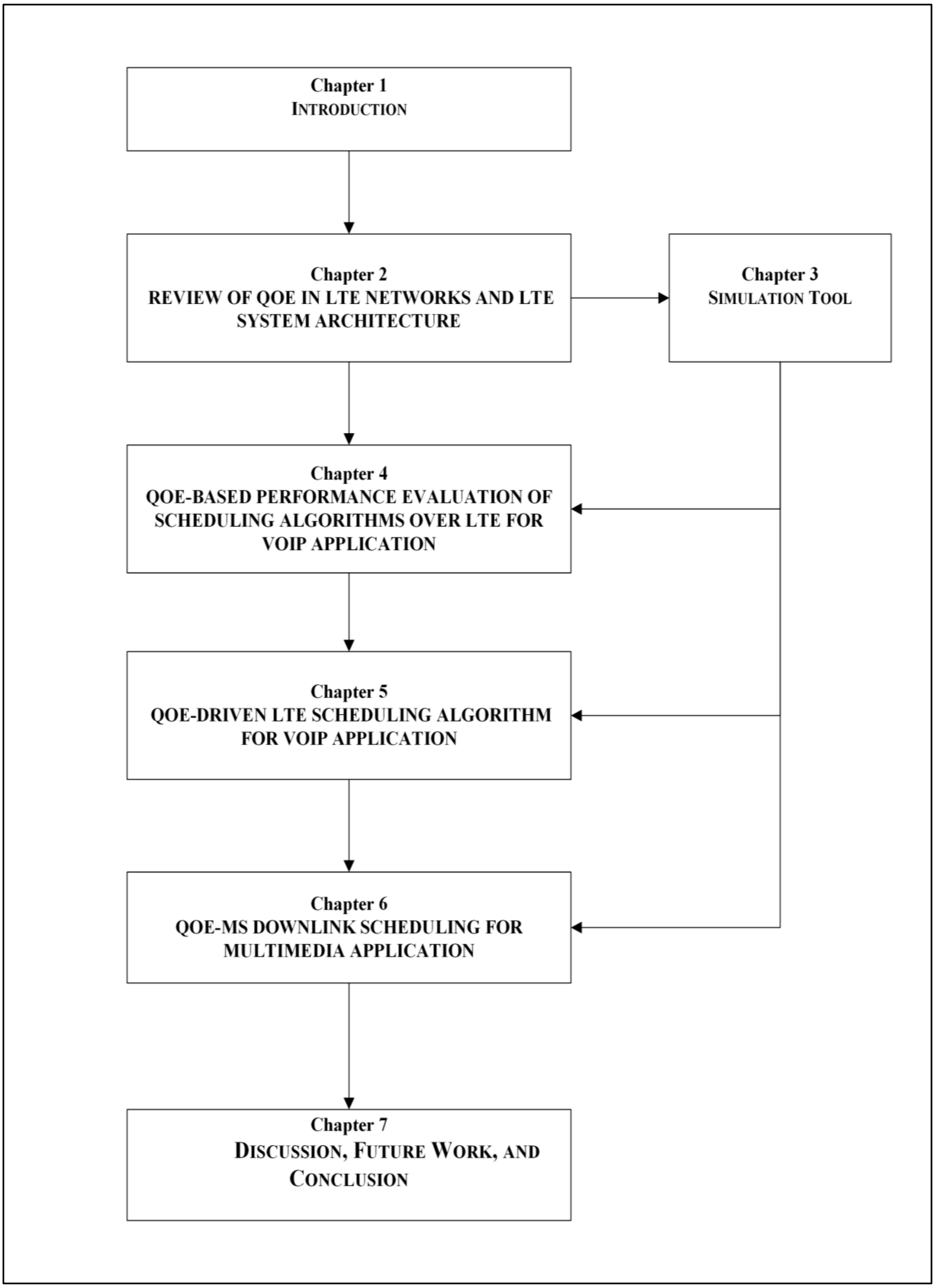


Figure 1.1: Outline of thesis

Chapter 2 REVIEW OF QOE IN LTE NETWORKS AND LTE SYSTEM ARCHITECTURE

2.1 Introduction

In today's market, the wireless communication system is more widespread than wired system because high mobility and offer cheap deployment. From the user's perspective, it can be seen that these systems are widely used for most of the user's personal communication. These wireless communication systems were intended to fulfil the least possible requirements of users (i.e. voice calls), which resulted in the introduction of first generation (1G) mobiles. But soon after the requirements of users became greater than before, which stemmed further improvement in these systems and from 1G to 4G mobiles. During this progress, the most important elements to be considered, from the user's perspective, were the quality of user experience (QoE) and the quality of service (QoS). These two elements need to be maintained in order to achieve the high level of users' satisfaction. Therefore, in the wireless communication systems, QoS and QoE managements play a noteworthy role, especially in the design phase [17]and [18].

According to Rao, et al. [17], the quality of service is considered a significant concern in designing any system from both the system provider and the user's perspective. As users have expectations to receive the best quality services by using these systems and providers are keen to deliver the best quality service from the system, it is important from both standpoints and for the wireless communication systems as well. There are different factors of quality of service that are significant variably for different users and different applications. The different factors of QoS also determine how satisfied the users are, so different factors decide the different users' satisfaction levels. Therefore, QoS and QoE can be defined differently on the basis of various applications and the user's satisfaction levels. However, this chapter aims to define these two terms, QoS and QoE, from two distinct viewpoints; that is, from the technical viewpoint and the nontechnical viewpoint. Fundamentally, QoS is such a quality that is delivered to the users and can be described as the capability of a network with a guaranteed level of service, and the QoE is basically defined as the end users' opinions about the functionality of the services to which extent the services are useful [17], [19], and [20].

The aim of this chapter is to provide an analysis of important aspects associated with QoE in LTE networks and an overview of LTE system and its design. Section 2.2 provides basic knowledge of QoS and QoE. Section 2.3 describes the LTE system design. Section 2.4 introduces LTE system architecture. Section 2.5 discuses the scheduling algorithms in LTE system. Section 2.6 describes the QoS and QoE of LTE networks. Section 2.7 presents VoIP quality assessment. Section 2.8 provides an overview of video quality assessment. The chapter warp up in section 2.9 with summary.

2.2 The Basics of Quality of Service (QoS) and Quality of Experience (QoE)

The Internet has been present for 40 years, and during these years, it has progressively advanced, commencing from a minor network in which the connectivity was crucial, and advanced to a huge media-rich network wherein the crucial aspect is the user. The users have become significant, as they are not only involved in consuming the content, but also in producing the new content dynamically. In accordance with the advancement of Internet from a small network of media-rich sites, the requirements and demands of the users also increased, as the users exceeded the demands over connectivity, and now they count on the services that must be provided to them in equivalence to their requirements on quality. Subsequently, in contemporary years, much of the research has been devoted to the ways to measure the user QoE [21].

There are a number of standards regarding QoE that have been suggested, and some are in the process of development in the International Telecommunication Union (ITU). The QoE is defined by the ITU-T Focus Group on IPTV (FG IPTV) [22] as the complete acceptability or satisfactoriness of a service or application in the same way as recognized instinctively by the end-user. QoE, in that way, comprises the comprehensive end-to-end system outcomes (network, user, services infrastructures, terminal, etc.), where the complete acceptability of service or application can be affected by the context and the end-user anticipations. This delineation of QoE clearly denotes it to be a subjective measure, which means that, in order to measure QoE accurately, there must include tests with the real-life users, and the testing with real users would be an expensive and time-consuming procedure. Therefore, a network or service provider must preferably have some tools that can measure it objectively by indicating the results within the rational precision of the subjective mean opinion score by the end users [20], [23].

According to Laghari & Connelly, “quality of experience (QoE) is such a subject that is emerging promptly and is involved with multiple disciplines, such as economics, social psychology, engineering science and cognitive science, driven by comprehending the general human quality requirements.” [24] Thus, it can be regarded as an outline of total human quality requirements and anticipations. Conventionally, technology-based methodologies driven by QoS factors have been put to use to guarantee the delivery of service quality to users. QoE enlarges its boundaries, where it also tries to fulfil users’ appeals and also the lofty requirements. Thus, QoE can be described as a system of all human objective and subjective quality requirements and encounters resulting from the user’s interface with technology as well as with corporate units in a specific perspective [24].

Nowadays, a predominant area for research is to embrace a complete comprehension of the quality as the end users expect (QoE). In the case of low service quality provided to the user, it is too expensive for a service provider to wait for users' criticisms. As found in an Accenture survey [25], almost 90% of end users simply switch to the other providers rather than complaining about the quality, which makes it too expensive for the provider to wait for complaints. Thus, it is of high significance that service providers must have ways to recurrently compute the QoE from time to time in order to advance it, if required. There are a number of elements on which the perceived quality depends and is affected by, such as the terminal performance, the reliability of the network, and the procedure of the content production. The multimedia issuing services on Internet Protocol networks' QoS are dependent on many mutually dependent factors. Among these, some factors can be corrected after issue is identified, for example image resolution and bandwidth, whereas other factors cannot, such as delay and packet loss ratio. Such important, but neglected, factors need to be considered for improving the satisfaction level of the end user. However, it is not only the QoS factors that affect the user's level of contentment, but also a number of subjective factors related to QoE, for instance an end user's interests, expectations, and experience. Many researchers use diverse methodologies on the basis of media type, such as image, video, and voice, since on the basis of these media types there are many measurement approaches with diverse computational and functioning requirements [26], [27].

Two central quality measurement approaches are applied, which are known as objective and subjective measurement. When the quality of experience is measured and rated for a video application by users, then it is quite subjective measurement. Among the subjective measurement approaches, the Mean Opinion Score (MOS) is the most prevalent for assessing quality. This approach is standardized in the ITU-T recommendation [11]; in addition, it is demarcated as a numeric value starting from 1 (used for poor quality) to 5 (used for excellent quality). But there are some disadvantages of this methodology: it is quite expensive, it requires a great deal of time, it cannot be repeated, and it also cannot be practiced in real time. Such let downs have stemmed the advancement of objective methodologies, which anticipate the subjective quality exclusively from bodily features. The objective method can be explained as being a comparative or a mathematical tool that produces a quantitative calculation of the quality of a one-way video application. Commonly, intrusive approaches are precise but not viable for observing live traffic due to the requirement for the original copy, for instance, full reference quality assessment. Non-intrusive techniques are not in need of a duplicate of the original [18] [28].

These two approaches have their own disadvantages that can be overcome by using these two collectively. The PSNR-mapped-to-MOS method is normally implemented to assess video QoE upon which the network conditions have effects. Many scholars have found this method to be imprecise with regard to the relationship to the observed visual quality. Nonetheless, for improving the estimation precision, many enhancements have been suggested. A number of practitioners have approved the new enhanced version of PSNR-mapped-to-MOS technique, for example the recommendations presented in ITU-T in J.144 and ANSI T1.801.03 Standard [28].

From the end-user viewpoint, there are three classes of techniques used to measure service quality [29]: Testing User-perceived (TUQ), Modelling Media Quality (MMQ), and Surveying Subjective QoE (SSQ). Among these, TUQ and SSQ are involved in gathering subjective information from end users, while MMQ is based on objective methodological assessments. Significant consideration is devoted to examine the quality on the basis of network-based QoS and end-user-based QoE factors. It highlights the requirement of getting an understanding of the major techniques, wherein the network QoS factors have their influences on the measurable quantities of QoE. As the QoE can be calculated and quantified, a map linking these QoE measures with the QoS factors can be developed. Therefore, a useful QoE-responsive QoS framework can be developed. There are many objective frameworks developed for approximating QoE. By concentrating on every framework, the International Telecommunication Union (ITU) has established the Focus Group on IPTV (FG-IPTV) to standard these frameworks [30]. Commonly, there can be five categories of models of objective quality valuation approaches [30]:

1. *Parametric planning model* uses inputs in terms of quality planning considerations for terminals and networks. Also, such a model entails theoretical data regarding the system under evaluation.
2. *Parametric packet-layer model* makes use of packet-header data in order to forecast QoE, regardless of managing the media signals itself. It faces issues examining the content reliance of QoE, because it does not take the payload information into account.
3. *Media layer model* makes use of media signals through SSIM in order to estimate the QoE, but if these signals are not there, then this model becomes impracticable.
4. *Hybrid model* is the most effective and a joint model of all those mentioned above; it can make the most of the information for estimating the QoE.
5. *Bit-stream model* is placed between the media layer and parametric packet layer models. It is a novel approach that stems the quality through obtaining and examining the data features as of the implicit bit-stream.

2.3 LTE system Design

For GSM (mobile communications) as well as Universal Mobile Telecommunications System (UMTS) standards, the LTE standard has developed beyond the global system and is generally known as the 4G system. The very first application, for which the information was improved, was voice communication. From the beginning, the flexibility and all-in-one handoff were demanded, in the same way that the central organization of all parts was demanded. Regarding the speed of LTE, it is said that it would be comparable to a home-based modem speed. Its architecture allows 150Mbps DL (downlink) and 50Mbps UL (uplink) within the across-the-board range. Despite the fact that LTEs' speculative maximum uplink speed is 150 Mbps, the existing bandwidth and the way the bearers organize their network would decide every user's bandwidth. The crucial design contest is to maintain high rates along with curtailing the power. There is a novel physical layer in LTE as it contains irregular modulation as well as UL and DL data rates. The LTE is devised for the maximum duplex procedure, with instantaneous sending and receiving information. As the transmitter at the central place has a lot of energy, the radio is enhanced for functioning on the DL. While the radio is enhanced further for energy intake rather than efficiency over the UL, the mobile battery power remained basically steady even with an upsurge in the processing power [27], [31], [32], and [33].

In comparison to the 3G systems, LTE is advancement, and its design is based on Internet Protocol (IP). Formerly, an isolated Radio Access Network (RAN) made up of the Medium Access Control (MAC), Radio Link Control (RLC), and Radio Resource Control (RRC), protocols were employed for interfacing with the user equipment in 3G systems. However, currently the eNodeB used in LTE manages all of these protocol tasks that make it employ few nodes, which help in lowering the system inactivity as well as enhance the whole performance. While the LTE's network design encompassed a basic network Evolved Packet Core (EPC) as well as the access network Evolved-Universal Terrestrial Radio Access Network (E-UTRAN), the eNodeB certifies the required QoS in the access network that must be met on the interface for a carrier. Every carrier contains a linked QoS class identifier (QCI), in which every class is categorized on the basis of bearable delay and packet loss and priority [34], [35], and [31].

There are two common categories of bearers, which are distinguished on the basis of their delivered QoS type: Guaranteed Bit-Rate (GBR) and non-GBR bearers, where the GBR are real-time bearers and non-GBR are not. LTE helps both the modes, Frequency Division Duplex (FDD) and Time Division Duplex (TDD), at the physical level. According to Sulthana & Nakkeeran [34], the evolving applications with different delay, bandwidth, throughput, and Packet Loss Rate (PLR) conditions call attention to the requirement of a network that can upkeep a number of services. This requirement was considered by the Third Generation Partnership Project, from which LTE was originated. Improving the data rate in order to deliver the radio resources for a number of services high in demand is the core purpose of the LTE network, whereas it also takes a contended level of QoS into account for all end users. LTE fulfils this demand by employing Single Carrier-Frequency Division Multiple Access (SC-FDMA) technology and Orthogonal Frequency Division Multiple Access (OFDMA) technology in the Uplink (UL) and Downlink (DL), respectively. Among these, the OFDMA technology function by splitting the existing bandwidth up into a number of sub-bearers and then assigns a cluster of these sub-bearers to the user according to its QoS needs. Based on the frequency and time areas, the existing bandwidth is split into a number of Resource Blocks (RBs), which are the tiniest apportionment units that can be controlled freely. RB contained 12 sequential sub bearers in frequency area, while there is a slot of only a 0.5ms time period and 2 slots being in the apportionment period made up a RB in the time area. One Transmission Time Interval (TTI) equalling to the 1ms time period is the scheduling period [34], [35], and [31]. The number of RB varies in a resource grid, depending on the size of the bandwidth. The LTE bandwidth and number of RBs are shown in Table 2.1.

Table 2.1: The LTE bandwidth and number of RB

Bandwidth (MHz)	1.4	3	5	10	15	20
Number of RBs	6	15	25	50	75	100

Therefore, to achieve the system execution goals, a planned procedure for effective resource apportionment is significant for real usage of radio resources. While allocation of parts of the spectrum mutually shared among users is performed by the packet scheduler at the radio base and the network functioning depends on the algorithms employed for the packet scheduler, it is a significant job to build an effective scheduler for distinguishing the performing of a wireless system from others [34] [36] [37].

2.4 LTE system Architecture

The LTE Architecture shown in Figure 2.1 describes the part that interrelates with the user equipment (UE) or a mobile at the maximum intimacy. Overall, the LTE architecture is quite complicated, where a thorough picture would explain the whole internet and further parts of system connectivity maintaining the handoffs including WiMax, 2G, 3G, and additional standards. It demonstrates the eNodeB (also called base station) and its crossing points with the user equipment. The whole network is the E-UTRAN, the certified name of the LTE standard [27], [32].

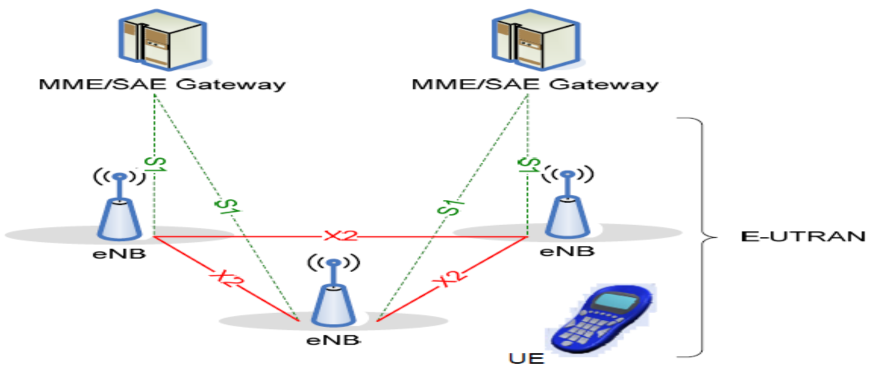


Figure 2.1 LTE Architecture [9]

The LTE PHY is normally a maximum duplex, as LTE is basically developed for maximum duplex functioning in the spectrum that is paired. On the other hand, the WiMAX functions in the partial duplex on the spectrum that is unpaired in which the data is sent only in a single route in a single point in time. Though LTE can maintain TDD functioning within the unpaired spectrum, this is not the main concern. The PHY works constantly for the DL with scattered sync, supplying manifold channels instantaneously with fluctuating inflection. Thus, the DL channel functions as an unceasing flow. In contrast to the IEEE® 802 family standards, no linkage can be found between the real service data unit (SDU) packages (approaching from the highest place of protocol packs) and the air interface (conveyed frameworks on the air). The idea of a resource block is adopted in LTE, where there is a block having 12 sub-carriers in a single slot, and the transport block is made up of these resource blocks with joint coding or modulation. The physical layer of LTE is basically a transport block that matches the information transferred in a single point in time of the apportionment for a specific UE. A radio sub-frame has a time period of 1 millisecond (ms), while every frame is 10ms. Many users can be provided with the service on the DL in a single transport block any time. The MAC is responsible for managing which information is to be transmitted any particular time [32], [33].

2.4.1 Control Plane Protocols – Radio Resource Control (RRC)

The Control Plane manages the tasks particular to the radio, and it is contingent upon the user equipment's states, whether UE is idle or it is connected. When the UE is in idle state, the user equipment makes groups on a cell when it has chosen or reselected a cell, and in this procedure, a number of aspects are carefully thought out, such as cell status, radio access technology, and link quality. Additionally, the user equipment scrutinizes a contacting network to identify received calls and obtain system data. In this state, the control plane protocol encompasses the procedure of selecting and reselecting a cell. Contrasting this state, when the UE is in its connected state, it provides the E-UTRAN with neighbouring cell data that contains added frequencies, radio access technology, and DL channel quality in order to allow the E-UTRAN to choose the most appropriate cell for the user. Here, the CP protocol comprises the RRC protocol. There are following practical areas embraced by the RRC protocol [32], [38], [39], [40].

2.4.2 System Information

It is responsible for dissemination of the system information about a category appropriate for the idle and connected states. It can be described as System Information Blocks SIBs) with diverse factors. In master information blocks (MIB), there are 8 SIBs described as having some quite often transferred factors that are crucial to user equipment's early entrance to the network. System information is plotted against diverse reasonable channels contingent upon the user equipment's state and data form [32], [38], [39], [40] RRC Connection Control.

It contains processes for creation, amendment, and issue of RRC networks for safety activation, calling, Signalling Radio Bearers (SBRs), handovers, Data Radio Bearers (DRB), and additional purposes for example arrangement of inferior protocol layers. For curtailing the E-UTRAN costs and dealing out user equipment which is recorded on the MME for protecting the battery power, the data associated with user equipment can be discharged quite a bit later, after the data idleness wherein the MME would recollect the user equipment situation and recognize carrier data throughout these idle times.

"These situations are denoted as EPS Connection Management (ECM) idle and connected states. At the MME, the stage of user equipment is taken through the EPS Mobility Management (EMM) state, besides it can be listed or unlisted" [32]. The changeover among the ECM Idle and Connected states happened when the RRC connection is made. The RRC organizes the inferior layers (RLC, PDCP, PHY, and MAC) while locating the DRBs. For instance, PDCP is commanded by RRC to put on the header compression for VoIP packs, or in other cases, RRC commands the MAC to put on the hybrid ARQ (HARQ) for postpone-bearing traffic and allot the Prioritized Bit-Rates (PBRs) to regulate the way a user equipment splits the UL resources among diverse radio carriers [39] [40].

2.4.3 Network Controlled Mobility

This element contains flexibility processes, security activation, as well as transmission of user equipment RRC background data [39], [40].

2.4.4 Measurement & Configuration Reporting

It is necessary to maintain flexibility purposes. Besides the RRC, the RLC and PDCP execute their jobs on the control plane information, as discussed below [39], [40]:

User Plane Protocols

The following layers are contained in a user plane:

- Packet Data Convergence Protocol (PDCP) Layer
- Medium Access Control (MAC) Layer
- Radio Link Control (RLC) Layer

2.4.4.1 Packet Data Convergence Protocol Layer

Letting the transfer of information along with signalling in an integrated manner is the core purpose of PDCP. It further controls the compression as well as de-compression of the header and data of IP file packages. It also helps in the coding and decoding of the information pieces within both the control plane and the user plane, an in addition it can organize the reliability defence and authentication in the control plane. A sequence number is entered in the header, which is used to recognize the PDCP Protocol Data Unit (PDU). It guarantees an in-sequence supply of upper-layer PDUs plus the removal of a replicated low layer Service Data Units (SDU) during the restoration of the lower layers (for example, delivery). A PDCP PDU has a maximum magnitude of 8188 bytes. It can be seen that every PCDP PDU can be distinctively recognized in a specified radio access network through areas such as LCID, RNTI, and PCDP. Or it can also be said that this triad recognizes the transfer of the information piece on a specified bearer of particular user equipment [32], [38], [39], [40]

2.4.4.2 Medium Access Control Layer

There are several crucial purposes fulfilled by the MAC layer, where the scheduler is also included that allocates the existing bandwidth to many active users. The employment of the scheduler differs among different sellers, where the main point of distinguishing is product performance. Another MAC job is the random access procedure control that is employed by the user equipment, which is not assigned to the UL radio resources for retrieving and coordinating through the network. The MAC furthermore implements the UL timing configuration, which avoids the overlapping of user equipment transmissions while getting to the central station. Additionally, the discontinuous reception is executed at this layer to protect the battery power through constraining the time during which the UE obtains DL networks at the extra price of additional potential. Moreover, it performs the HARQ process to retransfer and unite the obtained information pieces (transport blocks) as well as make NACK or ACK indicating whether the CRC let down happens. HARQ tries to modify information by joining manifold broadcasts of the information, even if each transmission has inaccuracies. If it does not go well in recovering the right information, the ARQ job at RLC is appealed to originate retransfers and re-division for any manipulated PDUs. LTE employs coexistent HARQ on the UL, where the retransfer happened on the already demarcated time compared with the early broadcast and nonparallel HARQ on the DL, where transference can happen at any time corresponding to the preliminary broadcast, thus necessitating extra data to specify the HARQ procedure number for allowing the receiver to link the re-transference with its matching transference. Even though the real maintenance and rearrangement of information is performed thru the PHY, the MAC does the organization and indicating. Also, the MAC layer plots the RLC data obtained via reasonable networks on transport networks linking the MAC layer with the PHY layer. The opposite process is performed on the receiver [32], [38], [39], [40].

2.4.4.3 RLC (Radio Link Control) Layer

In the transfer route, the RLC is charged with reconfiguring the PDCP PDUs, called division or/plus the concatenation, to suit the range necessary in the transport block or the MAC layer. While on the receiving course, the RLC rebuilds the PDCP PDUs. Five factors determine the transport block size, which include the kind of application, distance, modulation arrangement, power supplies, and bandwidth requisites. In the RLC PDU, the division as well as concatenation can be existed simultaneously. The RLC correspondingly rearranges information packages obtained via sequence through the HARQ process. It connects with the PDCP via the Service Access Point (SAP), while it also links with MAC via reasonable networks. The information is transferred in RLC via three modes: Acknowledged Mode (AM), Unacknowledged Mode (UM), and Transparent Mode (TM). The TM is a navigation mode that draws RLC SDUs into the RLC PDUs and contrariwise, exclusive of any operating costs or alterations made in data packages. It can merely be operated for a few control signalling, for example paging, messages as well as transmission system information. The UM mode functions for postponed complex traffic, for instance VoIP. The multimedia broadcast/multicast service (MBMS) is the point-to-multipoint traffic that employs the UM mode. The layer executes the division of RLC SDUs, reorganization and replica recognition of RLC PDUs, and recollections of RLC SDUs in the UM mode [32], [38], [39], [40].

The AM mode helps to maintain the forbearing of interruption however error vulnerable the traffic is (impractical applications like web browsing). It permits info transmission in both routes in which RLC can transfer and get information. It performs Automatic Repeat re-Quest (ARQ) for modifying the mistaken data packages beyond transfer of information (that is, RRC messages) that also employs this mode [32], [38], [39], [40].

2.5 Scheduling in LTE

The multiuser scheduling is the most prominent attribute in LTE systems, as it is responsible for ensuring the QoS of all active users. Some of the important scheduling mechanisms are discussed below:

2.5.1 Downlink scheduling algorithms

A resource block that comprises frequency and time domains contains the resources that are being transferred within LTE. The 3GPP LTE system contains components known as eNodeB, which are base stations. The scheduling packet is operating in conjunction with other radio resource management (RRM) mechanisms.

Each eNodeB would receive a notification of instantaneous downlink channel conditions such as signal-to-noise ratio (SNR) from the users within each of the TTI. Several parameters were used by the eNodeB packet scheduler to determine the user priorities. It is possible for eNodeB to obtain the full information regarding the quality of the channel; such parameters include head of line (HOL) packet delays, buffer status, and service types by utilising the channel state information (CSI) and channel quality indicator (CQI).

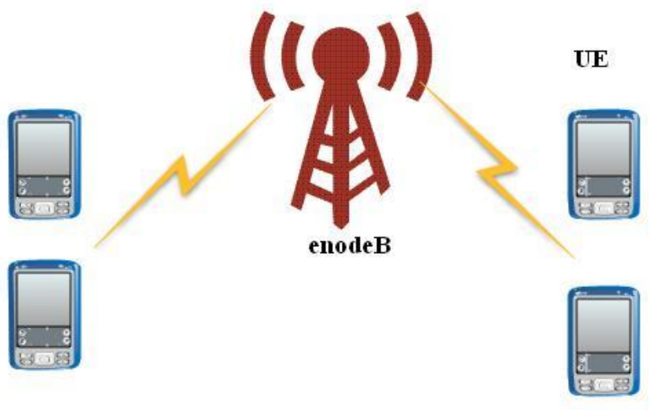


Figure 2.2: LTE downlink system

According to the criteria of first-in and first-out, packets approach in the buffer at eNodeB. These packets are queued for the transmission, and the HOL is calculated. Also, if any delay arises, it is computed as well, and in case the packet is delayed beyond a certain threshold of time, it is discarded and consider as packet loss.

The main purpose of LTE scheduling is to fulfil all of the demands regarding Quality of Service (QoS) for all users through attempting to achieve the most favourable compromise among effectiveness and impartiality. This objective is extremely daring, specifically due to the occurrence of present multimedia applications that are categorized by strict limitations on the data package delay and jitter. The system performance is enhanced through the augmenting of the network spectral effectiveness and impartiality, which is the objective of the radio resource allocation algorithms. Reaching the optimal level among effectiveness and justice between users is highly important. There are a number of categories and groups of algorithms present in the literature, where every group has a collection of algorithms with shared attributes [41], [42], [43].

2.5.1.1 Opportunistic algorithms

Opportunistic scheduling contemplates the user in which backlogs are constantly accumulated, where these safeguard settings are normally employed to demonstrate flexible or the finest effort flows. The central purpose of this algorithm is to optimize the whole system output. At present, a large number of algorithms employ this method, for example, Proportional Fair (PF), Proportional Fair Exponential (EXP-PF), and the Maximum Largest Weighted Delay First (MLWDF) scheduler, which are based on delay, and are opportunistic scheduler [41], [42], [43].

2.5.1.1.1 Proportional Fair (PF)

Proportional fair algorithms [44] are employed in High Data Rate (HDR) networks and implemented to opt between the total data rate and a fair data rate of each user. PFs assign radio resources after analysing the experienced quality of channel as well as the past user throughput, and for this property, PF is rendered as a perfect scheduling choice for non-real traffic. In order to ensure fairness among the flows, the objective is to maximize the entity, “total network throughput”.

2.5.1.1.2 Maximum-Largest Weighted Delay First (M-LWDF)

M-LWDF is a type of algorithm designed for the purpose of supporting multiple real time data users in CDMA-HDR systems [45]. M-LWDF considers instantaneous channel variations, and in the case of video service, it takes in consideration the delays in this service; thus, it configures multiple data users having different Quality of Services (QoS) requirements. M-LWDF makes use of the information about the channel state by striving to maintain a balance between the weighted packet delays.

2.5.1.1.3 Exponential Proportional Fairness (EXP/PF)

Exponential Proportional Fairness (EXP/PF) is an algorithm that configures the multimedia applications in a system of time division multiplexing (ACM/TDM system, adaptive modulation and coding). This type of algorithm can have both the real-time service user as well as non-real-time service one [46], and so it can enhance the priority of real-time flow with respect to no-real-time flow.

2.5.1.2 Fair algorithms

It is important to note that equity/justice does not signify equality, while the central purpose of fair algorithm is to attain justice and equity among users. Commonly, such algorithms have a deficiency with regard to the spectral proficiency. Numerous works have considered justice among users, such as, Max-Min Fair, Round Robin, and game-theory-based algorithms [41], [42], [43].

2.5.1.3 Round Robin (RR)

The round robin is a well-known algorithm that is generally employed in the radio resource apportionment due to its simplicity and low level intricacy, as it is devoted to consider the justice issue among users. Thus, the algorithm apportions a similar quantity of resource allocation. As the algorithms do not contemplate the described SINR standards while execution of the distribution procedure, this policy does not also have spectral proficiency and output performance [41], [42], [43].

2.5.1.4 Max-Min Fair

The algorithm apportions the resources between all users sequentially so as to multiply the data rate for every user. As soon as the user allocates resources needed to attain its necessary data rate, meanwhile the algorithm chooses alternative users for arrangement. And after contentment of all users and allocation of resources, the algorithm halts [41], [42], [43].

2.5.1.5 Throughput-Based Algorithms

The throughput-based algorithm attempts to optimize the objective role, which signifies the data rate, wherein this method deals with the real-time streams as well as non-real-time. In this, the size of every user's queue decides the resource distribution. For instance, the Max-Weight and EXP Rule is in [41], [42], [43].

2.5.1.6 QoS-Based Algorithms

Such an algorithm considers the spectral efficiency of immediate or non-flexible streams, such as VoIP and video, in fact, it attempts to amplify the objective role signifying the data rate. This method considers the real-time streams and also non-real-time, reflecting that there is no need to give these any precedence. The size of every user's line decides the resource distribution [13, 14, and 15].

2.5.1.7 Delay-Based Algorithms

These algorithms and HOL are central constraints of such schedulers. This method deals with the non-flexible streams. While a package surpasses it's HOL, and it would be eradicated from the line. M-LWDF is such an algorithm that is driven by delay and is simultaneously an opportunistic one as well [41], [42], [43].

2.5.2 Uplink Radio Resource Allocation

Contrasting to the DL scheduling, the UL scheduling is quite complex for many reasons. First, user equipment transmits the data to eNodeB, even though the UE has a narrow power source. Another reason is that it is quite challenging to foresee the quantity of radio resources, which are required by UE for interchanging the data with the central station. There are three classes of schedulers on the basis of objective role reflected and the traffic categories that are transmitted to the radio channels; the best effort scheduler is managing premium effort streams, which also considers the quality of service and those enhancing the power transfer [41], [42], [43].

2.5.2.1 Paradigms of matrix construction

Designed for the LTE UL radio resource distribution, the scheduler takes input as a UE-RB connection matrix for delivering the highest outcomes that enhance the system performance. While in the matrix production procedure, there are two chief arrangements or models, which are the proportional fair (PF) and channel dependent (CD), in this procedure, the CD model contemplates the channel state information (CSI) for the purpose of providing a chance to assign more resources for the users having greater CSI standards, this method provides greater output, but faces a malnourishment issue. In the meantime, the PF model employs the proportion of CSI as well as every user's data rate; therefore, justice is comparatively on CSI standards. This method attained beneficial outputs and, simultaneously, resolves the malnourishment issue [41], [42], [43].

2.6 QoS and QoE in LTE Networks

The previous downlink scheduling strategies are driven by QoS parameters without consideration of subjective perceived quality as discussed in above mention. For example, an Opportunistic Packet Loss Fair (OPLF) scheduling algorithm was proposed to enhance the performance in terms of throughput, PLR and fairness among users [47]. It depends on HOL packet delay, PLR and achievable instant downlink rate for each user to calculate a simple dynamic priority function. Considering the fact that maintaining and improving user satisfaction becomes very important to network providers; therefore, QoE-driven schedulers has received significant attention in both academic and industry such as; the QoE-oriented scheduling algorithm has suggested dynamically prioritizing YouTube users over other users if QoE degradation is imminent [48]. Aristomenopoulos et al. proposed a methodology for integrating QoE into a network's radio resource management (RRM) mechanism by exploiting network utility maximization theory [49]. The QoE-driven cross-layer optimization for wireless video streaming that takes into account two objectives: utility maximization MOS and utility max-min fairness [50]. Also many cross-layer optimization schemes were developed to maximize the QoE and network resource utilization (e.g. [51] , [52] and [53]). An enhanced QoE-oriented packet scheduling algorithm was proposed to reduce the pauses during video playback in a wireless environment. Both channel quality and the total of video data stored in the player buffer are considered in the process of resource allocation. The proposed scheme decreases the number of pauses during video playback and improves the QoE [54].

2.7 VoIP Quality Assessment

Lingfen Sun developed nonlinear regression models to predict perceived voice quality nonintrusive for four modern codecs (i.e., G.729, G.723.1, AMR, and iLBC). This model was verified using two schemes. For Scheme I is used PESQ/E-model to predict MOS for VoIP. However, Scheme II is used a nonlinear regression model [55]. This model was obtained high prediction accuracy from the regression models (correlation coefficient of

0.987 for Scheme I and 0.985 for Scheme II, respectively) using real Internet VoIP trace data. The MOS for VoIP traffic can be predicted using the following equation (Equation 2.1) to determine the MOS score from network packet loss and end-to-end delay [55]:

$$\boldsymbol{MOS = a + bx + cy + dx^2 + ey^2 + fxy + gx^3 + hy^3 + ixy^2 + jx^2y} \tag{2.1}$$

Where $\boldsymbol{x}$ represents the packet loss ratio (packet loss ratio in percentage) and $\boldsymbol{y}$ is the end-to-to end delay (delay in ms). The parameters for fitting surfaces with different codec are shown in table 2.1.

Table 2.2: SURFACE FITTING PARAMETERS FOR DIFFERENT CODEC [55]

Parameters	**AMR**	**iLBC**	**G.729**	**G723.1**
a	**3.91**	**3.64**	**3.61**	**6.45**
b	**-0.17**	**-5.25e-02**	**0-0.13**	**-9.99e-02**
c	**1.57e-03**	**2.45e-03**	**1.22e-03**	**1.28e-03**
d	**6.51e-03**	**1.34e-03**	**3.76e-03**	**2.73e-03**
e	**-2.40e-05**	**-2.71e-05**	**-2.29e-05**	**-2.31e-05**
f	**-7.53e06**	**-2.07e-05**	**4.71e-06**	**4.94e-06**
g	**-1.00e-04**	**-1.76e-05**	**5.16e-05**	**3.55e-05**
h	**2.62e-08**	**2.95e-08**	**2.54e-08**	**2.59e-08**
i	**1.38e-07**	**6.23e-08**	**1.28e-07**	**1.12e-07**
j	**-5.51e-08**	**1.12e-08**	**4.43e-08**	**9.63e-08**
$\boldsymbol{R^2}$	**0.9948**	**0.9964**	**0.9946**	**0.9946**

2.8 Video Quality Assessment

Mean squared error (MSE) and peak signal-to-noise ratio (PSNR) as reliability metrics were commonly used in image and video process community because they are as very simple to understand and implement as easy and fast to compute. As the PSNR is not devised for recognizing inadequacies and mistakes, this also stimulated the structural similarity method for assessing the image quality. But the PSNR is developed for identifying image patterns for taking out the organization or association of ordinary images. On the basis of this reflection, it can be said that a beneficial perceptual quality measure would stress the arrangement of acts on the illumination effects. This method is generally unresponsive to the alterations produced by illumination variations and the fluctuations in the mean and disparity of an image. Alternatively, the structural method is susceptible to alterations, and this alteration crashes the natural three-dimensional connection of an image, for example fuzziness, block compression objects, and noise. The PSNR can be calculated using the following equation [28]:

$$PSNR = 10 \log_{10} \frac{(2^B - 1)^2}{MSE} \qquad (2.2\ )$$

where $2^B - 1$ is the maximum single value that a pixel can take for B-bit representation, while the MSE is computed as the average quadratic pixel by pixel difference between the original video frame, $p(x, y)$, and the received video frame, $p'(x, y)$ as :

$$MSE = \frac{1}{MN} \sum_{x=1}^{M} \sum_{y=1}^{N} (p(x, y) - p'(x, y))^2 \qquad (2.3\ )$$

Where M and N represent the horizontal and vertical resolution respectively. Furthermore, when any application requires an image quality metric that is insensitive to three-dimensional transformation, this postponement of SSIM can be modified, as it is not very responsive to minor interpretations [41], [42], and [43]. The SSIM is regarded as a perception-based model which can predict better QoE, compared to traditional metrics such as MSE and PSNR, which are based on absolute errors between the reference and the degraded pixels and are less consistent with the perceptual quality of video [15].

SSIM measures the picture corruption in terms of perceived structural data change, accordingly taking into account those tight correlations between spatially adjacent pixels, which hold data about the objects in the visual scene. SSIM will be ascertained through factual measurements registered inside a square window about size N×N (typically 8×8), which moves pixel by pixel through the whole picture. The SSIM can be measured based on the middle of those relating windows X and Y of two pictures as the following equation (Equation 2.2) [56]:

$$SSIM(X,Y) = \frac{(2\mu X\mu Y + c1)(2\sigma XY + c2)}{(\mu^2 X + \mu^2 Y + c1)(\sigma^2 X + \sigma^2 Y + c1)} \quad (2.4\ )$$

with μ and σ^2 represent the mean and variance of the luminance value in the corresponding window, and c1 and c2 variables to stabilize the division with weak denominator.

The average values of SSIM for each video were taken. The range of the SSIM index goes from 0 to 1, which represents the extreme cases of totally different or perfectly identical frames, respectively.

2.9 Summary

This chapter provides a review of QoS and QoE for LTE networks, current literature related to the work presented in this thesis and an overview of LTE system architecture. There are a number of factors that have an impact on the QoS aspects of the LTE data download rate. Such factors include the conditions of the channel, resource allocation policies, available resources, and the delays that occurred as a result of the sensitivities of the traffic. A resource block that comprises frequency and time domains contains the resources that are being transferred within LTE. The 3GPP LTE system contains components known as eNodeB, which are base stations. A packet scheduler controls the allocation of the time and frequency resources among users for each TTI of 1ms. The packet scheduler resource allocation decision is based on many parameters at the scheduler itself and receives the CQI from users. The current downlink scheduling algorithms in LTE system were discussed. The majority of current downlink schedulers with QoE-aware are focusing in maximizing the QoE for the user. At finally quality assessments metric were presents MOS for VoIP and SSIM for video stream. This chapter is essential since it sets the background for developing QoE-driven LTE downlink scheduling.

Chapter 3 SIMULATION TOOL

3.1 Introduction

This chapter presents the simulation tool that was used to implement the experiments during the project. LTE-Sim provides an open-source platform upon which to simulate the LTE system from application layer to MAC layer. In the LTE-Sim, the algorithms that were scheduled by the downlink packet were implemented. Quite a few aspects of LTE networks are supported by LTE-Sim. From the layer down application to the tangible layer, for instance, the environments of multi-cell and single-cell are, user mobility, frequency reuse techniques, CQI feedback, QoS management, handover procedures, and multi-users environment [57]. LTE-Sim is inclusive of fundamental nodes such as S-GW, MME, UE, and eNodeB, and it is also supported by the trace-based nodes, like VoIP application and infinite-buffer traffic generators application, which are available at the data radio bearer management and application layer. Furthermore, the following are the downlink packet schedulers that come with the LTE-Sim by default Proportional Fair (PF), Modified Largest Weighted Delay First (MLWDF), and Exponential Proportional Fair (EXPPF).

This project makes a comparison of the results of simulation of real-time video and VoIP stream commencing from the eNodeB towards numerous UEs. In this comparison, the developed algorithms that are generated by the built-in packet schedulers, i.e. MLWDF, PF and EXPPF, in LTE-Sim will be evaluated and compared to the proposed algorithms. The function of the trace-based traffic application is to use the video trace file to send the video application packets. Here, it is used for the simulations with the four schemes of downlink schedules. Moreover, understanding of the Media Access Control (MAC) Queue, QoS, Radio

Bearer, and Packet components is required in the execution of the new developed algorithms in LTE-Sim.

The rest of this chapter is organized as follow. Section 3.2 provides an overview of Packet flows of the LTE-Sim. Section 3.3 describes the algorithm of default LTE-Sim downlink scheduling. Section 3.4 presents the QoE-driven class and method. Section 3.5 describes the simulation environment. Finally, section 3.6 summarizes this chapter.

3.2 Packet flows of LTE-Sim

When a downlink flow begins from the eNodeB to UE, the Radio Bearer and Radio Bearer class models activates in LTE-Sim. For every Radio Bearer command, the QoS requirements are defined by the QoS Parameters class for the flow. The packets, which are arranged by the Packet class, are transmitted by the application layer of the trace-based traffic generator, and the packets are transported via Radio Bearer. When it is delivered on the eNodeB, it reaches to user-plane protocol stack for the addition of headers of User Diagram Protocol (UDP) and IP, then it gets associated with a particular Radio Bearer, and lastly it gets queued on the MAC Layer. The IP-based packet classifier then maps the resultant IP diagrams to the Radio Bearers. The First In First Out (FIFO) queue, which is formed by the Mac Queue class, maintains each and every Radio Bearer. The relationship of these entities is depicted in Figure 3.1. With the help of the eNodeB physical layer, the packet can be sent to the logical channel. Then, it will be received by the UE and through the UE user-plan protocol stack; the packet will be delivered to the application layer.

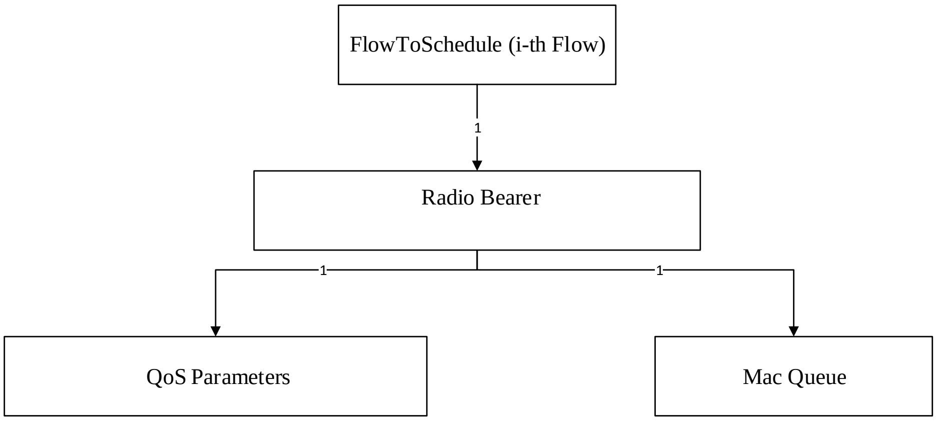


Figure 3.1: LTE-Sim brief class diagram

The flows that can be scheduled initially are nominated by the downlink schedulers and then with the highest metric flow, they assign each RB. This is how the work goes on LTE-Sim [57]. If the receiver UE is active and the data packets are ready to transmit in the MAC layer, then a flow can be scheduled. Scheduler processes every flow at every single TTI epoch, which can be scheduled, with the given metric. The period to transmit the data block from application layers to sub-frames, which are located on the radio link layer, is named as TTI. In the transmitter, the data is split into the blocks and the bits are encoded inside the block to prevent any errors that may occur due to the interference and fading.

Below is the scheduling procedure through which the metric is allocated to the i-th flow for the RB of j-th be the $m_{i,j}$, i.e., metrics[j] [i] in the code of LTE-Sim [57]:

1. A list of flows, which is included the packets to transmit, is created by the eNodeB (FlowsToSchedule). In that list, the CQI feedback and MAC queue length are stored.

2. As per the used downlink scheduler, the chosen metric is processed in the list of

FlowsToSchedule for each flow by using the function ComputeSchedulingMetric().

3. The highest metric flow is assigned to each RB by the eNodeB. The flow is deleted from the FlowsToSchedule list as soon as it sends the enqueued packets.

4. In every single scheduled flow, the data quota size is computed by the eNodeB first, and then it is transmitted during the current TTI at the MAC layer. Lastly, at the MAC layer, there is a dequeueing of packets that is invoked by eNodeB for all the scheduled flows.

3.3 Downlink Scheduling Algorithms of LTE-Sim

The process to compute the metric by the downlink scheduler of the built-in LTE-Sim, i.e. EXPPF, MLWDF and PF, is discussed in the following sections. These schedulers are dependent of R, i-th flow average transmission data rate or the returned value of the function call of GetAverageTransmissionRate in the calculation of the metric. Due to the following formula of weighted moving average, the value is updated every time on the TTI basis:

$$R_i(T) = 0.8R_i(T-1) + 0.2R_i(T) \tag{3.1 }$$

In the above formula, $R_i(T)$ is the rate of data that was achieved during the T-th TTI by the i-th flow. $R_i(T-1)$ is the rate of data in the last TTI. Now, the available rate of data of the UE receiver is needed to be known by the schedulers for the $r_{i,j}$, RB of j-th or the returned value by the following method:

GetSpectralEfficiency() is multiplied by the 180 kHz. The 180 kHz is the RB bandwidth in the domain of frequency. The information rate in seconds per bits, which can be transmitted in the specified bandwidth, i.e. 180000 Hz, is named as the spectral efficiency in the communication system of LTE. It has the ability to measure the efficiency of utilization of a spectrum that has a limited frequency, by looking at the CQI feedback update, which has been sent from the UE i-th flow for the RB of j-th.

3.3.1 PF Scheduling Algorithm

In the LTE-Sim class of DL_PF_PacketScheduler, the PF scheduler has been defined very well. The aim of PF algorithm is to maximize the overall throughput of the LTE network and guarantee the fairness among flows [57]. It is an excellent choice for the traffic of non-real-time (NR) [44]. Considering the past user amount and quality of experienced channel, the radio resources are assigned by the scheduler [9]. There are two ways to define the metric:

$$\boldsymbol{m_{i,j} = \frac{r_{i,j}}{R_i}}, \qquad \textbf{(3.2)}$$

in the DL_PF_PacketScheduler::ComputeSchedulingMetric()

3.3.2 MLWDF Scheduling Algorithm

In the DL_MLWDF_PacketScheduler class of LTE-Sim, the MLWDF is defined thoroughly. It aims to avoid the expiration of deadline for the wired networks and real-time operating systems and under the LWDF policy, it is known as the channel-aware extension [58]. It provides the delivering delay of bounded packets. With the highest delay, it prioritizes the real-time flows for the Head of Line (HOL) packet. In these packets, the first packet is passed on in the queue. Moreover, it is also the best channel condition that can be observed from the following metric [57]:

$$m_{i,j} = -\frac{log\delta_i}{\tau_i} D_{HOL,i} \frac{r_{i,j}}{R_i}, \qquad (3.3)$$

In the first function DL_MLWDF_PacketScheduler::ComputeSchedulingMetric(), δ denotes the maximum probability that the Head of Line packet delay ($D_{HOL,i}$) goes beyond the delay threshold(τ), for the flow of i-th real-time (RT).

The values of the three parameters (i.e. δ, τ and $D_{HOL,i}$) can be taken out by the GetDropprobability(), GetMaxDelay() from QoSParameters and GetHeadOfLinePacketDelay() from Radio bearer, respectively. The packets, which associate from the flow of real-time, of MAC queue are discarded, if they are not passed on at the right time and before their expiration. In this way, the wastage of bandwidth is avoided [5].

The metric reduces the PF scheduler if there are NR flows.

3.3.3 EXP-PF Scheduling Algorithm

In the class DL_EXP_PacketScheduler of LTE-Sim, the EXPPF scheduler is discussed very well. It is produced to allow the traffic of the real-time over the ones that are NR. The following equation is computed the metric for RT flow:

$$m_{i,j} = \exp\left(\frac{\frac{log\delta_i}{\tau_i} D_{HOL,i} - \gamma}{1 + \sqrt{\gamma}}\right)\frac{r_{i,j}}{R_i}, \tag{3.4}$$

where δ_i denotes the maximum probability that the Head of Line packet delay ($D_{HOL,i}$) goes beyond the delay threshold (τ), for the flow of i-th real-time and N_{rt} denotes the number of active flow of downlink real-time. It is similar to the MLWDF case:

$$\gamma = \frac{1}{N_{rt}}\sum_{i=1}^{N_{rt}} -\frac{log\delta_i}{\tau_i} D_{HOL,i}, \tag{3.5}$$

The metrics are computed jointly in the functions of LTE-Sim, ComputeAW()as follows:

$$AW(i) = -\frac{\log(dropProbaility(i))}{maxDelay(i)} HOLPacketDelay(i) \frac{spectralEfficien}{averageTransn} \quad (3.6)$$

$$averageAW = \frac{1}{nbFlows} \sum_{i=1}^{N_{rt}nbFlows} AW(i), \quad (3.7)$$

ComputeschedulingMetric() as follow:

$$m_{i,j} = \exp\left(\frac{AW(i) - averageAW}{1 + \sqrt{averageAW}}\right) \frac{spectralEfficiency(i,j) \times 18000}{averageTransmissionRate(i)}, \quad (3.8)$$

where the values of$HOLPacketDelay(i)$, $dropProbaility(i)$, and$maxDelay(i)$ for thei-th flow can be obtained by calling the functions, GetHeadOfLinePacketDelay, GetDropProbability, and GetMaxDelay, respectively.

3.4 QoE-driven Class and Method

The implementation for QoE-driven downlink scheduling was required to create a new packet scheduler class in LTE-Sim which was used in chapter 5 and 6. This class is called DL_QoE-driven_PacketScheduler. The new source and header files were created in packet-scheduler folder and called:

- ./lte-sim/src/protocolStack/mac/packet-scheduler/dl-QoE-driven-packet-scheduler.cpp
- . /lte-sim/src/protocolStack/mac/packet-scheduler/dl-QoE-driven-packet-scheduler.h

Moreover, the class called DownlinkPacketScheduler: was modified by adding constructor and methods as follow:

- DL_QoE-driven_PacketScheduler::~DL_QoE-driven_PacketScheduler()
- DL_QoE-driven_PacketScheduler::DL_QoE-driven_PacketScheduler()
- DL_QoE-driven_PacketScheduter::DoSchedule()
- DL_QoE-driven_PacketScheduler::RBsAllocation()

3.5 Simulation Environment

The single-cell scenario was design to run the performance experiments for LTE downlink scheduling. It consists of one eNodeB with a radius of 1 KM and a variable number of users from 10 to 100. The user mobility was set as a random way-point model [59]. The simulation simulated different users' speeds, which are 0, 3, 30, and 120 km/h. These were selected to characterize user movement static, pedestrian, and vehicular scenarios [15]. Each user has three types of flow running simultaneously, including VoIP, video, and best effort. The scenarios were designed and implemented by using a LTE-Sim simulator [57]. Table 3.1 summarized the simulation parameters used in the simulation. The traffic model used in this simulation is described in the simulation parameters (c.f., Table 3.1). There are two real-time flows: the first one is a trace-based application that used a video trace file at 242 Kbps from [60]; the second one is a VoIP application, which used G.729 codec with ON/OFF Markov model for VoIP flow. The period has an exponential distribution of 3s as the average value. The OFF period contains a truncated exponential probability density function, where the upper limit is 6.9s with the same average as the ON period of 3s [57].

Table 3.1: SIMULATION PARAMETERS.

Parameter	Value
Simulation length	180 s
Number of Cell	1 eNodeB
Cell layout	radius: 1 KM
Number of users	10, 20, 30, 40, 50, 60, 70, 80, 90, 100
User speed	0, 3, 30, 120 KM/H
Traffic model	Real-time flows type: H.264 at 242 Kbps, G.729 VoIP. Best effort flows: infinite buffer

3.6 Summary

LTE-Sim is an open-source framework that is used for the simulation of LTE networks. In the LTE-Sim, the algorithms that were scheduled by the downlink packet were implemented. Quite a lot of aspects of LTE networks are supported by LTE-Sim. Three different downlink packet schedulers are studied in this section with their features. The equations through which they can be computed are also discussed in this chapter. The implementation of a new downlink packet scheduler class QoE-driven and their constructor and methods are presented. Finally the simulation environment and parameters is discussed. The next chapter will discuss the QoE-based evaluation for LTE scheduling for VoIP applications.

Chapter 4 QoE-based Performance Evaluation of Scheduling Algorithms over LTE for VoIP Application

4.1 Introduction

Currently there are no 3GPP standards for scheduling algorithms to support both real-time and non-real-time applications in LTE-based 4G networks. Extensive research has been carried out on a variety of scheduling algorithms to improve system performance in terms of throughput, fairness, or other QoS metrics, such as packet loss and delay. It is unclear how these scheduling algorithms perform in terms of end-user experience or user-perceived quality. The QoE-aware scheduling algorithm is still in its early stages in 2012, and only a few papers have examined the QoE for video streaming [61] [62]. I argue that even other real-time services such as VoIP needs to be evaluated when designing a full QoE-aware algorithm.

In this chapter, The aim is to evaluate the performance of several scheduling algorithms for VoIP applications in terms of the Quality of Experience (QoE) metric (e.g. MOS score), which is more closely linked with end-user perceived quality, in addition to normal QoS metrics (e.g. packet loss and delay). This work would help to better understand how scheduling algorithms behave from the QoE point of view and help to develop QoE-driven scheduling algorithms to achieve maximized QoE for real-time and non-real-time multimedia applications in the future.

To compare the different algorithms, a simulation platform following the 3GPP LTE standard was built up based on an LTE-Sim simulator, which was described in chapter 3. Four different scenarios were developed using a single cell: static at user speed 0 KM/H, pedestrian at user speed 3KM/H, and vehicular scenarios using different speeds (30, and 120 KM/H, respectively). The single cell includes one eNodeB and a variable number of user equipment (UE), ranging from 10 to 100. A realistic scenario was adapted whereby every UE receives video and VoIP flows as well as best-effort flows with infinite buffers. Packet loss ratio and end-to-end delays were captured in the simulation, and I evaluated the QoE performance (in terms of MOS score) for the VoIP flows.

Preliminary results show that MLWDF has the best performance across different speeds by allowing the maximum number of user access to more than 40 users at the acceptable MOS score of 3.5. While MLWDF and EXP-PF performed very well in minimizing the end-to-end delay in all cases tested, their performance varied with the packet loss ratio after the user reached more than 30 users. The PF scored the largest end-to-end delays, but kept the packet loss ratio to a minimum compared to other algorithms.

The rest of the chapter is organized as follows. Section 4.2 discusses the downlink scheduling algorithms. Section 4.4 examines the simulation environment, and Section 4.5 reports the simulation results. Finally, Section 4.6 draws the summary.

4.2 Scheduling Algorithms

There are several scheduling procedures that are followed by the wireless networks, which may include selection of parameters such as available resource allocation. Such procedures could be the channel gain, average rate, and packets arrival delay. PF is the most suitable for elastic flows, as it can accommodate the ability to choose the users according to their channel state. In terms of real-time services, a proposal has been made for a packet delay algorithm such as M-LWDF. A proposal has also been made for the EXP/PF and EXP rule to accommodate mixed, real-time, and non-real-time flows. Both the M-LWDF and EXP/PF are specifically tailored for the scheduled algorithms. In addition, PF is included as a reference within the performance.

4.3 Simulation Environment

The performance of PF, M-LWDF, and EXP/PF within the LTE has been evaluated. A realistic single-cell scenario is being made. eNodeB has number between 10 to100 of UEs. According to [63], the speed of 0, 3, 30, and 120 km/h will be used in order to diagnose the static, pedestrian, and vehicular scenarios, respectively. Simultaneously, a video flow, VoIP flow, and best-effort flow are allocated within each UE. The scenarios parameters are described in chapter 3.

Five simulations were run for each number of users at different speeds to calculate the average packet loss ratio and delay. In total, 150 simulations were conducted to determine the final result for each LTE scheduling algorithm. The results in this chapter are focused on VoIP performance, but the traffic model used within the simulation is explained as in chapter 3 with the following flows: a video service with foreman 242 kbps source video rate was used. G.729 voice flows are created within the VoIP application in order to determine the VoIP flows and infinite buffer flows.

4.4 Simulation results

The performance of the LTE algorithms in a single cell with a set of users at different speeds is analysed. As one of the real-time flows, VoIP is considered to measure the impact on QoE. QoE performance for our experiments was measured using the MOS score. As the packet-loss ratio and delay were provided in the simulation results, the MOSc was determined using equation 2.1 [55].The parameters for fitting surfaces with G.729 codec from [55] were used.

The results for all experiments are shown in Figures 4.2 through 4.10. The figures indicate slight changes in results at different speeds. As the single-cell scenario user only has the ability to move around a small area, handover will not occur. However, the results are affected by increasing the number of users within the cell. MLWDF and EXP-PF keep the end-to-end delay to fewer than 50ms with all sets of users, but PF can only handle up to 30 users in the cell before the end-to-end delay increases rapidly, as the number of user increases in all test cases and follow the same pattern as (Figures 4.2, 4.5 ,4.8and 4.11).

MLWDF and EXP-PF consider the end-to-end delay in the scheduling algorithm, their very good results show that they effectively minimise the end-to-end delay across different speeds. On the other hand, the packet loss ratio (PLR) for MLWDF is the lowest until 60 users comparing to other scheduling algorithms, and then PF takes over as the lowest PLR until 100 users as shown in (Figures 4.3, 4.6 4.9 and 4.12) . The EXP-PF has the largest PLR compared to the other algorithms. The MOS score figures 4.4, 4.7, 4.10 and 4.13 almost reflect the PLR scores in that the lower the PLR, the better the MOS score. The acceptable voice quality is 3.5 in the MOS score, according to normal telecommunications applications. This also follows standards defined according to ITU-T Recommendations P.800 [55]. The cell capacity is 40 to 50 users with acceptable voice quality in MLWDF and PF scheduling algorithms.

Table 4.1: The maximum number of users access at acceptable MOS

User speed	PF	MLWDF	EXP-PF
static	20	40	25
3 KM/H	25	45	28
30KM/H	25	50	32
120 KM/H	30	50	30

In Table 4.2, the maximum number of users with an acceptable MOS score was captured. At higher speeds, the cell can accommodate more users comparing to slower speeds. PF has the highest number of users, but the end-to-end delay is very long, exceeding the recommended maximum delay by ITU-T Recommendations P.800 [11], which is 150ms. So the best suitable downlink scheduling algorithm is MLWDF, which has short end-to-end delay of less than 50ms with maximum number of users: more than 60 at the acceptable MOS score.

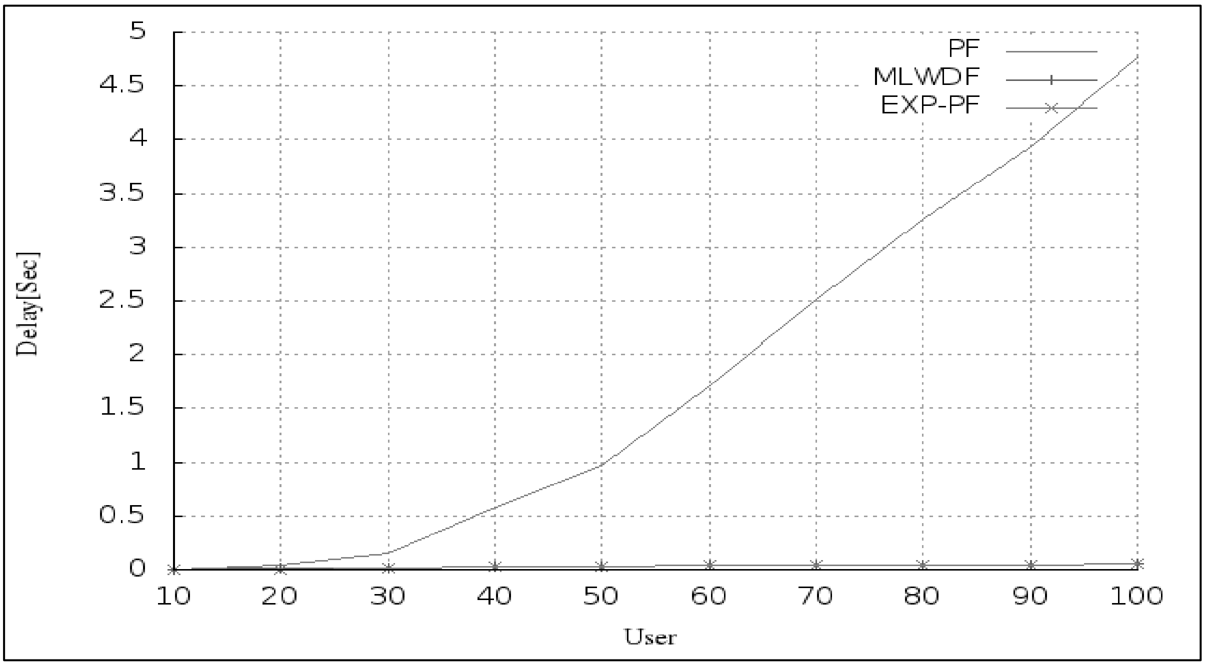


Figure 4.1: End-to-end delay time vs number of user at static

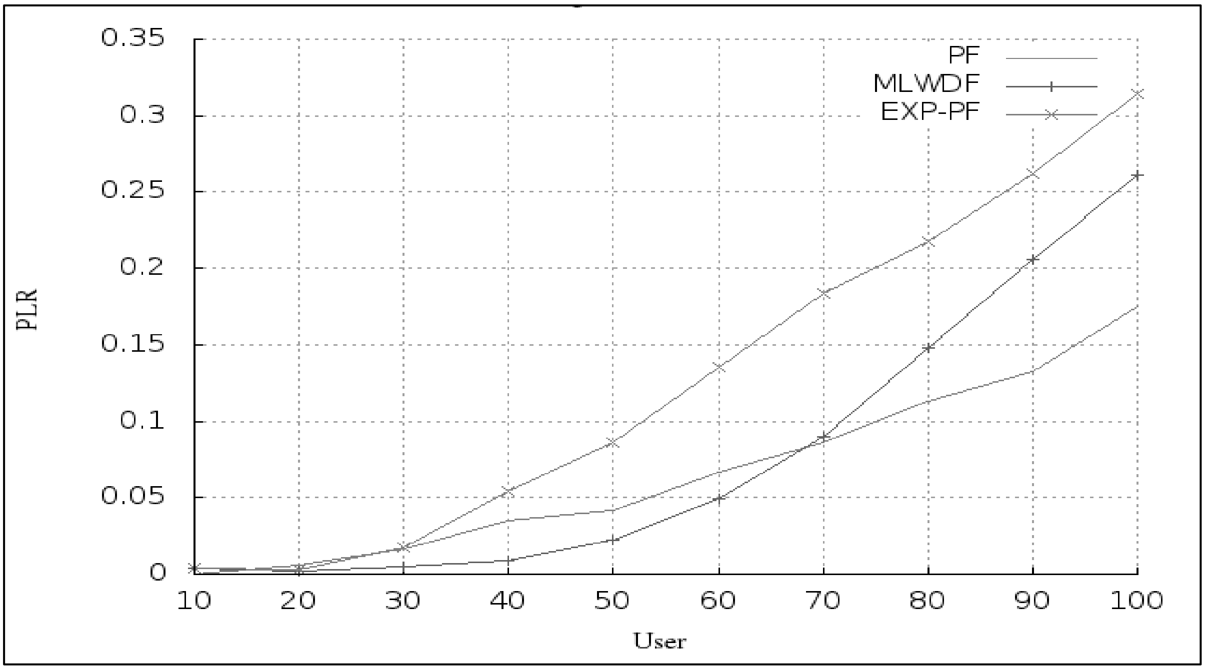


Figure 4.2: Packet Loss Ratio vs number of users at Static

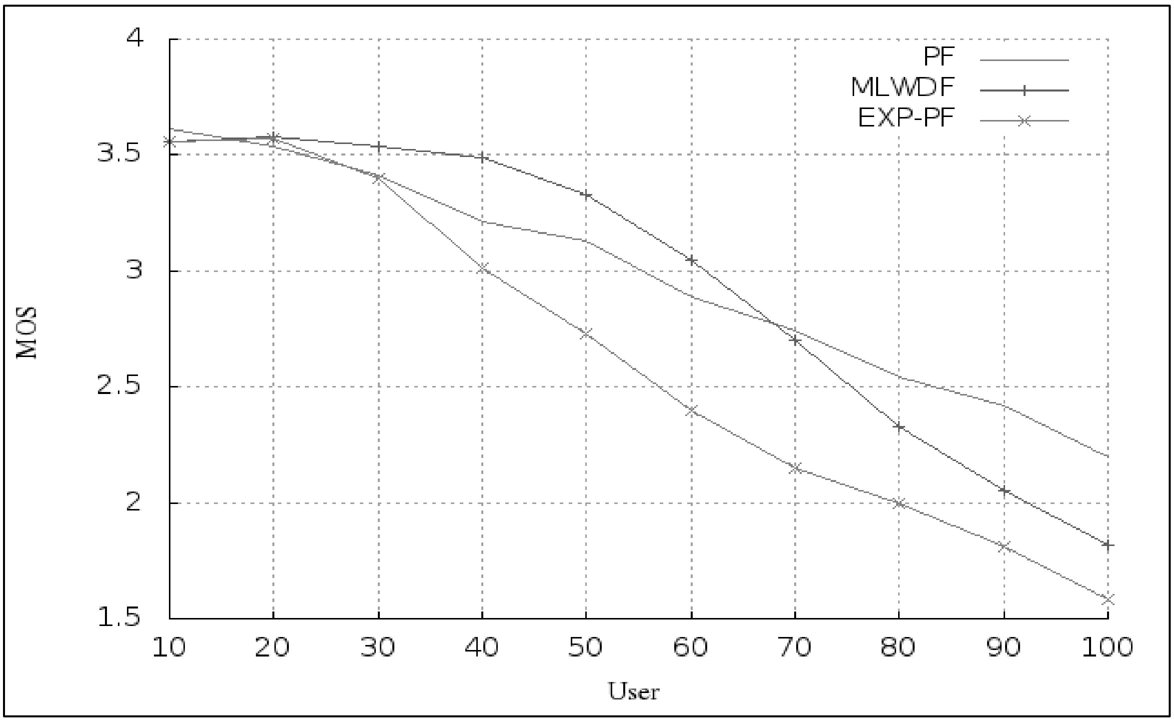


Figure 4.3: MOS Score vs number of users at Static

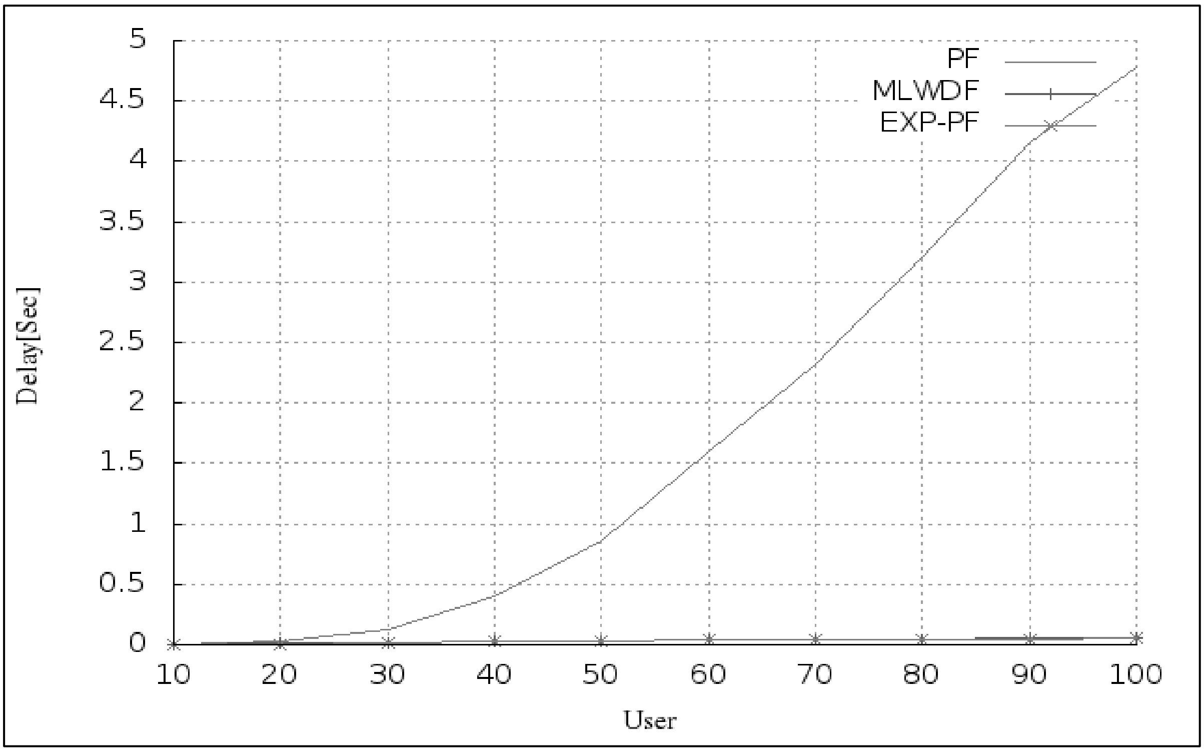


Figure 4.4: End-to-End Delay Time vs number of users at 3 km/h

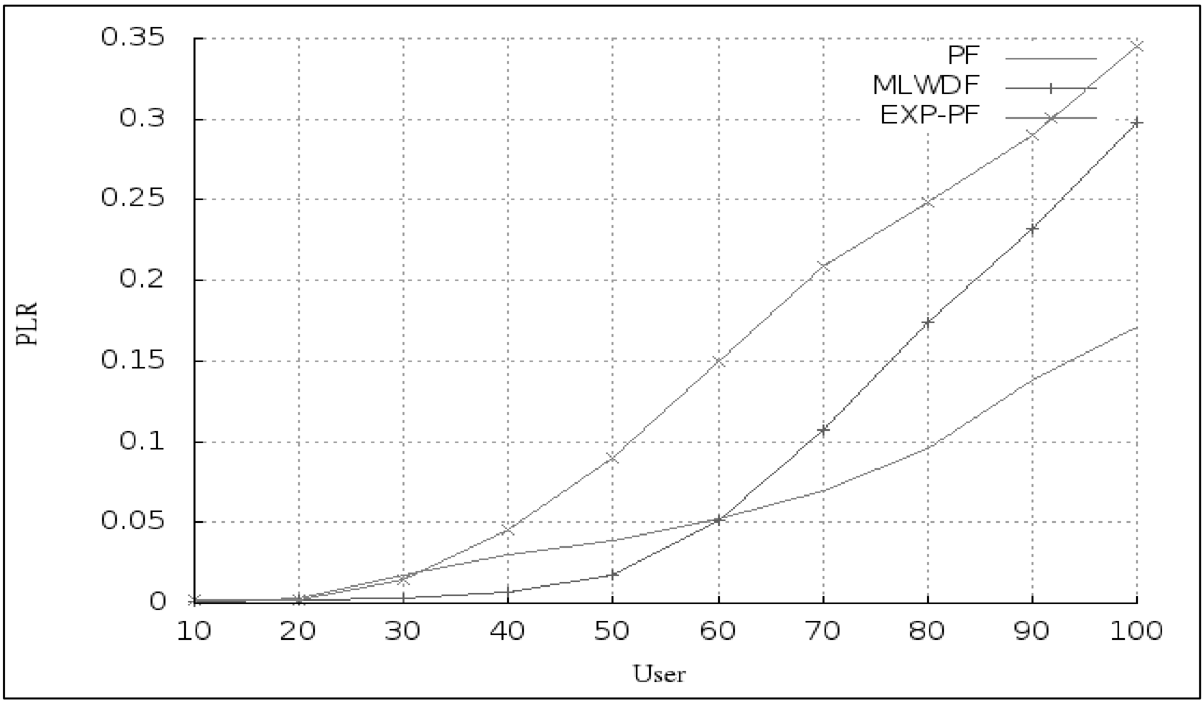


Figure 4.5: Packet Loss Ratio vs number of users at 3 km/h

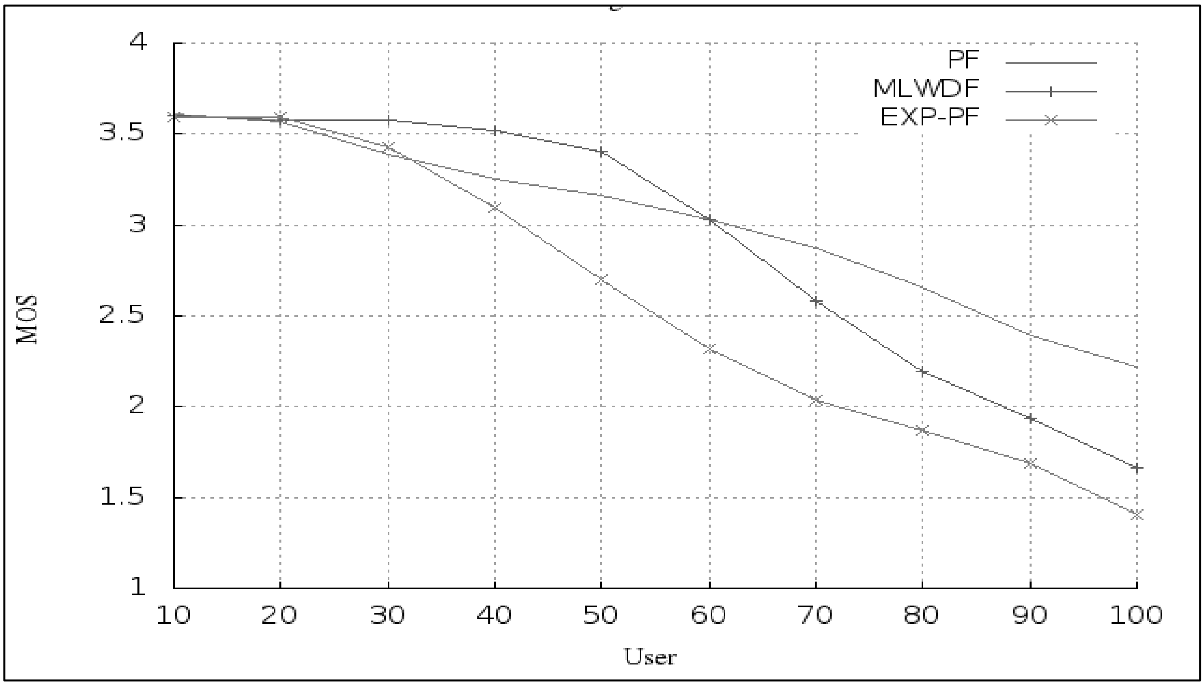


Figure 4.6: MOS Score vs number of users at 3 km/h

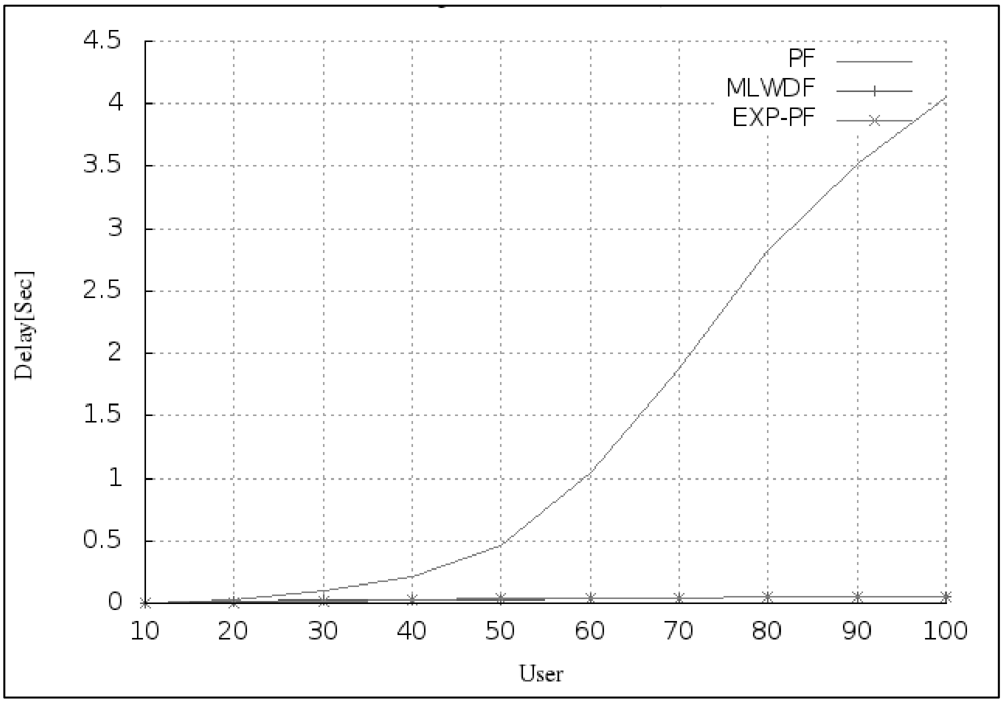


Figure 4.7: End-to-End Delay Time vs number of users at 30 km/h

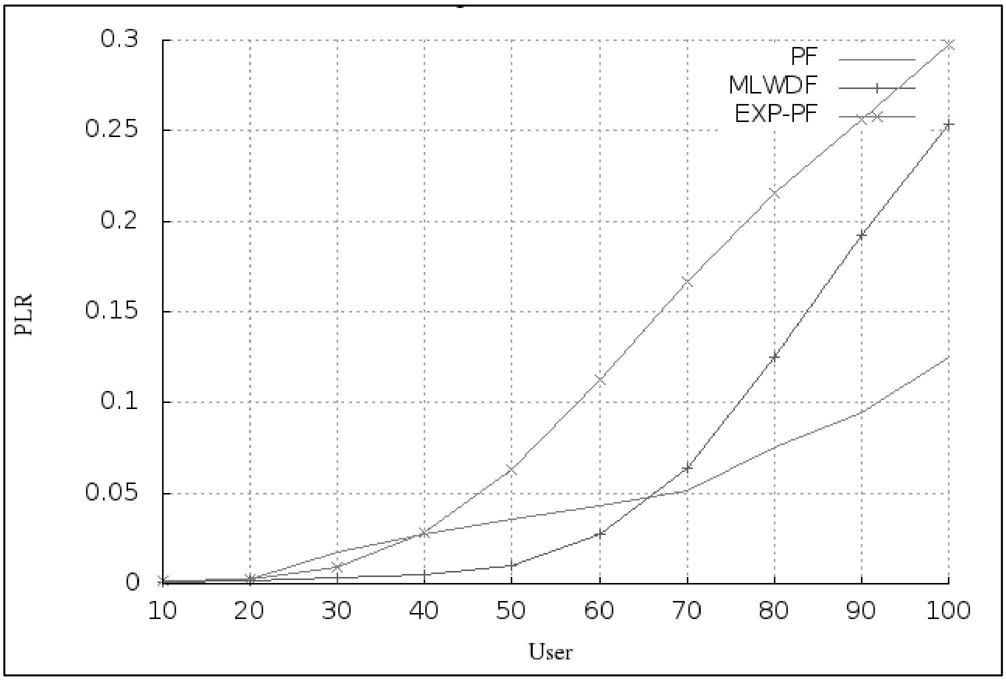


Figure 4.8Packet Loss Ratio vs number of users at 30 km/h

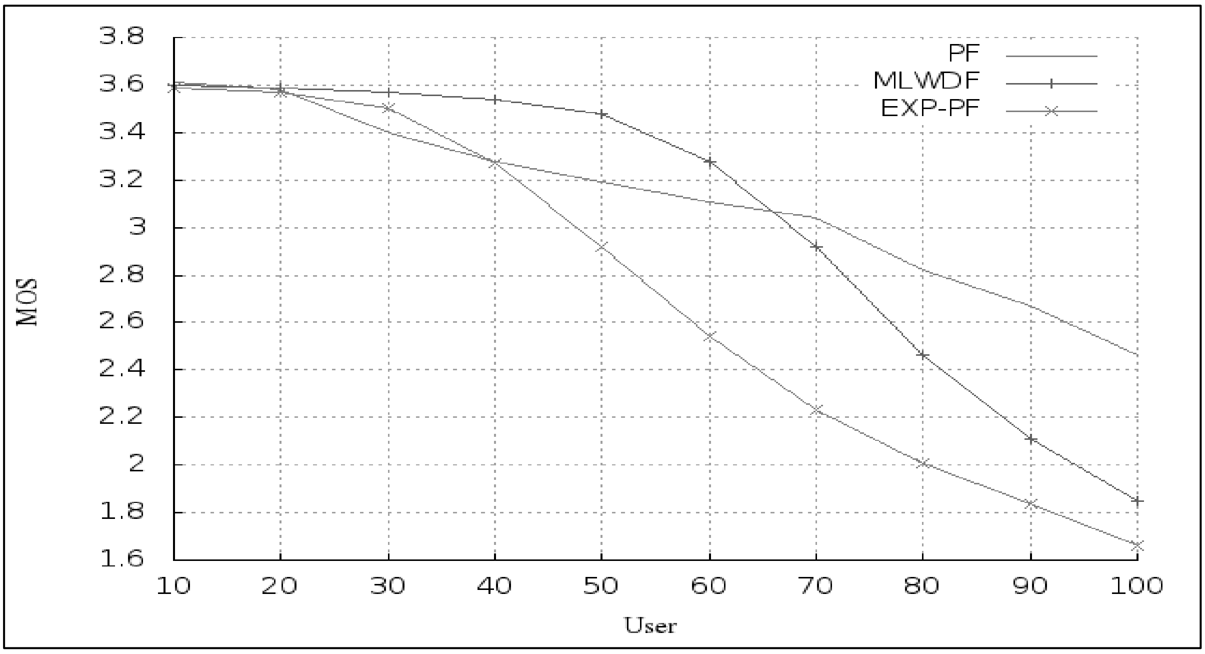


Figure 4.9: LTE Scheduling vs. MOS Score at 30 km/h

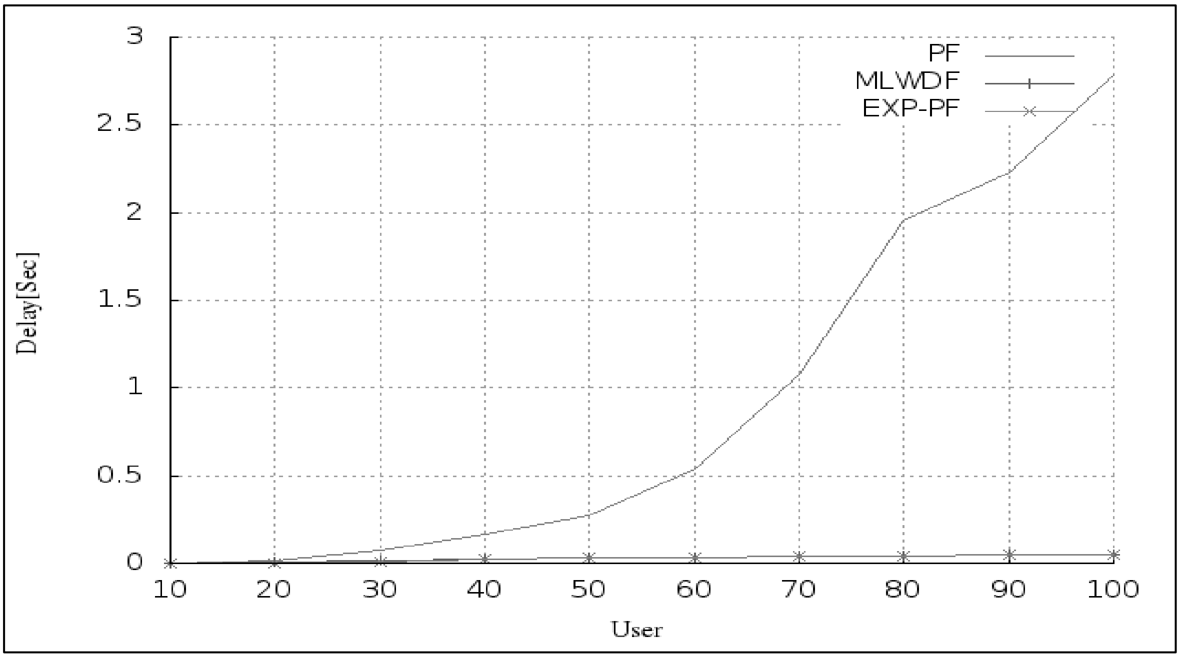


Figure 4.10: End-to-End Delay Time vs number of users at 120 km/h

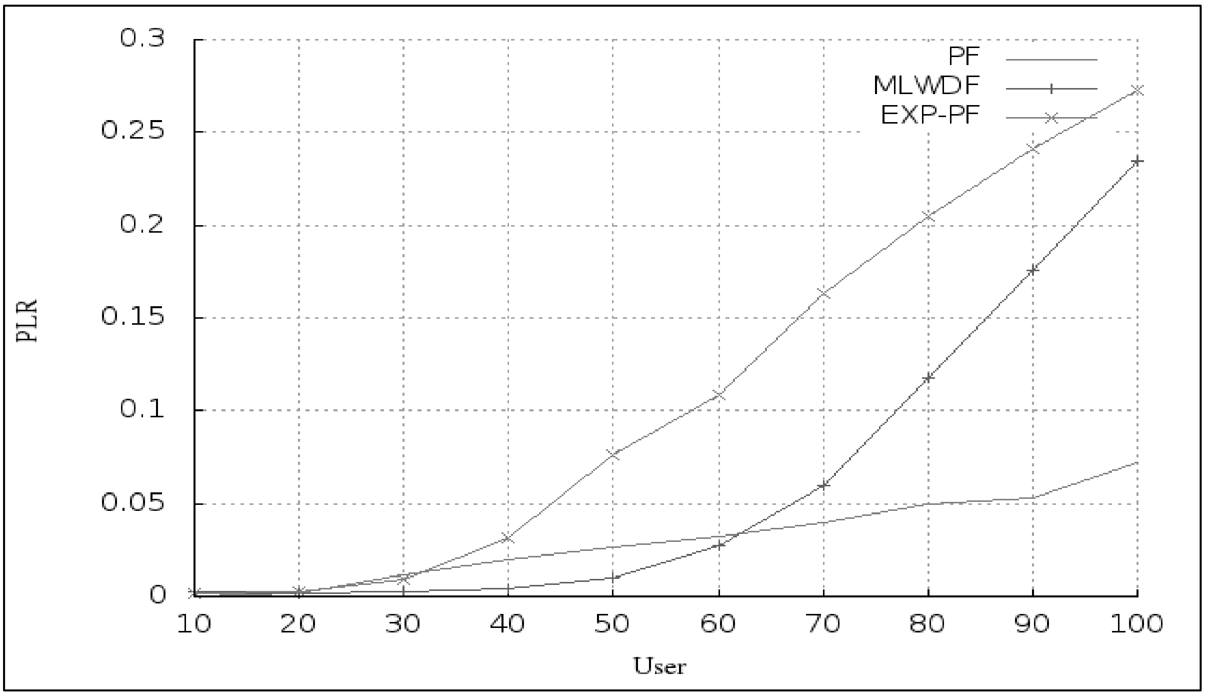


Figure 4.11: Packet Loss Ratio vs number of users at 120 km/h

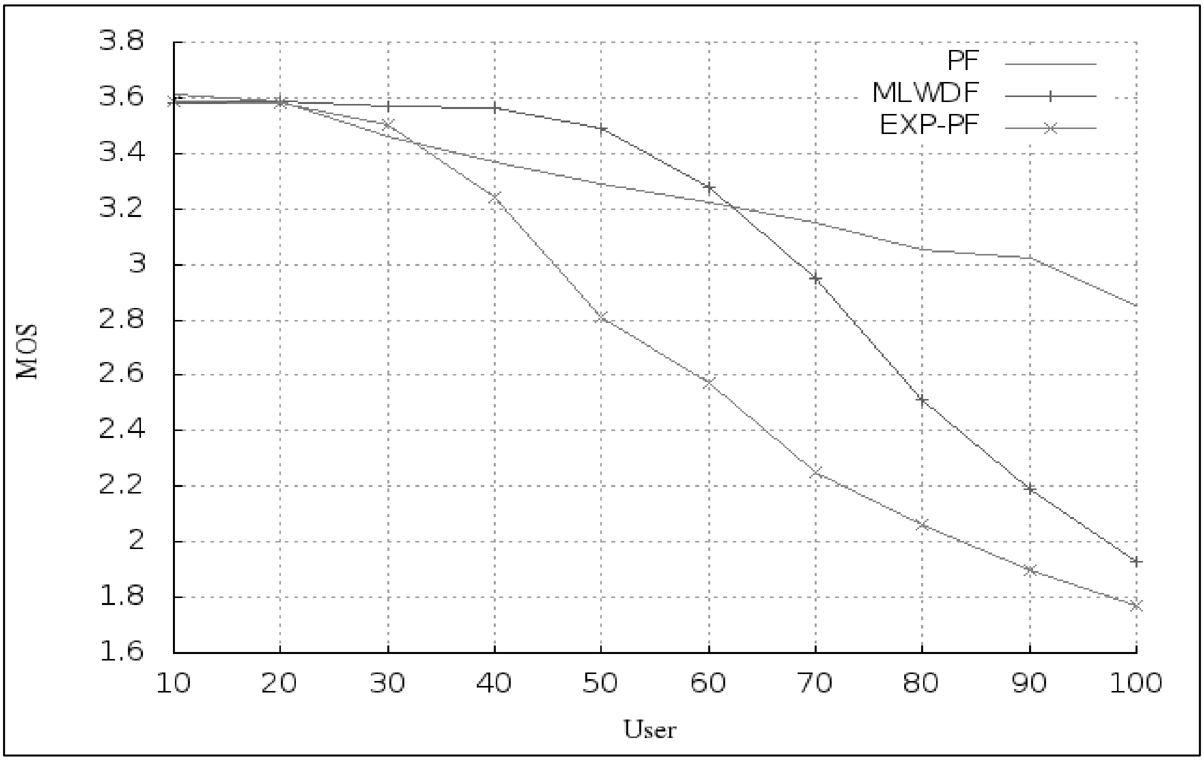


Figure 4.12: MOS Score vs number of users at 120 km/h

4.5 Summary

The LTE simulation was built based on LTE-Sim with three LTE scheduling algorithms (i.e., PF, EXP-PF, and MLWDF) for a single-cell scenario. Different speeds and numbers of users were used to monitor the voice quality of the LTE scheduling algorithms. EXP-PF is considered for real-time flows, demonstrating lower MOS scores with more than 30 users per cell. Also, PF has great end-to-end delay, which is not suitable with VoIP application. On the other hand, MLWDF handles 50 users with acceptable MOS scores (MOS over 3.5). The limitation of current well-known downlink scheduling was identified to provide clear image how to develop QoE-driven scheduling that will provide satisfied QoE for more users.

The next chapter will present the development of novel QoE-driven scheduling algorithms to increse the number of users per cell at an acceptable MOS score of 3.5 for VoIP application.

Chapter 5 QoE-driven LTE scheduling algorithm for VoIP application

5.1 Introduction

Traditionally, users' experiences of voice service can be evaluated through network measurement tools for availability, accessibility, and quality [64]. Nevertheless, in data services, statistics show that the correlation between network measurement tools and user benefits/interests are not as simple as could be perceived, because:

1. Performance of individual nodes and protocols through which information travels are more affected due to use of packet switching in the data system.
2. Radio resources are now being used and shared amongst different applications in computer networking.

Under these specific circumstances, the performance evaluation of data service is generally carried out through monitoring terminals in the existing real networking system. The end-to-end quality experienced by an end user results from a combination of elements throughout the communications protocol stack and system components. Therefore, a detailed performance analysis of the entire network is required for the evaluation of the network service operation (from the user computer system up to the application server or remote user system).

The Quality of Experience (QoE) is a popular metric to represent/measure the satisfaction level of user. When the service providers assess the QoS being provided to the user, the major objectives of the assessment are to optimize the operational system of the network so that the end user can get the finest quality and to meet the highest level of satisfaction of the user. The

QoE also discusses the context and use of applications keeping the perspective of the end user and his/her satisfaction level. In this regard, It can be seen the rapid growth of the smart phone industry, as these smart phones not only satisfy their users but also accomplish the QoE expectations of the users.

Several studies in the literature have primarily focused on QoS and QoE evaluation of LTE downlink scheduling algorithms, but not many investigate how to improve the end-user quality. Chapter 4 [14] was limited to an evaluation of only the QoE performance of the current well-known LTE scheduling algorithms, i.e. PF, EXP-PF, and MLWDF for VoIP application without further improving their performance. In this chapter, a QoE-driven LTE scheduling algorithm is proposed and implemented to provide an acceptable QoE for VoIP applications in order to increase the number of satisfied user in LTE cell. The investigation and evaluation of the performance of the proposed QoE-driven LTE downlink scheduler has been carried out against three popular LTE downlink schedulers, including PF, EXP-PF, and MLWDF. The performance evaluation is based on QoE metric parameters can be measured in MAC layer (i.e. estimation MOS score) instead of the standard QoS metrics (e.g. packet loss and delay) for VoIP application.

The main contribution of this chapter is that it maximizes the number of users per cell at an acceptable MOS score of 3.5 [11]. The preliminary results have shown that the proposed algorithm has improved the cell capacity by 75% at an acceptable MOS score compared to the MLWDF cell capacity and 250% compared to PF and EXP-PF algorithms in all three lower speed categories.

The rest of the chapter is organised as follows. Section 5.2 introduces the concept of the QoE-driven LTE downlink scheduling algorithm. Section 5.3 describes the simulation environment based on LTESim. Section 5.5 presents the simulation results for the proposed QoE-driven scheduling and three common scheduling algorithms in terms of QoE and QoS metrics.

Section 5.6 summarizes the chapter.

5.2 The concept of novel QoE-driven LTE downlink scheduling

The current LTE downlink scheduling algorithms were designed to satisfy QoS, such as delay, or to provide the maximum level of QoE for specific scenarios with a limited number of users. However, the proposed QoE-driven LTE downlink scheduling algorithm is designed to provide an acceptable QoE and hence maximize the number of users in a single cell. The QoE-driven LTE downlink scheduling algorithm is set and defined in Table 5.2:

Table 5.1: Definition of QoE-driven LTE downlink scheduling

Given	Delay $\boldsymbol{d}_{\boldsymbol{i}(\boldsymbol{t})}$,loss $\boldsymbol{p}_{\boldsymbol{i}(\boldsymbol{t})}$,buffer $\boldsymbol{b}_{\boldsymbol{i}(\boldsymbol{t})}$ and number of resource blocks $\boldsymbol{RB}_{(\boldsymbol{t})}$
constraint	$\boldsymbol{QoE_{min}}$, MOS score 3.5 for VoIP
Optimization	maximize number of users to access cell

Where, $d_{ij}(t)$ be the head of line (HOL) packet delay for user i at cell *j* at current scheduling instant *t*. $p_{ij}(t)$ and $b_{ij}(t)$ denote as the packet loss rate and the buffer size (containing contents) for user *i* at cell *j* at instant *t*. $RB_j(t)$ is the number of good available RBs at cell *j* at instant *t*. The constraint of this algorithm is that the received $QoE \geq QoE_{min}$ (I assume $QoE_{min} = 3.5$, or MOS of 3.5 which is widely used in telecommunications world for an acceptable quality for services such as VoIP). The optimization problem is to maximize the number of users for a given number of cells based on Equation 5.1 subject to Equation 5.2 and Equation 5.3.

$$\max \sum_{j=1}^{M} o_j \qquad \text{for } j = 1, \ldots, M \tag{5.1}$$

Subject to:

$$QoE_{ij} \geq QoE_{min} \tag{5.2}$$

$$\sum_{i=1}^{N} RB_{ij}(t) \leq RB_j(t)_{\max} \tag{5.3}$$

Where o_j denotes the number of mobile users for node *j*, M is the total number of cells, o_j is the number of users for cell j. N is the maximum number of users for a single cell. , and QoE_{ij} represents QoE achieved for user i at node j $(i = 1, \cdots N; j = 1, \cdots M)$. $RB_{ij}(t)$ is the RBs allocated to user *i* at cell j at time *t*. $RB_j(t)_{max}$ is the maximum accessible resource blocks at cell *j* at time *t*. The LTE scheduling algorithm for a single node case (*j=1* and *M = 1*) for simplicity is illustrated below. Iteration process is carried out at every TTI. The optimization for each single epoch has been implemented for simplicity.

Table 5.2: QoE-driven LTE scheduling for VoIP

Algorithm LTE Scheduling	
Input	Delay $d_i(t)$, PLR $p_i(t)$, buffer $b_i(t)$ for user i ($i \in [1, \cdots N]$) at time t
1:	Predict $QoE_i(t)$ and calculate the metric, $m_i(t)$ according to Equation 5.4
2:	If $QoE_i < QoE_{min}$, $m_i = m_i + 1$; or else $m_i = m_i$;
3:	Schedule a user with the highest metric the required RBs matching to its buffer size $b_i(t)$
4:	Check for remaining RBs and schedule the resource to the user with the 2nd highest metric
5:	Repeat Step 4 until the TTI end.

$$m_i(t) = \frac{r_i}{R_i} \cdot d_i(t) \cdot p_i(t) \tag{5.4}$$

Where $r_i(t)$ and $R_i(t)$ are expected throughput and average achieved throughput for user *i* at time *t*, respectively.

Figure 5.1 depicts a conceptual diagram of QoE-driven LTE downlink scheduling where an eNodeB allocates available mobile network resources, in the term of available Resource Blocks (RBs), to mobile users according to received channel information (Channel Quality Indicator, CQI) from a mobile user, network QoS (delay and PLR) and predicted QoE for a mobile user for a specified service at eNodeB.

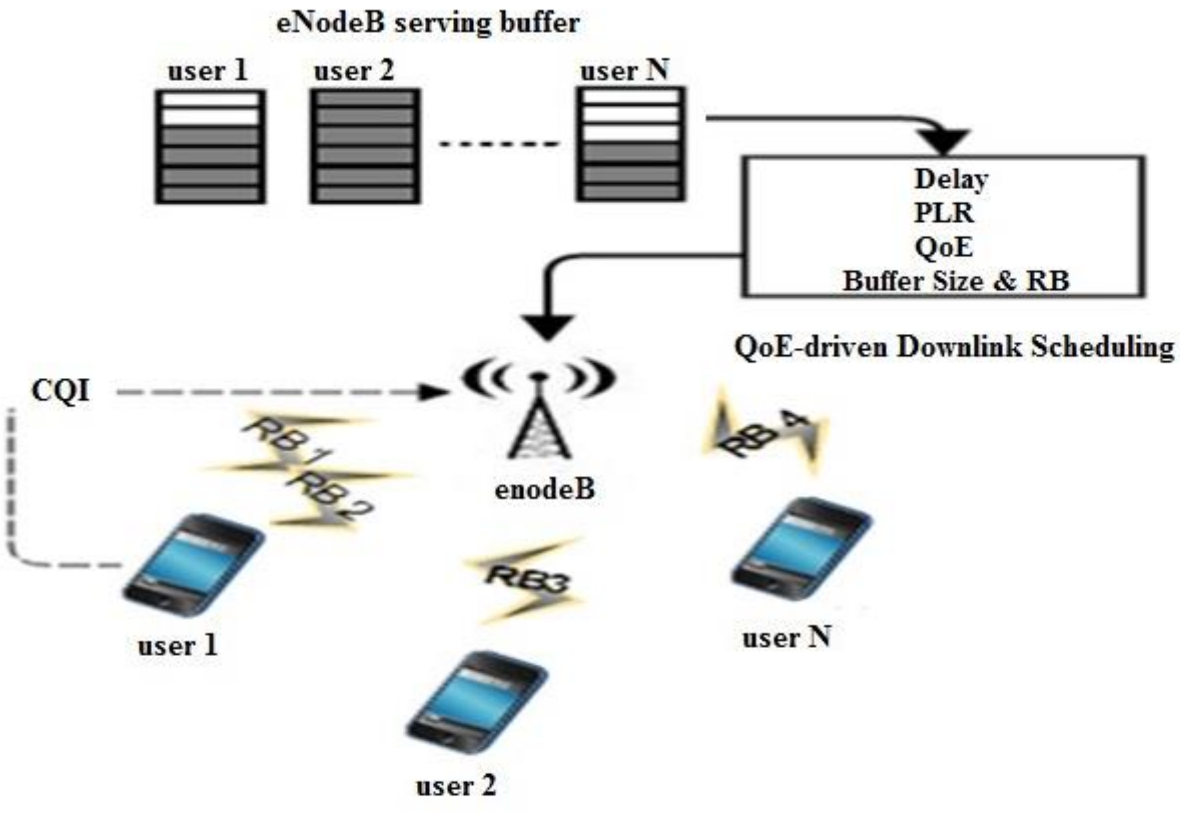


Figure 5.1: The model of QoE-driven LTE downlink scheduling

5.3 QoE-driven LTE downlink scheduling implementation

The QoE-driven scheduling is proposed to maintain $\boldsymbol{QoE_{min}}$ for real-time application over LTE, which is the border line of an acceptable QoE, such as an MOS score of 3.5 for VoIP application. The $\boldsymbol{d_{i_{(t)}}}$ and $\boldsymbol{p_{i_{(t)}}}$ are calculated for all $\boldsymbol{i}$ users at every TTI scheduling round.

These parameters were used to compute the metric to prioritized users based on the QoE requirements. Also, buffer $\boldsymbol{b_{i_{(t)}}}$ is calculated for all $\boldsymbol{i}$ users to determine the user requirement of RBs. The scheduler checks the accessible RBs at the current scheduling round. Then, the scheduler picks the user with the highest metric and checks his already allocated RBs. If the allocated RBs have served all packets in the user's buffer, the user will be taken out of the scheduling list at the current scheduling round, or else the scheduler allocates the user with the highest metric until it has either used all RBs available at this TTI or the allocated RBs for that user are served with the full buffer occupancy. Therefore, the scheduler checks if there are any available RBs to allocate to the second highest metric for the current scheduling round.

The QoE-driven scheduler flow chart is illustrated in Figure 5.2. This method ensures that each user has used all of his allocated RBs and, hence, there will not be wasted RB as in the case of current scheduling algorithms. Figure 5.3 depicts the efficiency use of RBs of the proposed algorithm compared to LTE-Sim scheduler resource allocation, as shown in Figure 5.4 [57].

For example, assuming there are 6 RBs for each TTI. At TTI (1), 6 RBs were assigned to user 1, which has the highest metric. Then the scheduler at TTI (2) checked the metric for all users. The user 1 still has the highest metric, but the allocated RBs in TTI (1) served the user 1 buffer, so the scheduler assigned 4 RBs for second highest metric, which is user 2. Also, there are still 2 RBs available, which were assigned to user 3, as shown in Figure 5.3. This illustrates that our proposed algorithm performs better than the existing normalized scheduler resource allocation (c.f., Figure 5.4) by saving 4 RBs, which can be used for other users.

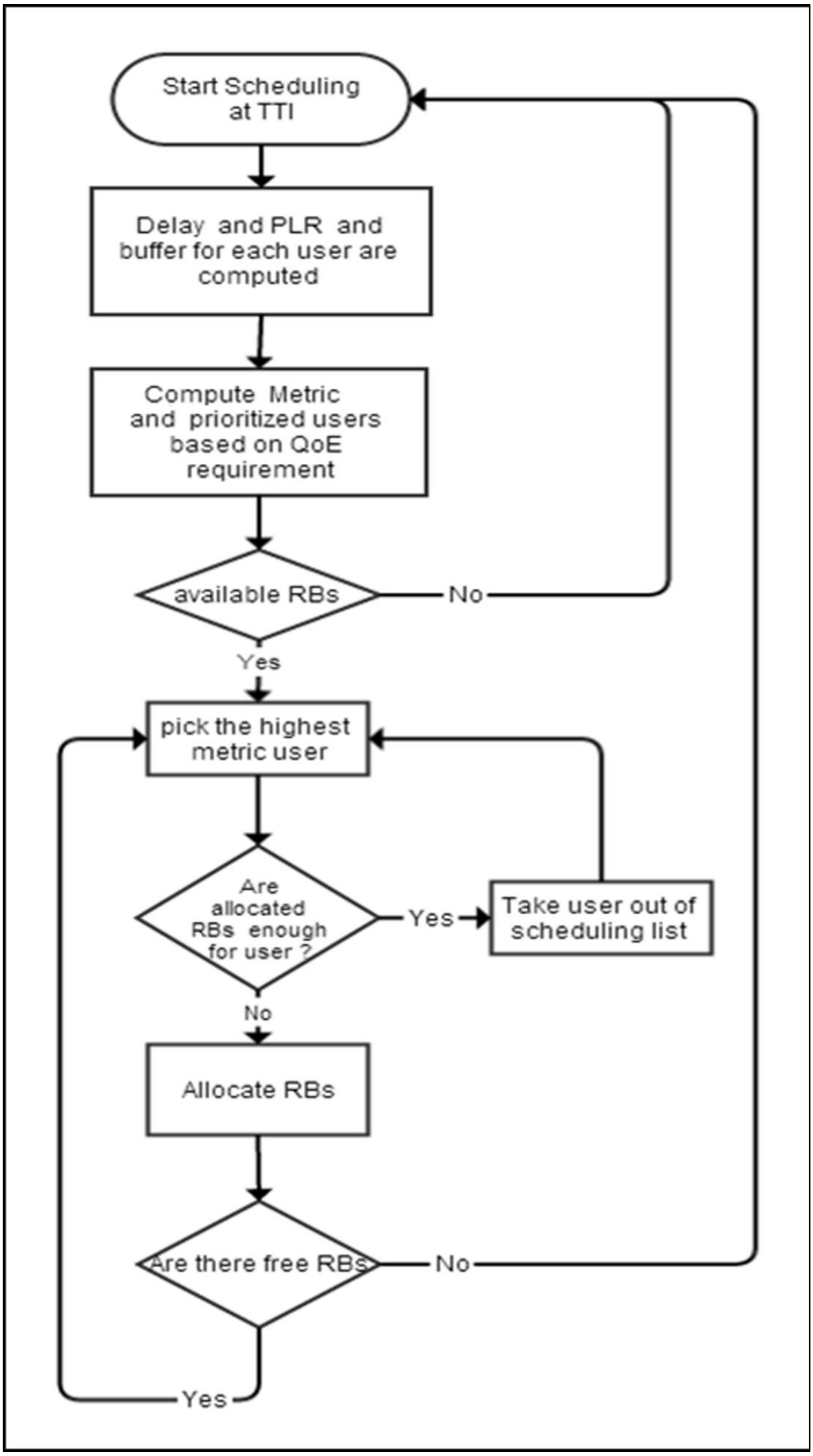


Figure 5.2: QoE-driven scheduler flow chart

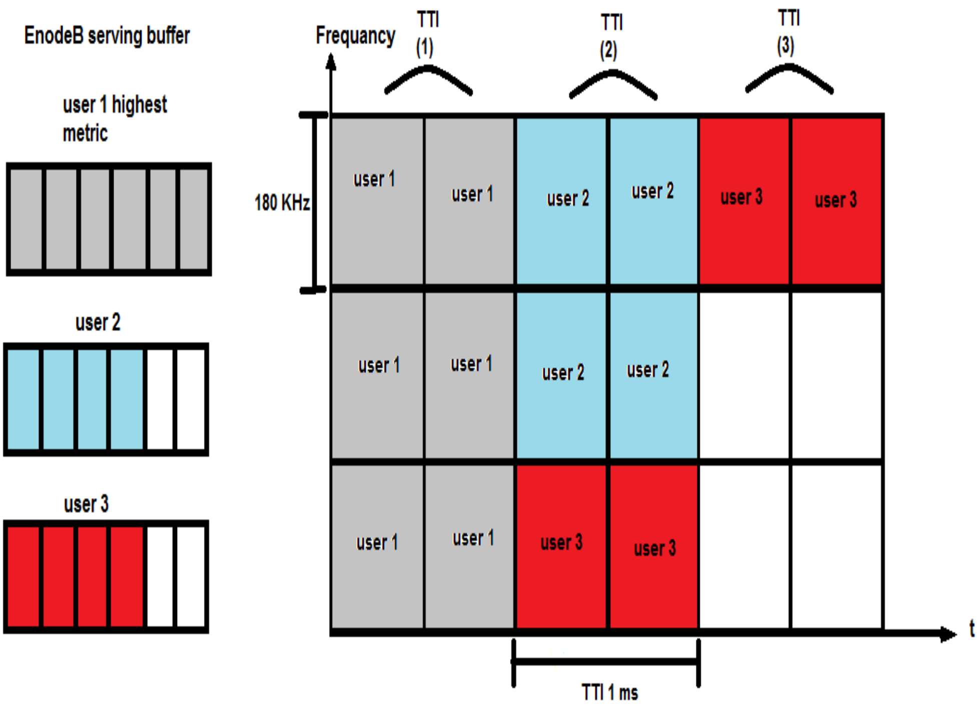


Figure 5.3: QoE-driven scheduler resource allocation

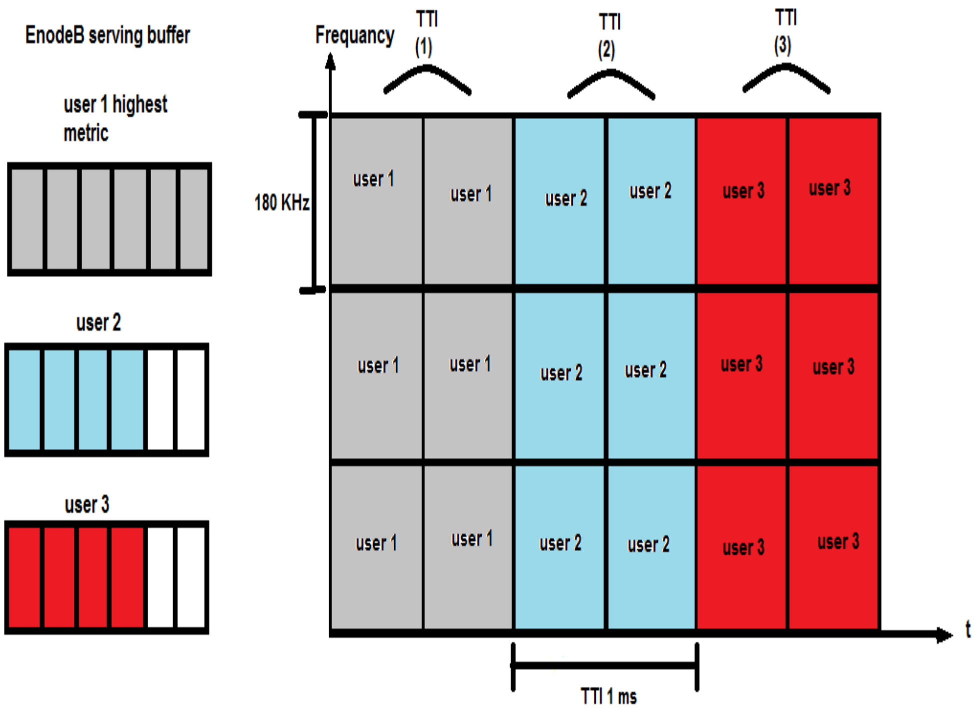


Figure 5.4: LTE-Sim scheduler resource allocation

5.4 Simulation Environment

The performance of the proposed algorithm against the popular LTE scheduling algorithms, i.e. PF, M-LWDF and EXP/PF within LTE network, has been evaluated. The simulation scenarios were adapted based on a single-cell scenario as described in chapter 3.

5.5 Simulation results

This chapter is focused on analysing the performance of the QoE-driven scheduling algorithm versus PF, MLWDF, and EXP/PF in a single cell with a set of users at different speeds. The VoIP flow was considered to evaluate the QoE performance. The MOS score was measured using the packet loss ratio and end-to-end delay that is provided in the simulation results. The equation from [55] was used to determine the MOS score for the VoIP application.

All experiment results are shown in Figures 5.5 to 5.13. The figures show that the results of the lower speed static, 3, and 30 KM/H scenario demonstrate minor changes in term of end-to-end-delay and packet loss ratio (PLR) and MOS scores, as the condition of channels has not being changed at lower speeds. The end-to-end delay results are fairly similar in all scenarios, as shown in Figure 3.5. The PF has the worst result as the end-to-end delay increases sharply, and it reaches above 150ms after 30 users. M-LWDF, EXP-PF and the proposed algorithm have maintained the end-to-end delay to less than 50 ms with all sets of users. The proposed algorithm shows the best performance in keeping end-to-end delay low.

Moreover, the proposed algorithm performed well and maintained a PLR of less than 1% for at most 70 users in most of the simulation scenarios, but it can only maintain this with 20 users at the 120KM/H scenario as the PLR is jumped over 1% after 20 users. The MLWDF has recorded slightly better handling the PLR at a speed of 120 KM/H. The highest packet loss ratio was reached with EXP-PF. The relation between MOS score and packet loss ratio is opposite. The higher the packet loss ratio, the lower the MOS score. According to standard telecommunications applications, the acceptable MOS score is 3.5. This also follows standards defined according to ITU-T Recommendations P.800 [11]. The proposed algorithm has improved cell capacity at lower speeds to allow 70 users receiving VoIP services with an acceptable MOS score compared to 40 users with MLWDF. While the proposed QoE-driven algorithm only reaches 20 users with an acceptable MOS score at speed of 120km/h scenario, MLWDF allows 50 users. Table 5.4 depicts the maximum number of users per cell with an acceptable MOS. Overall; the proposed algorithm shows good results in keeping end-to-end delay and PLR at the lower boundary through all scenarios. Also, the performance of the proposed algorithm in term of MOS score compares favourably.

Table 5.3: The maximum number of users access at acceptable MOS

User speed	PF	MLWDF	EXP-PF	QoE-driven
Static	20	40	20	70
3KM/H	20	40	20	70
30KM/H	20	40	20	70
120KM/H	30	50	20	20

This has improved the cell capacity by 75% compared to MLWDF and 250% compared to PF and EXP-PF in all three lower-speed scenarios. When the motion speed is at 120 KM/H, QoE-driven scheduling can achieve similar cell capacity as PF and EXP-PF, but it has lower cell capacity when compared with MLWDF. This is due to the higher packet loss, although delay did not have a great effect in this scenario.

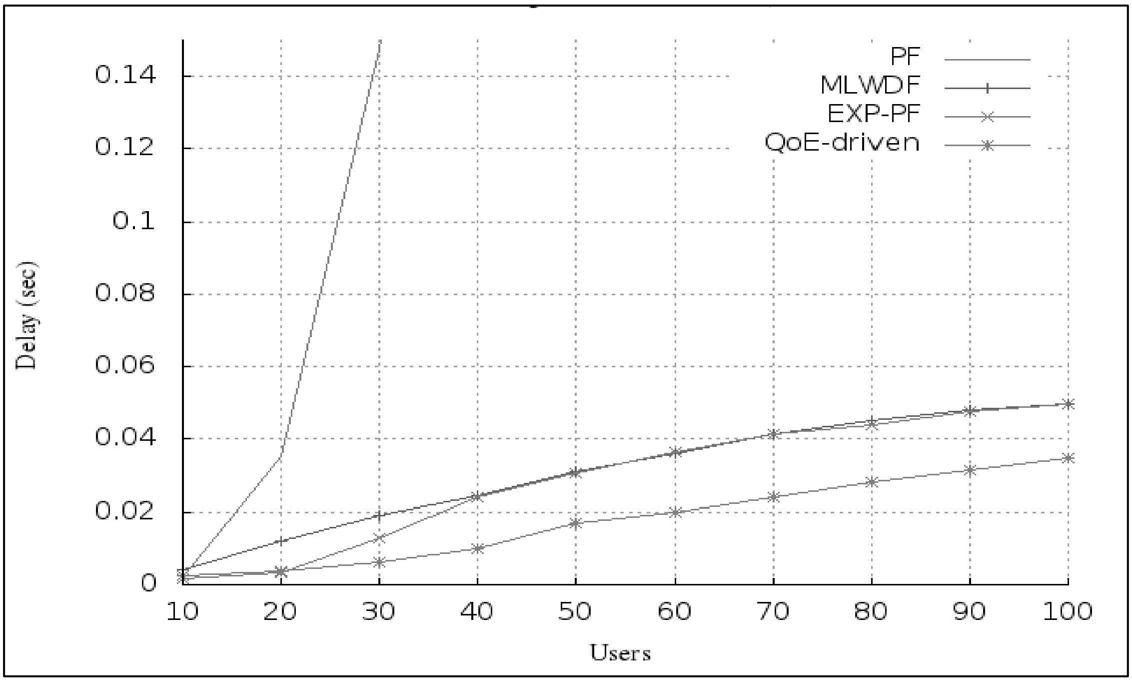


Figure 5.5: End-to-end delay vs. number of users at Static

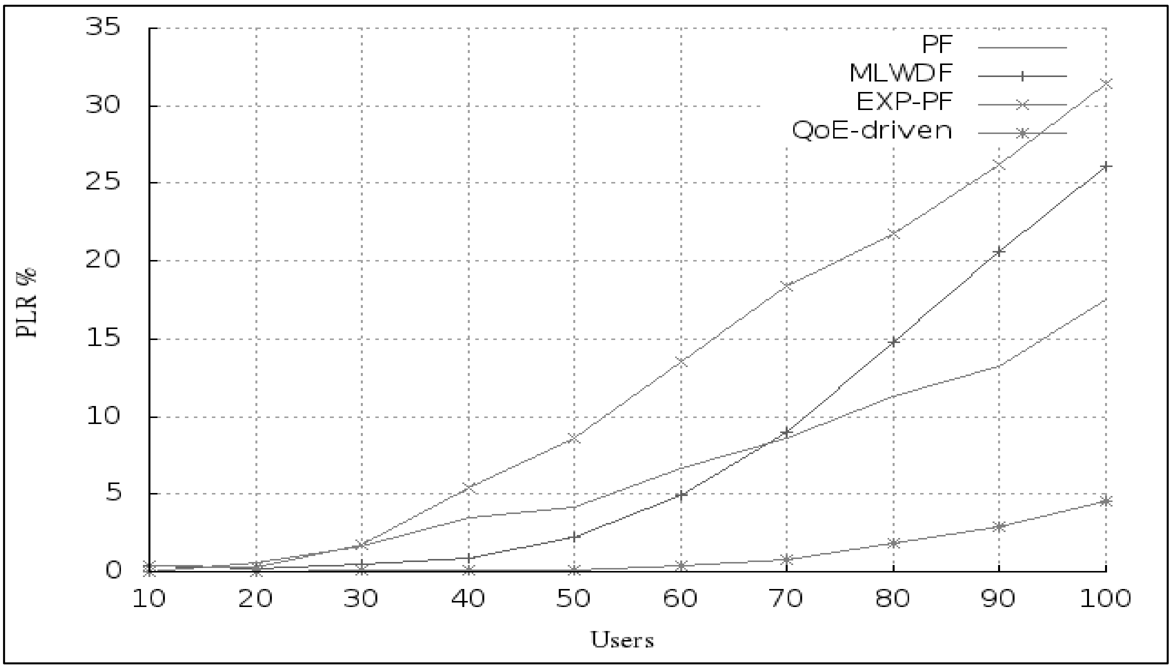


Figure 5.6: Packet loss ratio vs. number of users at Static

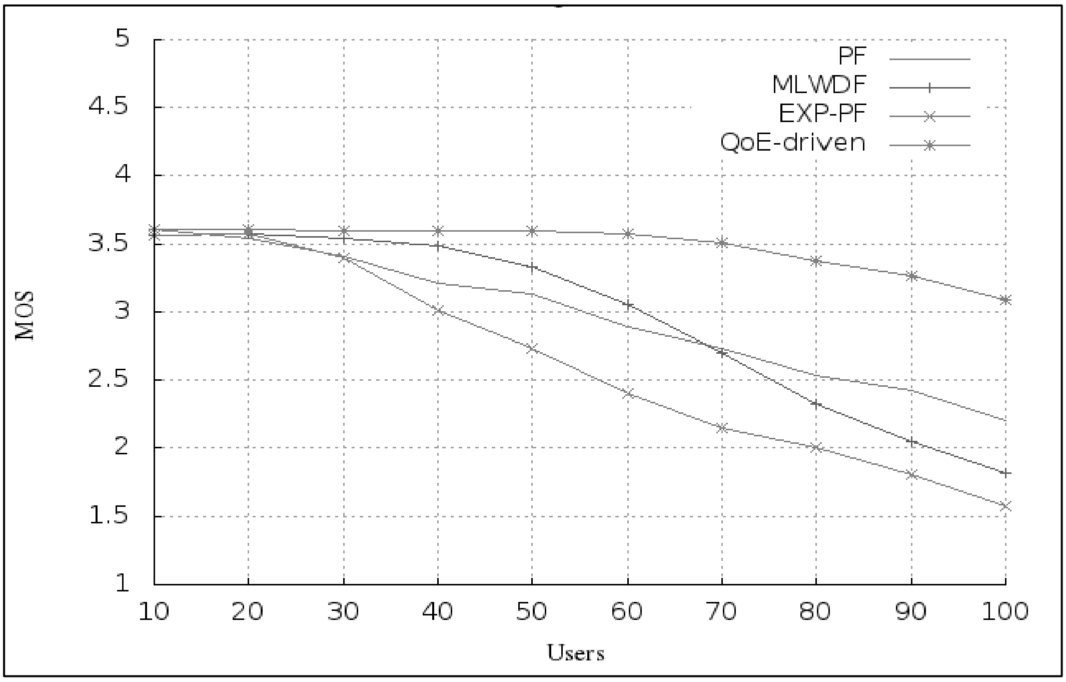


Figure 5.7: MOS Score vs. number of users at Static

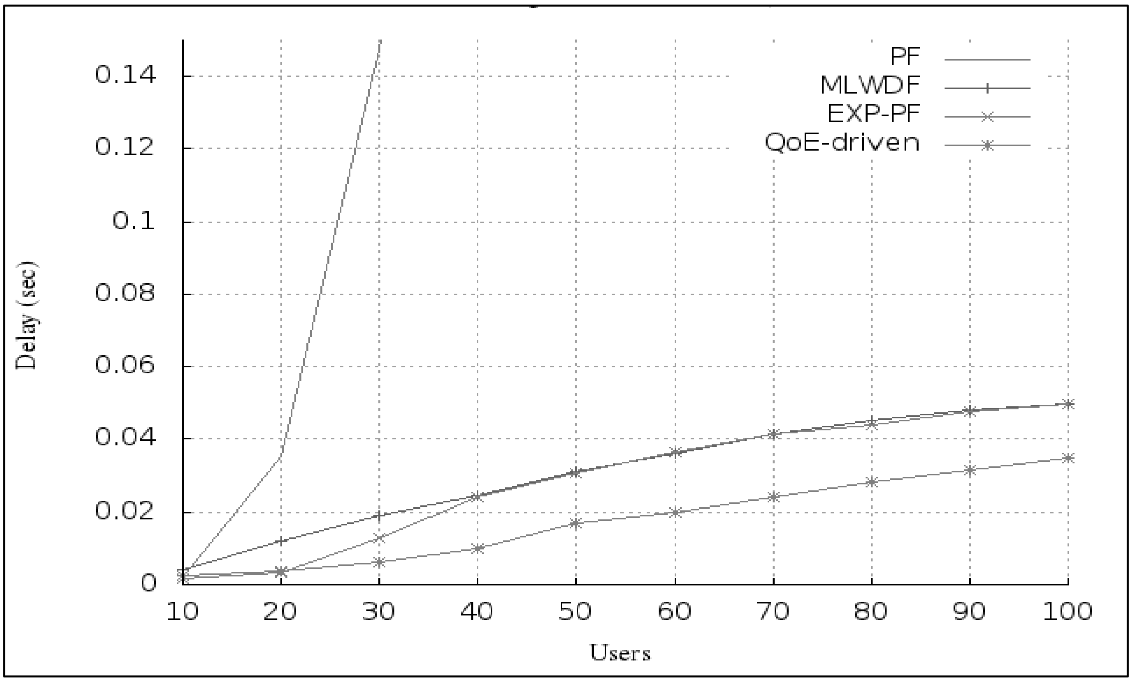


Figure 5.8: End-to-end delay vs. number of users at 3KM/H

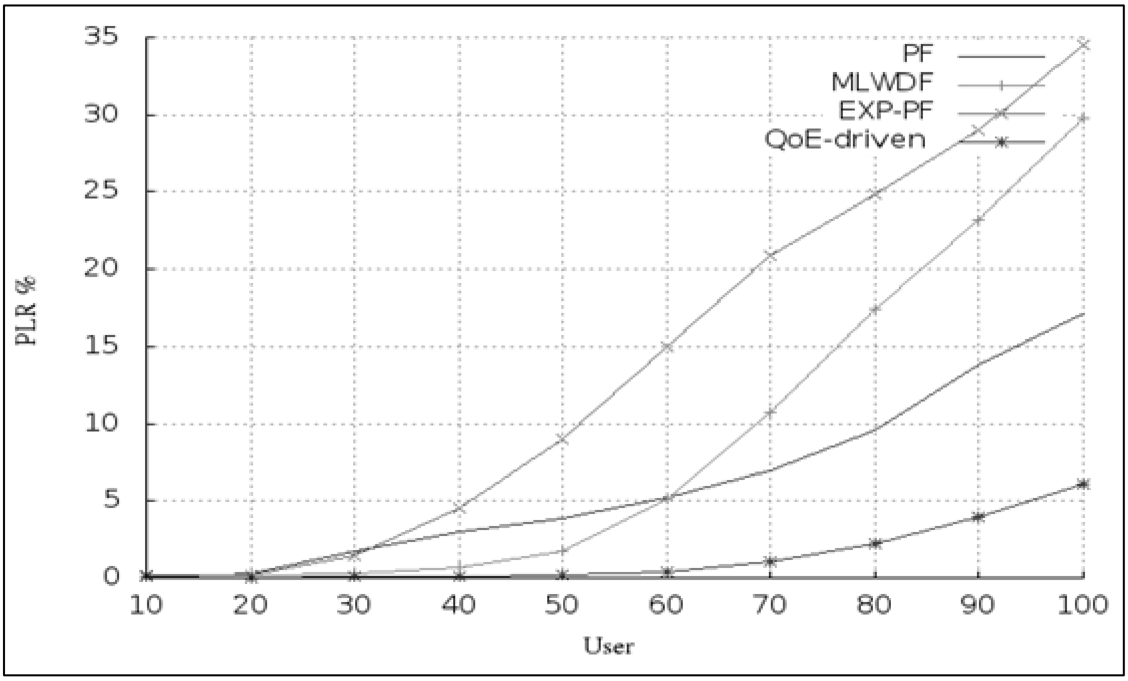


Figure 5.9: Packet loss ratio vs. number of users at 3KM/H

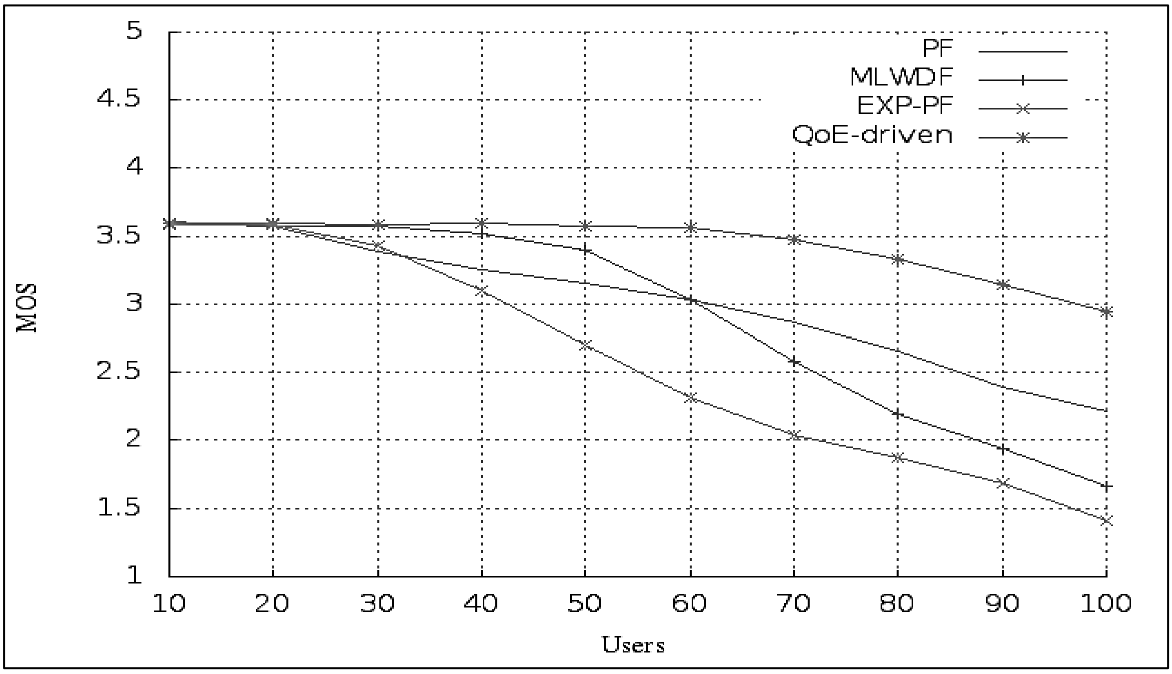


Figure 5.10: MOS Score vs. number of users at 3KM/H

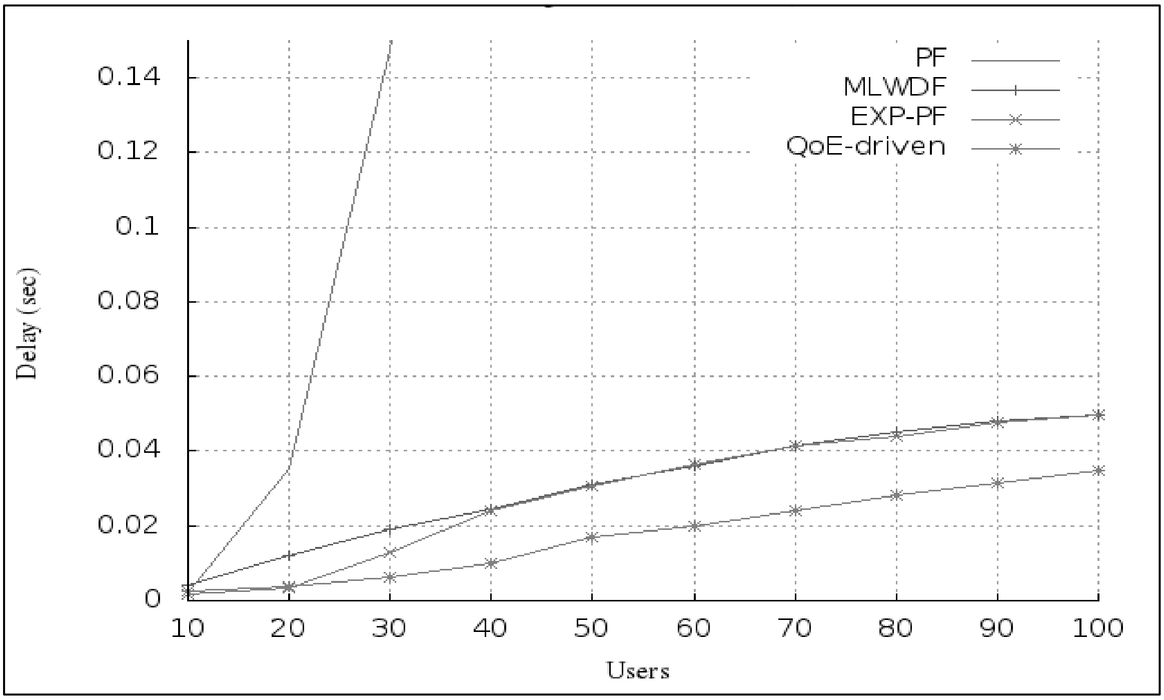


Figure 5.11: End-to-end delay vs. number of users at 120KM/H

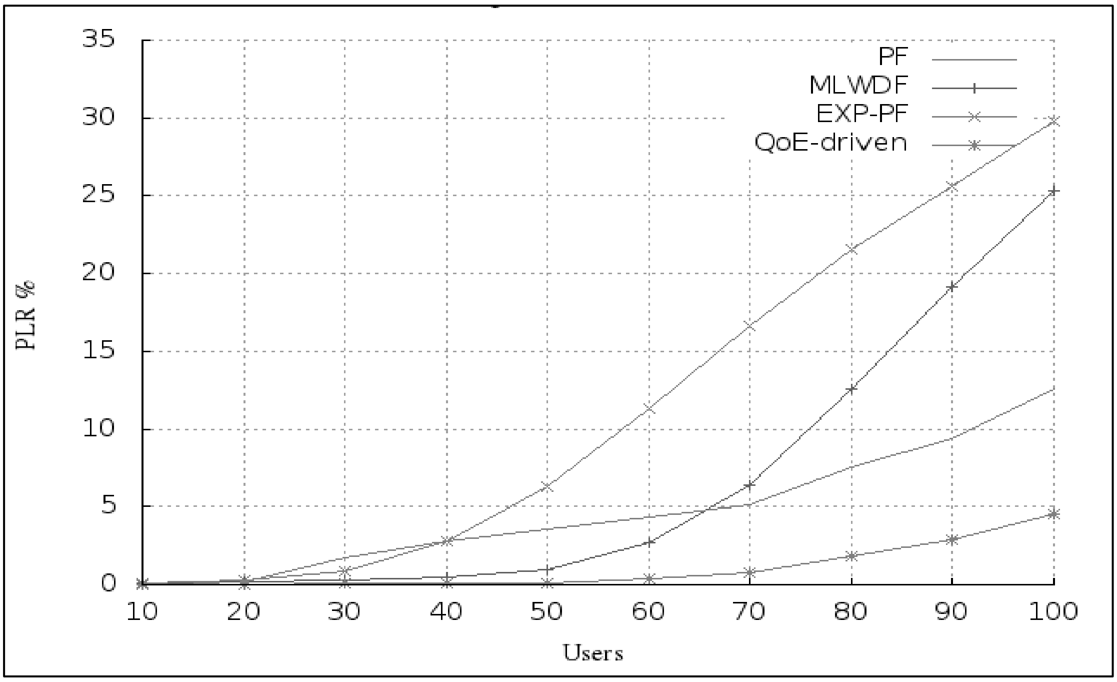


Figure 5.12: packet loss ratio vs. number of users at 30KM/H

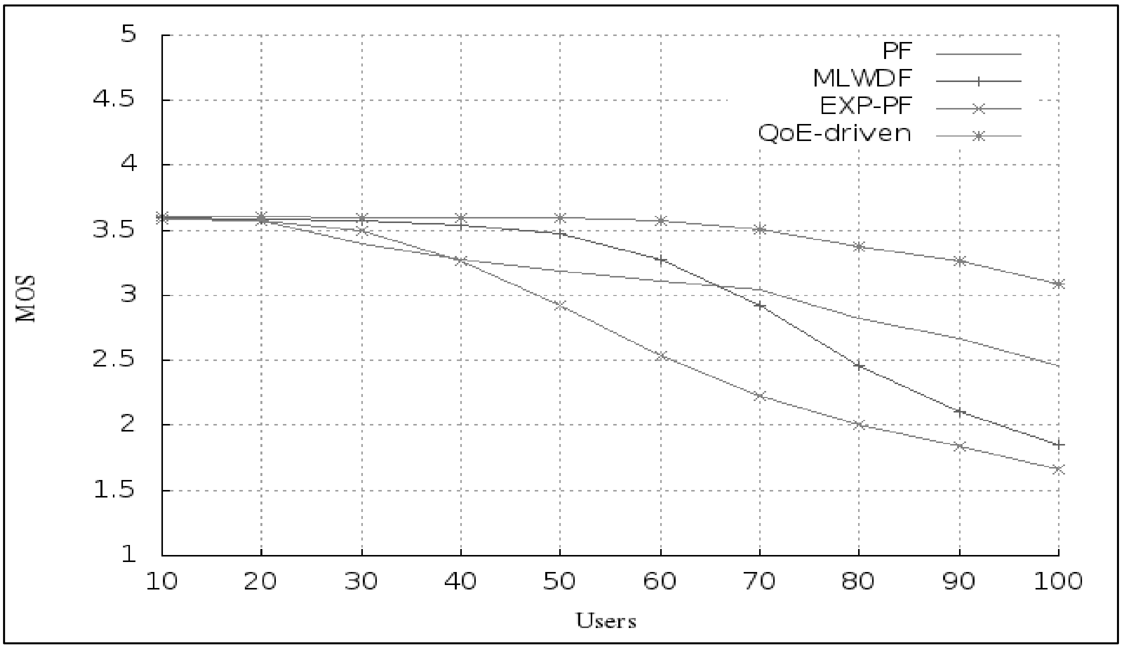


Figure 5.13: MOS Score vs. number of users at 30KM/H

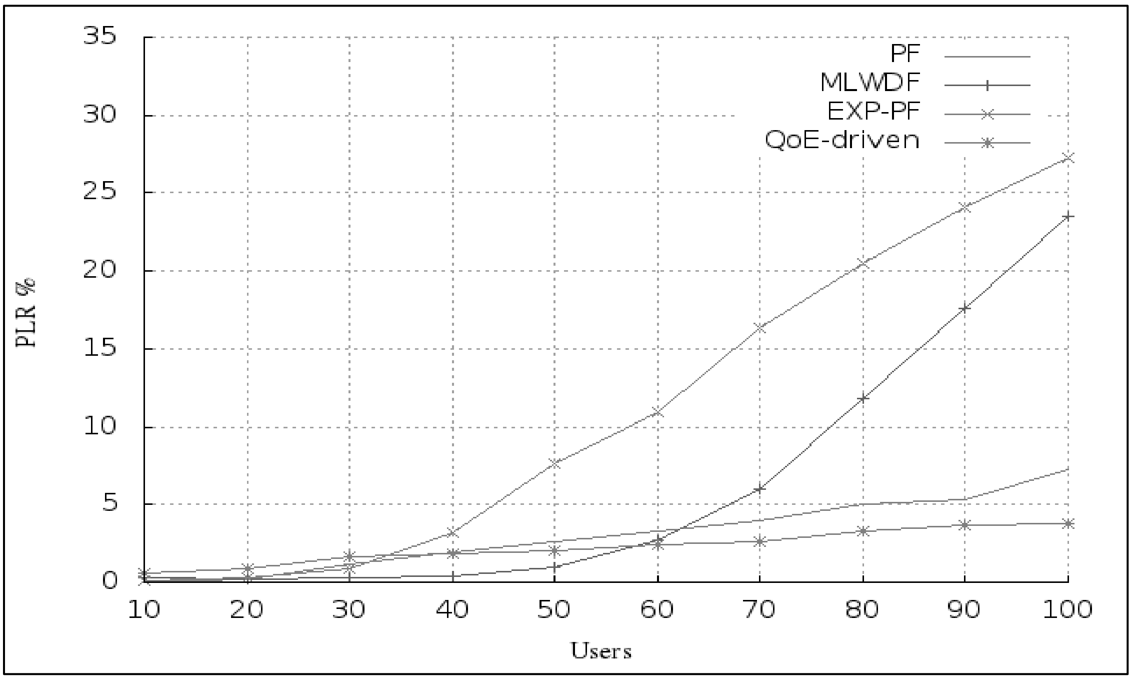


Figure 5.14: Packet loss ratio vs. number of users at 120KM/H

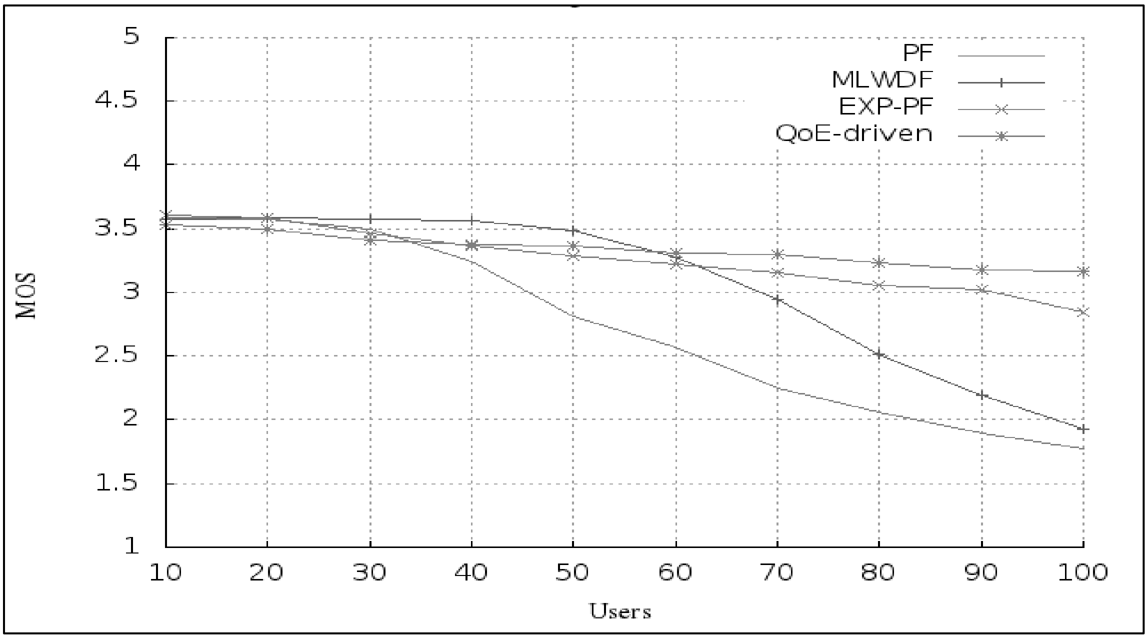


Figure 5.15: MOS Score vs. number of users at 120KM/H

5.6 Summary

The proposed QoE-driven scheduling algorithm was evaluated and compared with three LTE scheduling algorithms (i.e., PF, EXP-PF, and MLWDF) for a single-cell scenario by using LTE-Sim. Different speeds and numbers of users were simulated to monitor the voice quality of the LTE scheduling algorithms. The MOS scores were calculated using an equation based on network parameters (packet loss and delay) and codec type same as chapter 4.

The proposed algorithm achieved remarkable improvement in cell capacity by 75% at an acceptable MOS score comparing to MLWDF and 250% compared to PF and EXP-PF in all three lower-speed scenarios. When the motion speed is at 120 KM/H, QoE-driven scheduling can achieve similar cell capacity as PF and EXP-PF, but it has lower cell capacity when compared with MLWDF. It maintains PLR of at most 1% at lower-speed scenarios.

In the next chapter, the impact of QoE-driven scheduling on other services will be evaluated, such as video streaming. QoE-driven scheduling will also be improved to distinguish services based on the QoE requirement and be able to accommodate more users with multi-service scenarios.

Chapter 6 QoE-MS Downlink scheduling for multimedia application

6.1 Introduction

Many multimedia applications are developed for smart phones such as Vine, YouTube, and mobile video gaming, which motivate end users to heavily use wireless networks (LTE). According to the latest Global Mobile Data Traffic Forecast from Cisco [1], mobile video traffic exceeded 50% of total mobile data traffic in 2012 and increased to 55% in 2014. The video traffic will continue its growth by 75% between 2014 and 2019, accounting for 72% of the total mobile data traffic by the end of the forecast period (i.e. 2019).

The importance of QoE awareness in designing scheduling algorithms to incorporate different multimedia applications has gradually been recognized, and several studies have been published in the literature recently. It has been arguing that it may not be necessary to target a maximized QoE for all users under limited mobile resources from a business/cost point of view. End users may also not notice so obviously the increase of QoE at the high end, while the precious mobile resources could have been saved to serve for more end users.

In this chapter, a novel QoE-driven Multi-Service (QoE-MS) scheduling algorithm for multimedia application in downlink LTE systems is proposed to improve the cell capacity with satisfactory QoE for VoIP and video streaming services. The proposed algorithm predicts the QoE metric for each user and allocates LTE resource blocks to achieve a satisfied QoE for each user. This approach has not been taken by the previous research on LTE scheduling to the best of our knowledge. The proposed QoE-MS algorithm has also enhanced on QoE-driven scheduling approach that proposed in chapter 5 regarding the following three actions: 1) support multimedia services including video, instead of just one VoIP service; 2) provide a guaranteed level of satisfied QoE for each user; and 3) determine the impact of different video content types (e.g. gentle or rapid movement video) on the cell capacity.

The proposed scheme was implemented in LTE-Sim, and its performance was evaluated and compared with two existing scheduling approaches (i.e. MLWDF and EXPPF). Preliminary results show that the proposed scheme achieved very good performance at pedestrian scenario by improving cell capacity with Rapid Movement (RM) video stream to double. For Medium Movement (MM) video stream, the cell capacity increased by 20% when compared to MLWDF and 65% compared to EXPPF. In the vehicular scenario, it can still manage to enhance the cell capacity for the MM video stream.

The remainder of the chapter is organized as follows. Section 6.2, the QoE prediction and evaluation models used for voice and video services are introduced. Section 6.3, the proposed QoE-driven Multi-Service (QoE-MS) scheduling scheme is presented. Section 6.4, the detailed simulation scenario and parameter setting are discussed. The performance evaluation and comparison of the proposed scheme with other approaches are addressed in Section 6.5 and finally the summary is given in Section 6.6.

6.2 QOE PREDICTION AND EVALUATION MODELS

This chapter discusses a non-intrusive QoE metric to estimate the level of users' satisfaction for the multimedia applications from network- and application-related parameters such as packet loss rate, frame rate, and sender bit rate for video. The mean opinion score (MOS) is a common metric to capture users' satisfaction. Although the MOS was originally proposed for voice quality assessment and provided a numerical measure of the quality of the received human voice, it has been extended as a user-perceived quality metric to other services such as video. In this section, I will first discuss the QoE prediction model for VoIP service, the QoE prediction model for video streaming service, and finally the QoE evaluation model based on SSIM for video service.

6.2.1 VoIP QoE Prediction Model

The MOS for VoIP traffic was predicted and evaluated using the equation 2.2 to determine the MOS score from network packet loss and end-to-end delay.

6.2.2 Video Streaming QoE Prediction Model

The video content from raw video is classified into different categories by extracting the temporal and spatial features [65]. This is achieved by using cluster analysis, which can group samples that have several features into similar groups. The three content types for QoE prediction model for video streaming services are set and described as follows:

A. Content type 1, Slight Movement (SM), includes sequences, with a minor moving region of interest (face) on a static background.
B. Content type 2, Medium Movement (MM), includes wide-angled clips in which both background and content is moving.
C. Content type 3, Rapid Movement (RM), includes a professional wide-angled sequence in which the entire picture is moving regularly; e.g. sports videos.

The QoE prediction model for video streaming is calculated using the QoS parameters, which are obtained by nonlinear analysis of the QoS parameters both in the application and network layer, and is given below (Equation 6.1 [65]).

$$MOS = \frac{a_1 + a_2FR + \ a_3 \ln(SBR)}{1 + a_4PLR + \ a_5(PLR)^2} \tag{6.1}$$

Where SBR denotes Sender Bite rate, FR denotes Frame Rate, and PLR denotes Packet Loss Rate. The re-fitted metric coefficients a1, a2, a3, a4, and a5 along with the goodness of fit for all three video groups are listed in Table 6.1.

Table 6.1: The re-fitted metric coefficients [65]

Coeff	SM	GW	RM
a1	2.797	2.273	-0.0228
a2	-0.0065	-0.0022	-0.0065
a3	0.2498	0.3322	0.6582
a4	2.2073	2.4984	10.0437
a5	7.1773	-3.7433	0.6865
R_2	90.27%	90.99%	99.57%

6.3 Video Streaming QoE performance Evaluation Matric

In the above section, the QoE prediction model is based on a non-intrusive approach which is able to predict QoE from network- and application-related parameters. In order to assess the performance of different scheduling schemes, intrusive or full-reference QoE evaluation model based on the structural similarity (SSIM) index is utilized, which is able to predicate/evaluate QoE by comparing the reference and the degraded video clips as defined in equation 2.4.

Table 6.2 shows the mapping between the SSIM and MOS scale [66]. Several studies have shown that the *average* SSIM computed over a sequence of frames of a video clip is generally a good QoE index for the video as well [67], [68], and [66].

Table 6.2: Mapping from SSIM to MOS

SSIM	MOS	Quality
> 0.99	5	Excellent
0.95, 0.99	4	Good
0.88, 0.94	3	Fair
0.5, 0.87	2	Poor
<0.5	1	Bad

6.4 QoE- DRIVEN MULTI-SERVICE (QoE-MS) scheduling

The LTE scheduler is set at eNodeB, which is responsible for resource allocation among mobile users at each transmission time interval (TTI) of 1ms. The smallest resource unit is called Resource Block (RB). Assume that a mobile network contains a single cell which can support up to *N* mobile users. The QoE-MS scheduling becomes an optimization problem to maximize the number of users which the mobile node can support subject to satisfactory QoE achievable by all users within the cell.

Assumption 1:

Let user i (i=1 , …, N) and service k (k=1,…, K) and cell j (j=1,.., M) and Maximum accessible resource block RBmax.

For single cell case j=M=1. Each user i, can access K services (k=1,.., K)

$$\sum_{i=1}^{N}\sum_{k=1}^{K} RB_{ik} \leq RB_{max} \tag{6.2}$$

Where RB_{ik} represents the required RB for user i, with service k.

Assumption 2:

Each services has their share of RBmax (i.e. $RB_k = RB_k = Pk.RBmax \ \sum_{k=1}^{K} Pk = 1$)

Where Pk is the pre-defined.

Let $d_{ij}(t)$ be the head of line (HOL) packet delay for user i at cell j at current scheduling TTI t . $p_{ij}(t)$ and $b_{ij}(t)$ denote the packet loss rate and the occupied buffer size for user i at cell j at TTI t. $RB_j(t)$ is the number of accessible RBs at cell j at time TTI t. The constraint of this scheme is that the received $QoE_{ij} \geq QoE_{min}$ (I assume $QoE_{min} = 3.5$, or MOS of 3.5, which is widely used in the telecommunications world for an acceptable quality for services such as VoIP and $QoE_{min} = 3$ for video streaming). The optimization problem is the maximization of the number of users (i.e. user representing an active flow such as VoIP and video) for LTE cell based on Equation 6.3 subject to Equation 6.4 and Equation 6.5.

$$\max \sum_{\mathrm{k}=1}^{\mathrm{K}} \mathrm{O}_k \tag{6.3}$$

Where O_k represents the number of users at service k

Subject to:

$$QoE_{ik} \geq QoE_{\min(k)} \tag{6.4}$$

$$\sum_{i=1}^{N} RB_{ik}(t) \leq RB_{max}(t) \tag{6.5}$$

The assumption is that resource allocation will be based on the priority of a service e.g. if the VoIP has priority of 1, all VoIP service will assign resource first. After VoIP is allocated then move to service 2 with priority of 2 during each resource allocation. If QoE_{ik} reaches $QoE_{\min(k)}$ the user will be allocated the required resources.

An LTE scheduling scheme for a single cell case (*j*= *M* = *1*) with video trace based and VoIP as real-time services for simplicity is illustrated below as scheme for LTE scheduling. The iteration process is carried out at every TTI t.

Table 6.3: QoE-MS LTE scheduling scheme

Scheme: QoE-MS LTE Scheduling	
Input	Delay $d_i(t)$, PLR $p_i(t)$, and buffer $b_i(t)$ for user i ($i \in [1, \cdots N]$) at time TTI t with SB and FR
1:	Predict $QoE_i(t)$ for video and VoIP flows
2:	calculate the metric, $m_i(t)$ according to Equation 6.6
3:	Schedule a user with the highest metric the required RBs matching its buffer size $b_i(t)$
4:	Check for the remaining RBs and schedule the resource to the user with the 2nd highest metric
5:	Repeat Step 4 until the TTI end.

The scheduling metric equation as follow:

$$m_i(t) = ((\frac{r_i(t)}{\mathrm{R}(t)} \cdot d_i(t) \cdot p_i(t)) - g_i(t)) \tag{6.6}$$

Where $r_i(t)$ and $R_i(t)$ are expected throughput and average achieved throughput for user *i* at time *t*, respectively.

g is the function to guarantee the scheduling maintain QoE_{min} for multimedia services

$$g_i(t) = \min(0, (QoE_i(t), -QoE_{min}) \tag{6.7}$$

6.5 Simulation scenario

The simulation scenarios for a single cell were implemented as described in chapter 3. The performance of the QoE-MS scheduling scheme was evaluated and compared with that of MLWDF [69] and EXP/PF for both VoIP and video flows. The QoE metric (in terms of MOS) for VoIP was measured using the QoE model derived from packet loss and delay, as shown in Equation 2.1. The QoE metric for video was predicted using the MOS model derived from packet loss rate, frame rate, and sender bit rate, as shown in Equation 6. 1. The video QoE of the simulated system was also evaluated by calculating the average SSIM of the received video and then mapped to MOS as illustrated in Equation 2.2 and Table 6.2.

Table 6.4 summarizes the parameters used in the simulation. Each simulation scenario was run 5 times, and then the average results were taken.

Table 6.4: Simulation parameter

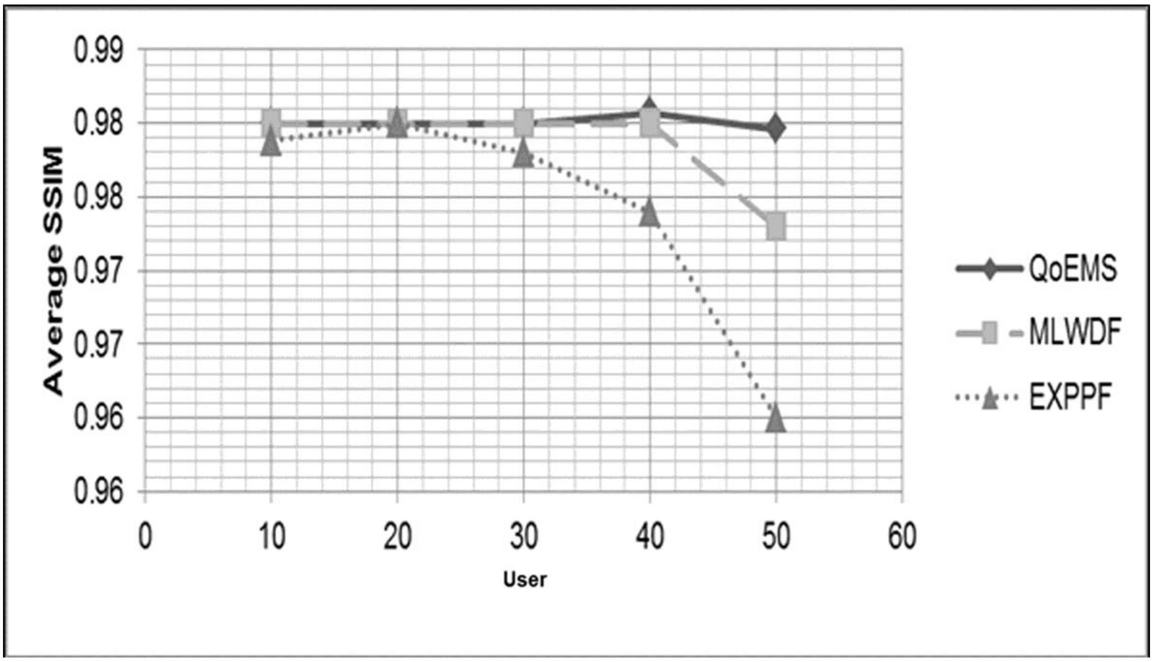
Parameter	Value
User speed	3, 30 KM/H
Traffic model	Real-time flows: H.264 for AKIYO, FOREMAN, and STEFAN video traces at 242 Kbps and 30 fps VoIP G.729. Best effort flows: infinite buffer

6.6 Performance of the proposed QoE-MS Scheduling

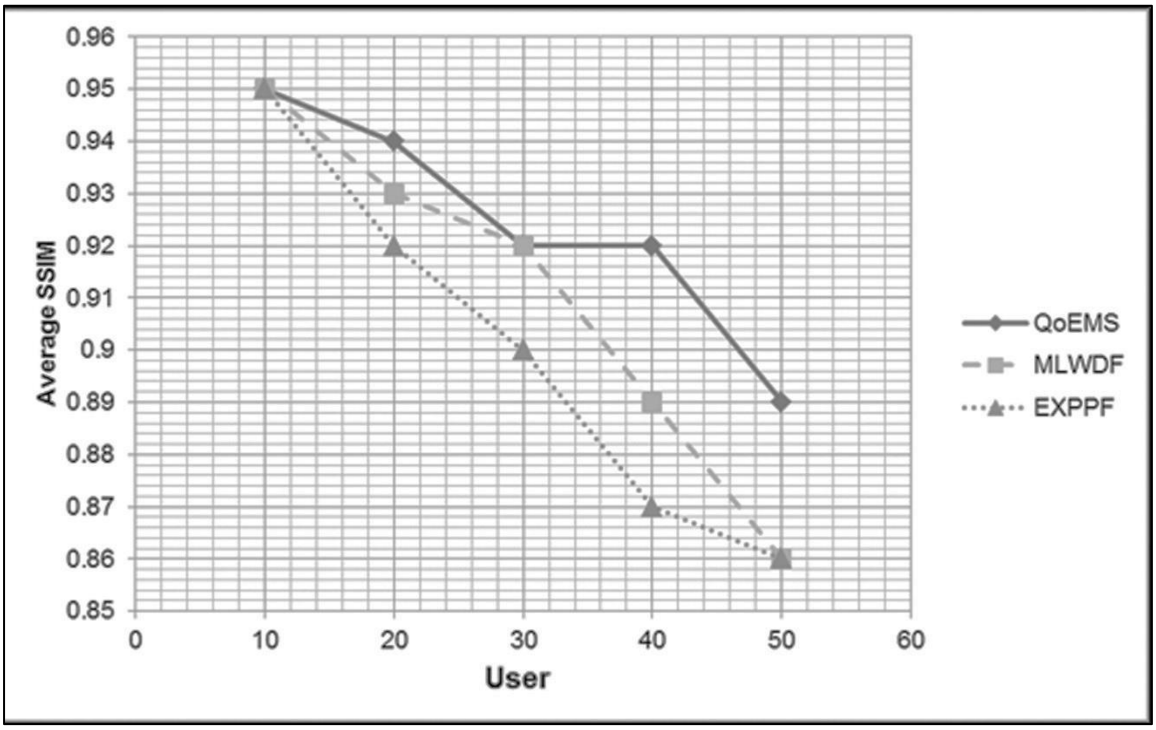
In this section, the proposed QoE-MS scheme is compared with other two scheduling algorithms (i.e. MLWDF and EXPPF). In the simulation, three types of services are running in the network: three types of videos mentioned in Table 6.3 as the video streaming traces, VoIP and infinite buffer for best effort data flow. The number of users is increased from 10 to 50 at two different speeds (pedestrian at 3 KM/H and vehicular at 30KM/H).

The average SSIM values of the received videos were calculated for pedestrian and vehicular cases, as shown in Figures 6.1 and 6.2, respectively. It can be observed that the average SSIM value achieved by QoE-MS is better than other algorithms in all scenarios. Also, average SSIM decreased rapidly as the number of user increases, especially with MM foreman and RM Stefan videos. This indicates that the more movement in the video required, the more resources are needed to achieve satisfactory QoE.

a) **AKIYO video**



b) **FOREMAN video**



c) **STEFAN video**

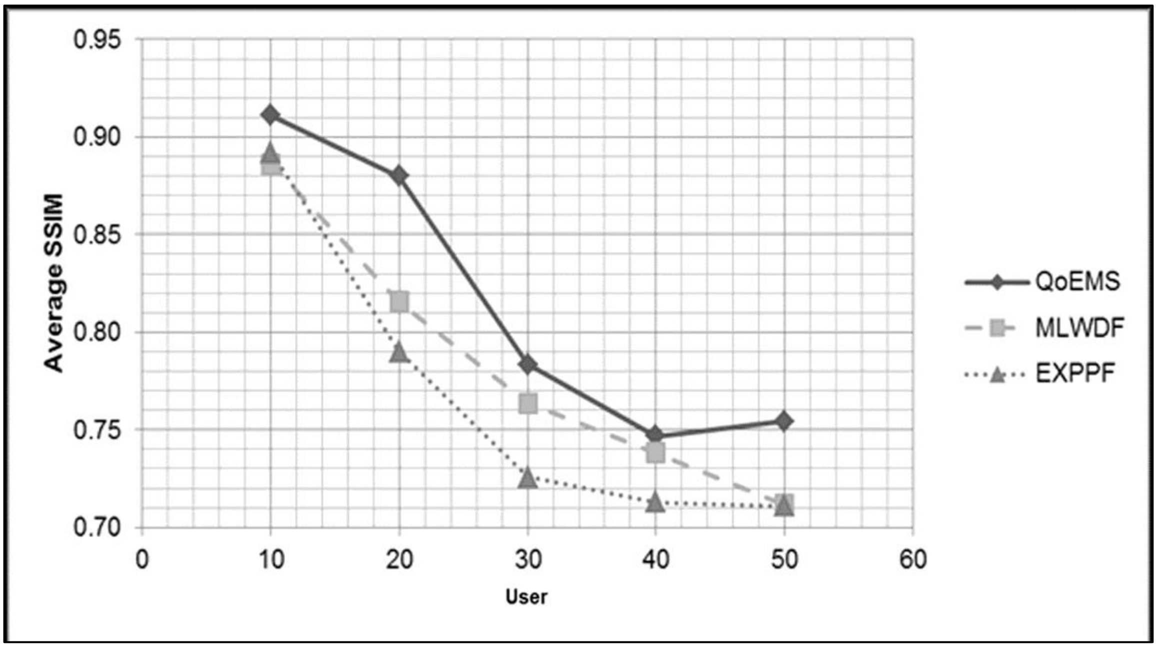
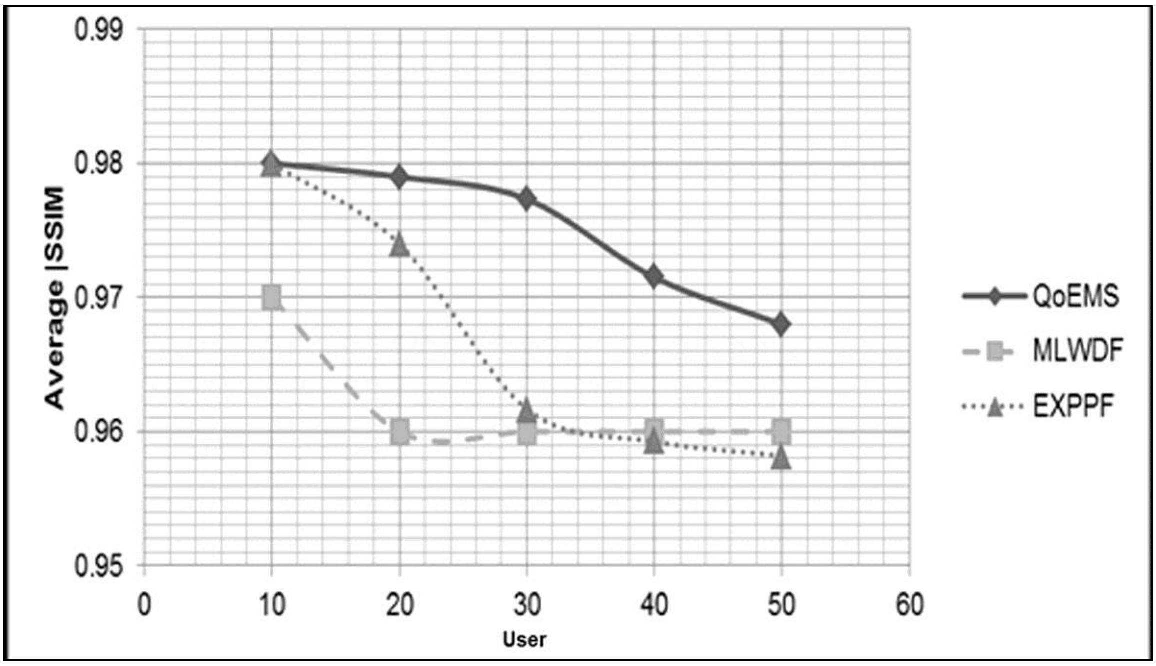
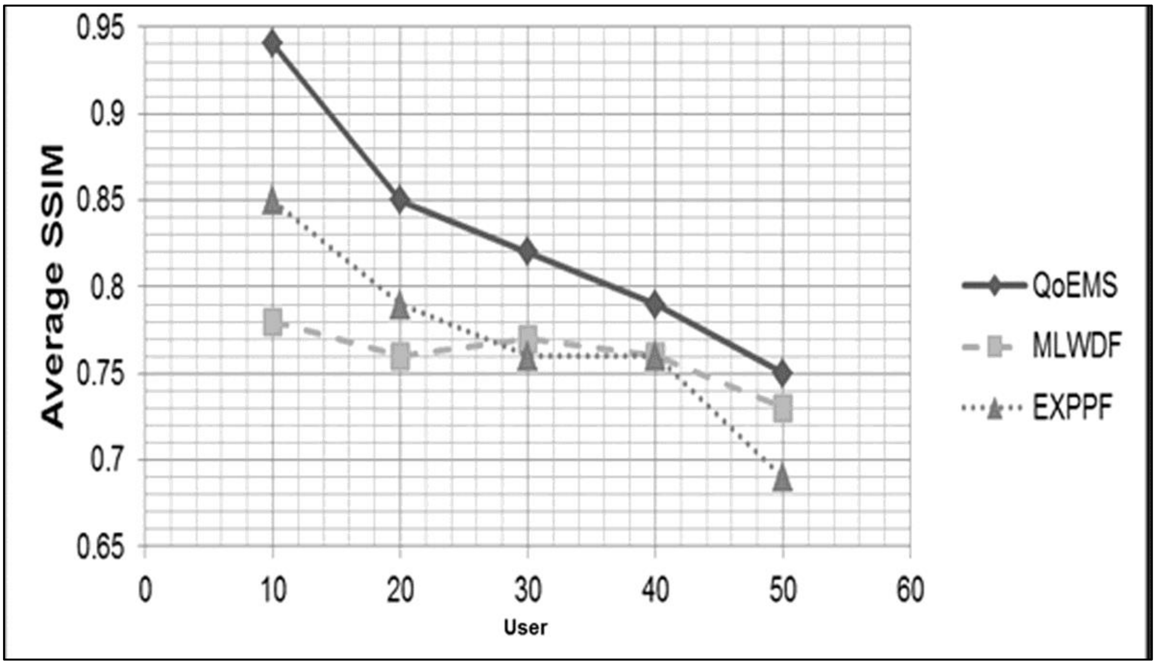


Figure 6.1: Average SSIM values for the three videos scenarios a) AKIYO b) FOREMEN c) STEFAN at user speed of 3 KM/H, varying the number of users

a) **AKIYO video**



b) **FOREMAN video**



c) **STEFAN video**

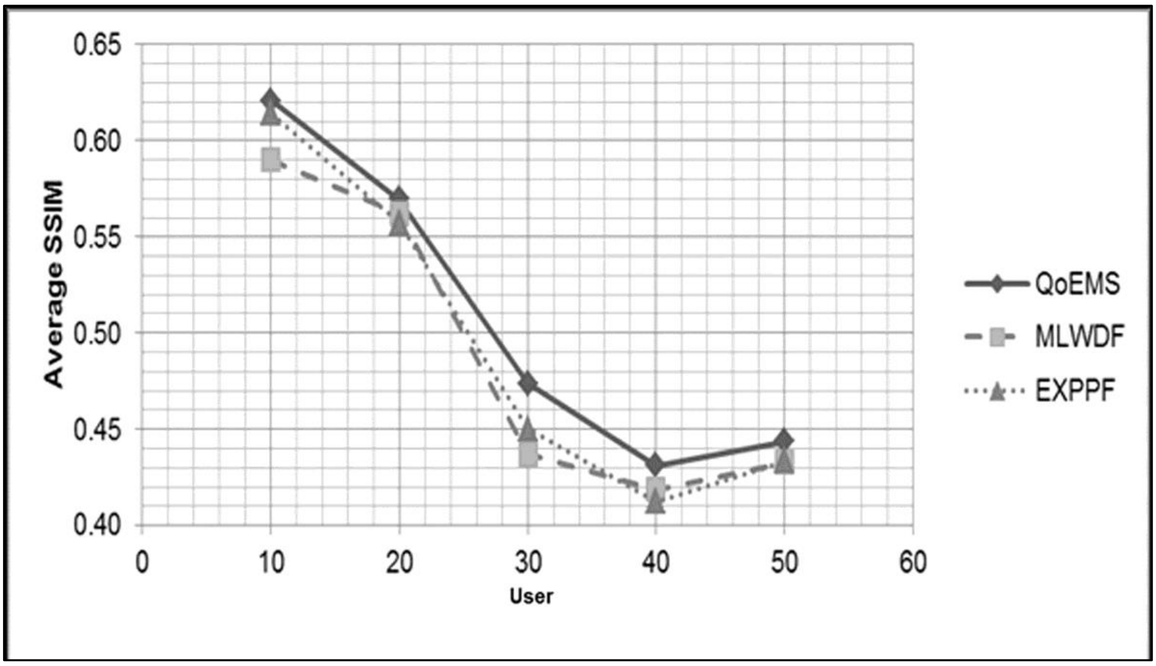


Figure 6.2: Average SSIM values for the three videos scenarios a) AKIYO b) FOREMEN c) STEFAN at user speed of 30 KM/H, varying the number of users

Overall, the proposed algorithm performed very well in the pedestrian scenario by improving cell capacity with Rapid Movement (STEFAN) video stream to double (from 10 users to 20 users), as shown in Figure 6.3. While for Medium Movement (MM) video stream (FOREMAN) the cell capacity has increased by 20% compared to MLWDF (from 40 users to 50 users) and 65% (from 30 users to 50 users) comparing to EXPPF, in the vehicular scenario, it can only manage to enhance the cell capacity for MM video stream, as shown in Figure 6.4 (from zero to 10 users). The Slight Movement (SM) AKIYO video stream has the same performance across all scenarios. In vehicular scenarios, the results show that the average SSIM dropped sharply, which reflects the decreased number of satisfied users. This raised the point that user speed and video content type are factors that impact the cell capacity with the satisfied QoE user.

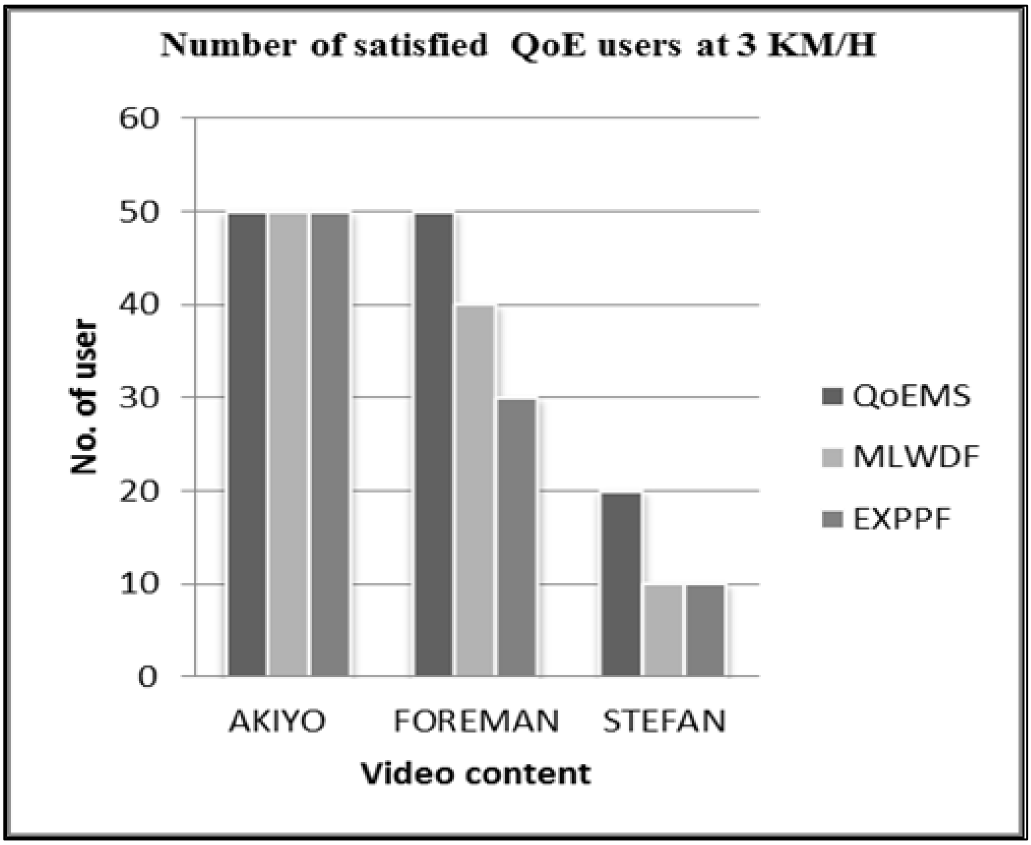


Figure 6.3: Number of satisfied QoE users for video at pedestrian

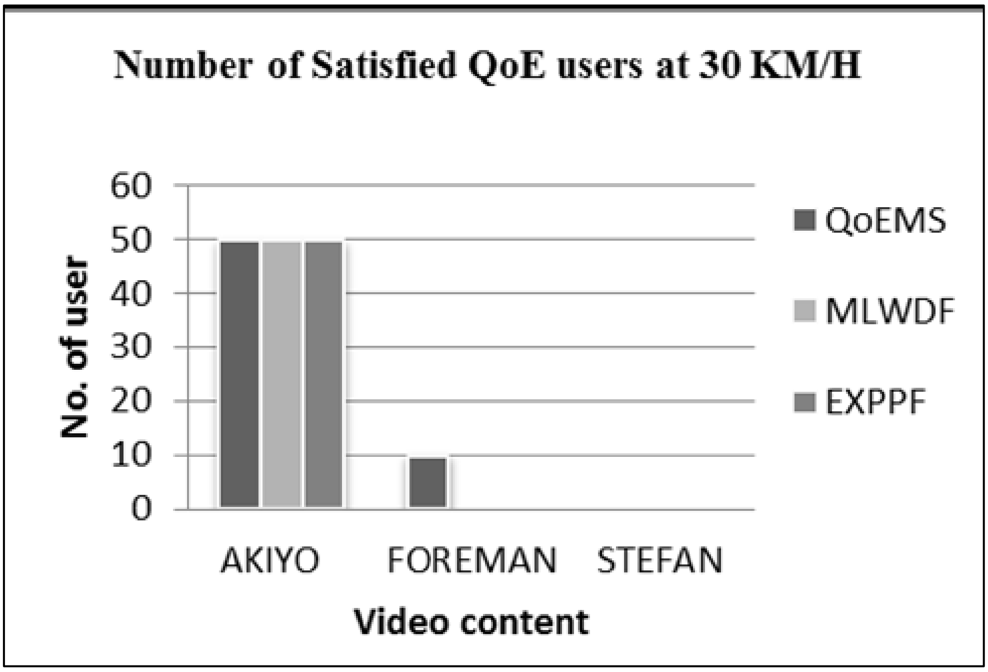
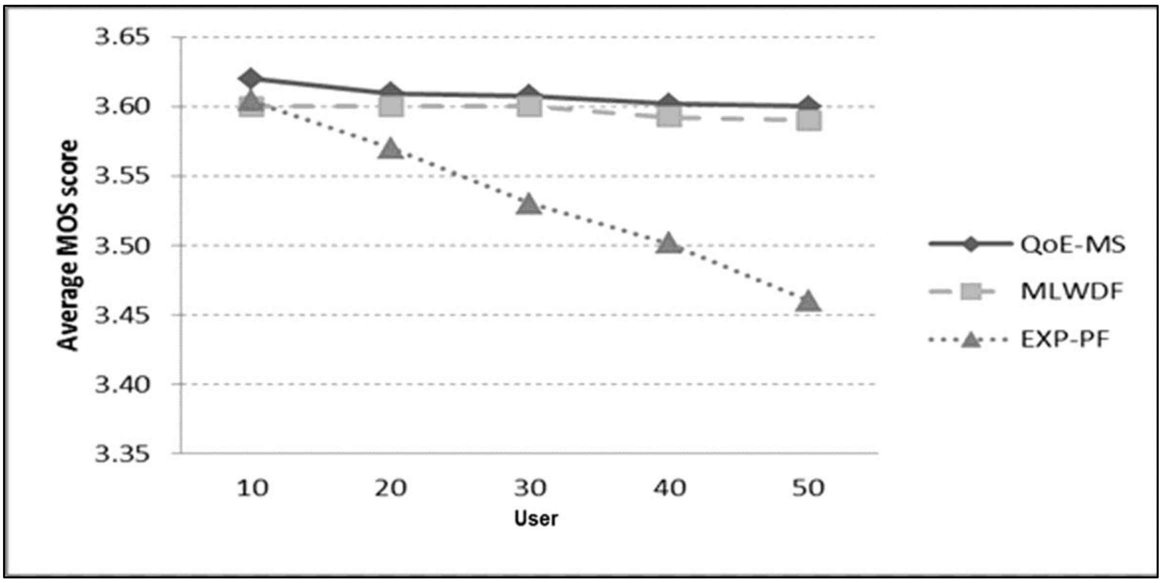


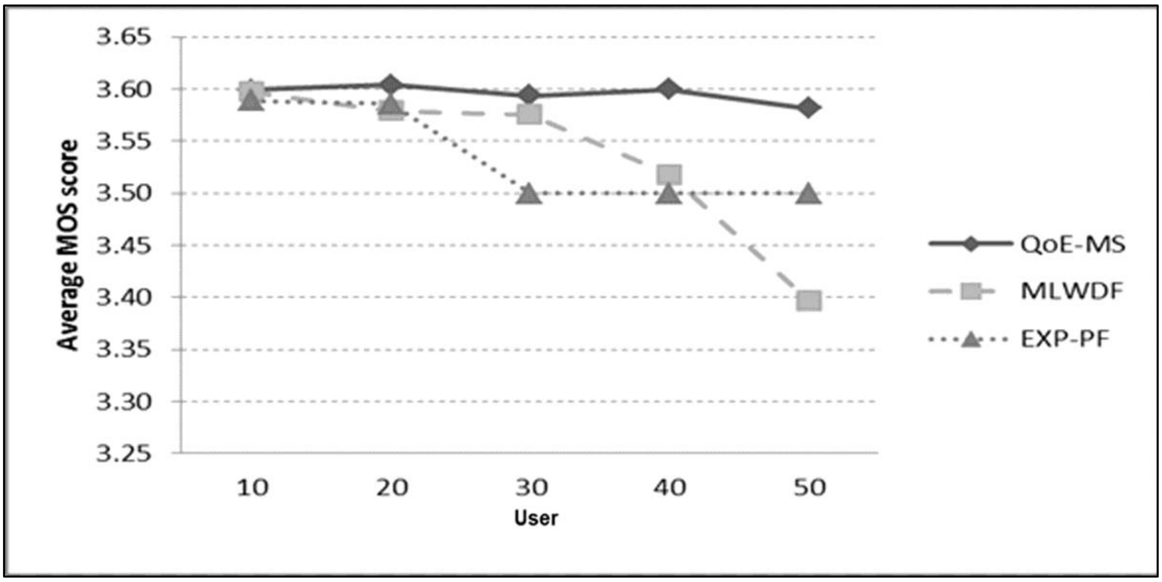
Figure 6.4: Number of satisfied QoE users for video in vehicular scenario

In terms of VoIP, the proposed algorithm has improved cell capacity in pedestrian and vehicular scenarios to allow 50 users receiving VoIP services with an acceptable MOS score compared to 40 users with MLWDF in FOREMAN and STEFAN video content types. On the other hand, QoE-MS and MLWDF have a similar number of access users with acceptable MOS but QoE-MS still has slightly better MOS score as is shown in Figures 6.5 and 6.6.

a) **AKIYO video**



b) **FOREMAN video**



c) **STEFAN video**

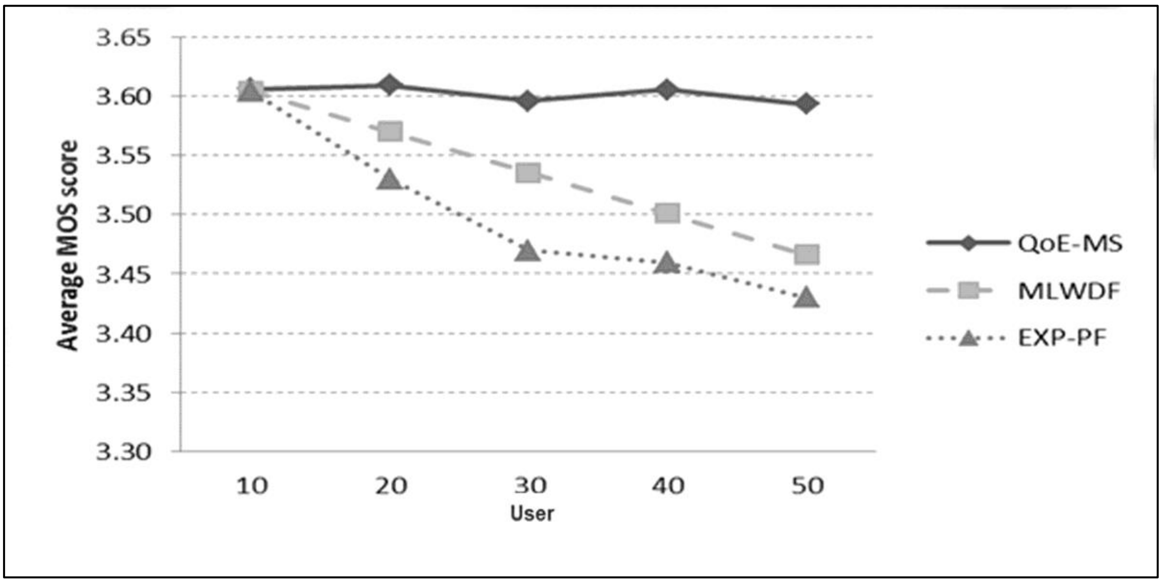
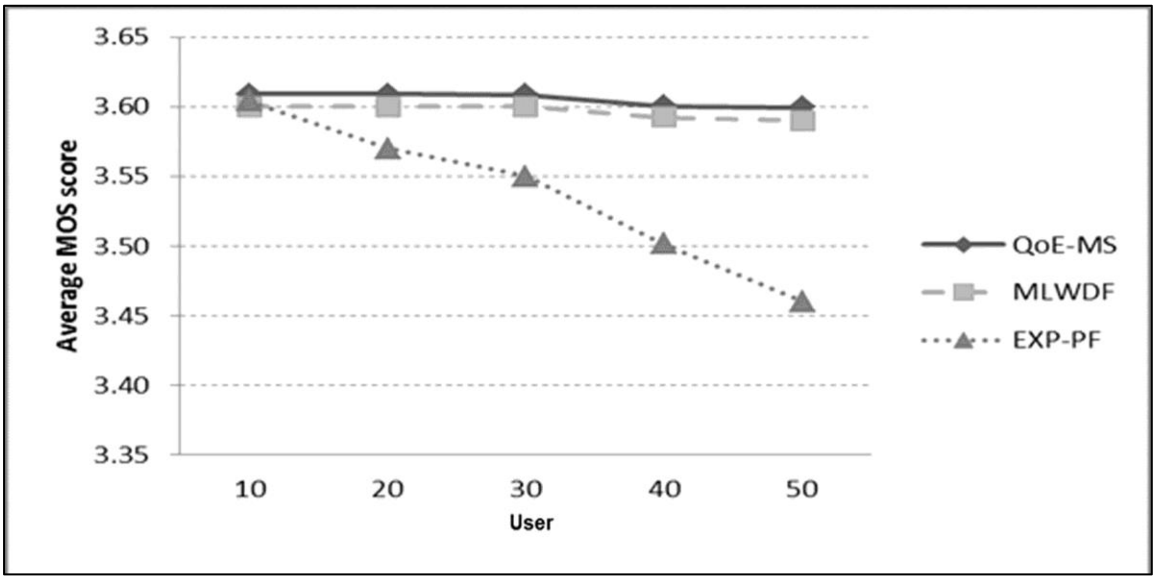
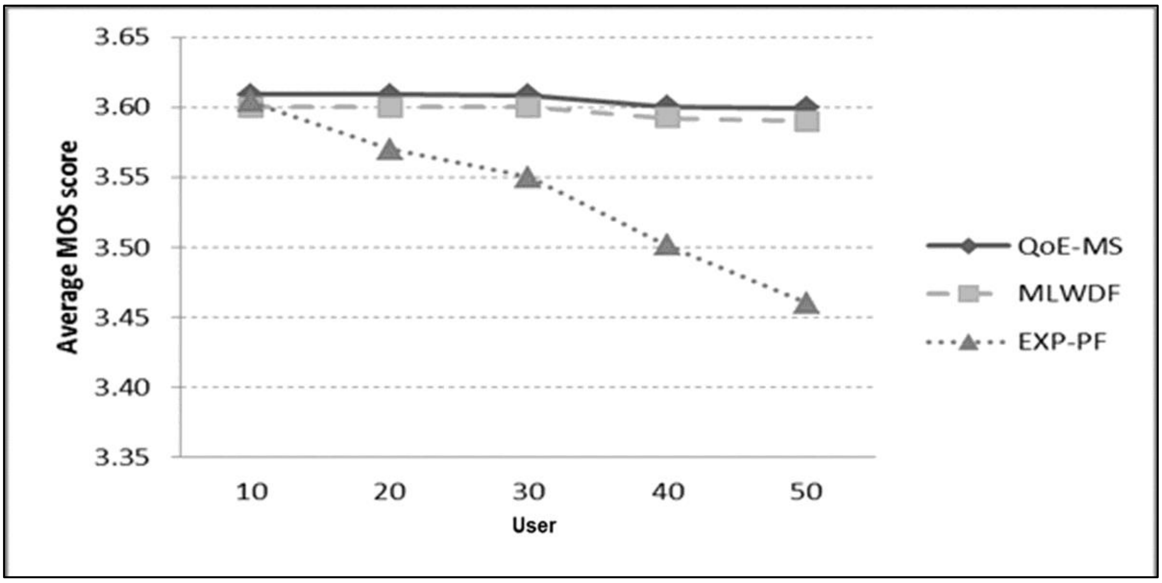


Figure 6.5: Average MOS score for VoIP with the three videos scenarios a) AKIYO b) FOREMEN c) STEFAN at user speed of 3 KM/H, varying the number of users

a) **AKIYO video**



b) **FOREMAN video**



c) **STEFAN video**

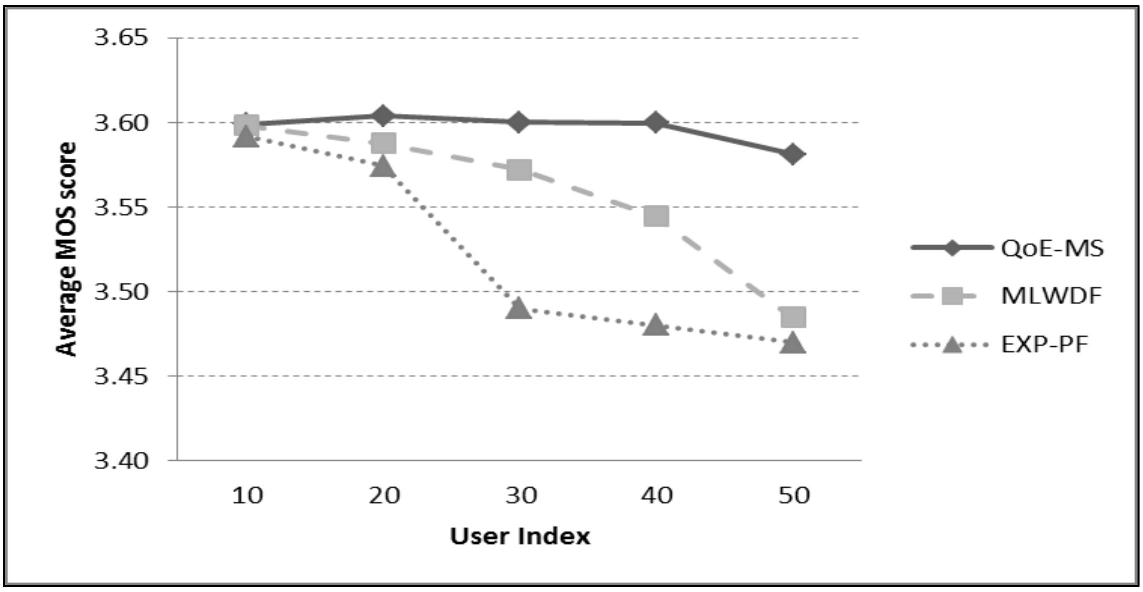


Figure 6.6: Average MOS score for VoIP with the three videos scenarios a) AKIYO b) FOREMEN c) STEFAN at user speed of 30 KM/H, varying the number of users

Table 6.5 depicts the maximum number of users per cell with an acceptable MOS. Overall, the performance of the proposed scheme in terms of MOS score compares satisfactorily by improving the cell capacity by 60% compared to EXPPF and 20% compared to MLWDF. The video content types have slightly impacted the cell capacity for VoIP service, as it is different from the SM AKIYO video scenario.

Table 6.5 The maximum number of users with access at acceptable MOS for VoIP

User speed	QoE-MS	MLWDF	EXP-PF
3KM/H	50	40	20
30KM/H	50	40	20

6.7 Summary

This chapter proposed a QoE-MS scheduling scheme for LTE networks. The proposed scheme was designed to improve the cell capacity with satisfactory QoE for multimedia application (VoIP and video streaming). The maximization of the LTE cell capacity was achieved by prioritizing users based on the predicted QoE performance for the user. The performance evaluation has been carried out on LTE-Sim. Pedestrian and vehicular mobility scenarios were implemented for VoIP and videos streaming users.

The proposed scheme has improved the LTE system capacity for pedestrian scenarios by 20% for MM video, while it achieved superior improvement by 100% for RM video compared with MLWDF. Moreover, it achieved a 65% capacity increase for MM video when compared to EXPPF. Although only 10 satisfied users were served with QoE-MS scheduling, MLWDF and EXPPF fail to get any satisfied users with MM video in vehicular scenarios. The SM video stream has the same performance across all scenarios. The QoE-MS increases the system capacity by 60% compared to EXPPF and 20% compared to MLWDF.

In Term of VoIP application the proposed scheme was enhanced the number of satisfactory user by 60% compared to EXPPF and 20% compared to MLWDF. The video content types have slightly impacted the system capacity for VoIP service, as it is different from the SM AKIYO video scenario.

Overall, QoE-MS scheme is enhanced the LTE system across different video content type scenarios. However the results show the impact of user speed and video content type on the performance of QoE-MS scheme.

Chapter 7 DISCUSSION, FUTURE WORK, AND CONCLUSION

7.1 Introduction

Today the growth of multimedia applications over the mobile network continues, with rapid revolution of both devices (e.g. powerful smart phones) and applications (e.g. Netflix with super HD videos). Hence, users' perceived expectation will be higher; this creates a great challenge for mobile network operators to provide an acceptable QoE for a huge number of mobile users (i.e. 7.4 billion mobile devices in 2019 predicted by Cisco [1]). The trade-off serves limited users with superior quality or a maximum number of users with acceptable quality. The poor quality will lead the user to unsubscribe from the service. QoE investigation has attracted interest from various disciplines and has been studied from different approaches. In order to meet the users' QoE requirements, mobile network operators should predict the perceived quality and take action in advance. In this thesis, QoE of MAC layer was estimated in the LTE network system as the scheduling plays a key role in resource allocation among network users. The LTE downlink scheduling can improve the QoE by distributing the Resource Block among users with the acceptable QoE requirement (i.e. MOS >= 3.5 for VoIP [11]).

The main aims of the project were(1) to investigate and evaluate the impact of different LTE scheduling on the perceived quality of experience for multimedia applications,(2) to develop novel QoE-driven scheduling schemes for improving system capacity in multiservice downlink 3GPP LTE. This is to move away from the traditional QoS-driven scheduling schemes to a novel QoE-driven approach, and (3) to apply developed a novel QoE-driven scheme for multimedia application over LTE networks.

This chapter is organized as follow. Section 7.2 presents the main contributions of this work and highlights its novelty. Section 7.3 discusses the limitation of current work and future works. Section 7.4 draws the conclusions of the thesis.

7.2 Contributions to Knowledge

The key contributions presented in this thesis are

[1]

The work contributes to the performance analysis of the LTE scheduling algorithms in terms of QoE. VoIP was considered as a real-time flow to measure the impact of different scheduling algorithms (i.e. PF, EXPF and MLWDF) on QoE. Also, four different scenarios were developed using a single cell: static, pedestrian at 3KM/H, and vehicular 30KM/H and 120 KM/h scenarios to study the user speed impact on QoE in addition to the scheduling algorithm. The results highlighted the gap between of QoS driven and perceived user quality and showed the need of developing a novel QoE-driven scheduling to meet user QoE.

The work has been contributed to the research community in the following publication [14]. The work is described in Chapter 4

[2] A novel QoE-driven LTE downlink scheduling algorithm was proposed and VoIP was chosen as an example to prove the concept. The investigation and evaluation of the performance of the proposed QoE-driven LTE downlink scheduler was carried out against three popular LTE downlink schedulers, including PF, EXP-PF and MLWDF. The performance evaluation is based on a QoE metric (i.e. MOS score) for VoIP applications. The preliminary results have shown that the proposed algorithm has improved the cell capacity by 75% at an acceptable MOS score (MOS score of 3.5) compared to the MLWDF cell capacity, and 250% compared to PF and EXP-PF algorithms, in all three lower speed categories (i.e. static, 3 and 30 KM/H). The work has contributed to the research community in the following publication [15]. The work is described in chapter 5

[3] The proposed QoE-MS scheme performed well at the pedestrian scenario by increasing system capacity to double for video stream with Rapid Movement (RM) content. For Gentle Waking (GW) video content, the capacity has increased about 20% when compared to MLWDF and 60% when compared to EXP-PF. In the vehicular scenario, the proposed scheme managed to enhance the system capacity for GW video stream case. Although only 10 satisfied users were served with QoE-MS scheduling, MLWDF and EXPPF fail to get any satisfied users with GW video in vehicular scenarios. The SM video stream has the same performance across all scenarios. In addition the performance of the proposed scheme for VoIP application in terms of satisfactory MOS score has enhanced. The system capacity was improved by 60% compared to EXPPF and 20% compared to MLWDF. The video content types have slightly impacted the system capacity for VoIP service, as it is different from the SM AKIYO video scenario.

The work has contributed to the research community in the following publication [16]. The work is described in Chapter 6.

7.3 Limitation of the current work and future works

The work carried out in this thesis has a number of limitations that should be addressed in future study. Firstly, LTE scheduling algorithms were implemented and evaluated using LTE-Sim [57]. While it is beneficial from controlled scenarios, repeatable, cost-effective, and easy to implement, the real network is unpredictable. For example, not all users will demonstrate the same behaviour and the same application. VoIP flows were created by VoIP application. Specifically, an on/off Markova chain was used as a model within the voice flow. The duration of ON period has an exponential distribution of 3s as the average value. The duration of OFF period contains a truncated exponential probability density function, where the upper limit is 6.9s, with the same average as the duration of ON period of 3s [40]. Moreover, the proposed Algorithms were implemented only in single cell scenarios for proof of concept, which is simple scenario compared to complex multi-cell scenarios. As more issues will be raised such as the handover and admission control for each cell.

For that reason, the research can be improved and extended further in future study by performing the following actions:

1. **1.** Conduct the evaluation using real test-bed

The proposed scheduling algorithms can be implemented in a real test-bed in which real mobile devices can generate real VoIP flows and can be evaluated subjectively. Also, live video streaming; which has different imperilment can be implemented to investigate cross-layer QoE optimization.

2. Advance enhancement for QoE-MS

The video content type plays a big role as it was observed from the results obtained. The developed QoE-MS scheduling scheme in this thesis can be extended to recognise different content types by ad tag in the packet of extending in the cross layer. Hence, the MM video has higher acceptance then the rest of the video types. Then the RB available can be used to serve RM video. Besides, other video content type can be tested, and mixed content can also be evaluated. Furthermore, the scheme can be evaluated to different VoIP codecs as the current evaluation was only restricted on G.729 codec. This will help to observe the impact of different VoIP codecs on the QoE and compare it with video content type impact.

3. Multi-cell scenarios for QoE-MS

QoE-MS can be implemented and evaluated using the multi-cell scenarios. Challenges (i.e. handover, admission control) can be observed to provide full solution for improving system capacity with an acceptable QoE performance for multimedia application over LTE networks. Also, an intelligent can be introduced to collaborate between different cells to provide optimal QoE for all users across the LTE networks,

7.4 Conclusions

The enormous growth of mobile users (e.g. on smart phones, tablets), which are using multimedia applications motivated this work to investigate and understand LTE scheduling. The main goal was to provide pleasant perceived quality of multimedia for many mobile users with the existing infrastructure of mobile network providers. The initial stage of this project intended to simulate the current existing LTE scheduling algorithms and evaluate their QoE performance at four user speeds. It was defined that scheduling algorithms and user speed have an impact on the QoE.

Afterward, a novel QoE-driven scheduling algorithm for VoIP application was developed to improve the LTE system capacity. A single-cell scenario was adopted as proof of the concept that this algorithm will work and serve more QoE satisfied users. It achieved outstanding results by improve cell capacity by 250% compared to PF and EXPF and 75% compared to MLWDF. PF was considered out of the league for real-time flows due to high end-to-end delay.

Finally, a novel QoE-MS scheduling algorithm was developed to support the multi-service of multimedia by considering the case of crowded areas such as shopping malls or intercity areas. Also, different video content types were introduced to understand the impact on QoE. The proposed scheme has improved the LTE cell capacity for pedestrian scenarios by 20% for MM video, while it achieved superior improvement of 100% for RM video compared with MLWDF. Moreover, it achieved a 65% capacity increase for MM video when compared to EXPPF. However, only 10 satisfied users were served with QoE-MS scheduling, MLWDF and EXPPF fail to get any satisfied user with MM video at vehicular scenario. The SM video stream has the same performance across all scenarios. The QoE-MS increases the cell capacity by 60% compared to EXPPF and 20% compared to MLWDF. However user speed and video content type have impact on the performance of QoE-MS scheme.

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APPENDICES

I. Published papers:

A. QoE-based Performance Evaluation of Scheduling Algorithms over LTE

This paper was published in IEEE GC'12 Workshop: Quality of Experience for Multimedia Communications

GC'12 Workshop: Quality of Experience for Multimedia Communications 1

QoE-based Performance Evaluation of Scheduling Algorithms over LTE

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***Abstract*— The LTE specification provides Quality of Service (QoS) of multimedia services with fast connectivity, high mobility, and security. However, 3GPP specifications have not defined scheduling algorithms to support real-time and non-real-time application services, this has led to the development of a variety of scheduling algorithms in the recent years. Performance is a key factor for assessing the scheduling algorithms. However, the performance analysis and evaluation for existing scheduling algorithms are limited to QoS parameters (e.g. in terms of throughput, packet loss and delay) and/or fairness metrics. It is not clear how these scheduling algorithms perform in terms of Quality of Experience (QoE) metrics (e.g. MOS score) which are more closely linked with end user experience or user perceived quality. In this paper, we investigate and evaluate the performance of three popular LTE downlink schedulers (proportional fairness (PF), exponential proportional fairness (EXP-PF), and modified largest weight delay first (M-LWDF)) in terms of QoE metric (i.e. MOS score) in addition to normal QoS metrics (e.g. packet loss and delay) for VoIP applications. Simulation results (based on LTE-Sim) show that the best suitable downlink scheduling algorithm is MLWDF, which has short end-to end delay (less than 50ms) and allowing maximum number of user access (more than 50 users) at acceptable MOS score (MOS over 3.5). The widely used scheduling algorithm (PF) will increase its end-to-end delay over 200ms, when number of user access is over 30. This will limit its application for VoIP when user access number increases. This work would help to assist the design and development of QoE-driven scheduling algorithms for mobile multimedia applications in the future.**

***Index Terms*— LTE, packet scheduling, QoE, QoS**

I. Introduction

There has been significant technological development such as the Internet, multimedia and real-time services; this leads to the expectation that the 3GPP Long-Term Evolution (LTE) should be able to achieve a high data rate to support multimedia applications (e.g. VoIP, video gaming and video streaming) over mobile/wireless networks for anytime and anywhere access. The LTE utilises orthogonal frequency division multiple access, also known as "OFDMA" as a means for downlink data transmission. OFDMA provides high data rates through dividing the frequency band into a cluster of common orthogonal subcarriers. This would enhance the capabilities of the system as it could deliver high data rate, supporting diverse range multi-users as well as creating resistance to frequency selective fading of radio channels.

The quality of service (QoS) of the LTE must be able to meet users' expectation through optimising the balance of usage and fairness. Real-time and non-real-time services would need to have outstanding QoS and minimum bit-rate respectively. In order to be able to meet the QoS demands for different services, many packet–scheduling algorithms have been developed to allocate limited frequency and time resources efficiently and fairly to real-time and non-real-time traffic for all data transfer devices including mobile and wireless networks. Examples of such scheduling algorithms include Proportional Fairness (PF), Exponential Proportional Fairness (EXP-PF), and Modified Largest Weight Delay First (M-LWDF) [1][2][5].

The Proportional Fair (PF) [1] introduces some certain PFs, by making use of the current channel status to average throughput ratio. This way the PF provides fairness to the users and PF scheduling optimizes the long term user throughput. However, the PF scheduling algorithms lack the characteristics to support the real-time services, such as Voice over IP (VoIP) service and the real-time video streaming service. The Exponential Proportional Fairness (EXP-PF) algorithm is the enhanced version of PF with priority for real-time flow with respect to non-real-time ones, which provides better QoS performance. Finally, the Modified Largest Weight Delay First (M-LWDF) [2] was designed to support multiple real-time flows by considering a balance between the weighted packet delays in the current channel status. As far as the scheduling algorithms such as M-LWDF and EXP-PF, these provide a much larger delay period and live up to the delay requirements more effectively [2][5]. M-LWDF has a drawback, however, as it is not easy for it to look for the best solution for traffic service.

Currently there are no 3GPP standards for scheduling algorithms to support both real-time and non-real-time applications in LTE-based 4G networks. Extensive research has been carried out on a variety of scheduling algorithms to improve system performance, in terms of throughput, fairness, or other QoS metrics, such as packet loss and delay. It is unclear how these scheduling algorithms perform in terms of end user experience or user perceived quality. The QoE-aware scheduling algorithm is still in its early stages, and only a few

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papers have examined the QoE for video streaming [9][10]. We argue that even VoIP needs to be evaluated when designing a full QoE-aware algorithm.

In this paper, we aim to evaluate the performance of several scheduling algorithms for VoIP applications, in terms of Quality of Experience (QoE) metric (e.g. MOS score), which is more closely linked with end user perceived quality, in addition to normal QoS metrics (e.g. packet loss and delay). This work would help to better understand how scheduling algorithms behave from the QoE point of view, and help to develop QoE-driven scheduling algorithms to achieve maximised QoE for real-time and non-real-time multimedia applications in the future.

To compare the different algorithms, a simulation platform following the 3GPP LTE standard was built up based on an LTE-Sim simulator [4]. Four different scenarios were developed using a single cell: static, pedestrian, and vehicular scenarios using different speeds (0, 3, 30 and 120 km/h, respectively). The single cell includes one enodeB and a variable number of user equipment (UE), ranging from 10 to 100. A realistic scenario was adapted whereby every UE receives video and VoIP flows as well as best effort flows with infinite buffers. Packet loss ratio and end-to-end delays were captured in the simulation, and we evaluated the QoE performance (in terms of MOS score) for the VoIP flows.

Preliminary results show that MLWDF has the best performance across different speed by allowing the maximum number of user access (more than 40 users) at acceptable MOS score of 3.5. While MLWDF and EXP-PF performed very well in minimizing the end-to-end delay in all the cases tested, their performance varied with the packet loss ratio after the user numbers reached more than 30. The PF scored the largest end-to-end delays, but kept the packet loss ratio to a minimum compared to other algorithms.

The rest of the paper is organized as follows. Section II provides basic background on LTE scheduling. Section III discusses the algorithms. Section IV examines the simulation environment, and Section V reports the simulation results. Finally, Section VI draws the conclusions.

II. DownLink System Model

There are a number of factors that have an impact on the QoS aspects of the LTE data download rate. Such factors include the conditions of the channel, resource allocation policies, available resources as well as the delays that occurred as a result of the sensitivities of the traffic. A resource block that comprises of frequency and time domains contain the resources that are being transferred within LTE. The 3GPP LTE system contains components known as "enodeB" which are base stations. The scheduling packet is operating in conjunction with other radio resource management (RRM) mechanism.

The bandwidth itself consists of several 180 kHz physical resource blocks (RBs) with 0.5 ms durability. Each of these blocks contains 6 or 7 symbols within the time domain and in addition, it also has 12 consecutive subcarriers within the frequency domain. The TTI has a definitive resource allocation; precisely every two successive resource blocks. Through this method, resource allocation is completed depending on the resource block pair.

Each enodeB would receive a notification of instantaneous downlink channel conditions such as signal-to-noise ratio (SNR) from the users within each of the TTI. Several criterions were used by the enodeB packet scheduler to determine the user priorities. Such criterions include, head of line (HOL) packet delays, buffer status, and service types. By utilising the channel state information (CSI), it is possible for enodeB to obtaining the full information regarding the quality of the channel.

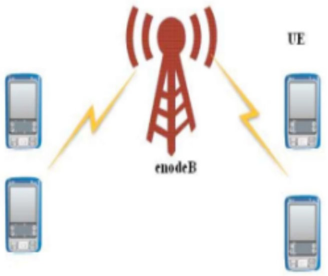

Fig1: LTE downlink system

On the criteria of first-in and first-out, packets approach in the buffer at enodeB. These packets are queued for the transmission, and the HOL is calculated. Also if any delay is arisen it is computed as well and in case the packet is delayed beyond a certain threshold of time it is discarded.

III. Scheduling Algorithms

There are several scheduling procedures that are followed by the wireless networks, which may include variety of flows such as resource allocation. Such procedures could be the channel gain, average rate as well as the packets arrival delay. PF is most suitable for elastic flows as it can accommodate the ability to choose the users according to their channel state. In terms of real-time services, proposal has been made for packet delay algorithm such as M-LWDF. Proposal has also been made for EXP/PF and EXP rule to accommodate mixed, real-time and non-real-time flows. Both the M-LWDF and EXP/PF are specifically tailored for the scheduled algorithms. In addition, PF is included as a reference within the performance.

A. Proportional Fair (PF)

Proportional Fair algorithm [1] belongs to those algorithms that are employed in High Data Rate (HDR) networks and was implemented to opt between the total data rate and a fair data rate of each user. PF assigns radio resources after analysing the experienced quality of channel as well as the past user throughput and for this property PF is rendered as a perfect scheduling choice for non-real-time traffic. In order to ensure fairness among the flows, the objective is to maximize the entity "total network throughput".

B. *Maximum-Largest Weighted Delay First (M-LWDF)*

M-LWDF is a type of algorithm designed for the purpose of supporting multiple real-time data users in CDMA-HDR systems [2]. M-LWDF considers instantaneous channel variations and in case of video service it takes in consideration of the delays in this service, thus, it configures multiple data users having different Quality of Services (QoS) requirements. M-LWDF makes use of the information about the channel state by striving to maintain a balance between the weighted packet delays.

C. *Exponential Proportional Fairness (EXP/PF)*

Exponential Proportional Fairness (EXP/PF) is a sort of algorithm, which configures the multimedia applications in a system of Adaptive Coding & Modulation/Time Division Multiplexing (ACM/TDM) system. This type of algorithm can have both the real-time service user as well as non-real-time service [5] and so it can enhance the priority of real-time flow with respect to no-real-time flow.

IV. Simulation Environment

The performance of PF, M-LWDF, and EXP/PF within LTE has been evaluated. A realistic single-cell scenario is being made, which has a radius of 1 km and the cell itself has one enodeB and a random number between 10 to100 of UEs. The UEs' movement travelling cell is being elaborated with the random way-point model [6]. According to [11], the speed of 0, 3, 30 and 120 km/h will be used in order to diagnose the static, pedestrian and vehicular scenarios respectively. Simultaneously, a video flow, VoIP flow and best-effort flow is allocated within each UE. The scenarios will be run through software called LTE-Sim simulator [4].

TABLE 1
SIMULATION PARAMETERS.

Parameter	Value
Simulation length	180 s
Number of Cell	1 EnodeB
Cell layout	Radius: 1 KM
Number of users	10, 20, 30, 40, 50, 60, 70, 80, 90, 100
User speed	0, 3, 30, 120 KM/H
Traffic model	Real-time flows type: H.264 at 242 Kbps, G.729 VoIP. Best effort flows: infinite buffer

The simulation parameters are summarized in Table 1. Five simulations were run for each number of users at different speeds to calculate the average packet loss ratio and delay. In total, 150 simulations were conducted to determine the final result for each LTE scheduling algorithm. We focus on the VoIP results in this paper, but we explain the traffic model used within the simulation as the following: a video service consists of 242 Kbps source video rate was used. The traffic used is known as trace-based application, which delivers packets that are based on the realistic video trace files that are found in [4]. G.729 voice flows are created within the VoIP application in order to determine the VoIP flows. Specifically, On/OFF Markov chain was used as a model within the voice flow. The ON period has an exponential distribution of 3s as the average value. The OFF period contains a truncated exponential probability density function; where the upper limit is 6.9s with the same average as ON period of 3s [4].

V. Simulation Result

In this paper, we focus on analysing the performance of the LTE algorithm in a single cell with a set of users at different speeds. As one of the real-time flows, VoIP is considered to measure the impact on QoE. We measure QoE performance for our experiments using the MOS score. As the packet-loss ratio and delay were provided in the simulation results, we used the following equation to determine the MOS [8]:

$$MOSc = a + bx + cy + dx^2 + ey^2 + fxy + gx^3 + hy^3 + ixy^2 + jx^2y$$

where x represents the packet loss ratio and y is the end-to-end delay. The parameters for fitting surfaces with G.729 codec from [8] were used.

The results for all experiments are shown in Figures 2 through 10. The figures indicate slight changes in results at different speeds scenarios. As the single-cell scenario user only has the ability to move around a small area, handover will not occur. However, the results are affected by increasing the number of users access within the cell. MLWDF and EXP-PF keep the end-to-end delay to less than 50ms with all sets of users, but PF can only handle up to 30 users in the cell before the end-to-end delay increases rapidly as the number of users increases in all test cases and follow same pattern as Figure 2. As MLWDF and EXP-PF consider end-to-end delay in the scheduling algorithm, their very good results show that they effectively minimise the end-to-end delay across different speed. On the other hand, the packet loss ratio (PLR) for MLWDF is low until 60 users, and then PF takes over as the lowest PLR until 100 users. The EXP-PF has the largest PLR compared to the other algorithms. The MOS score figures almost reflect the PLR scores in that the lower the PLR, the better the MOS score. The acceptable voice quality is 3.5 in the MOS score according to normal telecommunications applications. This also follows standards defined according to ITU-T Recommendation P.800 [12]. The cell capacity is 40 to 50 users with acceptable voice quality in MLWDF and PF scheduling algorithms.

TABLE 2
The maximum number of user access at acceptable MOS

User speed	PF	MLWDF	EXP-PF
Static	20	40	25
3 KM/H	25	45	28
30KM/H	25	50	32
120 KM/H	30	50	30

In Table 2, we have captured the maximum number of user access with an acceptable MOS score. At higher speed we see that the cell can occupy more users comparing to slower speed. PF has the highest number of user but when we look at the end-to-end delay, it is very long exceeding the recommended maximum delay by ITU-T Recommendation P.800 [12], which is 150ms. So the best suitable downlink scheduling algorithm is MLWDF, which has short end-to-end delay less than 50ms with the maximum number of user (more than 50 users) at acceptable MOS score.

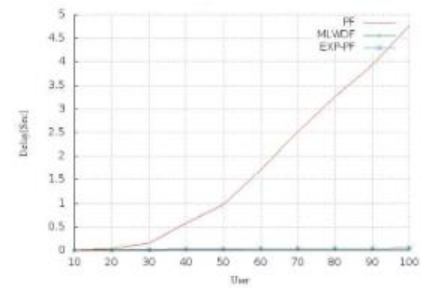

Fig 2: end-to-end delay vs. number of user at Static

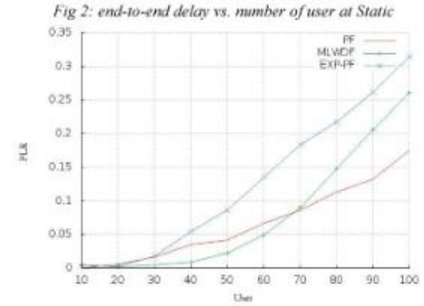

Fig 3: packet loss ratio vs. number of user at Static

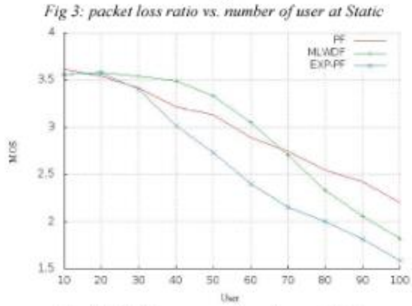

Fig 4: MOS Score vs. number of user at Static

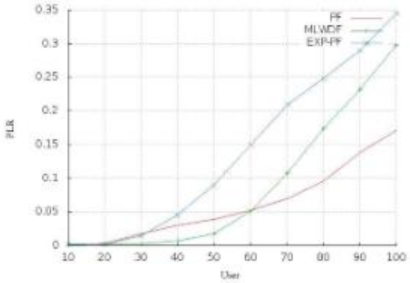

Fig 5: LTE scheduling vs. packet loss ratio at 3 km/h

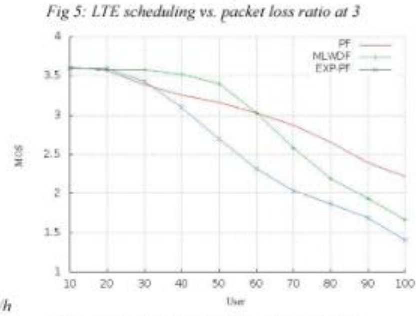

Fig 6: MOS score vs. number of user at 3 km/h

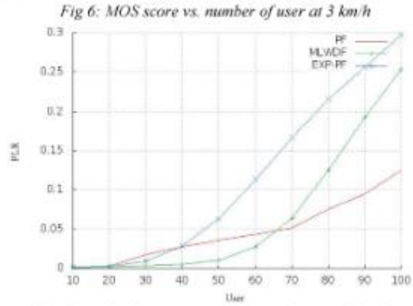

Fig 7: packet loss ratio vs. number of user at 30 km/h

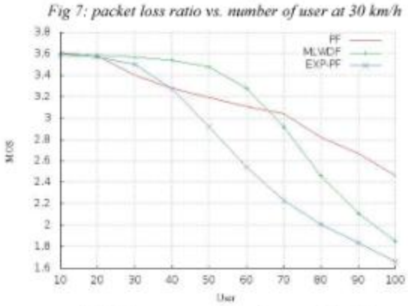

Fig 8: MOS score vs. number of user at 30 km/h

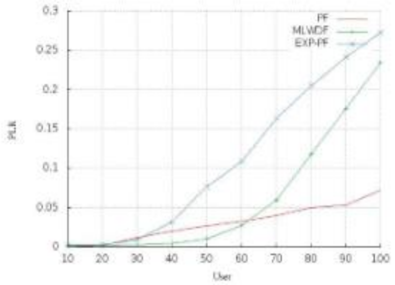

Fig 9: packet loss ratio vs. number of user at 120 km/h

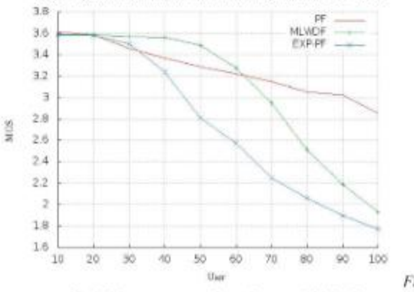

Fig 10: MOS score vs. number of user at 120 km/h

VI. CONCLUSION

The LTE simulation was built based on LTE-Sim with three LTE scheduling algorithms (i.e., PF, EXP-PF, and MLWDF) for a single-cell scenario. Different speeds and numbers of user were used to monitor voice quality of the LTE scheduling algorithms. The MOS scores were calculated using equation based on network parameters (packet loss and delay) and codec type that is not covering the speed parameter. EXP-PF is considered for real-time flows, demonstrated lower MOS scores with more than 30 users per cell. Also, PF has great end-to-end delay, which is not suitable with VoIP application. On the other hand, MLWDF handles more than 50 users with acceptable MOS score (MOS over 3.5) at 120 km/h.

Future work should expand experiments using multi-cells and consider handover issues. Both real-time and non-real-time flows should be considered. QoE-driven scheduling algorithms will be developed to achieve a maximum QoE for both real-time and non-real-time applications

ACKNOWLEDGEMENT

The work reported here is supported in part by the EU FP7 GERYON project (contract No. 284863)

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B. QoE-driven LTE Downlink Scheduling for VoIP Application

This paper was published in IEEE CCNC at Las Vegas, USA 2015

2015 IEEE 12th Consumer Communications and Networking Conference (CCNC): CCNC 2015 - Poster Session

QoE-driven LTE Downlink Scheduling for VoIP Application

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Abstract— This paper proposes a QoE-driven LTE downlink scheduling algorithm which is able to maximise the number of users per cell while maintaining an acceptable QoE for VoIP applications. The QoE-driven scheduling algorithm was implemented in LTE-Sim and its performance was compared to common LTE scheduling algorithms. The main difference between QoE-driven scheduling and existing scheduling mechanisms was prioritisation of users for resource allocation by taking into account their QoE requirements for VoIP applications. Preliminary results have shown that QoE-driven LTE downlink scheduling improved the cell capacity by 75% compared to MLWDF and 250% compared to PF and EXP-PF.

***Index Terms*— LTE, packet scheduling, QoE, QoS, VoIP**

I. Introduction

In 4G LTE mobile networks, there are currently no 3GPP standards for downlink or uplink scheduling algorithms to support both real-time and non-real-time services. Extensive research has been carried out in recent years on a variety of downlink and uplink scheduling schemes in order to improve the performance of delivered services. The importance of QoE awareness in designing scheduling algorithms to incorporate different multimedia applications has gradually become recognised, and there have been several recently published studies, such as a QoE-oriented scheduling algorithm [1] and a methodology for integrating QoE into a network's radio resource management (RRM) mechanism which was proposed in [2]. Both of these studies focused on maximisation of the QoE, rather than the cell capacity. In this paper, we try to answer the question on how to allocate limited mobile resources to as many end users as possible in a mobile cell while maintaining a satisfied user experience (or an acceptable QoE) for all users in order to maximise profits for the service providers. We choose VoIP as an example to prove the concept. The QoE-driven LTE downlink scheduling algorithm is proposed and implemented in LTE-Sim [3]. The investigation and evaluation of the performance of the proposed QoE-driven LTE downlink scheduler was carried out against three popular LTE downlink schedulers, including PF, EXP-PF and MLWDF [4]. The performance evaluation is based on a QoE metric (i.e. MOS score) for VoIP applications. The preliminary results have shown that the proposed algorithm has improved the cell capacity by 75% at an acceptable MOS score (MOS score of 3.5) compared to the MLWDF cell capacity, and 250% compared to PF and EXP-PF algorithms, in all three lower speed categories (i.e. static, 3 and 30 KM/H).

II. The Concept of QoE-driven LTE Downlink Scheduling

The LTE downlink physical resource is based on Orthogonal Frequency Division Multiplexing (OFDM) [5]. It can be represented as a time-frequency resource grid, containing several resource blocks (RB). A scheduler controls the allocation of the time and frequency resources between users for each transmission time interval (TTI) of 1ms. The scheduler resource allocation decision is based on several parameters such as delay, RB, buffer size and packet loss rate obtained at the scheduler itself and channel quality indicators (CQI) received from users' mobile devices.

The current LTE downlink scheduling algorithms were designed to satisfy QoS, such as delay, or to provide the maximum level of QoE for specific scenarios without consideration of increasing the cell capacity. However, the proposed QoE-driven LTE downlink scheduling algorithm is designed to maximise the number of users in a single cell while providing an acceptable QoE for all users. The QoE-driven LTE downlink scheduling algorithm is set and defined in Table I,

Table I: Definition of QoE-driven LTE downlink scheduling

Given	Delay $d_{ij_{(t)}}$, PLR $p_{ij_{(t)}}$, Buffer $b_{ij_{(t)}}$ and number of resource blocks $RB_{ij(t)}$
Constraint	QoE_{min} , MOS score at least 3.5 for VoIP
Optimization	maximise number of users to access a cell

where $d_{ij_{(t)}}$ is the head of line (HOL) packet delay for user i at cell j at the current scheduling instant TTI (t). $p_{ij_{(t)}}$ is the packet loss rate for user i at cell j at instant t. Buffer $b_{ij_{(t)}}$ is the buffer size at enodeB for user i at cell j at instant t. $RB_{j(t)}$ is the number of good available RBs at cell j at instant t. The constraint of this algorithm is the received $QoE \geq QoE_{min}$ ($QoE_{min} = 3.5$, widely used in VoIP applications as an acceptable quality). The optimisation problem is to maximise the number of users for a given number of cells using the following equation:

Given $d_{ij_{(t)}}$, $p_{ij_{(t)}}$, $b_{ij_{(t)}}$ and $RB_{ij_{(t)}}$,

$$\max_{T} \sum_{j=1}^{M} o_j \quad \text{for } j = 1, \ldots, M$$

subject to

$$QoE_{ij(t)} \geq QoE_{min} \text{ , for } i=1,\ldots,N;\ j=1,\ldots,M$$

$$\sum_{i=1}^{N} RB_{ij_{(t)}} \leq RB_{j(t)\,\max}$$

where M is the total number of cells, o_j is the number of users for cell j. N is the maximum number of users for a single cell. $QoE_{ij(t)}$ is the QoE achieved for user i at cell j at time t. $RB_{ij_{(t)}}$ is the RBs allocated to user i at cell j at time t. $RB_{j(t)max}$ is the maximum accessible resource block at cell j at time t. T is the total period of time for resource allocation

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(e.g. T=180 sec in simulation settings used in the paper). In our implementation we consider only a single cell, i.e. M=1, j=1 and $o_1 = N$, for simplicity.

The QoE-driven LTE downlink scheduling scheme is illustrated in Figure 1. The enodeB allocates RBs to users 1,..., N based on CQI, loss, delay, RB buffer size and acceptable QoE.

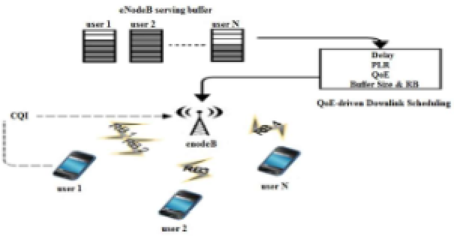

Figure 1: The model of QoE-driven LTE downlink scheduling

III. Simulation Environment

The simulation scenarios were adapted based on a single-cell scenario. User mobility was set as a random way-point model [5]. Table II summarises the parameters used in the simulation.

TABLE II: SIMULATION PARAMETERS.

Parameter	Value
Simulation length	180 s
Number of Cell	1 EnodeB
Cell layout	radius: 1 KM
Number of users	10, 20, 30, 40, 50, 60, 70, 80, 90, 100
User speed	0, 3, 30, 120 KM/H
Traffic model	Real-time flows type: H.264 at 242 Kbps, G.729 VoIP. Best effort flows: infinite buffer

The traffic model used in this simulation is described in the simulation parameters (c.f. Table II). There are two real-time flows: the first one is a trace-based application that used a foreman video trace file at 242 Kbps; the second one is a VoIP application, which used a G.729 codec with On/OFF Markov model for VoIP flow.

IV. Simulation result

In this paper, we focus on analysing the performance of the QoE-driven scheduling algorithm versus PF, MLWDF and EXP/PF in a single cell with a set of users at different speeds. VoIP flow was considered to evaluate the QoE performance. The MOS score was measured using the packet loss ratio and end-to-end delay parameters that were obtained from simulation results [6].

According to standard telecommunications applications, the acceptable MOS score is 3.5. This also follows standards defined according to ITU-T Recommendations P.800 [7]. Table III depicts the maximum number of users per cell with an acceptable MOS. The proposed algorithm has improved cell capacity at lower speed to allow 70 users receiving VoIP services with an acceptable MOS score, compared to 40 users with MLWDF. While the proposed QoE-driven algorithm only reaches 20 users with an acceptable MOS score in a scenario at a speed of 120 KM/H, MLWDF allows 50 users. .

TABLE III: The maximum number of user accesses at acceptable MOS

User speed	PF	MLWDF	EXP-PF	QoE-driven
Static	20	40	20	70
3 KM/H	20	40	20	70
30 KM/H	20	40	20	70
120 KM/H	30	50	20	20

This has improved the cell capacity by 75% compared to MLWDF and 250% compared to PF and EXP-PF in all three lower speed scenarios. When motion speed is at 120 KM/H, QoE-driven scheduling can achieve a similar cell capacity to PF and EXP-PF, but has a lower cell capacity when compared with MLWDF. This is due to a higher packet loss rate, although the delay did not have much effect in this scenario.

V. Conclusion

The proposed algorithm achieved remarkable improvement in cell capacity, by 75% at an acceptable MOS score compared to MLWDF and 250% compared to PF and EXP-PF in all three lower speed scenarios. When motion speed is at 120 KM/H, QoE-driven scheduling can achieve a similar cell capacity to PF and EXP-PF, but has a lower cell capacity when compared with MLWDF. Future studies will evaluate the impact of QoE-driven scheduling on other services, such as video calls and web browsing. QoE-driven scheduling will also be improved to distinguish services based on the QoE requirement and be able to achieve high performance in higher speed scenarios.

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C. QoE-driven Management Schemes for Multimedia Services

This paper was published in IEEE COMSOC MMTC E-letter

IEEE COMSOC MMTC E-Letter

QoE-driven Management Schemes for Multimedia Services

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1. Introduction

According to Cisco's forecast [1], IP video traffic will reach 79 percent of global Internet traffic and traffic via mobile/wireless access will exceed traffic via wired access by 2018. This trend indicates an increasing popularity of Internet video services such as YouTube and Netflix, and people would prefer to watch Internet video using a mobile device, such as a tablet or a smart phone due to its convenience. This fast-growing, large amount of video traffic (from SDTV, HDTV to UHDTV) and consumers' desire for a better perceived user experience, or Quality of Experience (QoE) pose real challenges to content/service providers and network operators on how to deliver multimedia services to end users over the Internet and/or resource-constrained mobile networks with a satisfactory QoE. The QoE becomes one of the most stringent factors influencing the success of multimedia services (including existing, new and emerging ones) delivered over the Internet.

The Quality of Experience (QoE) management generally refers to any control and management approaches which aim to optimize an achievable QoE for multimedia services such as VoIP and video streaming over fixed and/or mobile networks. In the past, the QoE management relied mainly on the network Quality of Service (QoS) parameters, for example, to reach a target packet loss rate below 2 percent for an RTP/RTCP-based video service or a target end-to-end delay less than 150 ms and jitter below 30 ms for a VoIP application. These QoS related parameters are not directly linked with the perceived user experience or QoE, and it is difficult to judge how well the QoE is for a delivered service under certain network QoS settings. In recent years, many novel approaches have been proposed to assess, control and manage QoE efficiently and effectively for multimedia services [2]. For example, QoE-oriented management/control schemes aim to automatically control the video sender bit rate according to the available network bandwidth, and to efficiently and intelligently allocate mobile resources according to different user and service needs. These QoE-oriented management approaches differ from the QoS-focused ones in the sense that the QoE is brought into the management/control cycle directly in order to optimize certain QoE targets.

In this paper, we will first present a general approach to apply the QoE assessment model for QoE-driven management schemes for multimedia services. We will then discuss in detail an application of the QoE model in an LTE downlink scheduling scheme for multimedia services (e.g. VoIP and video streaming) and further demonstrate how the proposed QoE-driven approach can increase the number of users a mobile node can support when compared with existing scheduling schemes.

2. Application of QoE Model in QoE Management

Considering a media delivery chain from media generation to media consumption as illustrated in Figure 1, the QoE as quality experienced by an end user is affected by both encoding and decoding/display techniques at a media server or a terminal device, and the transmission channel or the network the media is delivered over. Even under the same encoding /decoding and network settings, different video content may have a different impact on the QoE. As shown in Figure 1, the QoE metric is predicted from network, server/terminal (encoding/decoding) and/or media content related parameters non-intrusively via the QoE modelling block. The predicted QoE could be used for the QoE management in controlling encoding, transmission and/or decoding processes to optimize the delivered QoE or achievable QoE to an end user.

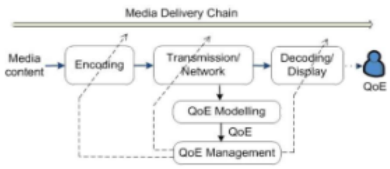

Figure 1: QoE Modelling and QoE Management

In QoE modelling, the network/transmission parameters may include average packet loss rate (PLR), mean burst loss length, delay variation (jitter) and delay. The server/terminal related parameters may include codec type (e.g. PCM or AMR for speech and H.264 or H.265/HEVC for video), quantization parameter (QP), sender-bit-rate (SBR), frame-rate (FR) and/or resolution for video, Forward Error Correction (FEC), concealment mechanisms and jitter buffer algorithms used in the server/terminal. The content parameter refers to video content type which could be classified to a fixed number of types (e.g. three types for fast, medium and slow moving video contents). A general QoE model can be expressed by Eq. (1) and a simplified video QoE model is presented by Eq. (2) [3].

$$QoE = f(network, terminal, content) \qquad (1)$$

$$QoE = \frac{\alpha + \beta FR + \delta \ln(SBR)}{1 + \mu(PLR) + \sigma(PLR)^2} \qquad (2)$$

where the coefficients (e.g. α and β) are video content dependent.

It has to be noted that Eq. (1) has only considered technical related parameters along the media delivery chain which may have an impact on QoE. Non-technical related parameters such as user preferences, environmental impact or context (e.g. whether the media content is consumed in an office, at home or on a bus), and business impact (e.g. whether the service is offered free of charge or at a premium-rate) which may have an impact on the QoE are beyond the scope of this paper.

In our previous work, we have demonstrated the use of the QoE model in HTTP-based adaptive video streaming services which could be applied to MPEG Dynamic Adaptive Streaming over HTTP (DASH) schemes. The proposed approach is able to automatically and adaptively select a video segment at certain sender bit rate which can provide an optimized or a satisfactory QoE based on the current estimated network bandwidth and playout buffer status at the terminal [4]. We have also applied the QoE model, together with relevant network QoS parameters in a QoE-driven video sender bit rate adaptation scheme for video streaming over UMTS networks based on RTP/RTCP protocols [5]. QoE-driven adaptation schemes in [4] and [5] have both demonstrated more stable, optimized and satisfactory QoE under changing network conditions when compared with other non-QoE-driven approaches.

The QoE model has also been used in a power-driven VoIP quality adaptation scheme for video call over WLAN networks [6], where an acceptable QoE could be maintained by automatically changing video sender bit rate in order to conserve power in mobile devices and hence, prolong a VoIP communication session. The power saving between 10 – 30% of the total system power was achieved when the QoE-driven power saving approach was taken.

3. QoE-driven LTE downlink scheduling

Due to limited mobile network resources, it is always a tradeoff between the QoE delivered over mobile networks and the number of users a mobile node can support. Existing work on LTE scheduling are focused mainly on optimization of QoE for a fixed number of users (e.g. maximization of a sum of QoE for all users) within the limited network resources. In this section, we present an algorithm for LTE downlink scheduling to optimize the number of users a mobile node can support subject to satisfactory QoE achievable for all users. This approach is particularly useful in emergency mobile communications or at major events (e.g. a favorite sport) where it is crucial to increase mobile capacity and at the same time to meet satisfactory QoE for all mobile users.

Figure 2 depicts a conceptual diagram of QoE-driven LTE downlink scheduling where an eNodeB allocates available mobile network resources, in terms of available Resource Blocks (RBs), to mobile users according to received channel information (Channel Quality Indicator, CQI) from mobile users, network QoS (e.g. delay and PLR) and predicted QoE for mobile users for specified services at eNodeB.

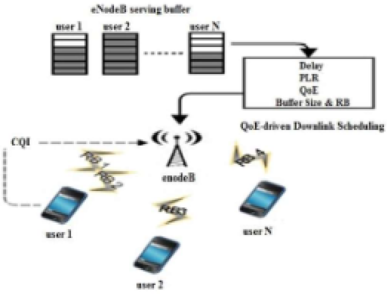

Figure 2: QoE-driven LTE scheduling

The LTE scheduler at eNodeB controls the allocation of RBs between mobile users at each transmission time interval (TTI) of 1ms. Assume a mobile network contains M eNodeB and each node can support up to N mobile users. The QoE-driven scheduling becomes an optimization problem to maximum the number of users which M mobile nodes can support subject to satisfactory QoE achievable by all users.

Let $d_{ij}(t)$ be the head of line (HOL) packet delay for user i at cell j at current scheduling instant t. $p_{ij}(t)$ and $b_{ij}(t)$ denote as the packet loss rate and the occupied buffer size for user i at cell j at time t. $RB_j(t)$ is the number of good available RBs at cell j at time t. The constraint of this algorithm is that the received $QoE \geq QoE_{min}$ (we assume $QoE_{min} = 3.5$, or MOS of 3.5 which is widely used in telecommunications world for an acceptable quality for services such as VoIP). The optimization problem is to maximize the number of users for a given number of cells based on Eq. (3) subject to Eq. (4) and Eq. (5).

$$\max_{T} \sum_{j=1}^{M} o_j \quad \text{for } j = 1, ..., M \tag{3}$$

$$subject\ to:\ QoE_{ij} \geq QoE_{min} \tag{4}$$

$$\sum_{i=1}^{N} RB_{ij}(t) \leq RB_j(t)_{max} \tag{5}$$

Where o_j denotes the number of mobile users for node j, T is the total time for resource allocation, and QoE_{ij} represents QoE achievable for user i at node j ($i = 1, \cdots N; j = 1, \cdots M$). $RB_{ij}(t)$ is the RBs allocated to user i at cell j at time t. $RB_j(t)_{max}$ is the maximum accessible resource blocks at cell j at time t.

An LTE scheduling algorithm for a single node case ($j=1$ and $M = 1$) for simplicity is illustrated below. Iteration process is carried out at every TTI.

Algorithm: LTE Scheduling	
Input	Delay $d_i(t)$, PLR $p_i(t)$, buffer $b_i(t)$ for user i ($i \in [1, \cdots N]$) at time t
1:	Predict $QoE_i(t)$ and calculate the metric, $m_i(t)$ according to Eq. (6)
2:	If $QoE_i < QoE_{min}$, $m_i = m_i + 1$; or else $m_i = m_i$
3:	Schedule a user with the highest metric the required RBs matching to its buffer size $b_i(t)$
4:	Check for remaining RBs and schedule the resource to the user with the 2nd highest metric
5:	Repeat Step 4 until the TTI end.

Where:

$$m_i(t) = \frac{u_i(t)}{v_i(t)} \cdot d_i(t) \cdot p_i(t) \tag{6}$$

And $u_i(t)$ and $v_i(t)$ are accessible throughput and average achieved throughput for user i at time t, respectively.

4. Simulation Environment and Results

The simulation scenarios for a single-cell were implemented in LTE-Sim [7]. User mobility was set as a random way-point model [8][8]. The following table summarises the parameters used in the simulation.

Parameter	Value
Simulation length	180 s
Number of Cell	1 eNodeB
Cell layout	radius: 1 KM
Number of users	10 to 100 at the increment of 10
User speed	3, 30 KM/H
Traffic model	Real-time flows: H.264 for Foreman video trace at 242 Kbps and G.729 VoIP. Best effort flows: infinite buffer

The performance of the QoE-driven scheduling algorithm was evaluated and compared with that of MLWDF [9] and EXP/PF for both VoIP and video flows. The QoE metric (in terms of MOS) for VoIP was measured using the QoE model derived from packet loss and delay [10][10]. The video QoE was evaluated using the mapping between SSIM and MOS as described in [11][11]. The results are illustrated in Figures 3 and 4. The QoE-driven algorithm improves the overall cell capacity by the number of users with satisfactory QoE (satisfied users). In terms of VoIP flow, it increases the number of satisfied users by 75% compared to MLWDF in both user's speeds. While for video flow, the user's speed seems to have more impact on the cell capacity. The QoE-driven algorithm has enhanced the cell capacity by 30% for satisfied users at 3 KM/H speed when compared to the other two algorithms.

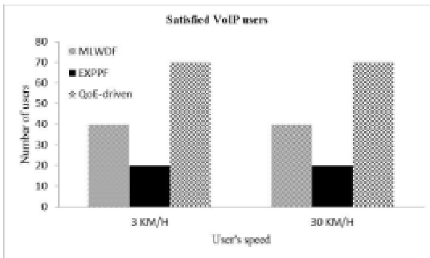

Figure 3: Satisfied VoIP users

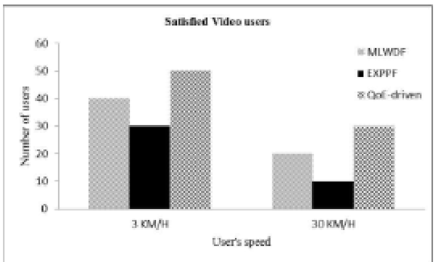

Figure 4: Satisfied Video users

5. Conclusions

In this paper, we presented a general approach to apply a QoE model in QoE management schemes for multimedia services. We also discussed an application of the QoE model in the LTE downlink scheduling to optimize QoE in terms of maximizing the number of users a mobile node can support while maintaining acceptable QoE for the mobile users.

Although QoE-driven management schemes have been proved to be effective in several multimedia applications, there are still many unsolved problems/issues which need to be addressed and investigated further by the multimedia QoE community. For example, how to predict and control QoE effectively and efficiently in rapidly changing network conditions; how to predict and manage QoE for new and emerging multimedia services such as mobile HDTV and online gaming; and how to achieve an overall end-to-end QoE optimization instead of segmented optimization focusing on one or two elements (e.g. encoding or network) over a complex system/network (such as the today's Internet).

Acknowledgement

The work reported here was supported in part by the EU COST Action, IC1003, QUALINET and the EU H2020 QoE-NET project (contract no: 643072).

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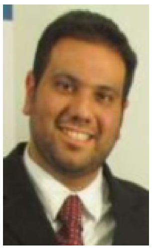

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