VoIP Packet Scheduling for LTE Advanced

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Declaration

I hereby declare that: (1) the above dissertation is my own unaided work, both in conception and execution, and that apart from the normal guidance of my supervisor, I have received no assistance apart from that stated below; (2) except as stated below, neither the substance or any part of the dissertation has been submitted in the past, or is being, or is to be submitted for a degree in the University or any other University.

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Abstract

The LTE-A (Long Term Evolution Advanced) technology constitutes a significant step in the evolution of 3G systems towards 4G. The performance targets set out for LTE Advanced make it an ideal solution to take care of the ever-increasing demand for wireless broadband services. LTE Advanced also comes along with new features and techniques that do not exist in LTE (Long Term Evolution), which is the 3G technology that precedes it. Such features include Carrier Aggregation, Coordinated Multipoint Reception and Transmission (COMP), relays and others. With the introduction of such features, Quality of Service (QoS) support for multiple data services such as voice and other multimedia applications is further enhanced. The Packet Scheduler (PS) which is a key entity located in the base-station/eNodeB plays a prime role as a key feature of LTE's Radio Resource Management in increasing the system's data rate and providing support for the diverse QoS requirements of mobile services. The Packet Scheduler should be able to distribute radio resources to mobile User Equipments (UEs) such that the LTE network is able to adhere to its performance requirements. Using dynamic system level simulations, this study performs an evaluation of a given set of scheduling algorithms. It seeks to determine whether these algorithms are capable of providing Quality of Service for VoIP traffic in a scenario where VoIP and web traffic compete for the same resources. In this study, two sets of algorithms were taken into consideration: QoS differentiated and non QoS differentiated packet scheduling schemes. It is found that those that take into consideration delay requirements of VoIP service presented the best overall performance. A performance evaluation is further carried out to determine the most suitable algorithm supporting VoIP traffic. It was observed that as regards VoIP service, Modified Largest Weighted Delay First Algorithm (MLWDF) outperforms other packet scheduling algorithms by offering a much higher system throughput, supporting more users and guaranteeing fairness at a satisfactory level. Based on this, an evolved MLWDF algorithm which is an enhancement to the existing MLWDF scheme is proposed in this study. The scheme proposed aims to minimise the packet loss of VoIP traffic that arises due to the delay of most packets at the user queues in the base station exceeding the delay threshold. The scheme does this by taking into account computation of parameters like packet arrival rate, buffer space capacity and packet size to reduce the packet loss for VoIP traffic. In the proposed scheme, priority is offered for VoIP traffic by performing resource reservation for VoIP packets

on the Component Carriers (small bandwidths of up to 20 MHz) available for scheduling. In this study, an investigation is carried out on the important problem of downlink resource allocation in recently enhanced LTE Advanced systems where a newly added feature Carrier Aggregation provides more flexibility in Radio Resource Management.

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God Bless you all.

Acronyms

CC	Component Carrier
CIR	Carrier to Interference Ratio
CSI	Channel State Information
DL	Down Link
EPC	Evolved Packet Core
EUTRAN	Evolved Universal Terrestrial Radio Access Network
EXP-PF	Exponential Proportional Fair
GSM	Global System for Mobile Communication
HSPA	High Speed Packet Access
ITU	International Telecommunication Union
LTE	Long Term Evolution
LTE-A	Long Term Evolution Advanced
MAC	Medium Access Control
MCS	Modulation and Channel Coding Scheme
MBMS	Multimedia Broadcast / Multicast Services
MR	Maximum Rate Algorithm
MIMO	Multiple Input Multiple Output

MS	Mobile Station
MSC	Mobile Switching Centre
MLWDF	Modified Largest Weighted Delay First
MME	Mobile Management Entity
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
PCEF	Policy and Charging Enforcement Function
PDCP	Packet Data Convergence Protocol
PDB	Packet Delay Budget
PF	Proportional Fair
PGW	Packet Data Network Gate Way
PLR	Packet Loss Ratio
PRB	Physical Resource Block
PS	Packet Scheduler
QAM	Quadrature Amplitude Modulation
QoS	Quality Of Service
RAC	Radio Admission Control
RB	Resource Block
RLC	Radio Link Control

RNC	Radio Network Controller
RR	Round Robin
RRM	Radio Resource Management
RRC	Radio Resource Control
S-GW	Serving Gate Way
SINR	Signal to Interference Noise Ratio
TB	Transport Block
TTI	Transmission Time Interval
UL	Up Link
3GGP	Third Generation Partnership Project
UMTS	Universal Mobile Telecommunication System
VoIP	Voice over Internet Protocol
WCDMA	Wide Band Code Division Multiple Access

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Chapter 1 Introduction

The great stride of mobile broadband began along with the emergence of Third Generation (3G) mobile phone networks at the start of the third millennium. The Third Generation Partnership Project (3GPP) brought forward Universal Mobile Terrestrial System (UMTS) standard which is 3GPP Release 5 as a follow up to the continuous success of its predecessor Second Generation (2G) Global System for Mobile Communications/Enhanced Data Rates for Global Evolution (GSM/EDGE) technology. UMTS greatly increased the data rates, allowing the user to download at about 2Mbps, which was a significant improvement to the EDGE Network's download speed. Some enhancements were also made to UMTS, resulting in a faster access technology, High Speed Packet Access, which was introduced in 3GPP Release 6 of UMTS [1].

HSPA offers download speeds of up to 14 Mbps and upload speeds of up to 5.8 Mbps, thus enabling the provision of a wide range of applications and services with reasonable QoS support [2]. This created a significant rise in the number of customers using these services. Due to the growth of 3G High Speed Packet Access, so many smart phones devices were produced for wireless data communication support and emerging mobile services such as media streaming and video calling.

Broadband subscriptions are predicted to arrive at the 3.4 billion mark by 2014, 80% of which will belong to mobile broadband subscriptions [3]. Due to the rise in mobile data services, the requirement for more advanced broadband mobile technologies created a need to support the predicted traffic volumes over the period of ten years or more. Hence, 3GPP initiated the move towards standardising the next generation of mobile broadband technology with the following aims in perspective [4]:

- To have 3GPP's 3G technology have a competitive edge in future markets.
- To cater for the increased demand for high data rates and improved Quality of Service support.
- To meet the demand for reducing Capital and Operational Expenses (CAPEX and OPEX) of mobile networks via moving into an optimised, fully packet switched network, and reducing the complexity in network protocols and network architecture.

In 2004, 3GPP started the work on Release 8 of mobile broadband standard as a 3G beyond solution, which was later termed as "Long Term Evolution (LTE)". In comparison to the mobile network technologies before 3G, LTE is pictured as a major step in the path of mobile broadband evolution in terms of its enhanced features and enabling technologies, which makes it a strong competitor to wired broadband networks such as cable and ADSL (Asynchronous Digital Subscriber Line).

The final adjustments to LTE Release 8 were made in December 2008, and this comprised setting up the system structure both at the access level and core level. 3GPP then introduced more enhancements to LTE in 3GPP Release 9, wbicb included new features such as relays and femto cells, which were concluded in December 2009 [6]. The Long Term Evolution technology is quite often referred to as "4G", but it's also claimed that LTE Release 10 (LTE Advanced) is the true 4G evolution step, with the first release of LTE (Release 8) then being labelled as "3.9G" [6]. This continuing race of increasing sequence numbers of mobile system generations is in fact just a matter of labels. What is of great importance are the actual system functionalities and how they have migrated or evolved which is the discussion of this chapter. More work has continued to be done on Long Term evolution with new enhancements being put forward in each release as illustrated in Figure. 1.1 below:

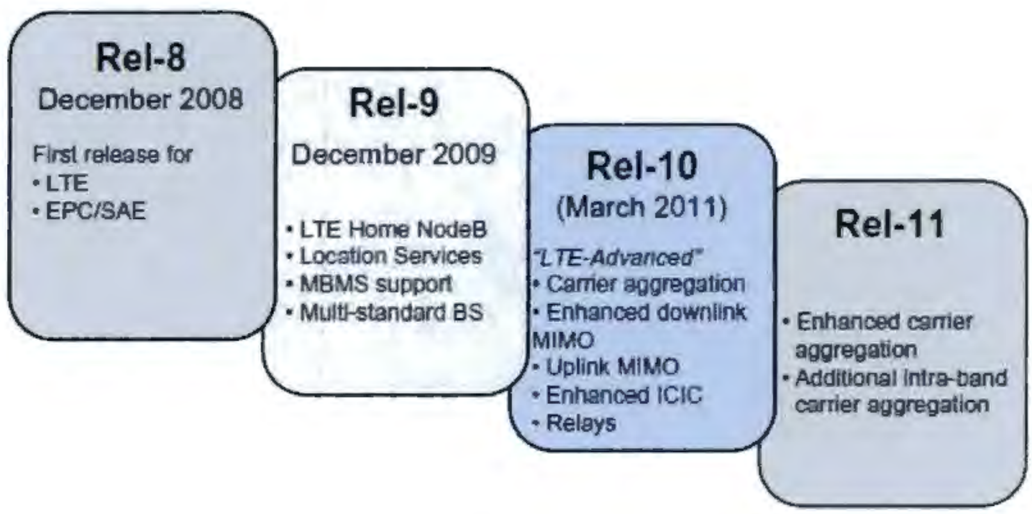

Figure 1-1. Releases of 3GPP specifications for LTE [6]

In this context, it must first be pointed out that LTE and LTE Advanced are the same technology [14], with the "Advanced" label primarily being added to highlight the relation between LTE release 10 (LTE-Advanced) and ITU/IMT Advanced, as will be discussed later. This does not make LTE Advanced a different technology from LTE and it is not in any way the final evolution step to be taken for LTE. Another important issue is that the effort on making enhancements on LTE to become LTE Advanced is work that is continuous within 3GPP, the same forum that developed the first 3G system (WCDMA/HSPA).

At the access level, an LTE Advanced network comprises mainly LTE Advanced base stations (BSs), referred to as evolved NodeBs (eNodeBs) as will be illustrated in Chapter 2. The LTE Advanced standard eliminated the Mobile Switching Controller (MSC) and Radio Network Controller (RNC) units that were present in 2G and 3G access networks, respectively. MSC and RNC units perform higher-level management of the radio access network, such as mobility management and Radio Resource Management (RRM). The eNodeB adopted some of the Radio Network Controller functionalities, such as mobility handling and Radio Resource Management, while other functions are set up at the packet core network. Radio Resource Management handles a very important function in LTE Advanced networks by management of the limited and often scarce radio resource such that the achievable data rates over LTE's radio interface become as high as possible. The focus of this work is on the Packet Scheduler, a very crucial Radio Resource Management component, which is positioned at the eNodeB. The Packet Scheduler is mainly responsible for the allocation of time and frequency resources among mobile terminals. The Packet Scheduler allocates radio resources both on the downlink (from the eNodeB down to the UE) and also on the uplink (from UE up to the eNodeB). System performance is enhanced by the Packet Scheduler. The Packet Scheduler in LTE determines which users are allowed to transmit or to be transmitted to and on what set of resources. It also plays a significant role in determining the transmission power of the terminals. Intelligent distribution of radio resources among active UEs optimises the system performance and reduces the cost per bit transmitted over the radio interface.

The functionality of the eNodeB's packet scheduler depends on its ability to detect several factors that are essential for high system performance. Firstly, the packet scheduler at the eNodeB has to have knowledge of the type of traffic flows that are going over the LTE Advanced interface and their respective Quality of Service requirements, e.g. , voice calls in LTE

Advanced are supported over a packet switched, IP-based platform as Voice Over IP (VoIP). The VoIP services have more stringent delay requirements than most other traffic types, and these requirements have to be handled to ensure a good service quality as seen by the end users. In addition, the LTE Packet Scheduler (PS) should be aware of the channel quality experienced by each user to be able to adjust the transmission rates on both the uplink and downlink directions. Rate adaptation dependent on channel quality has been improved due to the introduction of Orthogonal Frequency Division Multiplexing (OFDM) modulation technique. OFDM offers the scheduler the ability to take advantage of channel conditions in both time and frequency domains, where UEs are assigned sub channels over which these UEs experience good channel conditions.

1.1 Research Motivation

From a 2007 market research report it is stated that 80% of mobile revenues across the globe are still from voice services, the rest being covered by SMS [5]. From [5] it is stated that revenue from non voice services will grow but only reaching 25 % by 2014. So even in 2014, when we see LTE systems in the market, revenue from non-voice services will only be at 25% [5]. That implies that voice is becoming a commodity hence efficiency for voice offering is becoming key to maintaining potential profit for wireless operators hence making VoIP the focus of this study.

The focus of this research is on LTE Advanced, because it's a technology that is expected to serve more than 80% of all mobile broadband users in the near future [5][45]. Users' desire for high data rates has increased exponentially; hence it is expected that users will engage the network with more resource and bandwidth intensive applications. In that connection, efficient scheduling of radio resources in an LTE Advanced system becomes a key performance indicator and requirement for network providers. A study and analysis of the impact of resource scheduling algorithms on the performance of the LTE Advanced system for voice traffic under varying channel conditions becomes very necessary.

LTE Advanced is expected to reach peak data rates of 1 Gbps/500 Mbps (UL/DL) against 100/50 Mbps for LTE, and it will do so mainly by using a technique called Carrier Aggregation. In LTE, bandwidths of up to 20 MHz i.e. Component Carriers (CCs) can be used, however, LTE Advanced groups consecutive "normal" LTE bandwidths from larger ones up to 100MHz. Other techniques include both uplink and downlink transmissions to neighbouring base stations simultaneously.

With the introduction of Carrier Aggregation in LTE Advanced, terminals are no longer restricted to being scheduled across one Component Carrier. The introduction of such a technique which was not existent in the older releases of LTE, i.e. Release 8, means that current results obtained from simulation of scheduling algorithms for LTE will not apply for LTE Advanced. This means that new and intelligent multi-cell packet scheduling algorithms have to be studied and developed to address how packets will be scheduled across the Component Carriers.

1.2 Problem Definition

The growing need for wireless broadband access for different emerging user applications has increased the need for QoS guarantees and radio resource utilisation. Recent studies have shown that the proportion of VoIP users continued to grow from 28% of users in 2008 (up from 20% of user in 2007) to more than 50% in 2010 [6]. The continuous growth in users application devices coupled with limited radio resources available for users has increased the need for high QoS and better resource utilisation. Without packet scheduling, networks will not be able to provide QoS guarantees to real time applications like voice and efficiently utilise the network resources [7]. It is observed that existing wire line and wireless schedulers do not perform very well with respect to different traffic classes defined in the LTE standard. In addition, each of these traffic classes has a different scheduling requirement and consequently, it has become necessary to design appropriate scheduling schemes [8]. It is also observed that there exists a challenge to port VoIP services onto wireless networks while retaining the QoS of today's circuit-switched networks and the inherent flexibility of IP-based services. The problem of ensuring QoS is that of determining how to allocate available resources among users to meet QoS requirements [9].

The significance of packet scheduling is to ensure that when a user application is assigned network resources, the QoS requirements of such application are guaranteed. One of the contributions of this work is to identify the most suitable algorithm supporting VoIP traffic in the downlink LTE Advanced multicarrier system. The algorithm to be chosen should offer a high system throughput and low packet loss while maintaining fairness at a sufficient level.

In previous works, simulations of the existing MLWDF (Modified Largest Weighted Delay

First) Algorithm, which is the algorithm that this study aims to enhance, are based on the assumption that the buffer is infinite and no packet loss is caused by buffer overflow. But in the real communication system, buffer is finite and physical resource and the efficiency of all physical resources is equally important.

The traditional MLWDF guarantees allocation fairness within each CC (Component Carrier), but disregards the fact users (LTE Advanced) are assigned with different number of CCs. Therefore it cannot achieve the global fairness. E.g., consider two users with the same average channel quality and fast fading statistics, one is LTE Advanced and the other is Release 8. The traditional MLWDF algorithm gives an equal share of resources to the users on the CC that Rel'8 user is served. However, the LTE Advanced user is also scheduled on other CCs. Therefore, it gets much more resources than the Rel'8 user. This study thus investigates an important problem of downlink resource allocation in recently enhanced LTE Advanced systems where a newly added feature Carrier Aggregation offers more flexibility in Radio Resource Management.

1.3 Research Objectives

1. To evaluate whether the studied packet scheduling algorithms are capable of guaranteeing Quality of Service (measured in terms of packet delay) of the VoIP users in a scenario where the VoIP and web browsing services will compete for the same resources.

2. To carry out a performance evaluation to determine the most suitable algorithm for use in the downlink LTE Advanced multicarrier system supporting VoIP service. The algorithm to be identified should be able to provide low packet loss ratio while at the same time offering good throughput and fairness performance.

3. To propose and implement a scheduling scheme that is able to reduce the packet loss ratio for VoIP packets. The scheme proposed is an enhancement to the traditional Modified Largest Weighted Delay First (MLWDF) Algorithm. It takes into consideration critical parameters that aim to minimise packet loss for VoIP.

4. The scheme that this study aims to enhance should maintain resource allocation fairness between LTE and LTE Advanced terminals as both types of terminals will coexist in the same network. The LTE Advanced (release 10 and beyond) terminals have the ability to be scheduled across more than one Component Carrier and they therefore receive more resources than the LTE

terminals who have the capability to be scheduled on only one carrier. The scheme aims to address this issue.

1.4 Contribution

The contribution presented in our work is summarised as follows:

Conducted a survey on literature works for LTE/LTE Advanced downlink scheduling proposals, which can be classified into three groups of strategies: (i) channel-aware; (ii) channel-aware/QoS-unaware; (iii) Channel-aware/QoS-unaware; In this study the focus is channel aware approaches as the most recent Packet Schedulers that have been developed for LTE/LTE Advanced are dependent on channel conditions and are the most suitable for wireless environments.

Using the already existing Proportional Fair scheduling algorithm in the simulator, we extended the simulator to include the Queue Based Maximum Rate Algorithm (QBMC), Weighted Proportional Fair (WPF) Algorithm by extending and creating new scheduling classes. The study involved modification of all the respective algorithms selected for the performance evaluation to operate in a scenario with Carrier Aggregation. An evaluation was performed to determine whether these significant algorithms can guarantee QoS for VoIP and web traffic in a scenario where we have VoIP and Web traffic competing for the same time-frequency (Resource Block) resources.

Proposed and designed an evolved Modified Largest Weighted Delay First (MLWDF) algorithm which provides better performance for VoIP traffic by minimising the packet loss. The existing MLWDF algorithm was selected for this study as it proved to be the most efficient algorithm supporting VoIP traffic in terms of packet loss at all load conditions. The proposed algorithm in this study, takes the status of each buffer at the base station (eNodeB) into account incorporating traffic characteristic information such as the packet arrival rate and packet size while computing the priority as a new weight coefficient in the algorithm. The proposed scheme introduces an exchange of user past throughput across different Component Carriers to create a resource allocation fairness between LTE and LTE Advanced terminals. The scheme also addresses the issue of user scheduling across different Component Carriers which is a key feature for LTE Advanced terminals specifically offering a resource reservation for VoIP packets on the CCs so that they are given higher priority than Non Real Time packets so as to minimise packet loss. Motivated by the above points, the study thus introduces a downlink scheduling algorithm in such an environment with the feature that certain resources are reserved VoIP packets.

Analysed the simulation results obtained from the performance evaluation of the Round Robin, QBMC, Exponential Proportional Fair (EXP-PF), Weighted Proportional Fair (WPF) and MLWDF algorithms for VoIP traffic. The packet loss ratio and packet delay are utilised as evaluation metrics as they are the 3GPP QoS defined metrics for VoIP traffic but throughput and fairness are also considered as a suitable algorithm should be able to maximise one metric while offering minimal degradation on the other metrics used for measuring the LTE downlink performance. Performed a complete evaluation of the VoIP service for the representative schedulers in terms of QoS and scalability considering mixed traffic scenarios and provided recommendations on the LTE downlink scheduler design for future LTE Advanced packet scheduling proposals.

1.5 Paper Contributions

The major contributions of this research work are documented in the following peer reviewed conference publications:

[1] Ronnie Mugisha, Neco Ventura, "Packet scheduling for VoIP over LTE-A," Proceedings of the IEEE AFRICON Conference, Le Meridien-ile Maurice Hotel, Mauritius, 9-12 September 2013.

[2] Ronnie Mugisha, Neco Ventura, "LTE Advanced Packet Scheduling," Proceedings of South Africa Telecommunication Networks and Applications Conference (SATNAC), East London Convention Centre, South Africa, 4-7 September, 2011.

1.6 Dissertation Scope

The analysis in chapter 4 and the main simulations in this thesis were done for LTE downlink, the uplink was not considered for this study as it involves issues like power allocation and contiguous Resource Block allocation which limit the flexibility of the allocation scheme and are therefore not the focus of this study.

Two Component Carriers (CCs) within the same band were used in the study as non contiguous

Carrier Aggregation in different bands is still under discussion by the 3GPP and will be considered for discussion in later releases of LTE Advanced.

The Round Robin (RR), Proportional Fair (PF), Exponential Proportional Fair (EXP-PF), Modified Largest Weighted Delay First (MLWDF) and Weighted Proportional Fair (WPF) are selected for the performance evaluation as they representative of Real Time (RT) schedulers in LTE Literature and VoIP traffic is the focus of this study/research. Of all the available RRM functionalities, the focus as anticipated is going to be on the Packet Scheduler, though the interaction with other entities is often taken into account and discussed in specific sections. Specific algorithms are analysed, one is derived and the performance evaluated at system level.

1.7 Dissertation Outline

As for the remainder of the dissertation, Chapter 2 provides background information on the LTE Advanced standard that is related to its system and protocol architecture at the system access level. More details are provided on the physical layer frame structure as well as the rule of the Medium Access Control layer in Radio Resource Management (RRM) and QoS support via packet scheduling procedure. The background information provided in Chapter 2 is necessary to provide a foundation for the discussions on LTE schedulers. It also provides a literature review on recent LTE Advanced packet scheduling proposals. Chapter 3 presents the definition of the LTE downlink scheduling problem and provides a general discussion on the approach used to solve the scheduling problem being addressed in this study. In this chapter we clearly illustrate the specific research problem and illustrate the design of the proposed scheme to solve this problem. Chapter 4 then describes the simulation environment and the metrics against which the respective scheduling algorithms were evaluated. In Chapter 5, performance results are graphically presented for the different experiments along with the commentary on the obtained results. Chapter 6 provides concluding remarks on the findings from the performance evaluation, and gives recommendations for future LTE Advanced schedulers based on the result analysis.

Chapter 2 Background and Literature Review

This chapter provides a background on LTE Advanced packet scheduling and a literature review on previous work done in the area of packet scheduling that provided a foundation for this study.

In the background, an introduction to the Long Term Evolution (LTE) technology is provided that discusses the LTE architecture in terms of its system components. This introduction is meant to give an overall view of the LTE system architecture so as to illustrate the role of the eNodeB (which contains the packet scheduler), its location in the system architecture and how it interacts with other system components.

In this chapter, a discussion is also provided on the physical layer structure, frame structure and protocol architecture. The frame structure together with a discussion on the physical layer structure of LTE Advanced provide background information for discussing the Resource Block (RB) dimensions which is the resource element being assigned to users by the scheduler and other scheduling parameters. The protocol architecture is briefly discussed with key emphasis on the Medium Access Control (MAC) layer as the MAC layer is where the scheduling function is contained. The protocol architecture is 3GPP specified and defined and is the basis on which scheduling algorithms for LTE/LTE Advanced are developed, however, the development of scheduling algorithms is not specified and is left open for developers and research [45].

A description on the operation of the Packet Scheduler is provided while highlighting the new features (techniques) in LTE Advanced (Release 10 and beyond) that render the previous algorithms for LTE (Release 8) somehow irrelevant. A discussion is further provided on the principle of channel dependent scheduling as the scheme proposed in this study is classified as a channel dependent scheduler as will be discussed in the proceeding sections.

The chapter is concluded with a description of The Radio Resource Management (RRM) structure for an LTE Advanced multi carrier system. Information on the RRM framework is provided as it offers a framework on which the Radio Resource Management algorithms and ultimately packet scheduling will be carried out and developed in an LTE Advanced multi-carrier system with Carrier Aggregation.

2.1 LTE Architecture

In this section, a preliminary discussion on the LTE Network Architecture is provided. This offers an illustration of the interaction of the eNodeB (which contains the packet scheduler) and other network nodes. The 3GPP LTE Architecture consists of specifications for the Evolved Packet Core, the core network and the Evolved-UTRAN for the Radio Access Network.

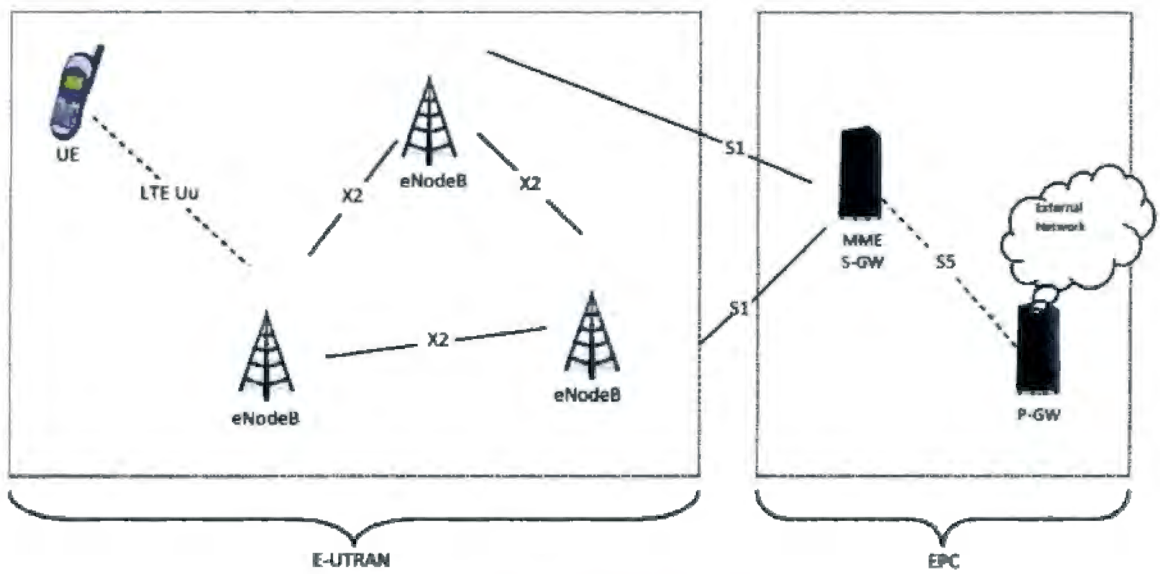

Figure 2-1. LTE Architecture

The Packet Data Network Gate Way (P-GW), Serving Gate Way (S-GW) and the Mobility Management Entity (MME) are the main components in the Evolved Packet Core (EPC). There are two main planes in the EPC which are the user plane and the control plane. The EPC components can be grouped within any of these planes. The MME is the core of the control plane while the S-GW is the core of the user plane. The signalling and connections with the Radio Access Network are managed by the MME. The forwarding and receiving of packets from the Radio Access Network (Network) is handled by the S-GW. The packet data interface is terminated at the P-GW and connects to the Packet Data Network [10][11].The eNodeB is connected to the MME and the SGW via the S1 interface. It transports user and control plane traffic between the Evolved UTRAN and the EPC. The EPC standards and architectural enhancements are described in the 3GGP specification document [12] [13].

The radio interface of the LTE system consists of only the eNodeB. The eNodeBs are linked via the X2 interface which reduces the packet loss due to mobility of UEs. The LTE-Uu interface links the eNodeB to the User Equipment and offers support for Radio Resource Management (RRM) functions. The LTE targets are reached through the use of Orthogonal Frequency Division Multiplexing (OFDM) for the downlink and Single Carrier Frequency Division Multiple Access (SFDMA) for the uplink. Orthogonal Frequency Division Multiple Access (OFDMA) is basically a multiple access technique where subsets of different subcarriers are assigned to different users and appear to be orthogonal to each other. SFDMA is a special type of OFDMA which uses less power when applied at the User Equipment.

Adaptive modulation schemes are utilised in LTE to achieve these very high data rates. There are three specific modulation schemes in the 3GPP specification i.e. QPSK, 16QAM and 64QAM. The channel coding adopted for the transport blocks is turbo coding [11].

The latency of LTE is divided into that for the control plane and for the user plane. The time delay faced by a user moving from a non active state into an active state is addressed by the control plane latency requirements. Two measurements are defined in this requirement, one measures the transition time from a camped state or idle mode state which has a latency of 100 ms and the other is the time taken to transit from a dormant state with a latency of 50ms.

The user plane latency requirement is defined as the time taken to send a packet from the User Equipment to the Radio Access Network edge node or vice versa. The 3GPP recommended one way transmission time should not exceed 5ms.

2.2 Physical Layer Interface and Structure

The physical layer interface and structure description is essential to be able to conceptualise the resource element (Resource Block) being assigned by the scheduler. The physical layer handles coding/decoding, modulation/demodulation, multi-antenna mapping and other physical layer functions. The physical layer offers services to the Media Access Control (MAC) layer in the form of transport channels. A description of the frame structure is discussed as it provides background information to understand and conceptualise the LTE scheduling parameters, such as the Transmission Time Interval (TTI), Resource Block (RB) size, dimensions and the number of subcarriers in each RB. The frame structure of the physical layer is described below:

2.2.1 The Frame Structure

The frame in LTE has duration of 10 ms. It consists of ten sub frames of the same size. Each sub frame consists of two time slots. Each slot comprises seven Orthogonal Frequency Division Multiplexing symbols including a cyclic prefix as illustrated in Figure 2.2.

The smallest resource unit that can be allocated by the scheduler is a Resource Block (RB), which has a dimension of 180 kHz in the frequency domain and 0.5 ms in the time domain as illustrated in Figure 2.3. This is different from WiMAX, which consists of 48 subcarriers per slot. A Resource Block comprises 12 contiguous subcarriers for one slot in duration [12] and one slot consists of 6-7 symbols depending on whether the cyclic prefix is normal or extended. The frame structure is seen in Figure 2.2 below:

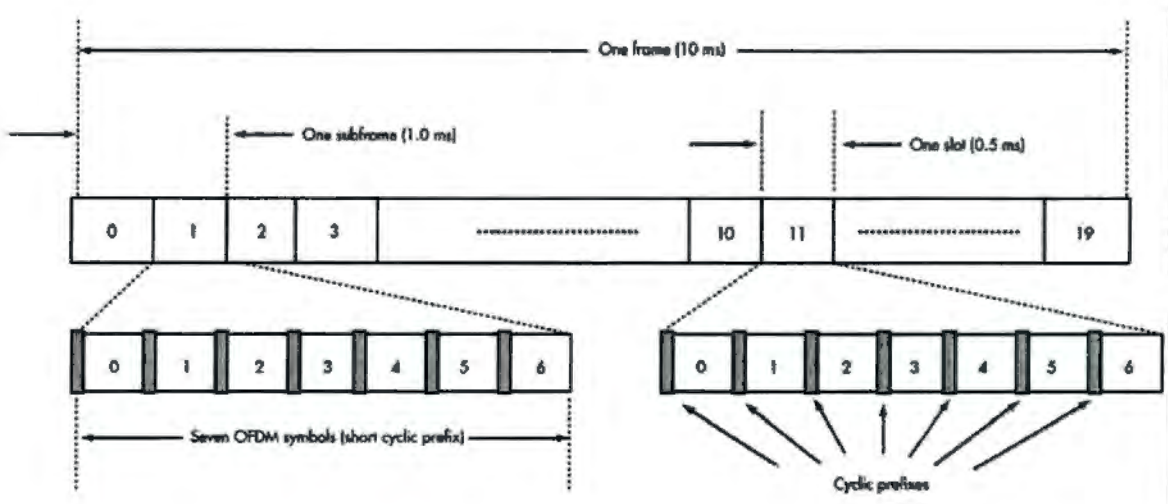

Figure 2-2. LTE Frame Structure [13]

There are two duplexing modes defined for the LTE standard to allocate LTE frames among the uplink and downlink directions. In Frequency Division Duplex (FDD) mode, each sub frame is treated as a whole unit for both the downlink and uplink as the uplink and downlink transmissions are separated into frequency bands. With Time Division Duplex (TDD), however, each sub frame comprises both uplink and downlink transmissions, where distributions of the sub frames between the two transmissions rely on the TDD configuration used, in addition to the presence of the special sub frame that provides a guard band between the uplink and downlink transmissions.

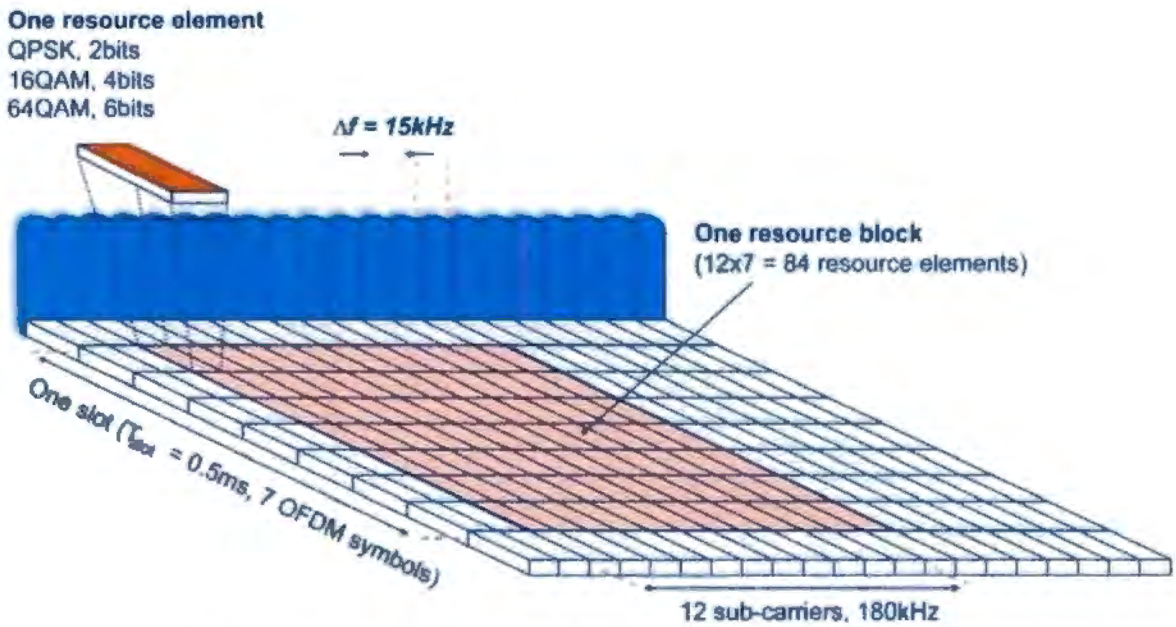

Figure 2-3. Downlink Time Frequency resource Grid [13]

2.2.2 Protocol Architecture

The protocol architecture differs between the downlink and the uplink. With respect to the packet scheduler, 3GPP made specifications for only the protocols while the actual scheduling algorithms are not specified and left open for development. It's based on this protocol architecture that the scheduling algorithms are designed. This dissertation is focused on the downlink scheduling, so only the LTE radio interface and protocol architecture for the downlink is discussed in this section. A more detailed discussion is provided for the protocol design with respect to the MAC layer as the scheduler entity, which is the focus of this study is located in the

MAC layer. The overview of LTE protocol architecture is shown in Figure 2.4 and thereafter discussed.

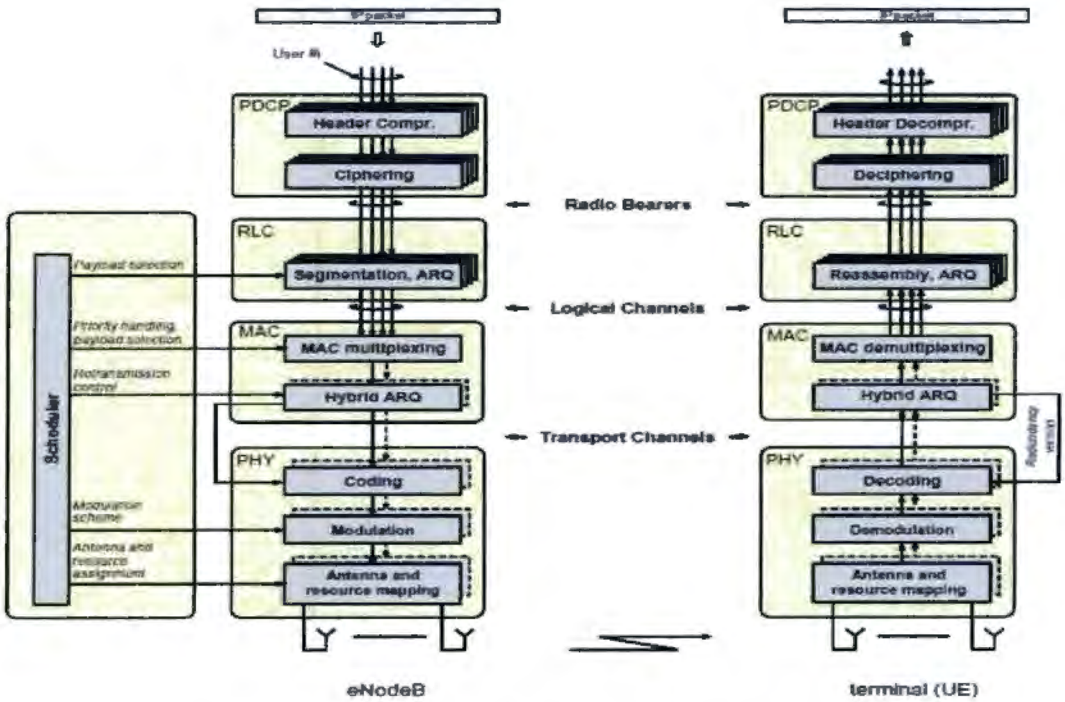

Figure 2-4. Overview of LTE Protocol Architecture [14]

RRC Layer

Radio Resource Control (RRC) handles control plane signalling. In addition to security and Quality of Service (QoS) management, RRC is in charge of establishment, maintenance, releasing of RRC connection and mobility management [14].

PDCP Layer

The Packet Data Convergence Protocol (PDCP) layer mainly handles header compression/decompression of IP packets that are received or transmitted to the IP layer above respectively. The header compression is important for reducing the overhead of data communication over the wireless interface, which also increases the system's spectral efficiency [14].

The PDCP also handles the in-sequence delivery of data packets either up to the IP layer or down to the Radio Link Control Layer. The in-sequence delivery ensures that PDCP detects missing packets that are to be discarded.

The PDCP also handles packet ciphering and packet encryption for the secure delivery of data packets over the radio interface.

RLC

Radio Link Control (RLC) is responsible for segmentation/concatenation, retransmission handling, duplicate detection, and in-sequence delivery to higher layers. The RLC provides services to the PDCP layer in the form of radio bearers. There is one RLC entity per radio bearer configured for a terminal.

MAC

The main role of the Medium Access Control (MAC) sub layer in LTE is to map between logical channels and Physical (PHY) transport channels. Logical channels are services offered by the MAC to the RLC sub layer above to handle different types of data exchange. Logical channels can be distinguished into data traffic channels, for transferring control signals between UE and eNodeB. Each logical channel service is provided to a certain RB from the RLC layer for proper RB prioritisation with respect to their QOS requirements.

MAC sub layer handles priority-handling operation to map user and control data flows from different RBs to their respective physical channels on the PHY layer via PHY's transport channels interface. Priority handling is also carried out either on RBs from different UEs, which is packet scheduling, or between RBs within the same UE.

The MAC layer is also responsible for detecting data transmission errors and correcting them via allocation of time and frequency resources for data transmission. Data transmissions are carried out at the MAC sub layer through a Hybrid Automatic Repeat Request, which is combination of forward error detection and correction via decoding process.

The MAC layer handles logical-channel multiplexing, hybrid-ARQ retransmissions, and uplink and downlink scheduling. It is also responsible for multiplexing/demultiplexing data across multiple Component Carriers when Carrier Aggregation is used. Figure 2.4 gives a summary of the flow of downlink data via all the protocol layers.

2.3 Key features of LTE Advanced

LTE Advanced should be a broadband wireless network that offers peak data rates that are equal or far greater than those in wired networks. The major requirements of LTE are reduced network cost, better provisioning and compatibility with other systems. LTE Advanced is backwards compatible with LTE and its requirements far exceed those of LTE. The data rates in LTE Advanced are however achieved through the combination of a few new advanced technologies. These new technologies are what render a few of the scheduling algorithms for LTE Release 8 irrelevant in an LTE Advanced multicarrier system and include the following:

Multiple Input Multiple Output (MIMO) – up to 8 layers

MIMO is a technology in a cellular system which involves the use of multiple antennas at both the receiver and transmitter sides [45]. The next generation cellular systems will be required to offer a large number of users high data rates and MIMO is a very essential technique to acquire these high data rates and spectral efficiency. Enhanced MIMO which comprises up to 8 streams at the sender and receiver side and is seen as one of the main features of LTE advanced that will allow the system to meet the IMT Advanced requirements as stated by ITU-R.

Cooperative Multipoint Transmission and Reception (COMP)

Cooperative Multipoint Transmission and Reception is a framework that refers to a system where several geographically spaced nodes cooperate with the aim of improving the performance of users in the common cooperation area. In LTE Advanced, COMP involves the simultaneous transmission by the User Equipment to or from several eNodeBs simultaneously. It is a very promising technique to reduce the inter-cell interference in the downlink by coordinating of transmissions.

Carrier Aggregation (CA)

The data rates promised by LTE Advanced can to a large extent be reached only through the extension of bandwidth. LTE Advanced uses a special technique called Carrier Aggregation where several bandwidths which we shall refer to as Component Carriers (CCs) are aggregated /joined together to create larger bandwidths as illustrated in the next section. With LTE Advanced, the bandwidth can reach as far as 100 MHz and there are plans to extend the bandwidth beyond this in later releases [6].

Relays

This is another technique that is included in LTE Advanced. According to 3GGP , the use of relays will provide coverage in new areas, offer temporary network deployment, increase cell edge throughput and offer coverage of high data rates[14].

2.4 Packet Scheduling in LTE/LTE Advanced

The Packet Scheduler is an entity located in the MAC layer of the eNodeB and is responsible for determining which terminal(s) to transmit and on which set of Resource Blocks (Figure 2.3). The Packet Scheduler is a very essential Radio Resource Management entity and to a large degree determines the overall behaviour of the system. This function is also referred to as channel dependent scheduling in the sense that the eNodeB at every 1ms interval transmits scheduling grants or information to the selected terminals while handling the uplink and downlink transmission activity. Scheduling is very closely linked to link adaptation which involves the adjustment of the data rates or power to adjust to the channel conditions and both of them are often seen as one function. The scheduling principles as well as which resources are assigned to various users may differ depending on the radio interface specifics for example whether we are considering the downlink (UE to eNodeB transmission) or uplink (eNodeB to UE transmission). In this section a detailed description of the sequence of operation of the downlink scheduler with respect to the downlink is provided as this study is focused on the downlink side.

2.4.1 Downlink Scheduling

The downlink scheduler is responsible for dynamically determining which terminals to transmit to and the set of Resource Blocks upon which this transmission will occur. The Packet Scheduler also handles transport-format selection and antenna mapping for each Component Carrier (CC) together with the logical channel multiplexing for the downlink transmissions as illustrated in Figure 2.5 below.

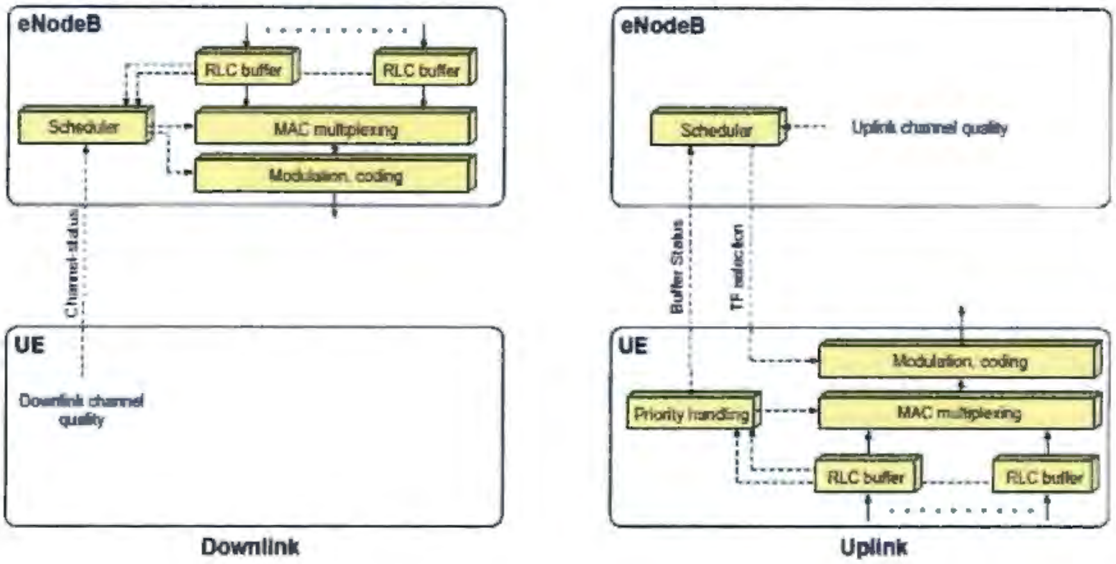

Figure 2-5. Transport format selection in downlink and uplink [14]

A user may not be able to utilise the full capacity of the cell due to several factors e.g. due to lack of sufficient data. In addition, the channel conditions may vary in the frequency domain, it is therefore imperative to transmit to different terminals on various parts of the spectrum. Multiple UEs can be scheduled for transmission simultaneously in a sub frame. The packet scheduler is in control of the instantaneous data rate and RLC segmentation. The scheduling strategy is implementation specific and is not specified by 3GPP. 3GPP only specifies the protocols for scheduling [45]. The schedulers for LTE aim to take advantage of channel variations between terminals and aim at scheduling terminals when their channel conditions are advantageous. The scheduling strategies take into consideration a number of factors which include:

- Channel conditions
- Interference situations
- Priority of the data flows

The data about the channel conditions at the terminal can be acquired in several ways. This is usually obtained through channel state reports. Other sources of channel information include e.g. using channel reciprocity to estimate the downlink channel quality from uplink channel estimates in TDD [14]. More on channel conditions is discussed in section 2.4.2. Downlink inter-cell interference coordination may also be incorporated in the scheduler this is however beyond the scope of this study.

2.4.2 Varying channel conditions

One key feature of mobile radio communication systems is the typically rapid and significant variations in the instantaneous channel conditions. There exist several reasons for these types of variations.

- Frequency- and time-selective fading will result in rapid and significant variations in the instantaneous channel conditions.
- Shadow fading and distance dependant path loss will also affect the average signal strength significantly and the interference at the receiver due interference from other cells will affect the signal strength.

Therefore, when discussing and comparing different scheduling algorithms it is important to distinguish the different types of variations in the channel [45]:

Fast variations in the channel quality can occur due to fast multipath fading and fast variations in the level of interference. For packet data applications these large short time variations are often acceptable and not even noticeable by the users.

The long term differences in the service quality between the communication links due to shadow fading and differences in the distance to the cell site.

Benefits of varying channel conditions

A significant rise in system capacity is obtained if the channel conditions are taken into account and this is referred to as "channel dependent scheduling" [45]. There is almost always a radio link whose channel quality is near its peak as illustrated in the Figure 2.6 below.

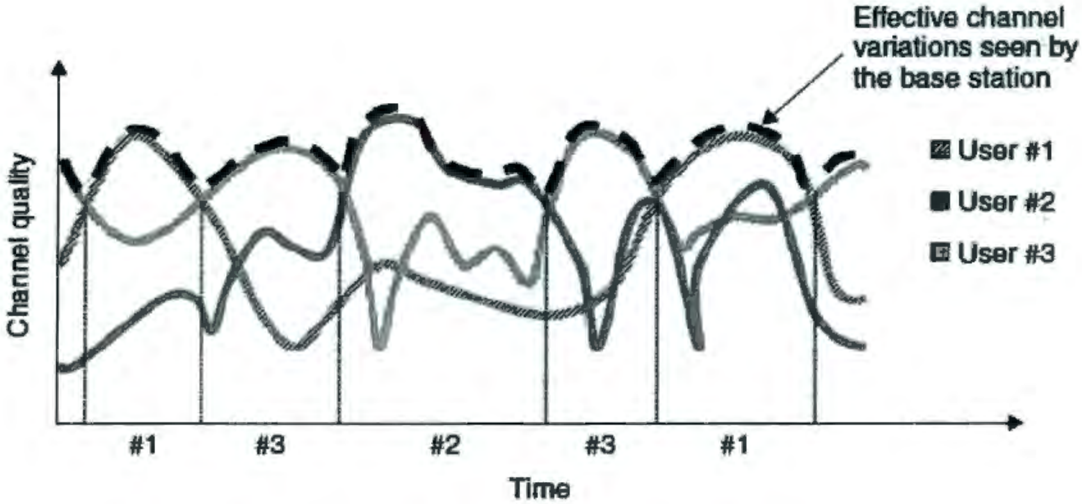

Figure 2-6. Channel dependent scheduling [45]

As this radio link is likely to have a good channel quality, a high data rate can be used for this radio link and this translates into high system capacity. "Multi user diversity' is the gain obtained by transmitting to users with favourable channel conditions [45]. The gains from multi user diversity are larger the larger the channel variations and the larger the number of users in a cell as this increases the chance of finding a user with favourable channel conditions. Hence, in contrast to the traditional view that fast fading is an undesirable effect which has to be combated it is instead beneficial and can be exploited [45].

2.4.3 LTE Advanced features with respect to packet scheduling

As highlighted in section 2.3, LTE Advanced comes with new technologies, the introduction of these new features means that most scheduling algorithms developed for LTE will not apply for an LTE Advanced multicarrier system. In this section, an explanation is provided on how each of these new features creates a scheduling challenge. A more detailed discussion is provided for the Carrier Aggregation feature as it is the feature studied in this research project. A discussion is also provided on the types of Carrier Aggregation to provide background information to further discuss the scheduling challenges that come with Carrier Aggregation and further discuss the assumptions set forth in this study.

Carrier Aggregation

Carrier Aggregation is one of the most distinguishing features of LTE Advanced which has been standardized by 3GPP as part of Release 10 and 11. This technique enables scalable extension of effective bandwidth delivered to a user terminal through the concurrent utilisation of radio resources across multiple carriers [15].

In general, there are three variations of Carrier Aggregation (CA) as discussed below: [15]:

- Intra-band Contiguous Carrier Aggregation. This is where a contiguous bandwidth wider than 20 MHz is used for Carrier Aggregation. This form of Carrier Aggregation will become possible when the 3.5 GHz becomes available in the future in various parts of the world [15].
- Intra-band non contiguous Carrier Aggregation. This is where multiple CCs belonging to the same band are utilised in a non contiguous manner. It's is expected in countries where spectrum allocation will be dispersed within a single band [15].
- Inter-band non Contiguous Carrier Aggregation. This is where multiple CCs belonging to different bands for example 2 GHz and 800 MHz band are joined together.

Note: due to the RF complexity that comes along with simultaneous transmission on CCs in different bands, it has been suggested by 3GPP to focus on intra band Carrier Aggregation for the uplink while on the downlink it has been suggested to concentrate on both intra and inter band carrier Aggregation [15].

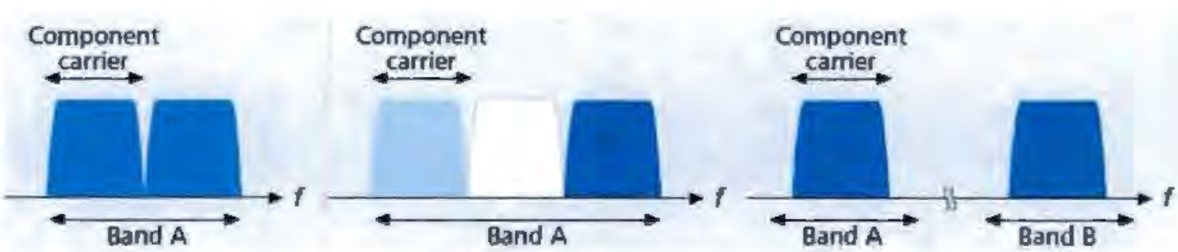

Figure 2-7. Intra band contiguous CA (b) Intra band non contiguous CA

(c) Inter band non contiguous CA [15]

The introduction of Carrier Aggregation comes with a number of packet scheduling challenges e.g. the designers of scheduling algorithms have to not only consider Resource Block allocation but also the allocation of users to different Component Carriers this is referred to as "carrier scheduling". Secondly, some LTE users cannot be scheduled on some CCs because of radio frequency issues and therefore the scheduler has to take this into consideration when allocating users to different CCs. Another issue that arises deals with the coverage of different CCs. Some CCs may have different coverage than others depending on the band from which the CC is assigned from. So the Packet Scheduler has to take this into account. There also arises the resource allocation unfairness between LTE and LTE Advanced terminals. The LTE Advanced terminals have the capability to be scheduled across all the Component Carriers while the LTE terminals can be scheduled on only one CC [6]. The LTE Advanced terminals therefore enjoy N-times more resources than the LTE terminal where N is the number of CCs that have been aggregated. This issue is considered in the design of the proposed scheme as will be discussed in chapter 3.

COMP

COMP as initially discussed in section 2.3, deals with the simultaneous transmission and reception to several base stations. The transmission and reception to several base station means that new types of interference come into play. In such an instance, new scheduling algorithms have to be coupled with interference management schemes /algorithms. In case of beam forming, beams of different cells might collide therefore neighbouring cells have to predict somehow the interference that will be experienced [17]. COMP would require the use of a master scheduler located in one eNodeB to schedule joint multiple cells in a centralised manner this is different from the LTE Release 8 type of scheduler where each eNodeB has a single scheduler. More details on this are provided in [17]. COMP is however not considered in the scope of this study and therefore a brief discussion is only provided.

MIMO (up to 8 layers)

MIMO technology as discussed in section 2.3, deals with having multiple antennas i.e. up to 8 sender and receiver antennas at both the transmitter and receiver end. This would require a completely different channel state reporting mechanism which introduces more signalling overhead. This implies that new scheduling algorithms would need to incorporate modified channel state reporting mechanisms [18]. Channel state reporting is a form of signalling where the Packet Scheduler acquires information about the state of the channel for each UE depending on its location. This information is used by the scheduler to determine which UE to schedule during a specified time interval. MIMO technology is also beyond the scope of our research and only a brief summary is provided.

Based on the features and information discussed in the preceding sections, a discussion on scheduling proposals for LTE Advanced in literature are discussed in the next section.

2.4.4 Radio Resource Management Framework for an LTE Advanced multicarrier system

Based on the introduction of Carrier Aggregation which is discussed in the preceding section, a new Radio Resource Management frame work was defined by 3GGP [16]. It is based on this frame work that new Radio Resource Management (RRM)/scheduling algorithms will be and have been designed.

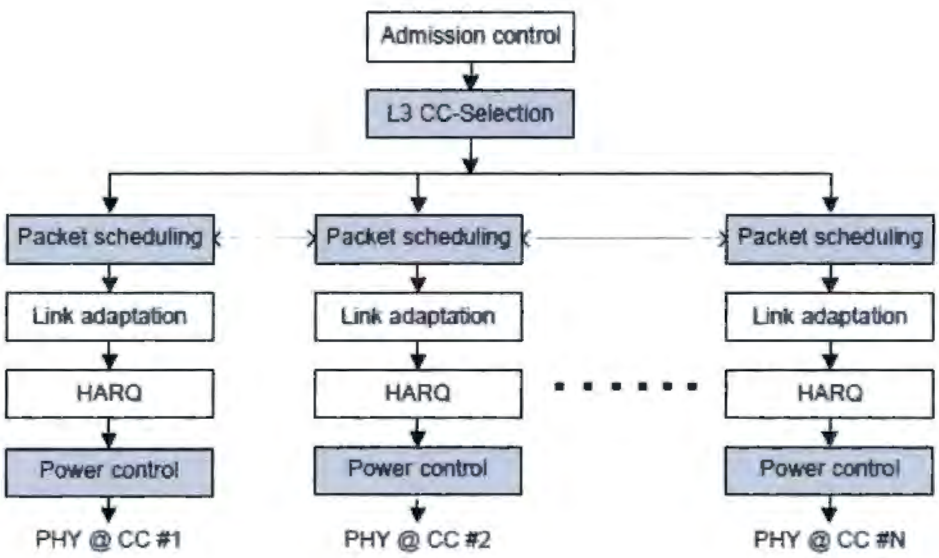

Figure 2-8. RRM Framework of a multi-Component Carrier LTE Advanced system [16]

The RRM framework for a multi Component Carrier system is demonstrated in Figure 2.8. As seen above there are separate RRM blocks which operate independently on each CC. It was agreed within 3GGP to adopt independent link adaptation per CC which takes into consideration LTE Release 8 assumptions. This maintains backwards compatibility such that the ordinary LTE terminal can function in an LTE Advanced system. The admission control module in the base station makes a decision on whether a connection is accepted or rejected followed by CC selection depending on the QoS requirements of a user [16].

2.5 Literature Review

Following the setup of the LTE protocols and radio interface/frame structure discussed in section 2 and the framework for RRM in LTE Advanced, various distinguishing works have been done as regards packet scheduling for LTE/LTE Advanced and are hereby discussed:

One of the earliest works that laid a foundation for the design of the LTE Advanced Packet Scheduler was carried out by Li Chen and Wenwen Chen in [19]. In their work, they performed an analysis and simulation for spectrum Aggregation in LTE Advanced. In their research, they proposed two scheduler structures for a system with spectrum aggregation (Carrier Aggregation) which are briefly discussed below:

The model of a disjoint queue scheduler

In this structure of the scheduler, the users each have a traffic queue on each Component Carrier (CC). They proposed a two layer scheduler in the sense that the outer scheduler allocates the traffic packets of users to its corresponding queue to await transmission on each CC. The second scheduler then assigns the packets in each corresponding queue to the different Resource Blocks as illustrated below in Figure 2.9.

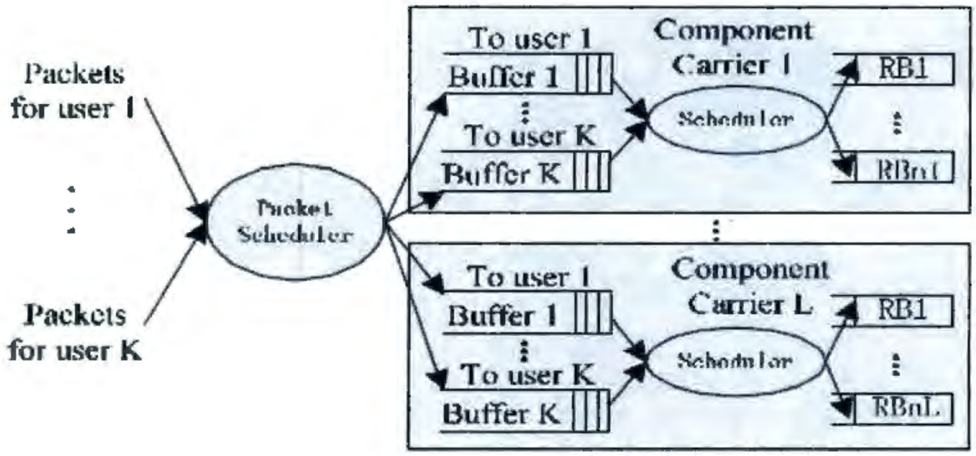

Figure 2-9. The model of a disjoint queue scheduler [19]

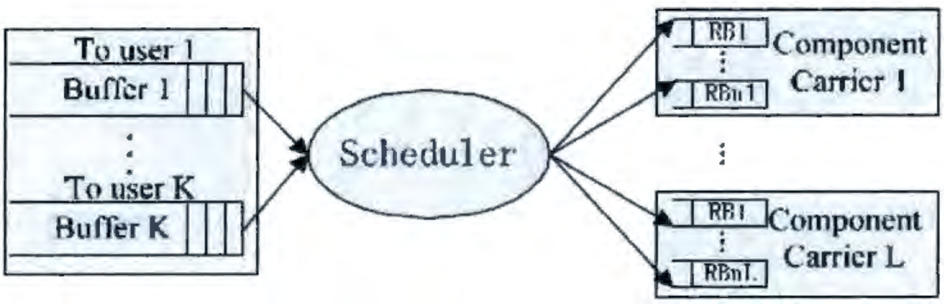

Figure 2-10. The model of a joint queue scheduler [19]

For simplification, they proposed a load based strategy for the outer scheduler and the Proportional Fair (PF) scheduler for the inner scheduler. The strategy behind the PF scheduler is discussed in the appendix. In this structure of the scheduler, they highlighted that the packets of a user can only be assigned to only one Component Carrier and therefore the efficiency is low especially when the packets are large.

The disjoint queue model that they proposed suffers several demerits. In a multicarrier system, the way to achieve high spectral efficiency is to assign the resources (RBs) to the appropriate user in the system who has data to send [45]. However, when using this type of scheduler, the selectable user set for each CC is just a subset of the total number of users which limits the "multiuser diversity gain" (refer to section 2.4 for a description of the multiuser diversity gain) and hence the spectral efficiency is low. On the other hand, for the scenario of elastic traffic input, this model of the scheduler will lead to the case that some CCs stand idle since the data of their serving users have been completely finished.

The model of a joint queue scheduler

In this structure of the model that they proposed there is only one scheduler that assigns users to the different Component Carriers as seen in Figure 2.10. Each user has only one queue for all the Component Carriers and the CCs share the joint queue of each user. The efficiency of this structure is much higher and the authors stated that the complexity was much lower than the disjoint queue scheduler.

The most significant merit of their work was the introduction of baseline schedulers for a multi carrier system. They also established a simulation platform where they were able to deduce that spectrum aggregation enhances the throughput and latency performance of LTE Advanced.

However, despite the significance of this structure of scheduler, it is limited as it doesn't cater for the older LTE terminals (Release 8). The LTE Release 8 terminals can only be scheduled on one CC and therefore cannot not be assigned a scheduler that distributes packets across all CCs such a scenario would require the terminals to have a complex RF circuitry which is not present in older releases of LTE at the same time, allocation of several CCs to a terminal would require a lot of power (more signals to process) yet the older release LTE terminals are more power limited than the LTE Advanced terminals [15]. It is imperative to make consideration for the existence of both LTE and LTE Advanced terminals in the same network. This type of scheduler can only cater for the high end LTE Advanced terminals which have high complex circuitry [6]. Their work was also limited in the sense that they did not address how real time traffic specifically VoIP traffic would be scheduled. They limited their work to developing a carrier scheduling scheme.

G. Aiyetoro et al in [20] performed a study where they investigated the performance of the existing Modified Largest Weighted Delay First (MLWDF) and Exponential Proportional Fair (EXP-PF) scheduling algorithms. In their study, they used a satellite LTE Air Interface which enables global coverage and hence supports the terrestrial one in the provision of LTE services to mobile users. In their work, they were able to determine how both schemes operate. However, in their study they were limited and did not consider other common and highly relevant Real Time (RT) scheduling schemes suitable for a terrestrial environment in LTE. There exist a considerable number of other common RT schedulers for VoIP traffic which offer QoS differentiation hence these schedulers have to be considered too. This study extends their work by considering the terrestrial scenario. This study goes on further to consider other representative Real Time schedulers for VoIP traffic such as the Weighted Proportional Fair (WPF) algorithm and the Queue Based Maximum Rate Algorithm (QBMC) in an LTE Advanced system multicarrier system.

In [21], authors presented an efficient quantised water-filling packet scheduling algorithm for high-data-rate nomadic users in such LTE-A systems. They focused on a scheduling scenario with Component Carriers (CCs) in different bands, as CCs aggregated in different frequency bands have more ability to ensure full flexibility in the process of dealing with the bandwidth extension to achieve the much higher data-rate transmission. They addressed the fact that with the Carrier Aggregation technology, it will be possible to schedule a user on multiple Component Carriers simultaneously. They state that a new problem should be considered in resource scheduling. In non adjacent inter band aggregation scenario, where the aggregated carriers belong to different frequency bands, the fading characteristics are different between carriers, such as the path loss and Doppler shift [27]. Consequently the coverage may vary from carrier to carrier. With different location in the cell, some users can only be scheduled on fewer carriers, while other users can be scheduled on the entire aggregated carrier, which has a negative impact on allocation fairness among users. Figure 2.11 shows that the coverage area of lower frequency band f1 (e.g. 800 MHz) is bigger than that of the frequency band f2 (e.g. 2000 MHz). Therefore, User 1 located further away from base station will operate on fewer CCs than the User 2 who is closer to the base-station

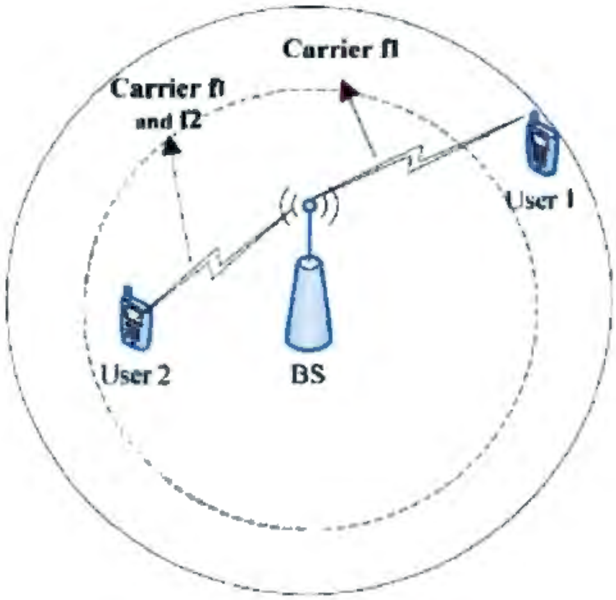

Figure 2-11. Coverage of different frequency bands [21]

In this regard, they proposed a Proportional Fair (PF) scheduling algorithm based on user grouping (UG-PF) for an LTE-Advanced system with Carrier Aggregation. The users are partitioned into some specific groups according to the number of carriers that they can be scheduled on. Then, the average channel access probability of the different user groups is analysed. Based on the analysis results, a modified scheduling policy for the UG-PF Scheduling algorithm was obtained. It can adjust the average channel access probability of different user groups proportionally so as to improve the allocation fairness. Their scheme was able to cope with the issue of CCs availability for users in a cell. Through analysis and simulation their scheme showed significant improvements in the overall delay performance when compared to the case without Carrier Aggregation. In their scheme, they considered the difference in the number of CCs users can be allocated based on coverage range of different CCs. This is different from the assumptions set forth in this study because this study considers difference in CC allocation based on terminal complexity. Their scheme was also lacking in the sense that it did not address delay constraints and QoS of Real Time packets or traffic such as VoIP. This study addresses this issue.

Mohammed et al in [22] performed an evaluation of uplink schedulers in LTE in mixed traffic scenarios. However, their focus was on a mixed traffic scenario consisting of VoIP, Video Streaming and FTP traffic. There objective was to evaluate the connection level performance of representative scheduling proposals, with focus on QoS aspects. They specifically utilised a mixed type of traffic flow as stated above and evaluated the schedulers in terms of per user throughput, packet loss and fairness. Their contribution was to offer an extendible environment in which uplink schedulers in LTE, in addition to other networks utilizing OFDM based techniques can be evaluated in a repeatable and practical manner. In their study, however they focused on uplink scheduling schemes. The results concluded for the uplink cannot apply to the downlink side because the uplink side involves power constraints and power reduction algorithms as the terminals which are transmitting to the scheduler are power limited this is however not the case in the downlink which involves transmission by the base station to users which has far more power than an ordinary terminal. The uplink multiple access scheme is based on Single Carrier Frequency Division Multiple Access which is a special type of OFDMA that requires contiguous resource block assignments to lower the Peak to Average Power Ratio (PAPR) and thus limits the scheduling flexibility. Hence the results for the uplink side cannot apply to the downlink. In their study, they concluded that the standards proposed in literature perform almost similarly.

Authors in [23] proposed an Adaptive Time Domain Scheduling Algorithm (ATDSA) for OFDMA based LTE Advanced Networks. The aim of their algorithm was to improve the QoS of different traffic types while keeping the system performance as high as possible and maintaining the fairness among all users at a good level. Their scheme uses a Hebbian learning process to learn the environment in terms of QoS, packet dropping rate of Real Time traffic and to allocate the bandwidth portion of RT traffic accordingly. The Hebbian learning process that they use in their algorithm is a form of artificial intelligence and it is based on a rule proposed by Hebb in 1949 [23]. It states how much weight of a connection between two points should be decreased or increased in proportion to the product of their activation. They integrated the Hebbian rule to make a decision on capacity allocation for Real Time traffic based on achieved QoS parameters. In their scheme the ATDSA, the QoS measure unit continuously calculates Packet Dropping Rate (PDR) of Real Time traffic during each transmission time interval and keeps this data in a vector named the PRB allocation priority vector. The learning process makes a comparison between the current and previous PDR values and changes the weight of Real Time traffic accordingly as such if the current value of the PDR is higher than the previous value; it increases the weight of Real Time traffic. In addition, they split the scheduler between the time domain and frequency scheduler having separate schedulers for control, real time traffic, streaming traffic and background traffic to handle the queues of the respect traffic types. In their simulation results, they were able to show that the proposed scheme reduces average delay, delay viability and PDR of RT traffic.

Their work however had two shortcomings: firstly, the introduction of separate schedulers for the different traffics in the time domain each for different traffics increases the complexity and thus the time in which the algorithms operates. The Transmission Time Interval (TTI) for LTE/LTE Advanced has been set to 1 ms and therefore scheduling algorithms have to be fast enough to operate within this time frame.

The second shortcoming is that they did not discuss how users will be scheduled across the CCs and did not incorporate any of the new LTE Advanced features or techniques such as MIMO or Carrier Aggregation in their work. In this study, the Carrier Aggregation feature in LTE Advanced systems is the feature considered in the scope of this research.

B. Wang et al in [24] in their study discussed the Packet Scheduler design and performance simulations for running VoIP services over High Speed Downlink Packet Access. A packet scheduling design incorporating VoIP packet aggregation and user multiplexing is proposed and the VoIP capacity is studied for a macro–cellular environment. The authors demonstrated that HSDPA is attractive for transmission of VoIP based on its high voice capacity. However, their work is limited in the sense that they only considered a single traffic scenario in their evaluation in addition their work was evaluated for an HSDPA system. The HSDPA system technology is different from the LTE Advanced System as the HSDPA Packet Scheduler schedules users on time while the LTE system schedules time and frequency resources as such the results cannot apply to LTE. This study extends their work by observing the VoIP performance in a mixed traffic scenario in an LTE Advanced multicarrier system.

In [46], the authors developed a scheduling scheme to reduce the scheduling delay in Carrier Aggregated systems. They were able to illustrate that it is possible to reduce the scheduling delay for users if the available Resource Blocks in a downlink transmission are organised into sets. The scheduling configuration they proposed is illustrated in Fig 2.11 below:

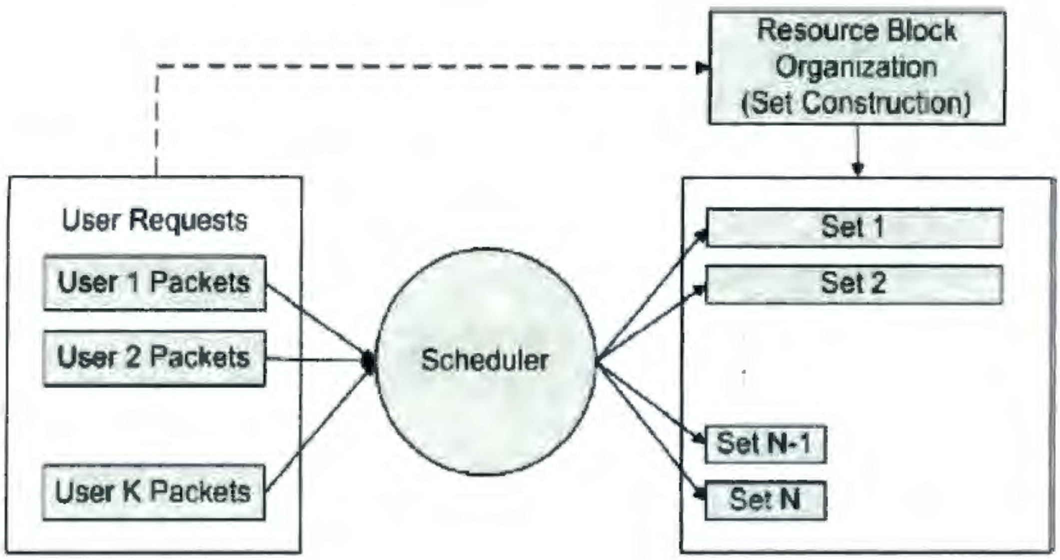

Figure 2-12. Proposed scheduler configuration for CA using Resource Block Organisation [46]

The authors mentioned that current scheduling strategies in literature rely on the individual assignment strategy of available Resource Blocks (RBs). They referred to this strategy as "RB by RB scheduling". They stated that as the number of Resource Blocks increases, the scheduling delay due to Resource Block (RB) allocation becomes significantly high. In this regard, they proposed a scheduling strategy whose configuration is illustrated in Figure 2.12 above. In their scheme, they proposed the use of a Resource Block organisation strategy/algorithm in conjunction with the scheduler. The scheduler in their scheme assigns sets of Resource Blocks to different users and they refer to this as "Set Scheduling".

Despite the significance of their scheme it suffers a significant amount of demerits which are hereby discussed. Their scheme is based on the assumption that the Resource Blocks are contiguous. It does not consider Inter-band non contiguous Carrier Aggregation and Intra band non Contiguous Aggregation which are illustrated in section 2.4.2. In both the two types of Carrier Aggregation RBs are non contiguous so a Resource Block organization strategy that organises Resource Blocks in sets wouldn't be possible.

In addition to this, the introduction of a Resource Block organisation strategy/algorithm increases the complexity of the scheduling algorithm. The scheduling interval specified for

LTE/LTE Advanced takes exactly 1ms so as to take advantage of the fast variations in channel quality due to fast multipath fading and fast variations in the interference level [45] which are beneficial if exploited as discussed in section 2.4.2. In this regard, the scheduling algorithm has to be fast enough to operate within this 1ms time interval.

Their scheme is further limited in the sense that it doesn't address how Real Time packets such as VoIP packets will be scheduled and only focuses on Resource Block organisation. In this study, the scheduling of Real Time packets such as VoIP is addressed. The scheme that is proposed in this study can be applied to Resource Blocks organised in a non contiguous manner.

Chapter summary

In this chapter, literature on packet scheduling regarding LTE and LTE Advanced has been discussed. A description of the LTE system architecture was initially discussed to give a holistic view and interaction of the LTE system components and radio interface. All main features of LTE and LTE Advanced have been presented to provide the necessary background relevant to the project. A description is provided to illustrate the resource sharing problem in LTE networks starting from the basics and then adding more and more details to explain complex aspects of the system. The LTE physical and frame structure is also explained in detail providing a description of the scheduling parameters as set forth by 3GPP. In addition, this chapter has presented a survey of proposed scheduling algorithms for LTE Advanced. Most of the algorithms were developed based on the new features of LTE Advanced and on the Proportional Fair algorithm which is discussed in Appendix A. Some challenges to implementing the scheduling algorithms are presented, as well as the drawbacks and merits of these studies highlighted. The work to be carried out after this chapter is to illustrate a new solution for VoIP packet scheduling in LTE Advanced in a Carrier Aggregation scenario. It includes modifying the well known existing Modified Largest Weighted Delay First (MLWDF) Algorithm, and evaluating the performance of packet scheduling algorithms such as Round-Robin, Max-Rate, Proportional Fair (PF), etc, in a multicarrier system and ultimately creating a suitable scheduling algorithm for VoIP traffic in an LTE Advanced system that reduces the packet loss.

Chapter 3 Proposed Packet Scheduling Scheme

The LTE technology uses advanced Radio Resource Management (RRM) techniques such as channel dependent scheduling to meet the high data rates set forth by the 3GGP project and hence achieve high system capacity. In Chapter 2, a discussion was given on how high system capacity can be achieved if channel conditions are taken into account when making a scheduling decision. The existing MLWDF first algorithm is one of such channel dependent schedulers. The existing MLWDF algorithm was designed to offer support for real time data services such as VoIP [44]. It has been utilised in legacy networks such as High Speed Access Networks (HSPA) and LTE as was discussed in chapter 1.

The introduction of Carrier Aggregation presents a scheduling challenge for the scheduling algorithms in literature that have been developed for LTE. There also arises resource allocation unfairness between the legacy LTE terminals and the most recent LTE Advanced terminals. This study proposes an evolved MLWDF scheme to address these issues.

The chapter begins with a discussion on VoIP traffic which is the traffic being considered in this study. The reason for selecting VoIP traffic is discussed with an illustration of the criteria for selecting packet loss as the main metric for evaluation. The existing MLWDF, which is designed to support VoIP traffic, is illustrated while highlighting its key features and drawbacks. A new scheme (evolved MLWDF algorithm) is then proposed with its key features highlighted.

3.1 VoIP (Voice Over Internet Protocol) Traffic

The LTE Advanced requirements for high data rates are achieved due to the fact this technology is only designed for Packet Switched Networks; hence there is no need for the circuit switched mode [42]. However, this design brings with it several technical challenges especially for a real time service such as VoIP.

The VoIP technology has grown rapidly due to different factors such as: low cost, the integration of voice and data traffic over the existing network infrastructure etc. VoIP services have attracted more customers as well as companies from circuit switch networks to packet switch networks [42]. However, the fact that VoIP uses a packet switched network implies that the QoS provided by VoIP is not as good as that of a circuit switched network. This is due to the fact that real time traffic such as VoIP is adversely affected by technical issues like end to end delay or latency, jitter and packet loss which affect the voice quality significantly [42].

The growing need for wireless broadband access for different emerging user applications has increased the need for QoS guarantees and radio resource utilisation. Recent studies have shown that the proportion of VoIP users continued to grow from 28% of users in 2008 (up from 20% of user in 2007) to more than 50% in 2010 [6]. This implies that VoIP is becoming a commodity hence efficiency for voice offering in the form of efficient Radio Resource Management techniques such as packet scheduling is becoming key to maintaining potential profit for wireless operators hence making VoIP traffic the focus of this research study.

3.1.1 VoIP QoS

QoS involves the delivery of data whilst meeting a combination of latency, jitter, error rate and maximum /guaranteed bit rate requirements [47]. VoIP is a very good example of a value-added service which requires efficient QoS support. The voice quality perceived by an end user can be measured using the Mean Opinion Score (MOS) which assigns a metric between 1(lowest quality) and 5 (highest quality) as displayed in Table 3.1 below. MOS is quite subjective, as it is based on figures that result from what is perceived by people during tests. However, there are software applications that measure MOS on networks [47].

MOS	Quality	Impairment
5	Excellent	Imperceptible
4	Good / Toll Quality	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

Table 3-1. Mean Opinion Score (MOS) definitions [47]

Figure 3.1 shows the effect of packet loss /error rate and end to end delay on Voice quality for the AMR codec typically used in 3G LTE/LTE Advanced.

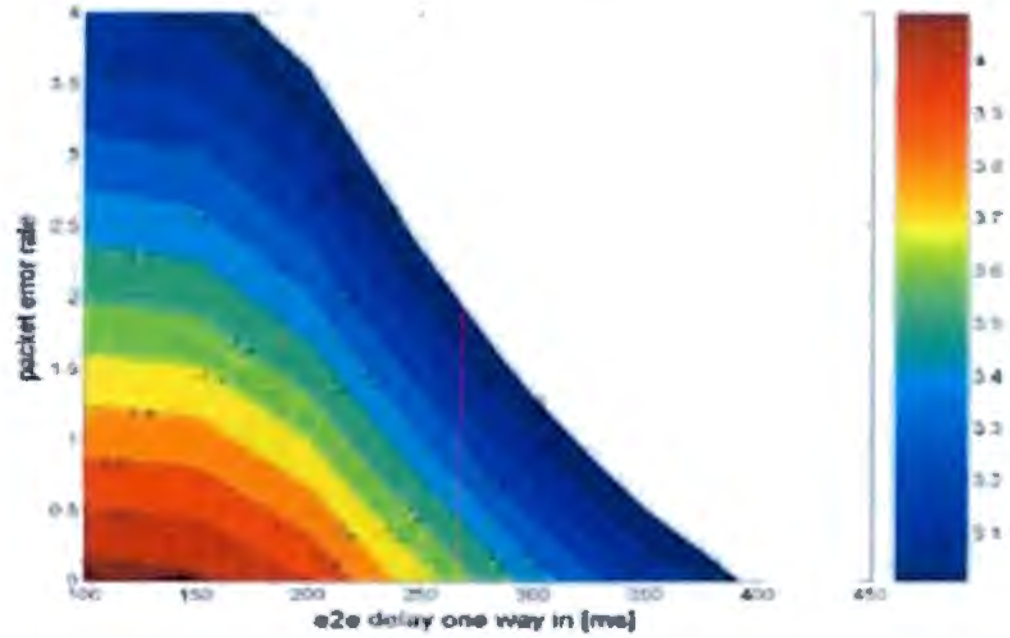

Figure 3-1. Mean Opinion Score (MOS) for Adaptive Multi- Rate (AMR) [47]

From the Figure 3.1, it can be seen that voice quality falls off rapidly for both delay and packet error/loss rate. Packet loss is thus a crucial metric to be taken into consideration when providing QoS for VoIP traffic.

One of the main characteristics of VoIP traffic is block/packet error rate that leads to packet loss and delay [42]. According to [42], the allowed maximum mouth to ear delay for voice is 250 ms with the basic assumption that the delay for the core network is approximately 100 ms, while the tolerable delay for the Radio Link Control, Medium Access Control (MAC) buffering, scheduling, and detection should be approximately lower than 150 ms as shown in Figure 3.2. Hence taking into account that both end users are LTE, the tolerable delay for buffering and scheduling must be lower than 80 ms. To better account for the variability in network end to end delays a delay of 50 ms from eNodeB to UE has been chosen for the 3GPP performance evaluation metric [47].

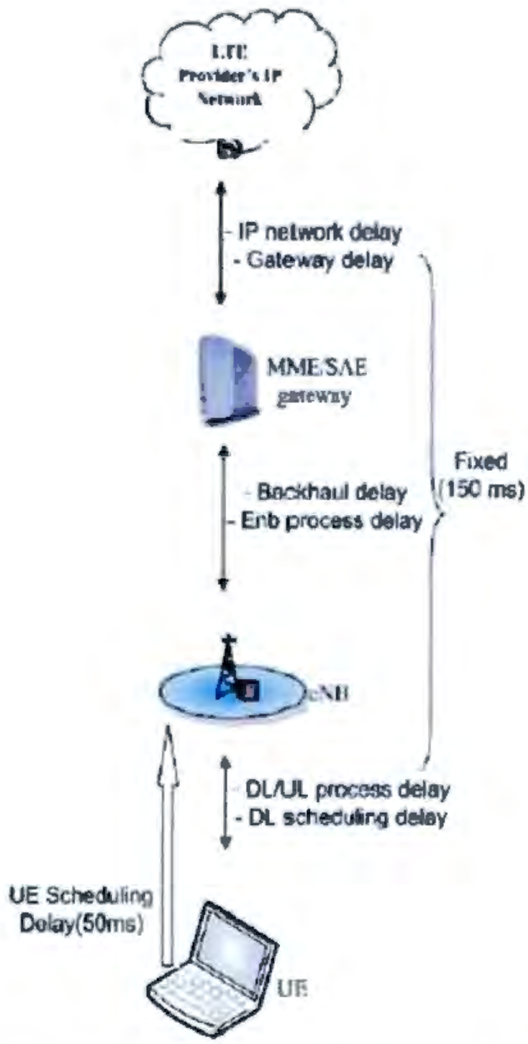

Figure 3-2. End to End delay components in E-UTRAN (LTE) [42]

3.1.2 VoIP packet scheduling

Providing QoS in particular meeting the packet loss and packet delay constraints of real time services such as VoIP is one of the requirements in emerging high-speed data networks such as LTE [42]. The most common real time schedulers that have been developed for VoIP are the Exponential Proportional Fair Scheduler (EXP-PF) and the Modified Largest Weighted Delay

First Algorithm (MLWDF) [45].The two types of schemes are both designed to offer support for multiple real time users in a wireless data system. Their strategies are discussed in detail in the appendix.

As part of this study, an initial performance evaluation was performed to determine the most efficient real time scheduler supporting VoIP traffic in terms of low packet loss in an LTE Advanced downlink multi-carrier system. Through the evaluation, it was found that the existing MLWDF offered the lowest packet loss ratio at all load conditions. The MLWDF scheme was thus selected as the scheduling algorithm for this study and hence selected as a base line scheduler for the evaluation of the new proposed scheme that is introduced in this research project. The existing MLWDF algorithm is thus further discussed in more detail in the next section.

3.2 The existing MLWDF Algorithm

Providing QoS for real time traffics such as VoIP is one of the key requirements in emerging high speed networks. This is very challenging in networks that include wireless links. The quality of a wireless channel is different for several users and randomly changes in time on both slow and fast time scales [44]. Wireless capacity is also a scarce resource that needs to be utilised efficiently. It is therefore imperative to find throughput efficient ways of scheduling.

One key feature of mobile radio communication systems is the typically rapid and significant variations in the instantaneous channel conditions. There exist several reasons for these types of variations:

Frequency selective fading will result in rapid and significant variations in the instantaneous channel conditions. Shadow fading and distance dependant path loss will also affect the average signal strength significantly and the interference at the receiver due interference from other cells will affect the signal strength.

When discussing scheduling algorithms it is important to distinguish the different types of variation in the channel [45]:

Fast variations in the channel quality which can occur due to fast multipath fading and fast variations in the level of interference. For packet data applications, these large short time variations are often acceptable and not even noticeable by the users. Fast variations due to multipath fading are considered in this study.

Secondly the long term differences in the service quality between the communication links due to shadow fading and differences in the distance to the cell site.

A significant rise in system capacity is obtained if the channel conditions are taken into account and this is referred to as "channel dependent scheduling" [45]. There is almost always a radio link whose channel quality is near its peak as illustrated in Figure 2.6. of Chapter 2.

As this radio link is likely to have a good channel quality, a high data rate can be used for this radio link and this translates into high system capacity.

The existing MLWDF Algorithm is an example of a channel dependent scheduler and is discussed in more detail below:

Consider a queuing model of a cell with two mobiles as shown below considering the transmission from the base station to the users (downlink transmission):

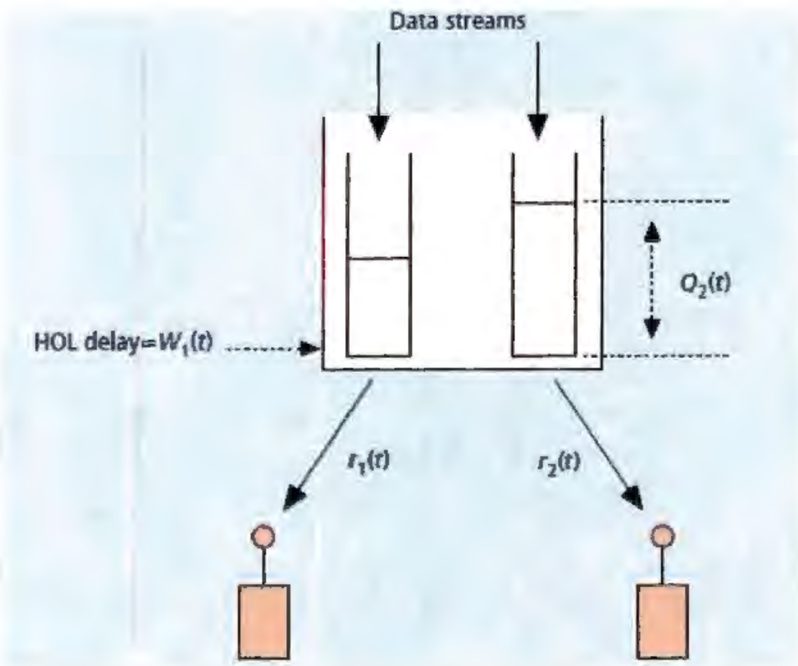

Figure 3-3. A queuing model of a cell with two mobiles [44]

The QoS of a data user can be defined in several ways. If the user is a real time user such as VoIP, the delays of most of the packets from the users' queues at the base station to the users in the cell have to be kept below a certain threshold. Specifically, the QoS requirement of a user is:

$$\Pr\{W_i > T_i\} \leq \delta_i \tag{3.1}$$

Where W_i is the head of line packet delay for a user queue i as shown in Figure 3.2 above and parameters T_i and δ_i are the delay threshold and the maximum probability of exceeding it, respectively. The MLWDF algorithm provides QoS in the form shown in Equation 3.1 as discussed below [45]. In each time slot t, it serves the user i for which Equation 3.2 below is maximal:

$$\alpha_i w_i(t) u_i(t) \tag{3.2}$$

Where $w_i(t)$ is the head of line packet delay for user i, $u_i(t)$ is the channel capacity/data rate with respect to user queue i that can be transmitted from the base station to the user. The data rate is determined through a set of channel quality feedbacks transmitted from the users to the base station via the uplink. $\alpha_i > 0$, i= 1,....,N, are weights which define the required level of QoS. α_i is an arbitrary positive constant. The main feature of this scheme is that it takes into consideration both the states of the queues and the channel conditions. The scheduler located in the base station time stamps the arriving packets of the users while keeping track of their queue lengths. A choice of the parameter α_i enables the packet control distributions to be controlled for different users. Increasing the parameter α_i for user i, while maintaining the α_is for other users reduces the packet delay for this user at the expense of delay increase for the other flows. The delay distributions can thus be shaped [42].

To guarantee the QoS of requirement of form shown in Equation 3.1, appropriate values of the parameter α_i are set. Simulation in [42] show that MLWDF scheduling, with $\alpha_i = a_i/\bar{u}_i$, $a_i =$

$-(\log\delta_i)/T_i$, and $\bar{u}_i$ being the average channel rate with respect to user i, performs well. The parameter a_i embodies the QoS requirement and provides QoS differentiation between different flows of different users. The parameter $\bar{u}_i$ is taken into consideration so that those users whose channel quality is poor probably due to being located far away from the base station or due to shadow fading are also given priority to schedule. The MLWDF rule with the above parameter setting chooses to transmit to the user i who has the maximal product:

$$i = \max\ a_i \frac{u_i(t)}{\bar{u}_i} w_i(t) \tag{3.3}$$

Hence the greater the user i current packet delay, channel quality/data rate relative to its average level and the higher the QoS requirement, the greater the chance of this user being scheduled. This algorithm thus approximately "balances" different users' probabilities of deadline violation relative to the maximum allowed values δ_i [44]. This scheme thus supports users with the desired QoS of form shown in Equation 3.1.

The scheduling principles as well as which resources are shared differ depending on the radio interface characteristics [45]. To discuss this further, in High Speed Packet Access (HSPA) it was assumed that all the radio resources assigned in the downlink were assigned to a single user at a time/in each time slot, that scheduling is done purely in the time domain using Time Division Multiplexing (TDM) between users as illustrated in the scheme shown in Equation 3.1 where each of the parameters is calculated for a specific time slot t.

However, TDM is complemented by Frequency Division Multiplexing (FDM) in the LTE Technology [45]. This is due to the use of Orthogonal Frequency Division Multiple Access (OFDMA) which is basically a multiple access technique where users are assigned mutually orthogonal subcarriers thus enabling access to the frequency domain. There are two reasons for not relying solely on the TDM in the downlink:

• In case of insufficient payload, that is the amount of data to transfer to a user is not sufficiently large to utilize the full channel capacity, and a fraction of resources could be assigned to another user, either through FDM or Code Division Multiplexing. In Time division Multiplexing, the user who is allowed to transmit is allocated the entire time interval, however if this user doesn't have sufficient data to transfer then the channel capacity/resource is wasted. The fraction of these resources can be utilised if a combination of TDM and FDM is utilised such that different users can transmit at the same time but on different frequencies.

• The benefits of channel variations (as discussed in section 2.4.1) in the frequency domain can also be exploited through FDM [45]. The larger the channel variations, the higher the chance of finding a user with better channel conditions to transmit which results in high system capacity. The use of FDM increases these variations because channel variations are not only exploited in the time domain but also in the frequency domain.

Hence the parameters defined for the scheme in equation 3.3 were redefined for the Long Term Evolution (LTE) Technology to make consideration for the Frequency Domain as illustrated below.

$$i = \max\ a_i \frac{u_i(t,k)}{\bar{u}_i} w_i(t) \qquad (3.4)$$

where $u_i(t,k)$ denotes the instantaneous data rate of user at time t on Resource Block k, with k denoting a Resource Block (RB). A RB (time-frequency resource element) as illustrated in Figure 3.4 is the resource element being assigned by the scheduler. It has resolution of 180 kHz in the frequency domain and 1ms in the time domain as illustrated in the Figure below:

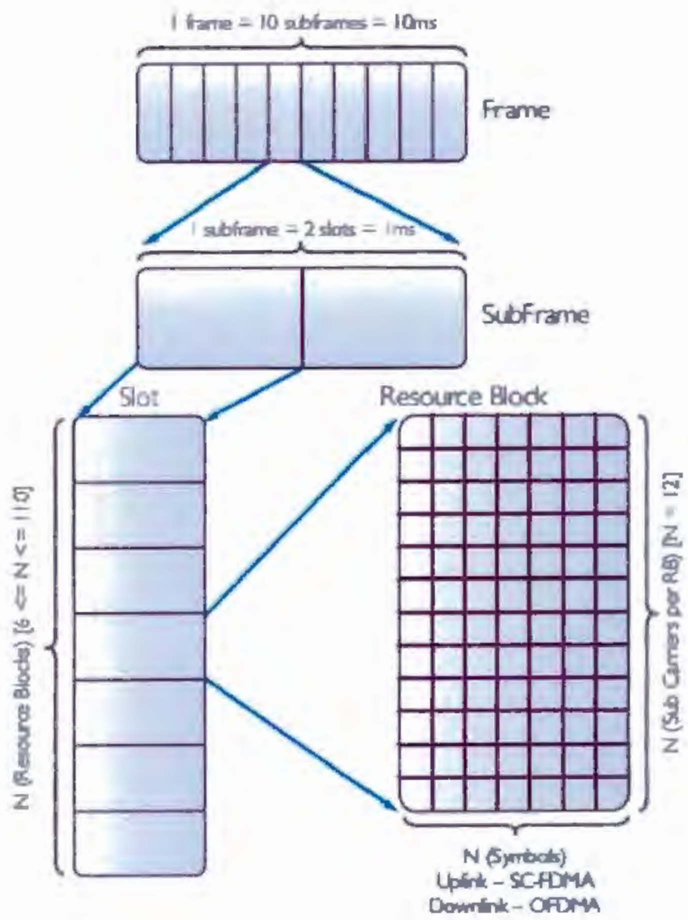

Figure 3-4. LTE Frame structure [48]

3.3 Drawbacks of the MLWDF Algorithm

MLWDF algorithm achieves a relatively low packet loss ratio (PLR) with a good throughput and fairness performance as it takes into consideration HOL packet delay together with Proportional Fair properties (e.g. the ratio of achievable data rate to the average data rate) when determining users priority [29]. However, despite the significance of the existing MLWDF algorithm it has some drawbacks. The design of the MLWDF in literature overlooks the following features. The following features minimise the packet loss for VoIP packets if taken into consideration.

3.3.1 Infinite buffer space

Firstly, infinite buffer space for the user queues in the eNodeB (base station) is assumed. It is based on the assumption that any arrived packet can be buffered and any packet loss will not occur due to buffer overflow. It does not consider packet loss caused by buffer overflow. In order to reduce the packet loss due to buffer overflow, intuitively a practical scheme should schedule those queues when their spare buffer space is small. The buffer space capacity thus has to be taken into consideration by the scheduling scheme. It is certain that buffer overflow will result in packet loss which happens when the spare space of the buffer is deficient when new packets arrive.

3.3.2 Packet length

In addition, it assumes that the length of packets is fixed which is of course not realistic. Packets of variable sizes occupy different buffer space and have different requirements in a practical sense. In a real communication system, physical resource and the efficiency of all physical resource are equally important. In order to reduce packet loss due to buffer overflow, a practical scheme should schedule those queues whose size of the HOL packet is large as the transmission of such packets means the release of more buffer space, which prevents buffer overflow when new packets arrive.

3.3.3 Packet Arrival Rate

On the other hand the traditional MLWDF algorithm doesn't make consideration for the different arrival rates of packets of different users. If the packet arrival rate of a user is high, the respective queue should be assigned higher priority for scheduling and for users with low packet arrival rates and they should be given lower priority. If the packet arrival rate is high a practical scheme should assign the corresponding queue higher priority for scheduling. The reason is that frequent arrival tends to lead to buffer overflow if the service rate cannot be guaranteed.

3.3.4 Consideration for Component Carriers in an LTE Advanced multicarrier system

The existence of Component Carriers (small bandwidths of approximately 20 MHz), which is a new feature in LTE Advanced system creates another challenge for the existing MLWDF algorithm. With the introduction of Component carriers users are no longer restricted to being scheduled on specific Resource blocks but also on specific Component Carriers. The introduction of such a new feature means that the parameters for the scheduling scheme in Equation 3.3 have to be slightly modified to enable the operation of the scheme in a multicarrier LTE Advanced system with Component Carriers.

3.3.5 Resource reservation for VoIP packets on Component Carriers (CCs)

The MLWDF algorithm is based on packet scheduling in a traditional single carrier system, which does not take into consideration the transmission on other CCs in an LTE Advanced multi-carrier system. In addition, Carrier Aggregation presents a further challenge because of the existence of Component Carriers (small bandwidths of up to 20 MHz aggregated together), it is imperative to reserve resources for the Real Time VoIP packets on the different Component Carriers depending on the number of CCs that are available so that they can meet their time deadlines. This is important so that the VoIP packets/users are protected or given priority when competing for resources with Non Real Time Packets. This is taken into consideration when setting the parameters in the design of the proposed scheme.

3.3.6 Resource allocation unfairness between LTE & LTE-Advanced

Resource allocation unfairness between LTE (Release 8) and LTE Advanced (Release 10 and beyond) terminals is a completely different issue from the drawbacks discussed above which revolve around reducing the packet loss. However, it is discussed and taken into consideration as both types of terminals will in the future coexist in the same network.

In this section, the resource allocation unfairness problem between LTE & LTE-Advanced terminals that would arise with the MLWDF algorithm is discussed. This problem arises due to the introduction of the Carrier Aggregation technique discussed in chapter 2 that comes in the most recent releases of LTE Advanced (Release 10 and beyond). A preliminary discussion is provided below on Carrier Aggregation to support this:

The International Telecommunication Union-Radio Communication Sector specified the new concept of IMT-Advanced, which aims to reach a peak data rate of 1 Gbps for low mobility and 100 Mbps for high mobility [30]. To fulfil this requirement, the 3rd Generation Partnership Project started a study item on LTE-Advanced [31] which supports the transmission over a far wider bandwidth than LTE Terminals. The bandwidth can go up to 100 MHz, which is formed by Carrier Aggregation of individual Component Carriers (CCs). Figure 3.5 shows a scenario where 3 CCs are aggregated to form a wider bandwidth required for the IMT -Advanced.

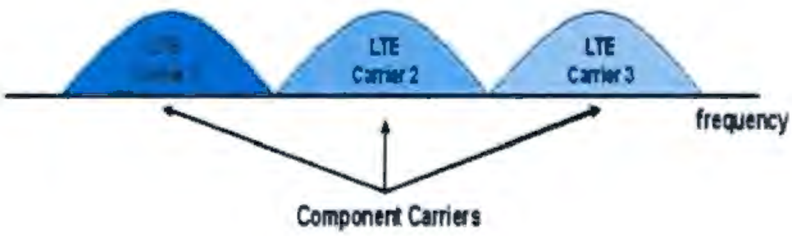

Figure 3-5. Carrier Aggregation of Multiple Continuous Component Carriers (CCs)

In a multi carrier LTE Advanced system, both LTE (Release 8) and LTE Advanced (Release 10 and beyond) terminals will coexist because of the backwards compatibility capability that comes with LTE Advanced [45]. The LTE Advanced users have the capability to be assigned on all CCs, whereas the Release 8 (LTE) terminals can transmit on only one CC. The reason for this is because the power a UE requires increases with the number of CCs it is assigned (more bandwidth and more signals to process) [38], yet the legacy LTE terminals are more power limited. Because the LTE Advanced users are scheduled on other CCs, they acquire overall N times more the resources used by Release 8 terminals, where N is the number of CCs assigned to the user. This creates a resource allocation unfairness problem between the two types of terminals. As was discussed in section 3.2, the existing MLWDF algorithm considers the average throughput in the metric to avoid starving the terminals with bad channel conditions. The issue of resource allocation is thus addressed by taking the past user throughput aggregated across all the CCs instead of just one CC. This reduces the metric for the LTE Advanced users over the LTE users. This is discussed in detail in the next section.

3.4 The Proposed Scheme

Based on the above mentioned points, an evolved MLWDF algorithm whose main goal is to reduce the packet loss for VoIP traffic is proposed. It is characterized by the following main features:

- It presumes the User Equipment's buffer capacities at the eNodeB are finite and considers buffer status when making scheduling decisions.
- Considers the packet characteristics i.e. packet size and arrival rate when making the scheduling decisions. It deals with the more practical case of where packets have different lengths.
- Provides resource reservation for VoIP packets across the Component Carriers so that they can be prioritised over non Real Time Packets.
- Guarantees resource allocation fairness between LTE and LTE Advanced Terminals.

3.4.1 Description of the proposed evolved MLWDF scheme

The system model is first introduced followed by a discussion of the operation of the proposed scheme.

System model

As shown in Figure 3.6, the downlink OFDMA based system is considered, where a single eNodeB serves I user equipments on N Component Carriers (CCs). Each user is able to receive data from multiple CCs, estimate the channel quality information (CQI) to the eNodeB. It is assumed that there are N adjacent CCs in the same frequency band taken to be aggregated in the base station for simplicity. All the CCs have the same bandwidth. The k^{th} (k=1,2,....,N) CC has d_i Resource Blocks. Therefore with CA, there are totally $d_{total} = \sum_{K=1}^{N} d_i$ RBs available for packet transmissions. The transmitted power on each RB is assumed to be the same.

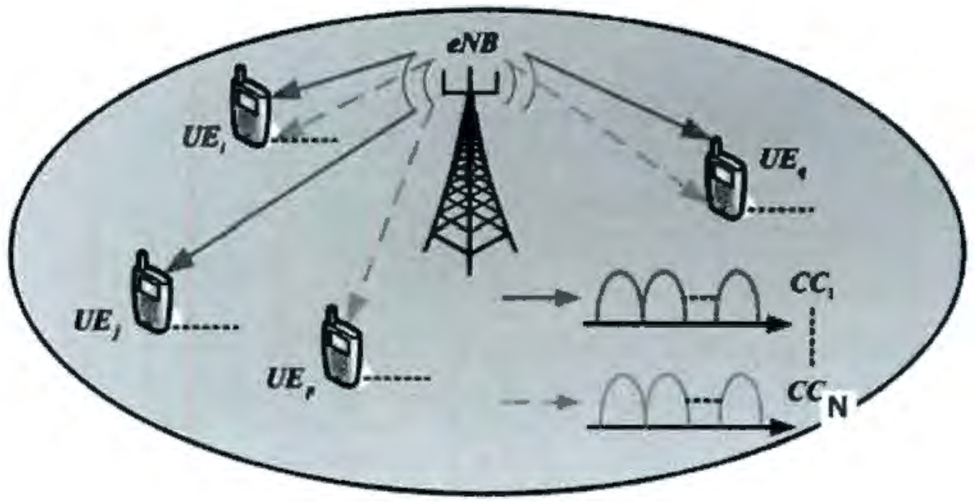

Figure 3-6. Illustration of the downlink OFDMA based LTE-A Carrier Aggregation system

S_i denotes the spare buffer space of user i at the instant of scheduling. In order to decrease the packet dropping rate due to buffer overflow, we schedule those user-queues at the eNodeB whose buffer space is small or the size of the HOL packet, p_i is large, since the transmission of such packets means the release of more buffer space, which prevents buffer flow when new packets arrive.

On the other hand if the packet arrival rate λ_i is high, the respective queue should be assigned higher priority for scheduling.

$u_{i,k,j}$ is the instantaneous throughput of a given user i on the kth CC (Component Carrier) on the jth PRB. This should be taken into account such that those queues, which have better channel conditions, should be scheduled to maximise the system throughput.

In order to guarantee some fairness, the average delivered throughput $\sum_{k=1}^{N} \bar{u}_{i,k}$ is considered. The average throughput in the past is $\bar{u}_{i,k}$ is calculated as: $\bar{u}_{i,k} = \left(1 - \frac{1}{\tau}\right) \bar{u}_{i,k} + \frac{1}{\tau} u_{i,k,j}$. Where τ is a constant for averaging the user i's data rate. The average throughput $\bar{u}_{i,k}$ is calculated over the averaging period τ. To ensure efficient usage of the short term variations which were discussed in Chapter 2 and at the same time limit the long term differences in service quality to a reasonably acceptable level , τ is the window size can be adjusted to maintain fairness.

N is the total number of CCs. The average throughput is considered to avoid starving the terminals with bad channel conditions. The difference this scheme has with the existing MLWDF algorithm which was shown in equation 3.3 is that it takes the past user throughput aggregated across all the CCs. The throughput within each CC is known by the eNodeB, therefore there is just one extra sum operation for the scheduling in all CCs. The LTE Advanced users thus have a reduced metric because their overall average throughput on all the CCs is higher than the throughput per CC, on the other hand the LTE users maintain their scheduling metric, because their transmission and reception is restricted to only one CC, they are therefore prioritised as compared with the LTE Advanced users in resource allocation which meets the objective of improving resource allocation fairness between LTE and LTE Advanced terminals.

The parameters discussed in the scheme above are summarised in Table 3.2 below:

symbol	meaning
s_i	Spare buffer space of user i at the eNodeB
λ_i	The arrival rate of user queue i
p_i	The packet size of user I (measured in bits)
$u_{i,k,j}$	The instantaneous data rate of user i on CC k on RB j (measured in bits/second)
$\bar{u}_{i,k}$	The average data rate of user i on CC k (measured in bits/second)
τ	Constant for averaging a user's average data rate
N	The total number of Resource Blocks
d_{NRT}	The number of resource blocks on which the Non Real Time Packets can be scheduled.
d_{total}	The total number of resource blocks available for scheduling
L	The Length of buffer of a specific user i at the eNodeB (measured in bits)
C	The amount of data (measured in bits) at the user's buffer in the eNodeB

Table 3-2. Parameters used in the proposed scheme

The packet scheduler tries to balance the weighted delays of different packets and to make use of the information concerning the channel state efficiently. At time slot t on RB k, it chooses the user i for transmission who maximises the following metric:

$$i = \arg\max \frac{a_i u_{i,k,j} \lambda_i p_i w_i(t)}{(\sum_{k=1}^{N} \bar{u}_{i,k}) s_i} \quad (3.4)$$

The rest of the parameters are the same as those defined for the scheme in Equation 3.3.

On the Component Carriers (CCs), the ratio of d_{NRT} the number of RBs that Non Real Time users can share to the total number of resource blocks d_{total} , is set to be a specified level α : that is to say $\alpha = \frac{d_{NRT}}{d_{total}}$. Where $d_{NRT} \in \{1,2\ldots,d_{total}\}$. The selection of those RBs that is to say the d_{NRT} RBs begins from the RB in sequence on the CC with index k=1. To be more precise, these Non Real Time packets can only be transmitted over αd_{total} RBs and the VoIP packets can be transmitted over d_{total} RBs. For Non Real Time packets they will be scheduled on only αd_{total} while all the VoIP packets can utilise d_{total} RBs. As such the VoIP packets are protected when the system is overloaded with NRT packets. t_{th} is defined as a period for delivering NRT packets in the transmission queue.

It is known that buffer overflow will cause packet loss. Those users whose spare buffer space at the eNodeB is small will be scheduled with higher priority to decrease the packet dropping. In the proposed scheme we integrate the buffer status into account when making the scheduling decision. The Buffer Status Report (BSR) in the downlink provides the packet scheduler with UE buffer data that includes the terminal buffer length L at the eNodeB and the amount of current received data C. When the data traffic process is established C will be initialized to 0. If C reaches the full buffer length it's flushed to 0 again. The BSR factor s_i which is considered in the scheduling decision (metric) is described as:

$$s_i = \frac{L - C}{L} \qquad (3.5)$$

The sequence of operation of the scheduling scheme is summarised in the following steps:

Step 1 Deliver Non Real Time (NRT) packets which are in the previous runs (if any) to the transmission queue, if the epoch of the current scheduling run is an integer multiple of t_{th} . Note that there are αd_{total} RBs available for the NRT packets that are delivered.

Step 2 Deliver VoIP packets which are in the buffer of the Real Time (RT) queue in the previous run (if any) to the transmission queue.

Step 3 Begin from the unused RB on the CC with the index k=1 to schedule packets in the transmission queue.

Step 4 Calculate the fairness vector, which is denoted as tuple (i*,k*j*), which is illustrated in equation 3.4 , and is expressed as:

$$\text{tuple } (i^*, k^*, j^*) = \arg\max \frac{a_i u_{i,k,j} \lambda_i p_i w_i(t)}{(\sum_{k=1}^{N} \bar{u}_{i,k}) s_i}$$

Step 5 Transmit the packet to user i* on CC k* on Resource Block j* in the current scheduling run.

Step 6 Update the average data rate for each and every user i according to :

$$\bar{u}_{i,k=} \left(1 - \frac{1}{\tau}\right) \bar{u}_{i,k} + \frac{1}{\tau} u_{i,k,j}.$$

Step 7 Check if there are still some resources that can be allocated in this run. If so go back to Step 3; otherwise go back to step 1 and begin the next run

Chapter summary

In this chapter, a preliminary discussion is provided on VoIP traffic (which is the focus of this study) and the key performance metrics taken into consideration when measuring the performance of VoIP traffic in high speed networks. Packet loss is selected as the main metric of evaluation and the criteria for selecting it discussed. Packet scheduling is discussed as one of the major Radio Resource management techniques to meet high performance for VoIP in high speed networks with a focus on LTE networks. The reason behind dynamic channel dependent scheduling in the downlink LTE system is also illustrated with an illustration of how high system capacity is achieved. The existing MLWDF algorithm designed for HSPA and LTE is introduced with a detailed discussion on its key features. Its shortcomings when applied to a multicarrier LTE Advanced system are highlighted. Based on these shortcomings an evolved MLWDF algorithm which is an improvement to the MLWDF algorithm is proposed and the system environment for it illustrated. It considers critical parameters which aim to minimise packet loss for VoIP traffic. The new scheme is also designed to offer resource allocation fairness between the LTE Release 8 terminals and the high end LTE Advanced (Release 10 and beyond) terminals as both terminals will in the future coexist in the same network.

Chapter 4 Simulation environment and Performance metrics

This chapter provides an introduction to the simulator used to perform simulation and the criteria used to adopt the existing simulator. A description of the traffic models and the important basic simulation parameters used in the simulation are also provided. Specific modules/blocks of the simulator are described while highlighting the modifications that were made to enable the experiments.

The simulations were run on a simulator called LTE-Sim [37]. It encompasses several aspects of LTE networks which include the Evolved Universal Terrestrial Radio Access (E-UTRAN) and the Evolved Packet System. It offers support for single and multi-cell environments, QoS management, multiuser environments, user mobility, handover procedures and frequency reuse techniques. In order to ensure modularity, polymorphism, flexibility and high performance, LTE-Sim is written in C++, using the object-oriented paradigm as an event-driven simulator [37].

LTE-Sim was selected as the platform for this study because it considers relevant aspects of LTE simulations such as realistic applications and a complete protocol stack with both uplink and downlink flows.

A new buffer status reporting module was added to the simulator as will be discussed later in succeeding sections and a new scheme (evolved Modified Weighted Delay First Algorithm) is implemented in the simulator as the main objective of this study. The proposed scheme that we implemented in the simulator is characterised by the following features:

1) Presumes that the User Equipment buffer capacity is finite at the eNodeB and takes into consideration buffer status/space, packet arrival rates and sizes when making a scheduling decision.

2) Prioritises VoIP packets when scheduling users across the Component Carriers.

3) Provides resource allocation fairness between the legacy LTE (Release 8) and the much recent LTE Release 10 and beyond terminals. The most common channel and QoS aware scheduling algorithms for LTE that is it say Max Rate algorithm, Weighted Proportional Fair Algorithm and the Queue Based Maximum Rate algorithm were added by extending the already existing

Proportional Fair Algorithm in the simulator.

In the next section, descriptions of the main components of the simulator that were utilised and description of the interaction between them is provided. An overview of how the schedulers (that were extended) compute their metric is also provided. This set of schedulers were utilised in a performance evaluation carried out in this study to determine the most suitable algorithm supporting VoIP traffic. The validation for the selection of the performance metrics is also discussed.

4.1 Description of the Simulator

In the diagram below in Fig 4.1, the interaction between the main components of the simulator is illustrated. The interaction between the eNodeB, buffer status reporting module, and the User Equipment is shown.

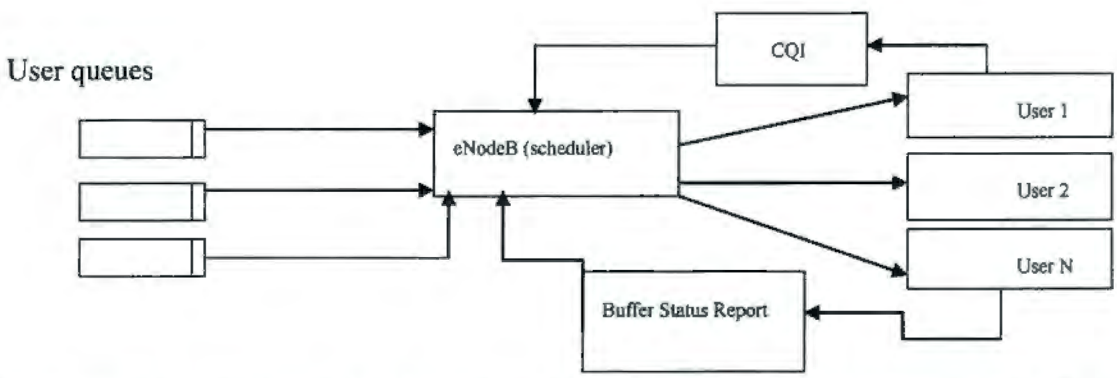

Figure 4-1. Diagram illustrating the interaction between the main components of the simulator

In the simulator three network nodes are modelled as shown in Figure 4.1 above: User Equipment (UE), evolved Node (eNB), and Mobility Management Entity/Gateway (MME/GW). The MME does not interact with the scheduling entity and is therefore not included in Figure 4.1. Each of the specified network nodes can be a source or destination of data flows described by the classical five tuple: sender and receiver ports, transport protocol type and source and destination IP addresses. Each network node is identified by a unique ID and its position in a Cartesian system is also defined. Each of the network nodes is described below. The eNodeB and User

Equipment modules are hereby described as they are the main entities that interact with the scheduler hence being the focus of this research study. However, a more detailed description is provided for the eNodeB as it contains the packet scheduling entity.

4.2 The eNodeB module

The eNodeB module handles Radio Resource Management for the evolved LTE radio access. It contains the MAC entity where both downlink and uplink scheduling strategies are defined. The LTE system uses the concept of channel dependent scheduling where radio resources are allocated to users while taking channel conditions into account. It takes advantage of the independent nature of fast fading between different users. When there is a large number of users, it is very likely to find one with either good or relatively good channel conditions at a specific time. Scheduling is related to the channel quality experienced by a specific user. Before the eNodeB performs scheduling, it has to acquire information regarding the channel quality of each user through a Channel Quality Indicator (CQI) sent to the eNodeB as illustrated above.

The eNodeB module in the simulator contains an implementation of a basic downlink scheduler and the Proportional Fair scheduler both of which contain an implementation of methods common to all scheduling strategies. The strategy behind the Proportional Fair scheduler is discussed in Appendix A.

In this study, the implementation of the already existing Proportional Fair Algorithm in the simulator was extended to include the Queue Based Maximum Rate Algorithm, Weighted Proportional Fair Algorithm (WPF) and Maximum Rate algorithm by creating new scheduling classes and extending the already existing ones. The above algorithms were used in an experiment to determine the most suitable algorithm for VoIP traffic. At the beginning of each sub-frame, the scheduler selects all flows that can be scheduled. A flow can be scheduled if and only if it has data packets to transmit at the MAC layer and the receiver UE is not in the idle state. At every TTI, the scheduler computes a given metric for each flow which can be scheduled. We will refer to $P_{i,k,j}$ as the metric assigned to the i-th flows for the k-th CC on the j-th RB. The scheduling algorithms differ in the way the metric is calculated. An overview is provided below on how each implemented scheduler computes its metric.

4.2.1 Max Rate Algorithm

This scheme schedules the users with the best instantaneous radio link condition and is often termed as max-CIR (or maximum rate) scheduling [33]. The radio link conditions for each channel or user vary independently of each other such that at any one time there always exists a user or radio link whose channel quality is near its peak. The scheduling metric $P_{i,k,j}$ for a specific user i is calculated based on the equation below:

$$P_{i,k,j} = \max R_{i,k,j} \tag{4.1}$$

The user who maximises the above metric is scheduled for transmission. Where $R_{i,k,j}$ is the instantaneous throughput of user i on the k-th CC on Resource Block j.

4.2.2 Queue Based Max CIR (QBMC)

This algorithm is a modification to the classic Max rate algorithm, by taking into consideration the information concerning the queue size of each user's transmission buffer [35]. Using this information, the scheduler gives preference to queues that are not being served due to its channel conditions. It uses the information about the delay of users indirectly. The priority calculation (metric) of a user is as below:

$$P_{i,k,j} = c_{i,k,j} s_i \tag{4.2}$$

Where $c_{i,k,j}$ is the measured Carrier to Interference Ratio of user i on CC k on resource block j and s_i is the buffer size in number of bits not yet transmitted. The user who maximises the metric in equation 4.2 is scheduled for transmission on a specific resource block.

4.2.3 The Weighted Proportional Fair (WPF)

This scheme behaves much the same way as the Proportional Fair scheduler which takes into consideration the ratio of the instantaneous data rate over the average data rate. They are distinguished by a fixed multiplicative weight w_{s_i} which is a QoS differentiation factor to be considered for a service class S_i of user i, as illustrated below.

$$p_{i,k,j} = w_{s_i} \cdot \frac{R_{i,k,j}}{T_{i,k}} \qquad (4.3)$$

Where $R_{i,k,j}$ is the instantaneous throughput of user i the kth CC on resource block j

Note: An overview of the implementation of the new proposed scheme is demonstrated and discussed in Chapter 3.

4.3 The User Equipment module

The User Equipment module performs Channel Quality Indicator (CQI) reporting to the eNodeB. With this channel state information, the eNodeB can implement adaptive modulation and coding to maximise the throughput on each carrier. Furthermore, this throughput information is passed on to the scheduler .The simulator offers support for CQI reporting. The channel quality is estimated by the User Equipment (UE) and changed into a set of CQI feedbacks which are reported to the eNodeB. The system provides a downlink signal of a predetermined structure, known as the pilot or reference signal which is used to estimate the downlink channel conditions. It is based on this channel quality that the packet scheduler makes a decision on which user to transmit to and is therefore an important scheduling factor. Both periodical and aperiodical CQI reporting are supported by the simulator. The simulator also offers support for full band and wide band reporting modes. The arrived packets are stored in the buffers of the UE which are taken to have varying lengths.

4.3.1 Buffer Status Reporting Module

A buffer status reporting module was added to the simulator. Buffer overflow results in packet loss, which decreases Voice quality. The UE queue buffer status at the eNodeB is taken into consideration as the scheduling decision is being made. The buffer status reporting message contains information concerning the terminal's buffer length at the eNodeB and the amount of received data. In the proposed scheme, the UE buffer status is considered when making a scheduling policy as it is computed as part of the scheduling metric to avoid packet loss extremely. Intuitively the users whose queues at the eNodeB have smaller spare buffer space will be scheduled with high priority to decrease the packet dropping/loss. The Buffer Status Report

(BSR) feedback in the downlink provides the packet scheduler with UE buffer data that includes the terminal's queue buffer length, L and the amount of current received data C. When the data traffic process is established C will be initialized to 0. If C reaches the full buffer length it's flushed to 0 again. The BSR factor Si which is a parameter in the computation of the metric for the new scheme introduced in this study is described as:

$$s_i = \frac{L - C}{L} \qquad (4.4)$$

4.3.2 User Mobility module

The simulator adopts a system level inter-cell handover. It offers support for two types of mobility models: Random Walk and Random Direction. With the Random Direction Model, the User Equipment randomly chooses the speed direction that is changed when it approaches the simulation boundary. With the Radom Walk model, the User Equipment alters its speed direction after covering a distance that depends on the user speed. The travel distances are 200,400 and 1000 m with users speeds equal to 3, 30, and 120 km/h, respectively. A Random direction model is used in this study as discussed in the succeeding section.

4.3.3 Flow management and the protocol stack

The EPS bearer as illustrated in figure 4.2 was introduced to offer QoS differentiation. The bearer maps a flow into a logical channel that is established between the UE and GW. The EPS bearer can either be a dedicated or a default bearer. The default bearer is created when the UE is connected to the network providing connectivity and exchange of control messages. It is maintained during the whole life time of the connection. The dedicated bearer however is established when a new service session is established. A dedicated bearer provides a dedicated tunnel to one or more specific traffic. A dedicated bearer can also be described as Guaranteed Bit Rate (GBR) or non GBR. A GBR bearer has a guaranteed bit rate and MBR (Maximum Bit Rate) while more than one non-GBR bearer belonging to the same UE shares an AMBR (Aggregate Maximum Bit Rate). Non-GBR bearers can suffer packet loss under congestion while GBR bearers are immune to such losses. Consequently these bearers are generally subject to admission control within the network. The current version of the simulator offers support for only dedicated bearers.

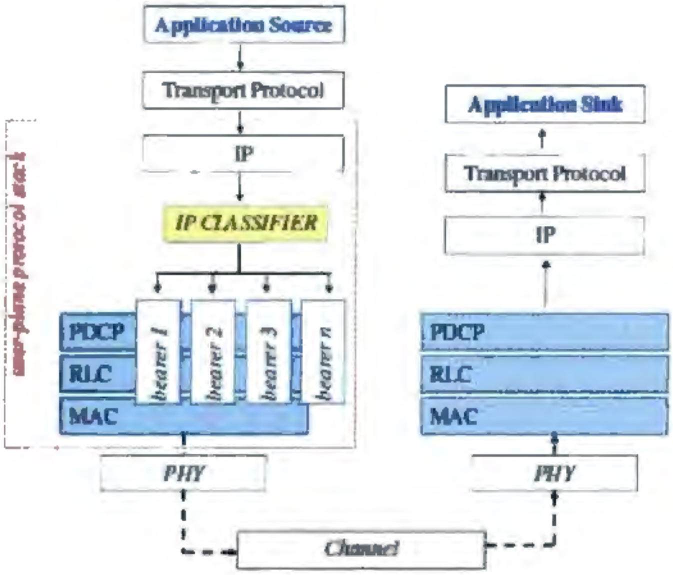

Figure 4-2. Protocol Stack implementation [37]

Functionalities of both user and control plane LTE protocol stack are implemented in the simulator. The RRC entity handles downlink and uplink dedicated radio bearers. It works with the classifier to classify a packet into a proper radio bearer. The header compression of packets from upper layers is handled by the PDCP entity. The unacknowledged data transmission at the RLC layer is modelled by the RLC entity. The MAC entity offers an interface between the UE / eNodeB and the Physical layer for the delivery of packets from the upper layers. An illustration of the interaction between the above entities is shown in Figure 4.2 above.

4.3.4 Channel Structure

The radio access uses Orthogonal Frequency Division Multiple Access providing support for flexible bandwidth from 1.4 MHz to 20 MHz. The simulator offers support for six channel bandwidths (i.e. 1.4, 3, 5, 10, 15, and 20 MHz).

The channel module of the simulator handles the transmission of packets and models the propagation loss through four different phenomena: (i) path loss, (ii) the penetration loss, (iii) shadowing and (iv) fast fading due to the signal multipath.

4.3.5 Frame Structure

The simulator supports two frame structures one is defined for FDD mode and the other is defined for TDD mode and are referred to as frame structure type 1 and frame structure type 2 respectively. For the latter, the frame is split into two consecutive half-frames lasting 5ms as illustrated below:

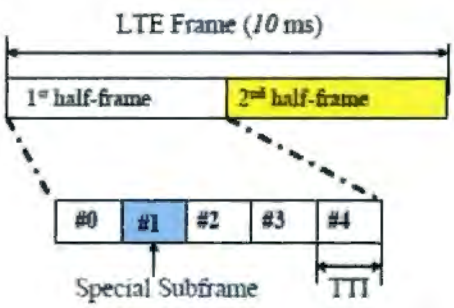

Figure 4-3. Frame type 2 for TDD mode [37]

4.3.6 Physical layer

Because of the high computational effort involved, the simulation of a complete physical layer is not suited for complex network scenarios which include the whole protocol stack. An analytical model approach is hence utilized in the system level simulation. The physical implementation of the physical layer in the computation of the SINR was hereby modified to enable Carrier Aggregation. The main function of the simulation platform physical layer is the calculation of the SINR which is obtained from the reference signals. The Physical layer has an interface with the channel module to know the state of the channel links. Both the SINR of reference signals and data packets sent by the eNodeB are calculated.

- When the i-th UE receives , the following steps are executed:

  The physical layer computes for each sub channel the SINR for the received signal considering the received power, the noise and the interference, as it follows:

$$SINR_{i,k,j} = \frac{P_{RX,i,k,j}}{(FN_oB) + I}$$

  Where the parameters k and j are the same as those defined in Section 4.2. F, N_o, B and I are the noise figure (default value 2.5), the noise spectral density (default value -174dBm), the bandwidth of the RB (180 KHz) and the interference respectively.
- The UE creates CQI feedbacks to send to the eNodeB.
- The physical layer determines that packets have been received correctly.

4.4 Simulation Parameters

Before describing the simulation set up, the characteristics of a Physical Resource Block (PRB) are described. The LTE system takes advantage of time and frequency resources. The time domain is split into 1ms Transmission Time Intervals, this comprises two slots each of duration 0.5 ms thus forming 1ms sub frames with each sub frame containing 7 OFDMA symbols. Each Transmission Time Interval contains 14 OFDMA symbols with 2 symbols reserved for uplink pilot transmission while the other 12 symbols are reserved for data and control information transmission. The minimum allocation unit in the time domain is referred to as a Transmission Time Interval (TTI) which has duration of 1 ms. In the frequency domain the minimum allocation unit is a Resource Block (RB) which constitutes 12 subcarriers of 15 kHz each. The cyclic prefix utilised determines the number of OFDMA symbols.

In this study, the type of cell used is a single urban macro cell which consists of a radius of 1 km, free from inter-cell interference. In the cell, a constant speed of 3 km/hr in random directions is maintained and this speed remains constant until the UE moves towards the boundary of the cell area. The UE choose a new speed direction when it approaches the end of the cell. Path loss, multi-path fading and shadow fading are used to form the LTE propagation loss model [39], and this is summarised as shown below [40].

- Path loss :128.1 +37.6log10(d), where d is distance between the UE and the eNodeB in Km
- Shadow fading: log-normal distribution with a mean value and standard deviation of 0 dB and 8 dB respectively
- Multipath: Jakes model and Penetration loss:10dB

Table 4-1. Simulation Parameters

Parameters	Value
Carrier Frequency	2 GHz
Bandwidth	10 MHz
Number of Subcarriers	600
Number of Resource Blocks	50
TTI Duration	1 ms
UE Speed	3 km/hr
Slot Duration	0.5 ms
No. of OFDM symbols /slot	7
Subcarrier Spacing	15 kHz
Frame structure	FDD

Assumptions

It is imperative to state that some assumptions were made in this study in order to reduce the complexity of the system level simulation. Power was assumed to be equally allocated among the subcarriers and these subcarriers are used for traffic transmission. Instantaneous SINR values on each RB are reported to the eNodeB at each TTI without error and delay [41]. Two adjacent CCs in the 2 GHz frequency band are assumed to be aggregated each with a bandwidth of 5 MHz. Considering the signal processing complexity and the power saving at the UE, the number of the CCs that a UE has to connect with is selected to be as low as possible. The value for averaging the users data rate is set to $\tau = 1s$.

4.5 Traffic Model

The VoIP traffic model is modelled using an ON/OFF Markov model. The ON/OFF periods are exponentially distributed with a mean value of 3 seconds and the off period has a truncated exponential pdf (probability density function) with an upper limit of 6.9 seconds, and an average value of 3 seconds. During the ON period the source sends 20 byte sized packets every 20 ms (i.e. the source data is 8.4 kbps) while during the off period the rate is 0 because of the presence of a voice activity detector [37].

4.6 Performance Metrics

The following metrics are measured in order to evaluate several aspects of the system performance under the operation of different schedulers. The main aim of our scheduling scheme is to reduce the packet loss. However, other metrics were considered for the performance evaluation and the criterion for selecting them is hereby discussed:

4.6.1 Packet Loss

Is a measure of the percentage of packets of a certain traffic class dropped in the transmission packet queue due to exceeding their packet delay budget. The packet loss metric was selected as one of the evaluation metrics because the main aim of the proposed scheme is to reduce the packet loss which has a direct impact on the voice quality.

4.6.2 Packet Delay

The delay per packet is measured from the time the packet enters the eNodeB user queue till the time it is successfully received by the UE. Packet delay is measured by collecting the delay stamps for all packets being sent within the entire simulation time, then determining the experienced average delay of all UEs. The packet delay measurements, along with the measurements from the packet loss, give an indication of the ability of a scheduling algorithm to adhere to the QoS requirements of VoIP traffic as was discussed in Chapter 3.

4.6.3 Fairness

Is a measure of the fairness among UEs within the same class of traffic. The measurement of the fairness is performed using the well-known Jain's Index, which is calculated as follows:

$$\text{Fairness} = \frac{(\sum x_i)^2}{(n \sum x_i)^2} \tag{4.4}$$

Where n is the number of users and x_i is the throughput of the i_{th} connection. Although Packet loss was chosen as the main metric for evaluation, it is imperative to consider other metrics because a suitable scheme should be able to maximise a specific metric while offering minimal degradation on other metrics that measure the LTE downlink performance [45].

4.6.4 System throughput

The system throughput or aggregate throughput is described as the sum of the data rates that are delivered to all terminals in a network. The scheme that this study proposes aims to design a channel dependant type of scheduler. Channel dependent schedulers experience the "multiuser diversity gain" discussed in Chapter 2 which involves the increase in channel capacity/throughput arising due to an increase in the number of users/offered load and the increase in channel variations as was discussed in detail in Chapter 2. System throughput is thus taken as a metric to determine this phenomenon.

Chapter Summary

In this section, an introduction to the simulator that was used for the experiments was provided. A description of the simulator is provided while highlighting the modifications that were made to suit the purpose of our experiment. The assumptions made during the study are also highlighted. The VoIP traffic model adopted is also discussed and the simulation parameters provided. The given performance metrics selected for the evaluation are also provided. Packet loss is selected as the main metric for evaluation. Other metrics are also discussed to analyse the performance of the selected algorithms under different conditions.

Chapter 5 Performance Analysis

In this section, the results obtained from the experiments that were conducted on the representative schedulers using the simulator discussed in Chapter 4 are presented. A description of strategies behind these algorithms is provided in appendix A. The experiments conducted on the LTE downlink schedulers were designed to look at the following aspects of the system performance under different scenarios. The metrics used in all the scenarios and the validation for choosing them are described in Chapter 4.

- The first experiment evaluates whether the studied packet scheduling algorithms are capable of guaranteeing the QoS of VoIP traffic measured in terms of average delay in a scenario where VoIP and web browsing services are competing for resources. The first part of the evaluation concerns a service mix where the VoIP traffic dominates in number of users (75% VoIP/25% Web). The other results are composed of both services in equal proportions (50% VoIP/50%Web). The last scenario is comprised of a domination of the web service (25% VoIP/75% Web).
- The second experiment evaluates the effect of representative scheduling algorithms on VoIP traffic under different performance metrics. The main aim of this experiment is to determine the most suitable algorithm supporting VoIP traffic in a multicarrier LTE Advanced network.
- The third experiment compares the performance of the proposed scheme (evolved MLWDF) against the MLWDF Algorithm using the packet loss as a main metric of evaluation.

5.1 Network performance for (75% VoIP + 25% Web):

The graph in Figure 5.1 displays the average delay for VoIP users and Figure 5.2 displays the average delay for Web-users under different scheduling policies. The strategy behind these schedulers is discussed in Appendix A.

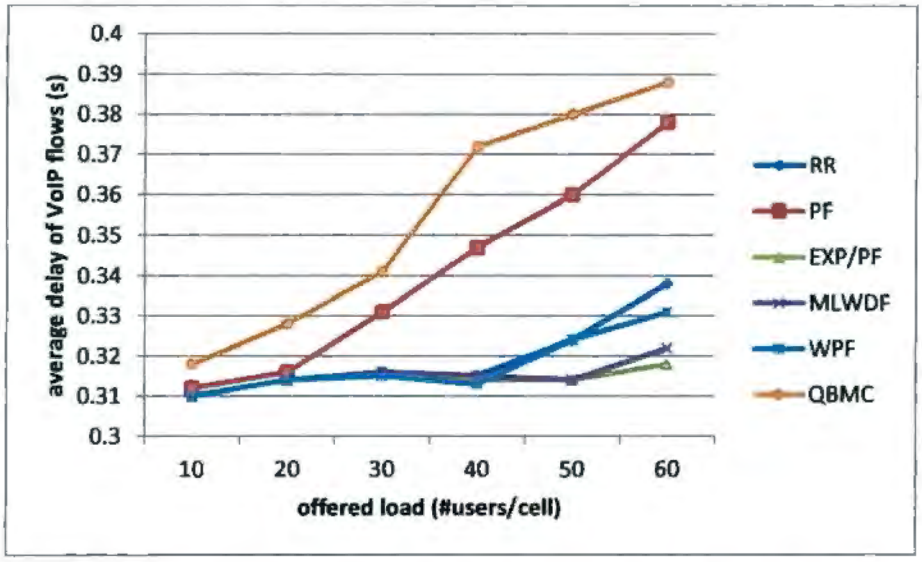

Figure 5-1. Average delay for voice users (voice 75%, 25% Web)

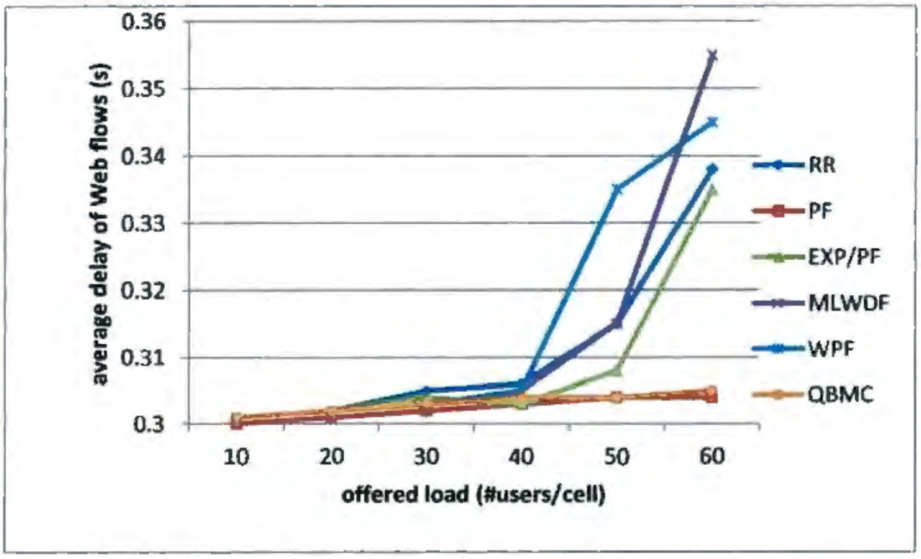

Figure 5-2. Average delay for Web users (75% Voice, 25% Web)

Figures 5.1 and 5.2 show the network performance for VoIP and Web users, respectively. There is a significant increase in the average delay of the QBMC algorithm for VoIP traffic as the number of users increases. This occurs majorly due to web browsing users having high buffer sizes, resulting in them having a higher priority according to the priority function described in

Appendix A. Both The EXP-PF and MLWDF have the same average delay except at 60 users and beyond. The WPF and RR schemes present a slightly poorer performance (than the above-mentioned algorithms) described by high average delay especially as the number of users increases, but much better than the Proportional Fair Algorithm. This occurs due to the fact that the WPF and RR schemes do not take into consideration the delay requirements of different packets in their metrics. As regards the Web service, The PF has the least delay and is followed by the QBMC algorithm. The higher weight given to VoIP traffic considered by the WPF causes a bad performance for Web users. The EXP-PF provides a better performance than the MLWDF with a distinctively higher delay as the number of users increases beyond 40. It also presents slightly higher delays in comparison to the PF and QBMC under high loads. In summary , the algorithms that provide better average delays for one traffic type, have worse performance regarding the other due to service prioritisation (EXP-PF,MLWDF,WPF, and QBMC) or due to resource distribution (RR and PF).

5.2 Network performance for (50% VoIP + 50% Web)

The graph in Figure 5.3 displays the average delay for VoIP users and Figure 5.4 displays the average delay for Web users.

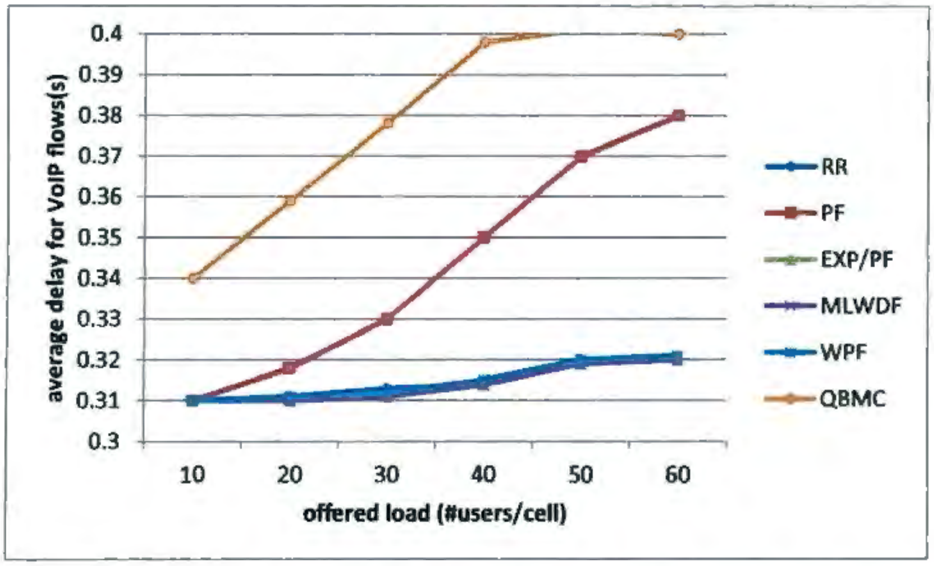

Figure 5-3. Average delay for VoIP users (50%VoIP, 50% Web)

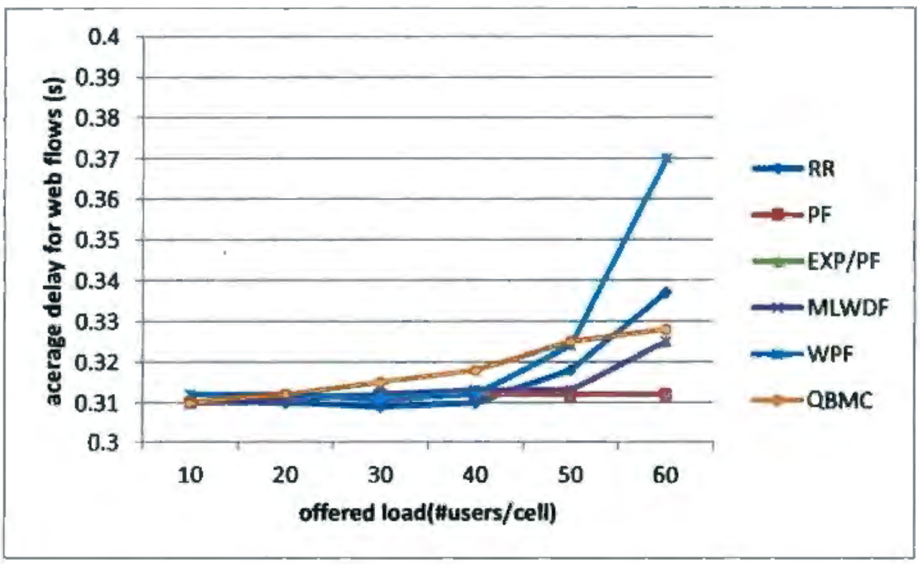

Figure 5-4. Average delay for web users (50% VoIP, 50% Web)

The performance of VoIP users is shown in Figure 5.3. It is evident that QBMC has the highest average delays, followed by the PF algorithm. The relative behaviour of both algorithms remains the same as the previous mix (see Figure 5.1) but the absolute performance is worse. On the other hand, the RR and WPF show a considerable improvement in the average delay experienced. Both the EXP-PF and MLWDF produced similar good results with considerably low average delay even as the number of users increases this arises due to the fact that the two schemes are designed to give high priority to packets when their delay thresholds are violated as the queues become loaded with more users.

For the Web users, the average delay is presented in Figure 5.4. The superior performance of the PF is not as evident as that obtained with a smaller proportion of the Web traffic in the offered load. Both the EXP- PF and MLWDF provide close to similar performance with the same average delay at all loads. The WPF and RR provide quite the same performance, with a difference in performance for overload conditions.

It is clear that, except for the case of QBMC and WPF, there is an observable improvement in average delay of the rest of the schemes in the performance of the Web service compared to the previous traffic mix (see Fig 5.4). This can be seen by the fact these algorithms provide a service prioritization to VoIP. Since there is a less amount of VoIP users, there is more resource left to the web service.

5.3 Network performance for (25% VoIP +75% Web)

The graph in Figure 5.5 shows the average delay for VoIP users and Figure 5.6 displays the average delay for Web users.

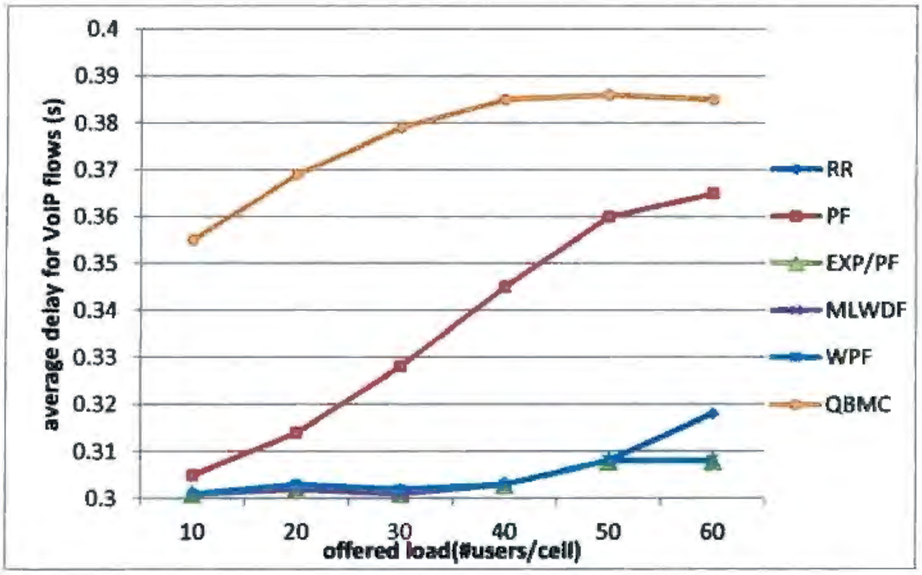

Figure 5-5. Average delay for VoIP users (25%VoIP, 75%Web)

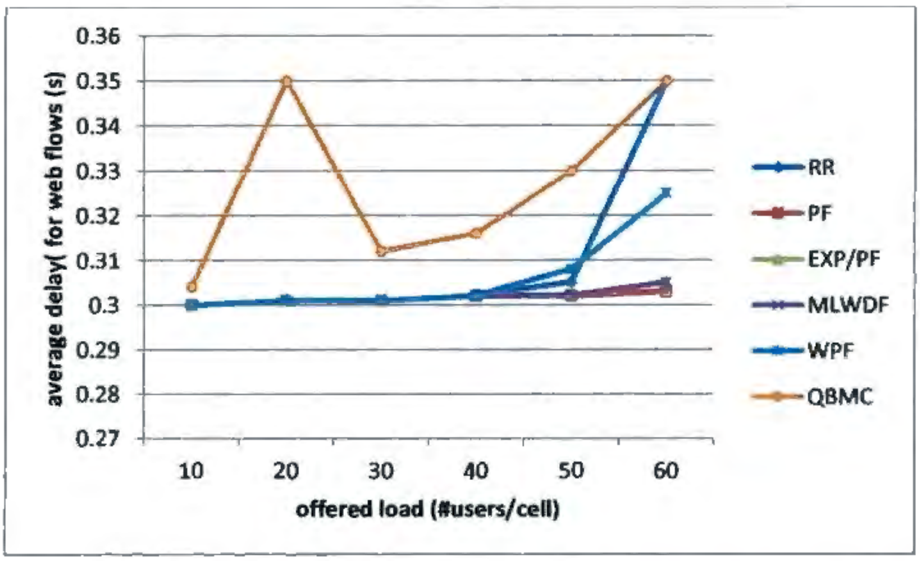

Figure 5-6. Average delay for web users (25% VoIP, 75% Web)

The performance of VoIP users is shown in Figure 5.5. The performance is almost similar compared to the instance with equal number of Web and VOIP users (see Figure 5.5). The EXP-PF, MLWDF and WPF show similar average delays while the QBMC is incapable of assuring a good QoS (measured in terms of average delay) during low and high load conditions. This arises due to the fact that it gives priority only to users with high instantaneous throughput and buffer size i.e. it doesn't consider the delay requirements of different users as their delay thresholds are violated. At the highest simulated load, the RR has a high average delay for Web traffic compared to the rest. For VoIP traffic, the PF shows a distinctively higher delay than the other algorithms which progressively increases as the number of users increase. However it has lower average delay than the QBMC at all load conditions. The QBMC algorithm provides a much higher delay compared to the previous traffic mix. One peculiar issue as regards this scenario is with The WPF. It has the capability to provide a reasonable performance for both services. It can be seen that, since there is a less amount of high priority users in the system, the scheduler provides a resource reservation to them and provides a good QOS measured in terms of the average delay to the other service (Web).

5.4 Performance evaluation of Packet Schedulers for VoIP traffic in a multicarrier system

The next set of experiments investigates the performance of well known packet scheduling algorithms for VoIP service to identify the most suitable algorithm supporting VoIP traffic. The most suitable algorithm should be able to offer low packet loss and packet delay for users. A suitable scheduler should however be able to maximise or minimise one metric while offering minimal degradation on the other metrics [45]. Two other metrics system throughput and fairness are considered.

(a) System Throughput performance

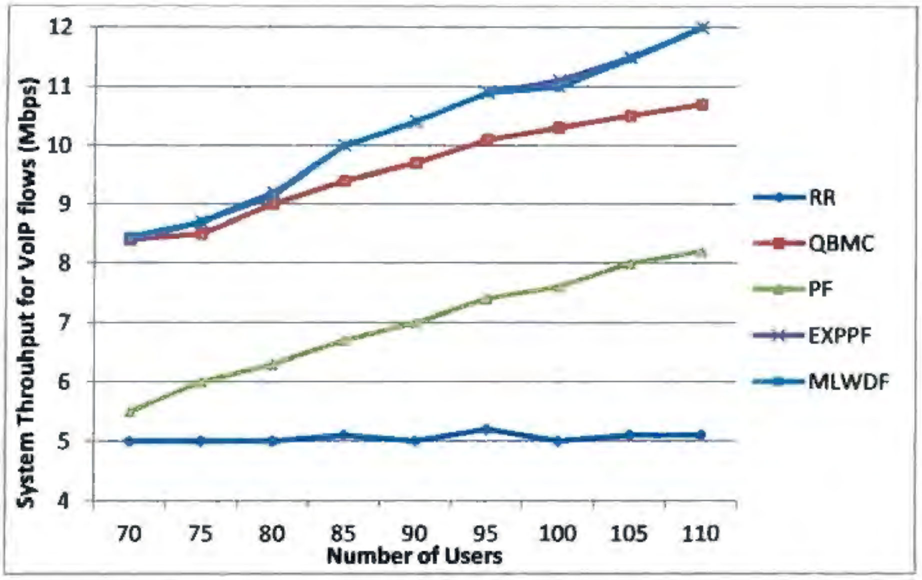

Figure 5-7. System throughput versus number of users

Figure 5.7 illustrates the system throughput of the given scheduling schemes. MLWDF and EXP/PF give better performance than QBMC, PF and RR algorithms as regards the system throughput. This is because both algorithms ensure that the packets whose delay deadlines are about to be violated are given higher priority thus minimising packet loss and resulting in high system throughput. The RR scheme produces the least throughput considering the fact that it doesn't take into account channel conditions when making scheduling decisions. In addition, a fixed data rate being assigned to all users in the simulation limits the throughput from increasing beyond the saturation level as the number of users increases. As channel conditions are taken into consideration when priotising users in MLWDF and EXP/PF, this enables the utilisation of the multiuser diversity gain. Multiuser diversity is achieved by user scheduling in multiuser channels where user scheduling permits the base station to choose high quality channel users so as to transmit information through a relatively high quality channel in time, frequency and space based on the channel quality feedback from all candidate UEs. This arises due to the fact that different users are located in different places leading to different channel gains for the same channels for different users. The higher the number of users, the higher the chance of finding users with good channels. It is therefore seen that the system throughput of these algorithms increases a great deal as the number of users increase.

Note: The graph was truncated at 110 users.

(b) Packet loss ratio performance

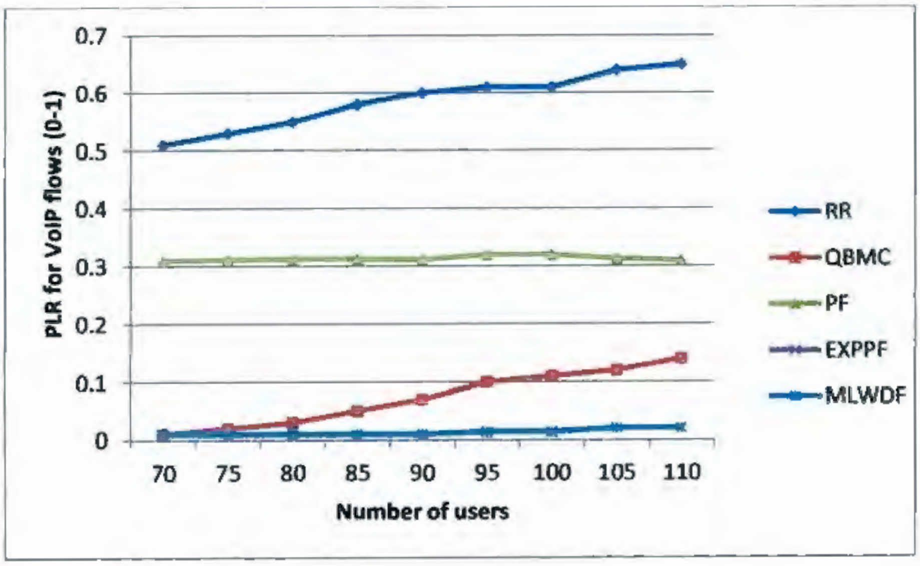

Figure 5-8. Packet Loss ratio versus number of users

With the VoIP service, it is imperative to keep the PLR below 1% such that the QoS requirement of VoIP users is satisfied [45]. In Figure 5.8, it can be observed that less than 70 users can be offered support by the QBMC, PF and RR scheduling schemes at the target PLR threshold with QBMC algorithm having better PLR performance than PF and RR algorithms. Both MLWDF and EXP-PF support a higher number of users at 1% PLR threshold. These same algorithms produce a significantly better performance in comparison to QBMC, PF and RR algorithms as they take into account packet delay information when making scheduling decisions. It is further seen that MLWDF and EXP-PF compete with each other in meeting the QoS requirements of VoIP users. As seen from the results approximately 96 and 100 users can be supported by both schemes at the target PLR threshold. As there is a slight difference between the number of users that can be supported by MLWDF and EXP-PF, the decision about which algorithm can support VoIP users in the downlink is inconclusive at this stage we therefore evaluated the performance with other metrics. A suitable algorithm should be able to maximize one metric while offering minimal degradation on the others.

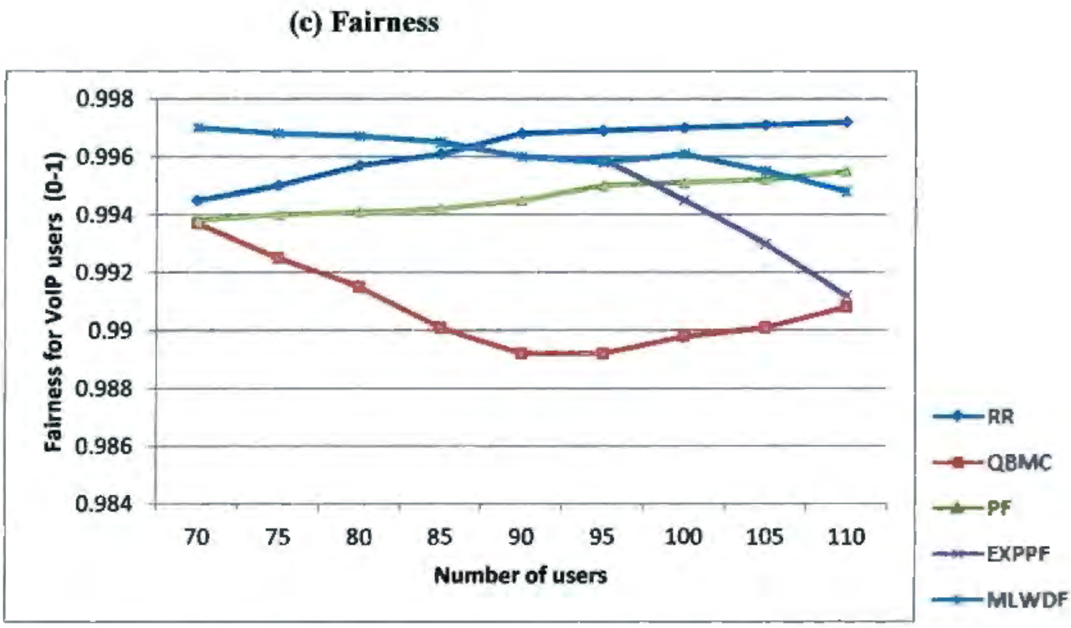

Figure 5-9. Fairness versus number of users

The fairness (which is defined in chapter 4) of each packet scheduling scheme is illustrated in Figure 5.9. Generally, the RR algorithm has the best fairness performance since the algorithm provisions equal share of packet transmission time to each user together with a fixed data rate assigned to each user within the system.

In comparison to the RR algorithm the fairness performance of MLWDF falls slightly lower than RR at regions with high number of users and better than RR at regions with low number of VoIP users. This is because at low number of users the MLWDF favours those packets with time constraints however as the number of users increases considerably the proportional fair properties of the MLWDF become more significant. This is a good indication that MLWDF does satisfy the fairness requirement of VoIP users at a sufficient level. The fairness results of EXP/PF decline a great deal with increasing number of users. The EXP-PF has a poorer fairness performance compared with MLWDF at high number of users.

5.5 Performance of the proposed scheme

In this last experiment, the performance of the proposed scheme (evolved MLWDF) is evaluated against the MLWDF algorithm. The MLWDF is selected as the baseline scheme for evaluation as it is the scheme that this study aims to enhance in terms of reducing the packet loss. The performance of our scheme is evaluated against the MLWDF algorithm only as it already proved to show lower packet loss ratio than other common real time schedulers.

(a) Packet loss ratio performance

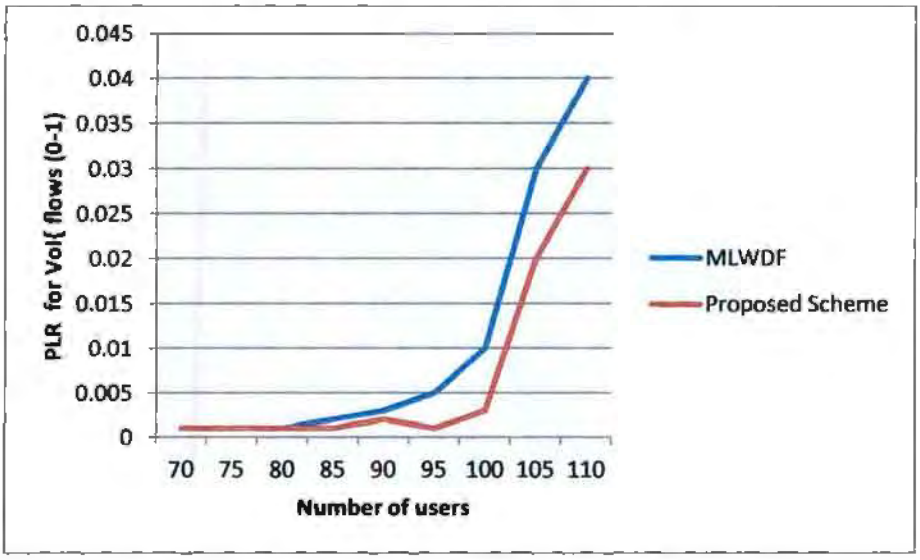

Figure 5-10. Packet loss ratio versus number of users

As seen from the graph in Figure 5.10, both schemes perform on a low packet loss value and have low packet loss for a low number of users in the network. This is because the queues are loaded with a moderately low number of packets from the different users and most of the packets can afford to meet their deadlines. As the number of users increases both schemes offer higher packet loss as the queues become loaded with more packets and an increasingly higher number of packets are lost due to buffer overflow and packet dropping. The MLWDF reaches the PLR threshold when the offered load is at 75 users with the proposed scheme reaching the 1% threshold at approximately 102 users thus supporting a higher number of users at the packet loss threshold. As the number of users increase beyond 100 users there is a drastic increase in the packet loss with our proposed scheme showing a relatively lower packet loss ratio than the MLWDF algorithm. This is because the proposed scheme takes into consideration arrival rates, packet size and queue size to minimize packet loss while at the same time reserving resources on the carriers for VoIP packets.

(b) System throughput performance

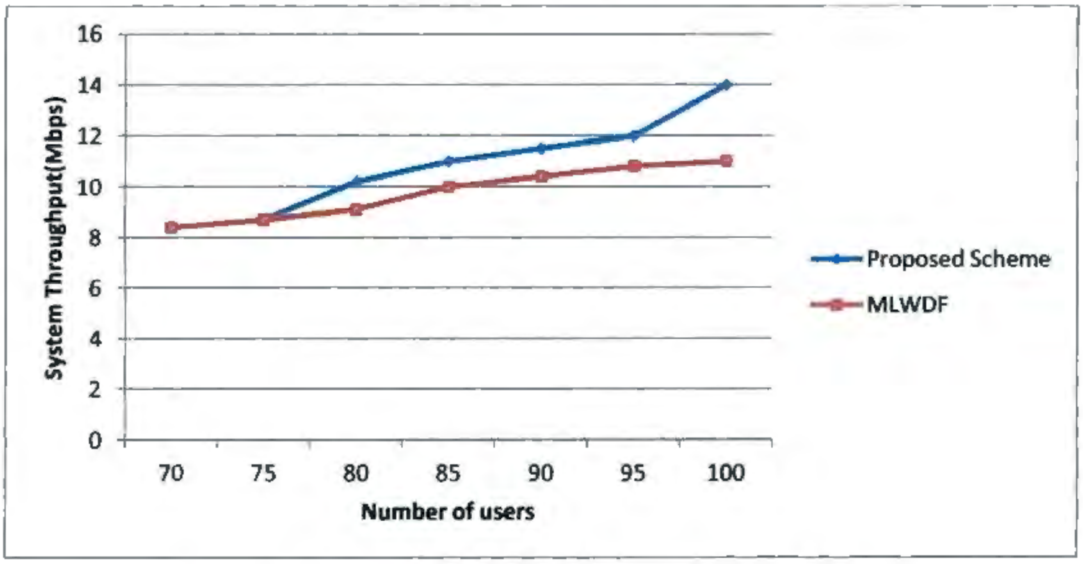

Figure 5-11. System throughput versus number of users

As seen from Figure 5.11, it is observed that as the number of users increases, the throughput of the system increases, this arises due to the multiuser diversity gain where by the higher the number of users being scheduled the higher the probability of obtaining a user with very good channel conditions and therefore high throughput. Multi-user diversity takes advantage rather than compensate for fading. As the load increases, our proposed scheme reflects a higher system throughput than the MLWDF algorithm. The significant difference in system throughput arises due to the ability of our scheme to limit the rate of buffer overflow when the queues start to get saturated with many packets.

Chapter summary

In this chapter, three experiments were performed; the first experiment was performed to establish whether a selected set of scheduling algorithms are capable of guaranteeing QoS for VoIP users in a scenario where VoIP and Web users are competing for the same resources. This was carried out by considering three scenarios where the service mixes were varied. The second experiment was carried out to determine the most suitable algorithm supporting VoIP traffic in a multicarrier LTE Advanced system. The final experiment compared the packet loss performance of the proposed evolved MLWDF algorithm against the MLWDF where the proposed scheme is shown to display low packet loss when the network load increases.

Chapter 6 Conclusions

The success of LTE deployment has caused a great change in the creation of more advanced mobile hardware and the upspring of faster internet services that are produced over mobile networks for example mobile browsing, video streaming and interactive gaming.

The great increase in the mobile traffic load will impact on the ability of LTE Advanced to accommodate forecasted volumes at a certain point.

LTE Advanced is looked at as a great step towards the change and evolution of currently used 3G technologies. 3GPP brought forward LTE to promote a competitive future mobile market where mobile broadband offers a platform to install mobile networks with high data rates at considerable costs. LTE Advanced promises performance levels that exceed its predecessor LTE in terms of high throughput, lower latency, more simplified architecture and easier deployment.

Packet Scheduling in LTE Advanced deals with determining which users to transmit to and on what set of Resource Blocks. The scheduling decision is repeated every 1ms which is known as a TTI. An LTE packet scheduler should optimally predict the needs of UEs with different requirements, such that a UE can satisfy its QoS requirement while utilizing the resources assigned to it as efficiently as possible.

The main focus of this study was to develop an evolved MLWDF algorithm. It's main objective is to reduce the packet loss for VoIP users when utilised in a multicarrier LTE Advanced system. The scheme introduced showed a low packet loss ratio performance than the MLWDF algorithm especially as the offered load increased. The low packet loss ratio of the proposed scheme is attributed to the utilisation of specific parameters in the computation of its metric.

Initially however, a performance evaluation was carried out to determine whether a specified set of given scheduling algorithms can provide QoS for VoIP service in a situation where the VoIP and Web users are competing for the same resources and also to perform a performance evaluation on various schedulers in literature to determine the most efficient scheduler for VoIP service.

The performance analysis on the LTE schedulers was conducted on a system-level simulator that adheres to the LTE standards. The performance metrics used to measure the performance levels of the schedulers were average delay, throughput, PLR and Fairness.

In conclusion it can be said that generally, packet scheduling algorithms that take into account delay requirements of the VoIP service (EXP /PF & MLWDF) presented the best overall capacity for all traffic mixes.

MLWDF algorithm was originally proposed to improve the performance for real-time services by taking HOL packet delay into account. This study mainly focused on the problem of packet scheduling for VoIP traffic. We introduced a new scheme with low complexity designed for VoIP traffic which is made flexible to adapt to the characteristics of flows from different VoIP users. The proposed evolved M-LWDF packet scheduling algorithm provides lower packet loss rate because it takes the buffer status into account for VoIP traffic to avoid packet loss. Two performance metrics namely packet loss throughput and fairness were selected to better weigh the performance of this new scheme.

LTE Advanced and its predecessor LTE are breakthrough technologies as regards the older generation of cellular networks as they use an all IP Architecture whose target is to offer support for multiple high quality services e.g. VoIP, online gaming and video streaming. The targets of LTE Advanced are rather ambitious as we discussed in Chapter 1 and can only be reached through the implementation of a series of procedures at the Physical and MAC layers. The Packet Scheduler is the entity that performs the major Radio Resource Management task by considering physical constraints and traffic flow requirements. To achieve the best spectral efficiency, the solution is to distribute resource blocks to the User Equipment that will use it best hence maximising the cell capacity. Other metrics such as fairness, cell edge coverage, QoS provisioning can be maximised at the cost of reducing the cell capacity. With this in mind, the design of an allocation strategy often is based on realising a good trade off among the system spectral efficiency and the targets that the network operator wants to achieve. Whatever algorithm that is implemented should have a low computational cost and should adhere to the fact that the transmission time interval in LTE Advanced is set to 1ms.

6.1 Recommendations and Future Work

As LTE Advanced involves more Resource Blocks and frequency carriers, it requires faster processing capacity for the different components and for the simulation program to model the scheduling task. There is therefore need to perform research experiments with this in mind.

User mode transition: as a result of the existence of new types of users and different modes, the scheduling task should now be able to handle several input attributes of each user and have the capability to handle the transitions of users to different modes. Take for example a user who is active during its connection may change from normal mode to COMP mode and may be later to relaying mode. The eNodeB should have the capability and intelligence to optimize the performance of this user and the entire system for all instances.

As most schemes are designed based on the existing proportional fair algorithm, there is a need to design a new algorithm that is optimized for LTE Advanced which includes all its new features and can be able to make consideration for the user and system requirements. In all the literature works it appears that each new feature is looked at separately.

With the introduction of the need for strong QoS support, the solutions offered in previous literature appear to be unsuitable for handling bounded delays and minimum data requirements. QoS aware solutions appear to be promising as they have the ability to describe flows' requirements and to meet the desired performance targets.

To conclude, the packet scheduling scheme proposed in this research item is suitable not only for LTE Advanced technology but also for any technology using multiple Component Carriers such as Wireless MAN-Advanced, the new 4G technology from WiMAX family.

Glossary

2G	*The second generation mobile phone technology.*
3G	*The Third Generation mobile phone technology.*
3GPP	*Collaboration between telecommunication associations also known as the organizational partners.*
Bandwidth	*It's is the rate of data transfer, bit rate / throughput*
Bit	*It's the basic unit of information*
BER	*The number of bits divided by the total number of bits transmitted*
BTS	*It's the GSM equipment which enables the communication between UE and the network*
CDMA	*Code Division Multiple Access. It's a multiple access technique where a transmission to different users is achieved by assigning users different codes.*
Cell	*Area of Coverage under a base station.*
COMP	*Coordinated Multipoint Transmission. A technique in LTE Advanced where a UE can receiver or send transmissions to several base stations simultaneously.*
EDGE	*Enhanced Data Rates for GSM evolution. The Last stage of evolution of GSM.*
eNodeB	*The evolved Base station in LTE Technology.*
EPC	*Evolved Packet Core. The main component of SAE.*
FDMA	*Frequency Division Multiple Access. Multiple Access technique where*

	users are separated based on frequency.
GBR	*Guaranteed Bit Rate. Used to distinguish between Real Time and Non real Time Services.*
GPRS	*General packet Radio Service. It's a standard of GSM, offering speeds of 115Kbps*
GSM	*Global System for Mobile Communications Mobile Phone Technology developed to operate on the 850MHz and 900MHz band.*
HARQ	*Hybrid Automatic repeat Request.*
HOL	*Head of Line Packet delay. Time interval between the current time and time if arrival of a packet*
HSPA	*High Speed Packet Access. The predecessor technology of LTE*
IMT-Advanced	*International Mobile Telecommunication Advanced/4 G*
LTE	*Long Term Evolution. A mobile telephone technology that is a migration of the GSM/ EDGE/ HSPA technologies*
LTE Advanced/LTE-A	*The Same technology as LTE with advanced label included to highlight the relationship between LTE-A and IMT Advanced*
MLWDF	*Modified Largest Weighted Delay First. Algorithm that's designed to offer priority for real time services*
NRT	*Non Real Time.*
PS	*Packet Scheduler. An element in the base station responsible for determining the way resources (time/ frequency/power are assigned to different users)*
QoS	*Quality of Service. A broad term to describe the attributes of an end to*

	end connection
RB	*Resource Block. A time frequency resource grid. The smallest resource element allocated by the scheduler to the UE*
RN	*Relay Node. Network element used to extend coverage in LTE*
TDD	*Time Division Multiple Access. A technique where the uplink and downlink transmissions are separated in terms of time slots*
UTRAN	*Universal Terrestrial Radio Access Network. It consists of Node B, RNC and air inter interface*
WiMAX	*Worldwide Interoperability for Microwave Access*

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Appendix A Description of the Algorithms used in the Performance evaluation

In today's current networks we have both VoIP and other types of traffic competing for the same resources. The first experiment evaluates whether a specific set of the most common packet schedulers for LTE are capable of guaranteeing QoS for VoIP traffic in a scenario where Web and VoIP traffic are competing for the same resources. Specifically, delay is a major source of voice quality degradation as the network grows and the load from different traffic on a network increases and is chosen as a metric for this evaluation. The analysis we carried out consists of three parts. The first part of the evaluation concerns a service mix where the VoIP dominates in number of users, the second part comprises a scenario composed of both services in equal proportions and the last part is composed web service. The scheduling algorithms selected for this evaluation are discussed in this section.

In this second experiment was carried out to determine the most suitable algorithm supporting VoIP traffic for LTE Advanced. The schedulers selected for this evaluation are also discussed below. The listed schedulers were selected for the performance evaluation as they are the most common real time schedulers for VoIP traffic which perform QoS differentiation. The most suitable algorithm should be able to offer low packet loss for users and packet delay. A suitable scheduler should however be able to maximise or minimise one metric while offering minimal degradation on the other metrics. Therefore two other metrics i.e. system throughput and fairness which monitor the LTE downlink performance were considered.

A.1 Round Robin Algorithm

This scheduling scheme allows the users to use the common resources in different turns, one at a time. At the start, the users are randomly selected in a queue. The new terminals are placed at the end of the queue. The first user is given all the available resources by the scheduling entity, and then placed at the back of the queue .The Round Robin scheme can be taken as fair scheduler because the same amount of time or resources is given to each specific user. The unfairness that results out of this scheme arises due to offering the same service quality to all communication links users [6]. Some terminals may be experiencing poor channel quality at a specific time and thus require more time.

A.2 Max Rate Algorithm

This scheme schedules the users with the best instantaneous radio link condition and is often termed as max-CIR (or maximum rate) scheduling [33].The radio link conditions for each channel or user vary independent of each other such that at any one time there always exists a user or radio link whose channel quality is near its peak. The priority function for a specific user i is calculated based on the equation below:

$$P_I = R_I \max \qquad (A.1)$$

Although, from a system capacity perspective, max-C/I scheduling is beneficial, this scheduling principle will not be fair in all situations. If all terminals are, on average, facing similar channel conditions and large variations in the instantaneous channel conditions are only due to, for example, fast multi-path fading, all users will achieve the same average data rate. Any changes in the instantaneous data rate are rapid and often not even seen by the user. However, in a real life scenario, different terminals will also experience differences in the (short-term) average channel conditions – for example, due to differences in the distance and shadow fading between the base station and the user equipments. In this case, the channel conditions experienced by one terminal may for a relatively long time be considerably worse than the channel conditions faced by other terminals. A pure max-C/I-scheduling strategy may then, in essence, "starve" the terminal with the bad channel conditions, and the terminal with bad channel conditions will never be scheduled [33].

A.3 The Proportional Fair (PF) Algorithm

The PF Algorithm is a compromise between the Round Robin and Max Rate algorithm [34]. It serves users with the highest instantaneous throughput relative to their average throughput. The priority value for a user i is determined based on the following priority function:

$$p_i = \frac{R_i}{T_i} \max \qquad (A.2)$$

The philosophy of the above priority equation is proposed by Kelly et al in [32] to balance between maximising the system throughput and maintaining fairness among all users. Basically with this strategy the resources are assigned to the user with the relatively based channel conditions. Moreover it was pointed out in [34] that the Proportional Fair Scheme is attractive for nonreal time traffics, since it can achieve substantially much larger system throughput than other representative packet scheduling schemes such as the Round Robin scheme. Also the PF scheme can provide the same level of fairness as the Round Robin Scheme in the average sense provided in [34].

A.4 Queue Based Max CIR (QBMC)

This algorithm is a modification to the classic Max rate/CIR (Carrier To Interference Ratio) algorithm, by taking into consideration the information concerning the queue size Si of each user's transmission buffer [35]. Using this information, the scheduler gives preference to queues that are not being served due to its channel conditions. It uses the information about the delay of users indirectly. The priority calculation of user is as below:

$$p_i = c_i s_i \qquad (A.3)$$

Where C_i is the measured Carrier To Interference Ratio and S_i is the buffer size in number of bits not yet transmitted

A.5 The Weighted Proportional Fair (WPF)

This scheme behaves much the same way as the Proportional Fair Scheduler. They are distinguished by a fixed multiplicative which is a QoS differentiation factor to be considered for a service class Si of user i, as can be seen below

$$p_i = w_{s_i} \cdot \frac{R_i}{T_i} \qquad (A.4)$$

This is simple way to establish a priority hierarchy between different service classes. The higher the priority of the service class the higher the weight value.

A.6 Modified Largest Weighted Delay First

This scheme is discussed in Chapter 3.

A.7 The Exponential Proportional Fair Algorithm (EXP-PF)

Exponential Proportional Fair is an algorithm that was developed to support multimedia applications in an adaptive modulation and coding and time division multiplexing (ACM/TDM) system, this means that a user can belong to a real-time service (RT) or non-real-time service (NRT). This algorithm has been designed to increase the priority of real-time flows with respect to non-real-time ones. At timeslot t, the EXP rule chooses the user j for transmission as follows:

$$j = \max\ a_i \frac{u_i(t,k)}{\overline{u_i}} \exp \frac{a_i w_{i(t)} - \overline{aw}}{1 + \sqrt{\overline{aw}}} \qquad \text{(A.5)}$$

Where all the corresponding parameters are the same as in the M-LWDF rule, except the term $\overline{aw}$ defined as

$$\overline{aw} = \frac{1}{N} \sum_i a_i\, w_i(t) \qquad \text{(A.6)}$$

When the HOL packet delays for all the users do not differ a lot, the exponential term is close to 1 and the EXP rule performs as the proportionally fair rule. If for one of the users the HOL delay becomes very large, the exponential term overrides the channel state-related term and the user gets a priority. In [36] it is proved that the exponential proportional fair algorithm is throughput optimal in that renders queues at the base station stable.

Appendix B Images of the simulation Tool

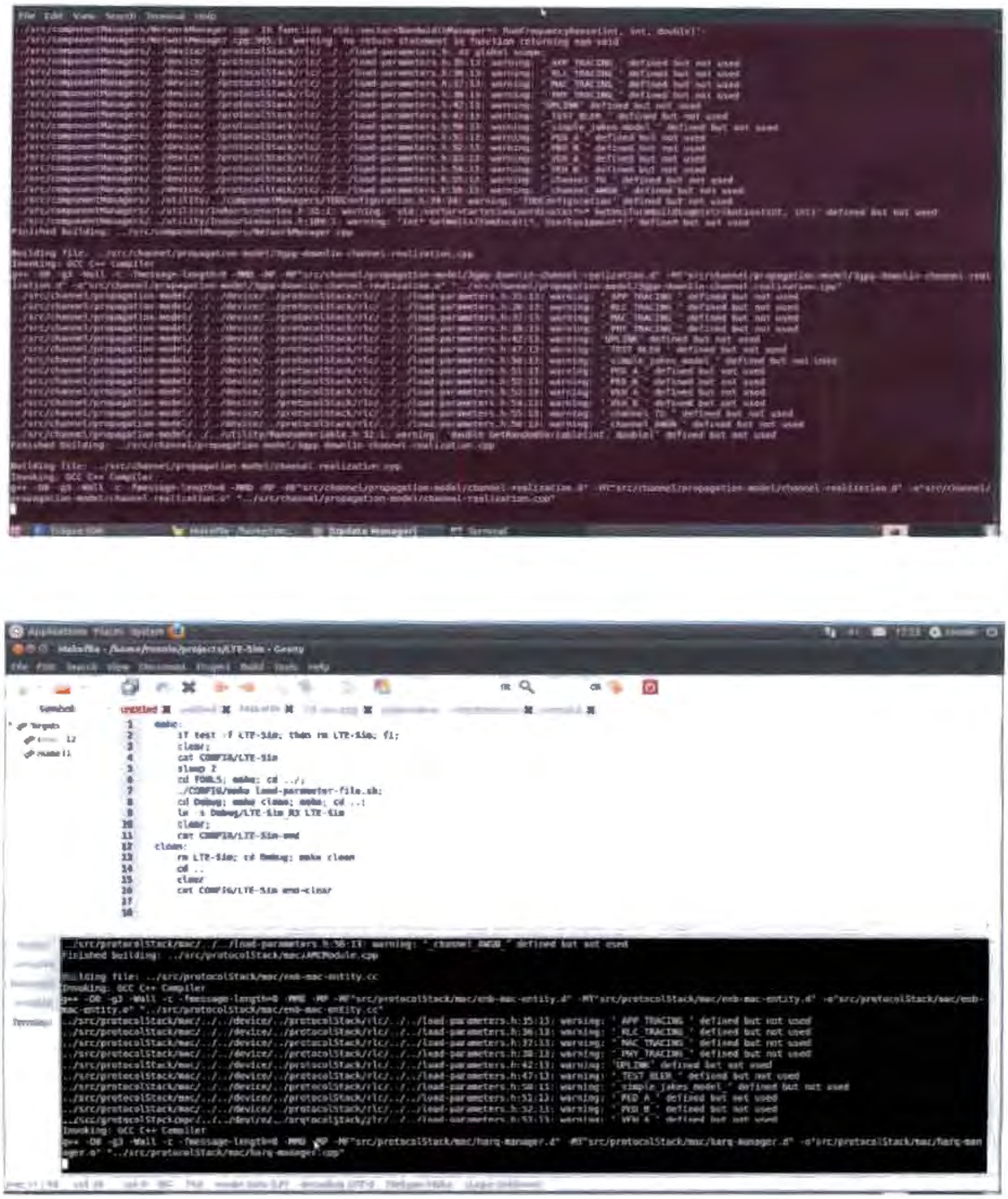

Fig B.1 A screen shot of the LTE-Simulator when being built

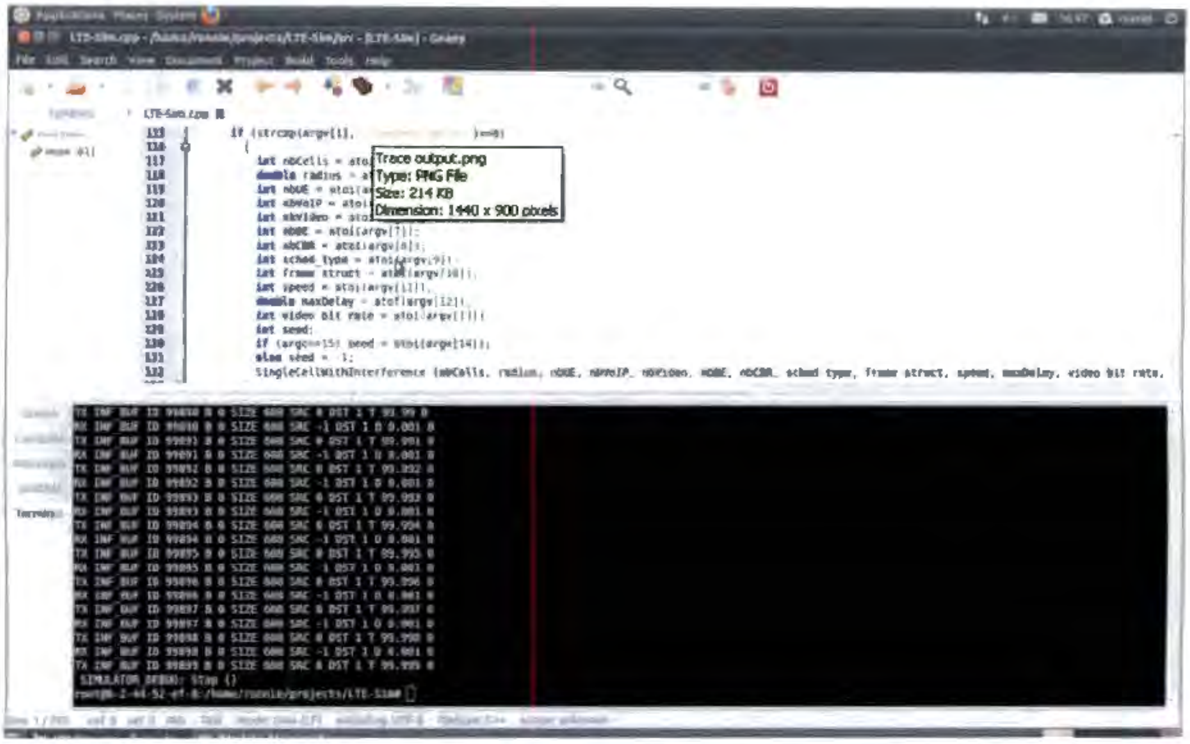

Fig B.2 Screen shot of the output trace in LTE-Sim

Fig A.2 shows an output trace that resulting after the execution of the simulation. The first field illustrates the event which has triggered the tracing. More specifically the rows Tx, Rx and DROP show the sent received and dropped. Other fields include:

ID=Identifier. Identifies each packet uniquely.

B= Bearer ID. Specifies the bearer used to map the packet

SRC=Source ID. Identifies the node that sent the packet.

DST=Destination ID. Specifies the node that receives the packet

T = Time. It represents the instant in which the packet is created.

D = Delay. It represents the delay of the received packet

Appendix D Accompany of CD-ROM

The CD-ROM included in this thesis contains the following files and information:

Research Literature: Electronic copies of relevant papers and other literature used during the research can be found in the directory labelled research literature can be found in a directory labelled "research literature".

Software: All software code used for the simulations can be found in a directory labelled "software"

Publications: Copies of papers which have been accepted to conferences can be found in a directory labelled "Publications".

Thesis: An electronic copy in pdf can be found in a folder "Thesis".

Results: The results obtained during the simulations can be found in the directory "Results".

Appendix C Hardware and Software Specifications

Hardware	
Operating system	Linux Ubuntu 11.04
System Model	Emerald 956G
Model Name	Intel (R) Core TM 2 CPU , 1.86GHz
Memory	0.98GB RAM
Software	
Name	LTE - Sim
Version	Version 2.0