

## إقرار

أنا الموقع أدناه مقدم الرسالة التي تحمل العنوان:

### **A Study of Packet Scheduling Schemes for VoIP and Best Effort Traffic in LTE Networks**

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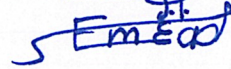
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**A Study of Packet Scheduling Schemes for VoIP and  
Best Effort Traffic in LTE Networks**

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## نتيجة الحكم على أطروحة ماجستير

بناءً على موافقة شئون البحث العلمي والدراسات العليا بالجامعة الإسلامية بغزة على تشكيل لجنة الحكم على أطروحة الباحث/ عماد منير رمضان صالح لنيل درجة الماجستير في كلية الهندسة قسم الهندسة الكهربائية - أنظمة الاتصالات وموضوعها:

دراسة لأنظمة جدولة حزم نقل الصوت عبر بروتوكول الانترنت مع أفضل جهد نقل في شبكات التطور على المدى البعيد

### A Study of Packet Scheduling Schemes for VoIP and Best Effort Traffic in LTE Networks

وبعد المناقشة العلنية التي تمت اليوم الاثنين 21 جمادى الآخر 1435هـ، الموافق 2014/04/21م الساعة الثانية عشرة ظهراً بمبنى القدس، اجتمعت لجنة الحكم على الأطروحة والمكونة من:

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واللجنة إذ تمنحه هذه الدرجة فإنها توصيه بتقوى الله ولزوم طاعته وأن يسخر علمه في خدمة دينه ووطنه.

والله ولي التوفيق ،،،

مساعد نائب الرئيس للبحث العلمي والدراسات العليا

أ.د. فؤاد علي العاجز





## **DEDICATION**

**I would like to dedicate this work to my beloved mother**

**To the spirit of my dear father**

**To my brothers**

**To my sisters**

**To my wife**

**To my daughter Talaa**

**To my son Muhammad**

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Praise Allah, thank you very much blessed as our Lord loves and is pleased with, Glory of our Lord does not count the praise you as you compliment yourself.

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## **ABSTRACT**

The Long Term Evolution (LTE) provides all services over Internet Protocol (IP) since it is an all IP network. To use available radio resources in an effective utilization, Packet Scheduling (PS) should be considered to enhance the Quality of Service (QoS) of Real Time (RT) and Non-Real Time (NRT) traffic.

In this thesis, the PS of both RT and NRT traffic is studied in LTE networks. A priority packet scheduling algorithm is proposed. The proposed algorithm has the ability to schedule the mixed traffic, RT and NRT, simultaneously. The objective of the algorithm is to maximize the Best Effort (BE) throughput while achieves the satisfaction QoS requirements of RT throughput. According to the obtained results of the thesis, the traffic should be differentiated and the services should be prioritized, when applying delay sensitive services.

A system simulation is performed to support the study for mixed services approaches with Voice over IP (VoIP) and a second BE service such as File Transfer Protocol (FTP). The performance of the proposed algorithm and the impact of the different factors on the overall system performance have been tested. The work is done at Medium Access Control (MAC) layer and Physical Layer (PHY). Finally, a good results are achieved that guarantee a good end to end performance for both voice and data services.

## ملخص البحث

نظام الإتصالات "تطور طويل الأجل" (LTE) يوفر جميع الخدمات عبر بروتوكول الإنترنت (IP)، لأنه شبكة تعتمد كلياً على بروتوكول الإنترنت. ولكي نستخدم الموارد المتاحة استخداماً فعالاً، يجب أن نطبق نظام جدولة الحزم (PS)، مما يؤدي لتحسين جودة الخدمة (QoS) لكل من الخدمات المباشرة (RT) و الخدمات الغير مباشرة (NRT).

في هذه الأطروحة، تم دراسة جدولة الحزم للخدمات المباشرة والخدمات الغير مباشرة ضمن نطاق شبكات (LTE). وتم تقديم مقترح خوارزمية أولوية جدولة الحزم، الخوارزمية المقترحة لديها القدرة على جدولة الخدمات المختلطة من خدمات مباشرة وخدمات غير مباشرة في نفس الوقت، الهدف من الخوارزمية هو زيادة وتعظيم الإنتاجية لخدمات أفضل جهد (BE) وهي الخدمات الغير مباشرة بينما يحقق متطلبات الارتياح في جودة الخدمة والإنتاجية في الخدمات المباشرة. خلصت الأطروحة إلى أنه ينبغي التمييز بين الخدمات المختلفة وأن تعطى أولويات حسب نوع الخدمة وخاصة عند تطبيق الخدمات الحساسة للتأخير.

لقد تم اجراء و تنفيذ نظام المحاكاة لدعم الدراسة المقدمة للخدمات المختلطة والتي تتألف من خدمة الصوت عبر الإنترنت (VoIP) وخدمة ثانية من خدمات أفضل جهد (BE) مثل بروتوكول نقل الملفات (FTP). تم اختبار أداء الخوارزمية المقترحة وتأثير العوامل المختلفة على الأداء العام للنظام. تم تطبيق العمل في طبقة التحكم بالوصول للوسط (MAC) و الطبقة الفيزيائية (PHY).

أخيراً، حققت الخوارزمية المقدمة نتائج جيدة تضمن تقديم أداء جيد للمستخدمين لخدمات الصوت والبيانات على حد سواء.

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## Abbreviations and Acronyms

<b>Acronym</b>	<b>Description</b>
2G	Second generation
3G	Third generation
3GPP	Third Generation Partnership Project
BE	Best Effort
BS	Base Station
CCH	Control Channel
CN	Core Network
CQI	Channel Quality Indicator
EDGE	Enhanced Data Rates for GSM Evolution
eNodeB	evolved Node B
FDD	Frequency Division Duplexing
FDPS	Frequency Domain Packet Scheduler
FTP/TCP	File Transfer Protocol/Transmission Control Protocol
GERAN	GSM EDGE Radio Access Network
GSA	Global mobile Suppliers Association
GSM	Global System for Mobile Communications
GSM BSC	Global System for Mobile Communications Base Station Controller
HARQ	Hybrid Automatic Repeat reQuest
HSPA	High Speed Packet Access
HTTP/TCP	Hypertext Transfer Protocol/Transmission Control Protocol
IEEE	Institute of Electrical and Electronics Engineers
IP	Internet Protocol
LA	Link Adaptation
LTE	Long Term Evolution
MAC	Medium Access Control
MAC	Medium Access Control



MAX C/I	Maximum Carrier to Interference
MCS	Modulation and Coding Scheme
MIMO	Multiple Input Multiple Output
MS	Mobile Station
MT	Maximum Throughput
NRT	Non-Real Time
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
PAPR	Peak-to-Average Power Ratio
PDCCH	Physical Downlink Control Channel
PDR	Packet Drop Ratio
PDSCH	Physical Downlink Shared Channel
PDU	Protocol Data Unit
PF	Proportional Fairness
PHY	Physical Layer
PRB	Physical Resource Block
PS	Packet Scheduling
QoS	Quality of Service
RAT	Radio Access Technology
RLC	Radio Link Control
RR	Round Robin
RT	Real Time
SAE	System Architecture Evolution
SC-FDMA	Single Carrier Frequency Division Multiple Access
SCS	Scheduling Candidate Set
SSSA	Service Specific queue Sorting and scheduling Algorithm
TDD	Time Division Duplexing
TDPS	Time Domain Packet Scheduler
TTI	Transmission Time Interval

UE	User Equipment
UMTS RNC	Universal Mobile Telecommunications System Radio Network Controller
UTRAN	UMTS Terrestrial Radio Access Network
UTRAN	Universal Terrestrial Radio Access Network
VoIP	Voice over IP

# Chapter 1

## INTRODUCTION

### 1. Introduction

The Long Term Evolution (LTE) is a new generation radio access network technology that is standardized in the 3<sup>rd</sup> Generation Partnership Project (3GPP). LTE has the ability to provide greater spectral efficiency, higher data rates and a lower latency. It is developed under a packet switching working. Therefore, the Core Network (CN) architecture is completely packet switched. It supports voice and data traffic simultaneously in the network, where the same radio and CN carry the traffic. Therefore, a strict guarantee of Quality of Service (QoS) could not be provided to each user.

The main goal of all Internet Protocol (IP) based networks usage is to offer data and Best Effort (BE) services. Hence, there are many challenges to support voice over LTE in order to get users satisfaction. To enhance overall performance of the 3GPP LTE, QoS of the system and obtain users satisfaction, Packet Scheduler (PS) must be used in order to distribute the resource blocks between the users. Therefore, a proper scheduling and prioritization of resources should be performed to increase voice capacity and to get a satisfied QoS as well as to maximize the BE throughput.

Several conventional scheduling algorithms are used to achieve the aforementioned goals, such as Round Robin (RR), Proportional Fairness (PF) and Maximum Carrier to Interference (MAX C/I). By using these algorithms, the system level performance are enhanced in terms of user fairness and system throughput, but the QoS of Real Time (RT) and Non Real Time (NRT) services can't be supported. Therefore, to guarantee QoS for both RT and NRT traffic, the researchers have proposed several scheduling algorithms for example semi-persistent scheduling [1] [2] [3], dynamic packet scheduling [3] [4] [5], an intelligent scheduling [6] and opportunistic scheduling [7].

## **1.1. Statement of the Problem**

The big challenge in the a packet switched systems like LTE networks is the ability to guarantee the QoS to all the system users, since these systems serve both data traffic and voice users at the same time. In other words, we should guarantee the QoS to each system user when both RT and NRT services are delivered simultaneously over the same radio and CN. In addition, to get the satisfied QoS, the BE throughput should be maximized. Therefore, a good scheduling and prioritization process of resources should be conducted. Also when a delay sensitive service such as, Voice over Internet Protocol (VoIP), and a delay insensitive service, as web traffic are running concurrently over packet switched network like, LTE network, differentiating the traffic and prioritizing the services that are necessary to be performed. The problem is how to find an efficient scheduling algorithm that can maximize NRT throughput while satisfy each RT users, concurrently and the overall system performance does not affected negatively.

## **1.2. Thesis Organization**

This thesis is organized in four chapters. The first chapter gives the thesis introduction. A background is introduced in chapter two. The new proposed packet scheduling algorithm architecture is explained in chapter three. Finally the conclusion and the recommendations for future works are presented in chapter four.

## **1.3. Motivation and Objectives**

The motivation of this work comes from the definition of the statement of the problem, where mainly IP based networks are designed for carrying best effort data services like Hypertext Transfer Protocol (HTTP) and File Transfer Protocol (FTP) services, in which strict guarantees on the quality of service demands are not provided. However, with the growth of mobile broadband technologies like LTE, carrying both voice and data in the same IP based network is needed. Since both data and voice are carried over the same PS network in LTE, a proper classification among them is needed for scheduling of network resources in the radio and core network domains. Since the future of the mobile broadband is LTE. In the future, it is expected that 80% of all mobile broadband users will be served by LTE. The key element in the evolved NodeB (eNodeB) is the scheduler since it determines to which users the resource blocks should



be assigned. A scheduler assigns the available resource blocks (time and frequency) among users terminals.

The objectives of this thesis:

- Studying the PS of both RT and NRT traffic simultaneously in LTE.
- Develop an efficient scheduling algorithm that maximizes the user satisfaction and the throughput, considering the different system parameters and traffic types.
- Investigate the performance of RT and NRT services in terms of throughput and capacity over LTE networks with mixed traffic.
- Explore the effects of the traffic differentiation and service prioritization when VoIP and BE traffic is in combination.

The proposed algorithm is implemented using a Matlab-implemented LTE system level simulation tool [8], openly and available free for academic and non-commercial use from Vienna University.

## **1.4. Contribution**

The main contribution of the thesis can be summarized as follows:

- Proposing a priority packet scheduling algorithm that has the ability to schedule the mixed traffic at the same time.
- The algorithm aims to maximize the BE throughput while achieves the satisfied QoS requirements of RT throughput.
- The negative impact of packets prioritization on the overall system throughput is minimized.

## **1.5. Thesis Outline and Chapters Structure**

In the sequel, a brief description of the thesis chapters is provided:

## **Chapter 1: Introduction of the thesis**

This chapter provides introduction and information concerning the study subject of the thesis. The scope of the thesis is presented and the statement of the problem is defined. The motivation for the thesis is presented while the thesis objectives are clarified. The contribution of the thesis is also summarized.

## **Chapter 2: Background**

The background chapter consists of three main sections; the first section is LTE network that presents information and overview about the LTE network. The motivation for the LTE is also highlighted. The LTE requirements in term of data rate, bandwidth and spectrum flexibility, throughput and spectrum efficiency, mobility, coverage, low latency and support for inter-operation and co-existence with legacy standards are introduced. A brief description about Orthogonal Frequency Division Multiplexing (OFDM) and Single Carrier Frequency Division Multiple Access (SC-FDMA) scheme are included. LTE Physical Resource Block (PRB) architecture is described.

The second section considers scheduling in LTE and provides general information about the concept of resources scheduling in general. A background about scheduling in LTE is also presented. Techniques used in Downlink/Uplink schedulers are described. The properties of a well-designed packet scheduling are presented. The two phases which most of the schedulers use are Time Domain Packet Scheduling (TDPS) and Frequency Domain Packet Scheduling (FDPS), these phases are explained in this section. Schemes of scheduling algorithms and classifications are also presented. Finally the famous scheduling schemes such as RR scheduling scheme, Best Channel Quality Indicator (Best CQI) scheduling scheme and PF scheduling scheme are described.

At the end of the chapter, literature review is provided to revise and survey the previous work in packet scheduling.

### **Chapter 3: The Proposed Packet Scheduling Algorithm Architecture**

This chapter consists of four sections; the first section is general introduction that provides information regarding the general background of the proposed packet scheduling algorithm. The description of the system model is presented and the frame structure is illustrated.

The second section discusses the related work to the thesis subject. Especially the algorithms those are designed to schedule a multi-service mixed traffic environment such as LTE and LTE-Advanced (LTE-A) systems.

The third section describes the proposed packet scheduling algorithm architecture and gives more details about the structure of the proposed algorithm and the steps of the procedures. The diagram of the proposed packet scheduling algorithm architecture, the queues prioritizing and sorting metrics for different traffic; RT traffic "VoIP users" and NRT traffic "FTP users" are illustrated and summarized. The work description of the Time Domain Scheduler (TDS) and Frequency Domain Scheduler (FDS) are presented in the section.

Finally, the last section in chapter three is devoted to the performance metrics, simulation model and results. The proposed packet scheduling algorithm is evaluated under performance metrics of QoS of both traffic types; RT and NRT, overall system throughput and user fairness. The simulation parameters and assumptions used for system level simulation are described in simulation model sub-section and it is mainly based on the 3GPP Universal Terrestrial Radio Access Network (UTRAN) LTE downlink specifications and recommendations. The last sub-section presents the simulation results, where the main results of the thesis are summarized.

### **Chapter 4: Conclusion and Future Work**

Thesis Conclusion summarizes the study results and introduces the recommendations for packet scheduling in LTE based on the thesis study. Furthermore, several points that could not be assessed during the thesis are discussed, a possible development and future research direction is suggested.

## **Chapter 2**

### **BACKGROUND**

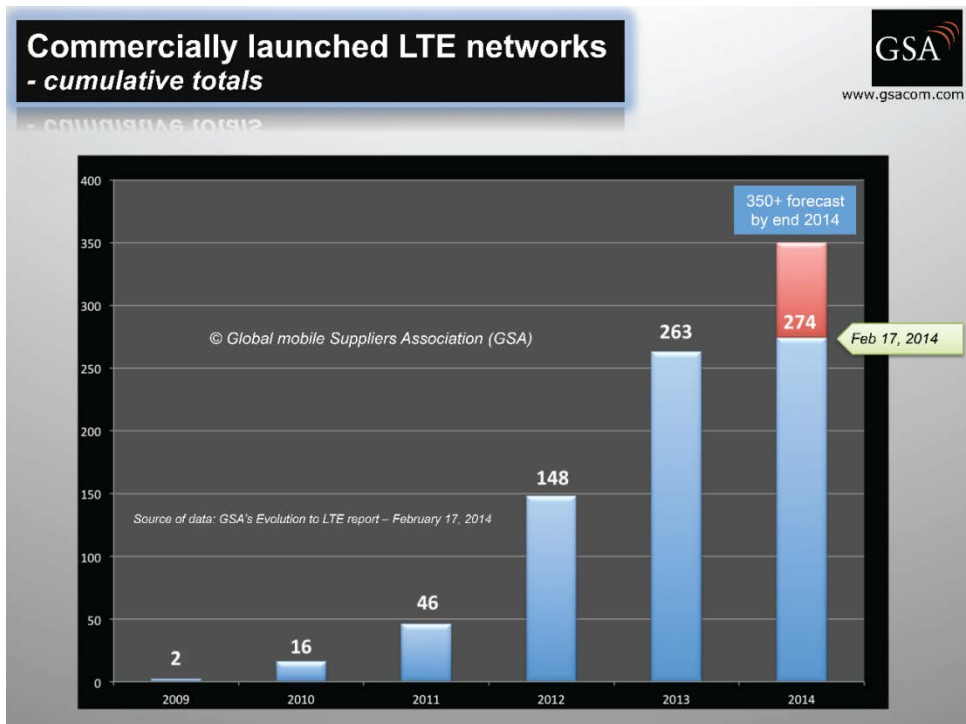
#### **2.1. LTE Network**

The LTE is a new generation radio access network technology that is standardized in 3GPP. The LTE is a mobile broadband radio access network that supposes to enhance spectral efficiency and radio utilization as well as improves data rates in terms of capacity and throughput, also to reduce the latency. LTE is developed under a packet switching working, so it is all-IP based network, which means that it is a system that based on an IP packet. It adopts OFDM as the downlink transmission scheme. The uplink transmission scheme is based on SC-FDMA to avoid a synchronization problem.

In the recent years, more attention has been given to multimedia applications, such as video streaming, mobile TV and online gaming. These applications need higher data rate. Since the LTE provide high data rate, these applications will be enhanced by LTE. LTE aims to be the global standard for the fourth generation wireless technology of mobile broadband. It is the fastest developing system in the history of mobile communication.

The first publicly available LTE service was launched by TeliaSonera in Oslo and Stockholm on December 14, 2009. And on February 17, 2014: From amongst the committed operators, 274 have commercially launched networks in 101 countries. Global mobile Suppliers Association (GSA) forecasts there will be 350 commercial LTE networks by end 2014 [9]. More details are illustrated in Figure 2.1.





**Figure 2.1: Commercial LTE network launches [9].**

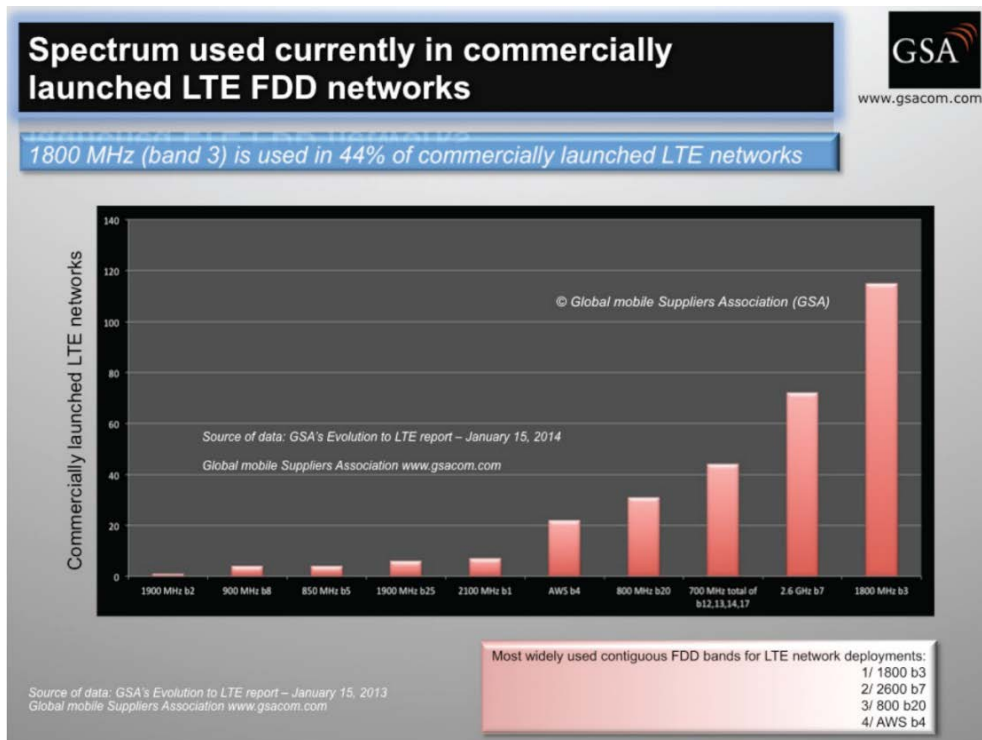
Release 8 in communication system is the first release of LTE, with a completely new radio interface and core network, enabling the system performance improvements compared with previous systems, features included:

- Provide up to 100Mbit/s downlink and 50Mbit/s uplink.
- Latency down to 5ms.
- Implementation in different bandwidths of 1.4, 3, 5, 10, 15 or 20MHz, to allow for different deployment scenarios.
- Orthogonal frequency division multiple access (OFDMA) downlink.
- Single carrier frequency division multiple access (SC-FDMA) uplink.
- Multiple input multiple output (MIMO) antennas.
- Flat radio network architecture, with no equivalent to the Global System for Mobile Communications Base Station Controller (GSM BSC) or Universal Mobile Telecommunications System Radio Network Controller (UMTS RNC), and functionality distributed among the eNodeBs.
- All IP core network, the System Architecture Evolution (SAE).

Furthermore LTE is developed to operate both Frequency Division Duplexing (FDD) and Time Division Duplexing (TDD). With TDD the uplink and downlink operate in same frequency band whereas with FDD the uplink and downlink operate in different frequency bands. LTE is developed for a number of frequency bands, ranging from 700 MHz up to 3.5 GHz. The majority of LTE operators have deployed the FDD mode of the standard. The most widely used band in network deployments continues to be 1800 MHz (band 3) which is used in 43% of commercially launched LTE networks. 117 operators worldwide have launched LTE1800 (band 3) systems, either as a single band system, or as part of a multi-band deployment. As 1800 MHz is the prime band for LTE deployments worldwide, it will greatly assist international roaming for mobile broadband services.

Where, 2.6 GHz (band 7) is the next most popular contiguous bands as used in 27% of networks, followed by 800 MHz (band 20) which is used in 13% of networks and AWS (band 4) that is used in 9% of networks in commercial service today [9]. The available bandwidths are also flexible starting with 1.4 MHz up to 20MHz. OFDM has been adopted as the downlink transmission scheme for the 3GPP LTE. A downlink is a transmission from the base station to the mobile station. OFDM divides the transmitted high bit-stream signal into different sub-streams and sends these over many different sub-channels. A Base Station (BS) is called an eNodeB in the LTE and a Mobile Station (MS) is called a User Equipment (UE) in the LTE.

According to information provided in GSA's Evolution to LTE report of February 17, 2014. There are currently in use 274 commercially launched LTE FDD networks and LTE TDD networks. Figure 2.2 shows the frequency bands of the LTE FDD networks that are commercially launched and currently in service. Also the frequency bands that are currently in use in commercially launched LTE TDD networks are shown in Table 2.1.

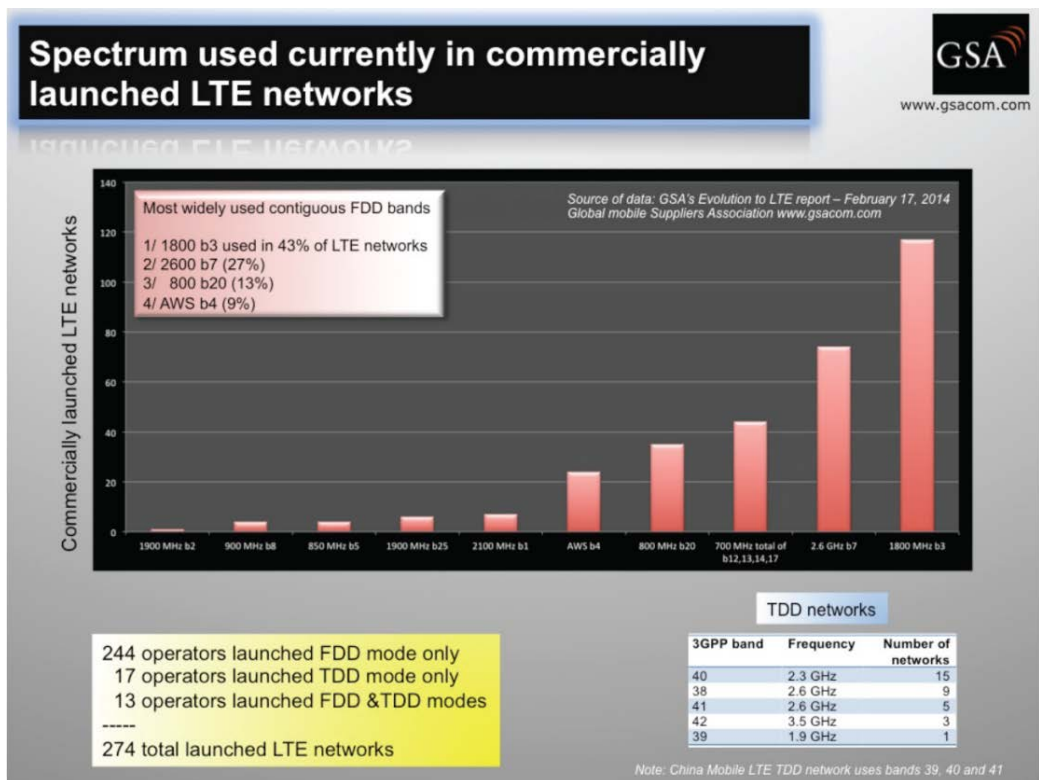


**Figure 2.2: Spectrum used in LTE FDD networks [9].**

**Table 2.1: Spectrum used in LTE TDD networks [9] .**

3GPP Band	Frequency	Number of Networks
40	2.3 GHz	15
38	2.6 GHz	9
41	2.6 GHz	5
42	3.5 GHz	3
39	1.9 GHz	1

We can summarize the following; the LTE Release 8 is one of the primary broadband technologies based on OFDM, which is currently being commercialized. LTE Release 8, which is mainly deployed in a macro/microcell layout, provides improved system capacity and coverage, high peak data rates, low latency, reduced operating costs, multi-antenna support, flexible bandwidth operation and seamless integration with existing systems. LTE provides scalable carrier bandwidths as well as TDD. And spectrum used currently in commercially launched LTE networks is summarized in Figure 2.3.



**Figure 2.3: Summary of spectrum used in LTE networks [9].**

### 2.1.1. LTE Motivations

There are many main issues are driving the move to LTE. The first issue is, a 2G and 3G operator has to maintain two core networks: the first network is the circuit switched domain for voice, and the other network is the packet switched domain for data. Provided that the network is not too congested, however, we can use VoIP techniques to transport voice calls over packet switched networks. By doing this, operators can move voice traffic to the packet switched domain, and can reduce both their capital and operational expenditure.

The second issue, 3G networks introduce delays of the order of 100 milliseconds for data applications, in transferring data packets between network elements and across the air interface. This is barely acceptable for voice and causes great difficulties for more demanding applications such as real-time interactive games. Thus, the wish to reduce the end-to-end delay, or latency, in the network is a second motivation.

The third issue, the specifications for UMTS and GSM has become increasingly complex over the years, due to the need to add new features to the system while



maintaining backwards compatibility with earlier devices. So the third motivation is the need for simple and low complex system.

We can summarize the main motivations for LTE as follows [40]:

- Continued demand for cost reduction of Capital Expenditure and Operational Expenditure (CAPEX and OPEX), cheaper infrastructure.
- Need to ensure the continuity of competitiveness of the 3G system for the future, more spectral efficiency than High Speed Packet Access (HSPA) Release 6.
- User demand for higher data rates can be achieved with new air interface defined by 3GPP LTE.
- User demand for new services and high quality of services, reduce round trip delay.
- Packet Switch optimized system, to evolve UMTS towards packet only system.
- Demand for Low complexity system; simplify architecture with reduced number of network elements.
- Avoid unnecessary fragmentation of technologies for paired and unpaired band operation.

### **2.1.2. LTE Requirements**

The most important requirements for LTE can be summarized as follows:

#### **All-IP Network**

One of the main requirements is the transition of circuit-switched and packet-switched networks into an all-IP network which can support different types of services with different QoS and which also provide the easy integration with the other communication networks. This will ultimately reduce the integration cost and provide with the users the seamless integration with other services.

#### **Data Rates**

LTE should support an instantaneous downlink peak data rate up to 100 Mb/s within a 20 MHz spectrum allocation, spectrum efficiency of 5bps/Hz and instantaneous uplink peak data rate of 50 Mb/s within a 20 MHz spectrum allocation, spectral efficiency of 2.5bps/Hz.

## **Bandwidth and Spectrum Flexibility**

LTE must have spectrum flexibility and operates in spectrum allocations of different sizes, including 1.4 MHz, 3 MHz, 5 MHz, 10 MHz, 15 MHz and 20 MHz in both the uplink and downlink. Operation in paired and unpaired spectrum shall be supported.

## **Throughput and Spectrum Efficiency**

The downlink average user throughput per MHz is about 3 to 4 times higher than that in the release 6. Target for spectrum efficiency (bits/sec/Hz) is about 3 to 4 times higher than in the Release 6. The uplink average user throughput per MHz is about 2 to 3 times higher than in the release 6. Target for spectrum efficiency (bits/sec/Hz) is about 2 to 3 times higher than in the Release 6.

## **Mobility**

Improved support for mobility, LTE performance should be optimized for low mobile terminals speed from 0 to 15 km/h. High performance should be provided for higher mobile speed between 15 and 120 km/h. Mobility across the cellular network should be maintained at very high mobile terminals speed from 120 km/h to 350 km/h (or even up to 500 km/h depending on the frequency band).

## **Low Latency**

Low data transfer latencies (sub-5 ms latency for small IP packets in optimal conditions), lower latencies for handover and connection setup time than with previous radio access technologies. Latency of 50-100 ms for C-plane and less than 10ms for U-plane.

## **Coverage**

The aforementioned LTE targets should be met for 5 km cells and some slight degradation in throughput and spectrum efficiency for 30 km cells.

## Inter-Operation and Co-Existence with Legacy Standards

LTE allows seamless integration with existing systems and Inter-working with 3GPP Radio Access Technology (RAT) GSM/EDGE Radio Access Network (GERAN/UTRAN) Co-existence in the same geographical area and co-location with GERAN/UTRAN on adjacent channels.

### Control Plane Capacity

At least 200 users per cell should be supported in active state for allocation of 5MHz spectrum.

### Multi-Antenna Configuration

The multi-antenna configuration will significantly improve the system performance and service capability and it would be used to achieve the transmit diversity, multi-stream transmission, and beam forming.

### 2.1.3. LTE Frequency Bands

The 3GPP define the following operating bands for LTE FDD and TDD mode in the specification "TS 36.101" [10]. Table 2.2 and Table 2.3 show the LTE frequency bands for LTE FDD and LTE frequency bands for LTE TDD, sequentially.

**Table 2.2: LTE frequency bands for LTE FDD**

Band	Uplink MHz		Downlink MHz		Width	Duplex	Gap
1	1920	1980	2110	2170	60	190	130
2	1850	1910	1930	1990	60	80	20
3	1710	1785	1805	1880	75	95	20
4	1710	1755	2110	2155	45	400	355
5	824	849	869	894	25	45	20
6	830	840	865	875	10	35	25
7	2500	2570	2620	2690	70	120	50
8	880	915	925	960	35	45	10
9	1749.9	1784.9	1844.9	1879.9	35	95	60
10	1710	1770	2110	2170	60	400	340
11	1427.9	1447.9	1475.9	1495.9	20	48	28
12	698	716	728	746	18	30	12
13	777	787	746	756	10	-31	21
14	788	798	758	768	10	-30	20

15*	1900	1920	2600	2620	20	700	680
16*	2010	2025	2585	2600	15	575	560
17	704	716	734	746	12	30	18
18	815	830	860	875	15	45	30
19	830	845	875	890	15	45	30
20	832	862	791	821	30	-41	11
21	1447.9	1462.9	1495.9	1510.9	15	48	33
22	3410	3490	3510	3590	80	100	20
23	2000	2020	2180	2200	20	180	160
24	1626.5	1660.5	1525	1559	34	-101.5	67.5
25	1850	1915	1930	1995	65	80	15
26	814	849	859	894	35	45	10
27	807	824	852	869	17	45	28
28	703	748	758	803	45	55	10

**Note:** Band 15 and 16 are reserved, also the uplink frequencies of band 13, 14, 20 and 24 are higher than the downlink frequencies.

**Table 2.3: LTE frequency bands for LTE TDD**

<b>Band</b>	<b>Frequency band</b>	<b>Bandwidth</b>	<b>Remark</b>
33	1900 MHz - 1920 MHz	20 MHz	3G core band
34	2010 MHz - 2025 MHz	15 MHz	3G core band
35	1850 MHz - 1910 MHz	60 MHz	PCS 1900 uplink band
36	1930 MHz - 1990 MHz	60 MHz	PCS 1900 downlink band
37	1910 MHz - 1930 MHz	20 MHz	
38	2570 MHz - 2620 MHz	50 MHz	3G extension band (EU, Africa, and Asia etc.)
39	1880 MHz - 1920 MHz	40 MHz	TD-SCDMA band in China
40	2300 MHz - 2400 MHz	100 MHz	To be deployed in China
41	2496 MHz - 2690 MHz	194 MHz	US 2.6 GHz band
42	3400 MHz - 3600 MHz	200 MHz	EU 3.5 GHz band
43	3600 MHz - 3800 MHz	200 MHz	EU 3.5 GHz band
44	703 MHz - 803 MHz	100 MHz	APT700

#### **2.1.4. Orthogonal Frequency Division Multiplexing (OFDM)**

The principle of the Orthogonal Frequency Division Multiple access (OFDMA) focuses on the usage of narrow, mutually orthogonal sub-carriers. OFDM has been adopted as the downlink transmission scheme for the 3GPP LTE. OFDM simply divides the total available bandwidth into multiple narrow sub-channels and transmits the data on these channels in parallel streams. Different levels of modulation, e.g. Quadrature Phase-Shift Keying (QPSK), Quadrature Amplitude Modulation (QAM) 16QAM and 64QAM are used to modulate each subcarrier. OFDM signal used in LTE consists of a maximum of 2048 different sub-carriers spacing typically 15 kHz regardless of the total transmission bandwidth. Spectral efficiency is boosted by LTE because the OFDM approach achieves high peak data rates in high spectrum bandwidth and also high flexibility in channelization.

#### **2.1.5. Single Carrier Frequency Division Multiple Access (SC-FDMA)**

Single Carrier Frequency Division Multiple Access (SC-FDMA) scheme is a new promising technology used for high data rate uplink communication direction and has been adopted by 3GPP for its next generation cellular system, called LTE, which is suitable to both FDD and TDD modes. SC-FDMA is a modified form of OFDM with similar throughput performance and complexity, SC-FDMA is suitable for broadband systems, because of its robustness against multipath signal propagation, the main advantage of SC-FDMA is its low Peak-To-Average Power Ratio (PAPR) compared to OFDM making it suitable for uplink transmission by user-terminals.

#### **2.1.6. Downlink vs. Uplink Transmission**

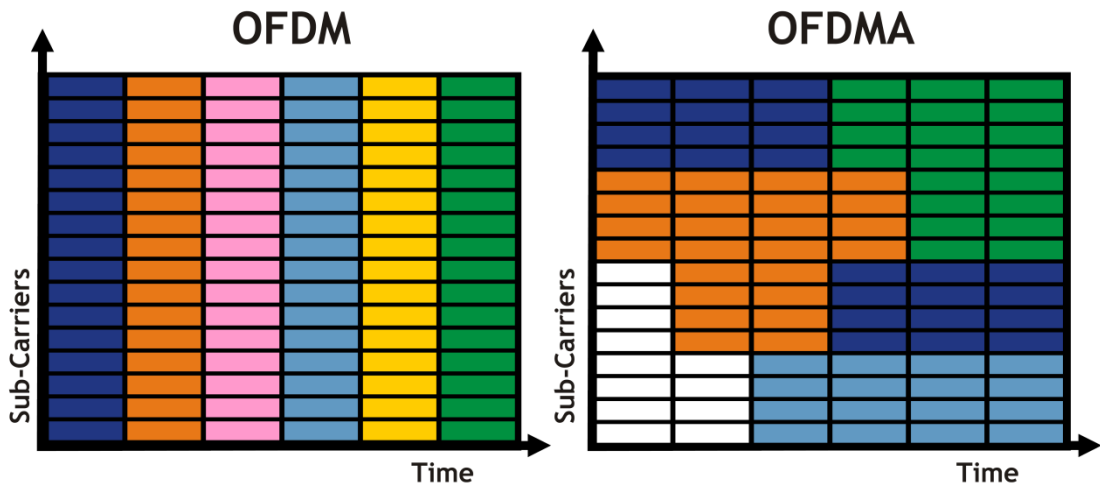
Physical layer downlink transmission is implemented using OFDMA while uplink transmission uses SC-FDMA. Both OFDMA and SC-FDMA use the same time-frequency grid, same time slots, sub-frames, frames, sub-carrier, and sub-band structure and so on. The differences between OFDMA and SC-FDMA are:

- OFDMA can achieve frequency diversity because random available subcarriers are combined in order to form sub-band, while adjacent subcarriers are combined to form sub-band in SC-FDMA, thereby no need for Cyclic Prefix (CP).

- Another difference is in the transmission of control signal. In OFDMA one subcarrier uses 7 OFDM symbols in one time slot to carry data transmission while in SC-FDMA two short blocks are reserved to carry pilot signal and 6 are used for data transmission.

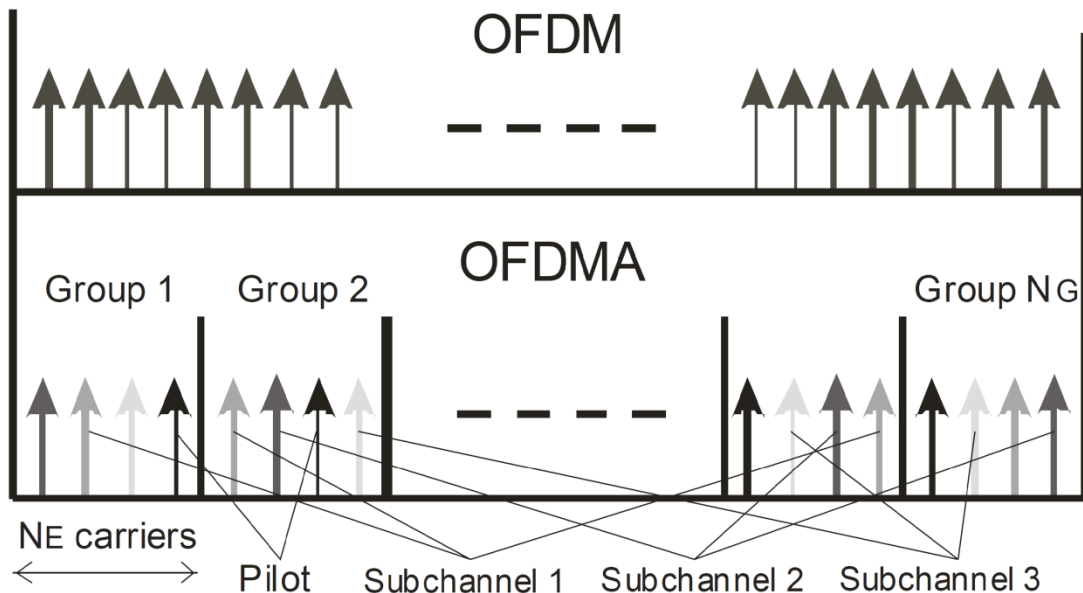
### **2.1.7. OFDM vs. OFDMA**

OFDM assigns all sub-carriers to one user in a symbol in the time domain. OFDMA is a combination of OFDM and FDMA, In OFDMA; the active sub-carriers are divided into groups termed as sub-channels. So the sub-channel is a subset of sub-carriers. The sub-carriers that form a sub-channel need not be adjacent. The main purpose of this sub-channelization is to support scalability, multiple access and advance antenna array processing capabilities. OFDMA also brings reduction of interference for user and improved None Line of Sight (NLOS) capabilities that are essential in mobile environment. Next, OFDMA assigns sub-channels with proper power to different users based on channel knowledge from CQI intending to maximize the system throughput. So the difference between OFDM and OFDMA can be concluded as that OFDMA has the ability to dynamically assign a subset of subcarriers to individual terminals, making this the multi-user version of OFDM, while OFDM assigns all the subcarriers to one terminal. Figure 2.4 illustrates the difference between channel allocation using OFDM and OFDMA scheme. To clarify the idea of OFDMA, we use different colors to make distinguish between the sub-carriers, and a unique color is assigned to each subset of sub-carriers that form a sub-channel. And then the sub-channels are assigned to different users by the OFDMA. Accordingly, the different colors indicate the sub-channels group given to the users.



**Figure 2.4: Channel allocation using OFDM and OFDMA schemes [11].**

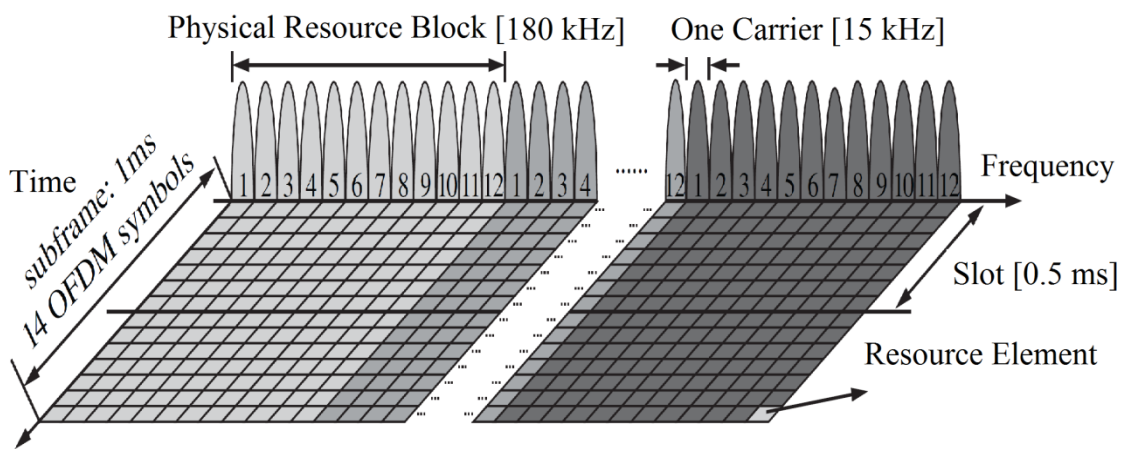
Figure 2.5 illustrates the principle of sub-channelization in OFDMA. The total number of the carriers is divided into  $N_G$  groups. Each group contains  $N_E$  carriers and thus  $N_E$  sub-channels are created. The type of coding and modulation are set separately for each sub-channel. The subcarriers can be allocated to different users (one sub-channel per one user) depending on the channel conditions. This characteristic is useful for operators who assign to these users most suitable subcarriers which leads to efficient use of resources.



**Figure 2.5: Comparison of OFDM and OFDMA [12].**

### 2.1.8. LTE Physical Resource Block (PRB) Architecture

OFDM is the core of LTE downlink transmission. LTE downlink physical resource can be represented as a grid of time and frequency as depicted in the Figure 2.6. The total system bandwidth is divided into number of physical resource block, the number of the PRB depends on the system bandwidth size, since the LTE support different bandwidth sizes from 1.4MHz to 20MHz, and so the number of PRB is from 6 PRBs to 100 PRBs. Each PRB has a bandwidth of 180 kHz and duration of 0.5ms and consists of 12 subcarriers; one subcarrier has 15 kHz bandwidth. One PRB has 7 OFDM symbols in the case of normal cyclic prefix with  $12 \times 7 = 84$  resource elements and 6 OFDM symbols in the case of extended cyclic prefix with  $12 \times 6 = 72$  resource elements. The resource grid refers to a number of resource blocks in the available bandwidth. Each entry of the resource block is called a Resource Element which represents one OFDM subcarrier during one OFDM symbol interval. The OFDM subcarrier spacing is 15 kHz.



**Figure 2.6: Time - Frequency grid of LTE [13].**

A set of possible bandwidths for the LTE standard are defined by 3GPP, which determine the number of RBs, data subcarriers and band guard size. The Table 2.4 shows available LTE system bandwidths and available resource blocks [14].



**Table 2.4: Possible bandwidths for LTE system and possible PRBs.**

Channel Bandwidth [ MHz]	1.4	3	5	10	15	20
Number of PRBs	6	15	25	50	75	100
Number of data subcarriers	72	180	300	600	900	1200
Transmission bandwidth [MHz]	1.08	2.7	4.5	9	13.5	18
Band guard size [% of Channel Bandwidth]	23%	10%	10%	10%	10%	10%

## 2.2. Scheduling in LTE

The OFDMA as the core radio access technology is chosen for the LTE download by the 3GPP. Packet scheduling is one of the key mechanisms for realizing the potential efficiency of this technology and for achieving optimal performance of the eNodeB, which coordinates the access to the shared channel resources. In OFDMA-based LTE systems this coordination refers to both the time dimension (allocation of time frames) and the frequency dimension (allocation of subcarriers), hence, packet scheduling in LTE is distinguish from that in earlier radio access technologies, such as High Speed Downlink Packet Access (HSDPA), is that LTE schedules resources for users in both the time domain (TD) and the frequency domain (FD) whereas HSDPA only involves the time domain. This additional flexibility has been shown to provide throughput and coverage gains. Also it balances maximum throughput and fairness by scheduling time slots, sub-channels, modulation and coding scheme and power with frequency diversity and multiuser diversity. Frequency diversity is achieved by utilizing the fact that each sub-channel suffers different attenuation in different time and frequency, due to many reasons such as shadowing, fast fading, multipath and so on. In a similar way, multiuser diversity is obtained by opportunistic user scheduling, since different users have different channel gains due to located in different places. Depending on CQI scheduling should be adjusted to keep optimal performance.

Recently, the development of packet scheduling algorithms has been a hot research topic. Till now, a lot of packet scheduling algorithms have been studied in many research papers. Most of them focus on maximizing the throughput, while others

focus on fairness, based on Maximum Throughput (MT), Proportional Fair and the Round Robin, as in [5] [15].

OFDMA allows multiple users to simultaneously share the OFDM sub carriers and thus leading to exploit multiuser diversity and to provide greater flexibility in resource allocation (scheduling). Scheduling decisions are taken by the eNodeB in LTE, each Transmission Time Interval (TTI) and radio resources are scheduled every 1ms in 3GPP LTE, and the decisions are potentially based on channel quality feedback provided by the UEs. The packet scheduler decides which users are served and how many resource blocks are assigned to each selected user. The scheduler is in the Medium Access Control (MAC) layer, but it controls MAC layer and Physical layer at the same time. Scheduler can be divided into downlink scheduler for downlink scheduling and uplink scheduler for uplink scheduling. So we can conclude that the process in which resource blocks are distributed among the UEs is a radio resource scheduling and before the eNodeB can assign the modulation technique and coding rate to an UE, based on the transmission channel condition, it must be assigned radio resource blocks. Due to the rapidly and instantaneously changing nature of radio channel quality the scheduling algorithm must be fast enough in order to follow variation and compensate the changing channel conditions.

### **2.2.1. Downlink Scheduler**

Channel Quality Indicator reports are feedback periodically from UEs to the eNodeB in order to report the downlink channel conditions. For channel-dependent scheduling, channel state, buffer status and priorities are taken into account by the downlink scheduler. Then it decides resource blocks allocation, the modulation scheme, and antenna mapping for terminals. As a result, downlink scheduler decision controls Radio Link Control (RLC) segmentation, MAC multiplexing and Hybrid Automatic Repeat reQuest (HARQ), Physical Layer (PHY) channel coding, PHY modulation and antenna mapping.

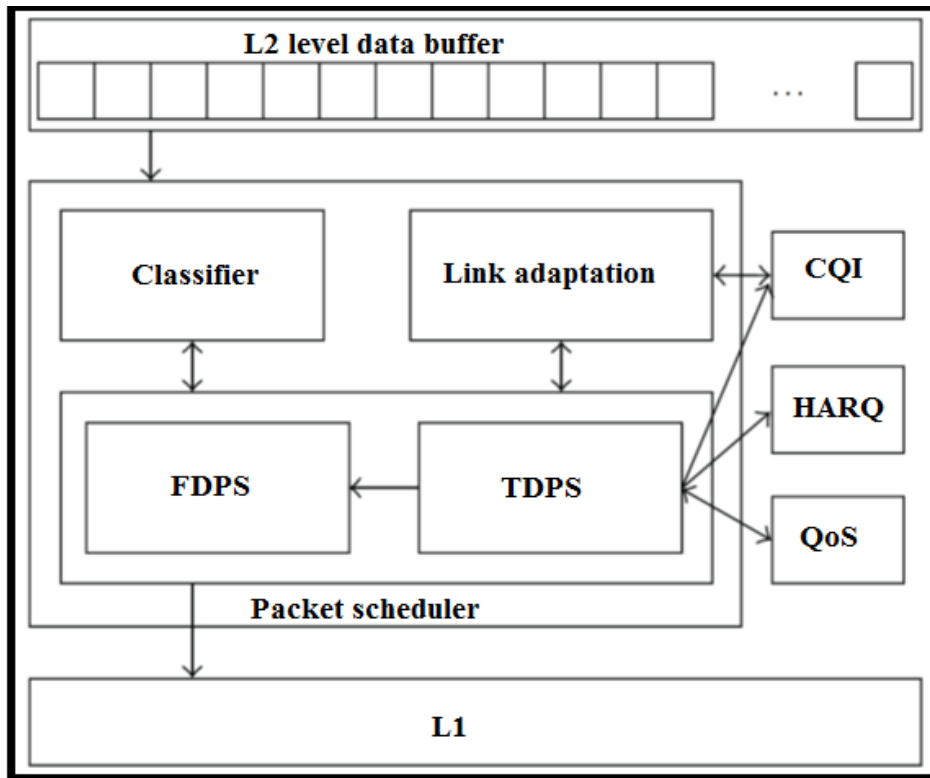
### 2.2.2. Uplink Scheduler

Similar to downlink, eNodeB uplink scheduler decides resource blocks allocation. However, logical channel multiplexing is controlled by terminals and channel state estimation is done for channel-dependent scheduling by eNodeB with reference signals transmitted from each terminal covered by this eNodeB. With the knowledge of channel conditions, uplink scheduler makes decisions to control channel coding and modulation scheme of terminals. Uplink Time/Frequency resources are scheduled in eNodeB based on QoS, CQI measurements on uplink, and UE capabilities and its buffer status. The uplink scheduling decision is transmitted to UE on Physical downlink Control Channel (PDCCH).

In order to have good scheduling decisions, a scheduler should be aware of channel quality in the time domain as well as the frequency domain. Ideally, the scheduler should have knowledge of the channel quality for each sub-carrier and each user. In practice, there are constraints at signaling resources, because it is limited, so that groups of sub-carriers are allocated in an OFDMA system. On the downlink in LTE systems, sub-carriers are grouped into Resource Blocks (RBs) of 12 adjacent sub-carriers with an inter sub-carrier spacing of 15 kHz. Each RB has a time slot duration of 0.5ms, which contains 6 or 7 OFDM symbols depending on whether an extended or normal cyclic prefix is used. A Scheduling Block (SB) is the smallest resource unit that a scheduler can assign to a user, which consists of two consecutive RBs, it forms a sub-frame time duration of 1ms.

A good packet scheduling must achieve maximum throughput and capacity, the QoS provision of users and certain level of fairness. The following properties should be in any packet scheduling algorithm:

- 1. Efficiency:** It is to achieve user satisfaction in terms of QoS requirements.
- 2. Protection:** It is to achieve flow isolation; it represents a sequence of input packets as providing individual virtual channels.
- 3. Flexibility:** Wide different users QoS requirements should be supported.
- 4. Low complexity:** The algorithm implementation should have reasonable computational complexity.



**Figure 2.7: The structure of RT and NRT traffic packet scheduler in eNB [16].**

A lot of schedulers operate in two phases, in order to reduce the complexity: Time Domain Packet Scheduler (TDPS), followed by Frequency Domain Packet Scheduler (FDPS) [5] [15]. The main structure of PS for RT and NRT traffic in evolved NodeB of the 3GPP LTE is illustrated in Figure 2.7.

### **2.2.3. Time Domain Packet Scheduling (TDPS)**

TDPS chooses the User Terminals (UEs) and groups them in a Scheduling Candidate Set (SCS); therefore, it determines the number of UEs for FDPS. The PRBs does not directly allocated by the TDPS. The TDPS chooses the SCS depending on priorities criteria such as the CQI, throughput and delay. Then the SCS information is relocates to the FDPS. Also the HARQ requests the retransmissions are automatically chosen in the SCS.

#### **2.2.4. Frequency Domain Packet Scheduling (FDPS)**

After the SCS relocated to the FDPS, the UEs are directly allocating the PRBs and they can transmit the data. The resource allocation to selected users aims to achieve the goal of the packet scheduling design such as improving the overall system performance and throughput and user fairness.

For example, if we have  $N$  users that have packets to be sent, and these users waiting in the queue, and that resources can only be allocated at the beginning of scheduling period which it is a predefined time period known as the TTI. So based on a certain priority metric, a set of users from the total of  $N$  users are selected in the TD scheduling. After the set of users have been selected, appropriate subcarrier frequencies and Modulation and Coding Schemes (MCSs) are then assigned by the FD scheduler. Note that the metrics used for TD and FD scheduling can be different in order to achieve the predetermined packet scheduling design target and to provide a greater degree of design flexibility.

An important issue in scheduling design, the type of traffic plays an important role in how scheduling should be done. For example, VoIP users are active only half of the time. Also, the size of VoIP packets is small equal 320 bits, and the corresponding inter-arrival time is fairly constant. While dynamic scheduling based on frequent downlink transmit format signalling and uplink CQI feedback can exploit user channel diversity in both frequency and time domains, it requires a large signalling overhead. Thereby time-frequency resources are consumed by this overhead, and thus the system capacity will be reduced. Many scheduling schemes are proposed to solve this problem such as persistent scheduling which aims to lower signaling overhead for VoIP-type traffic. The persistent scheduling lacks of flexibility in the time domain. This disadvantage has led to new scheme which called semi-persistent scheduling.

#### **2.2.5. Schemes of Scheduling Algorithms and Classifications**

There are many packet scheduling schemes proposed to support different types of traffic such as, (real-time, non-real-time or both) in wireless systems. Some of these algorithms such as proportional fair [17], Maximum Sum Rate (MSR) [18], Maximum

Fairness [19] and Modified Largest Weighted Delay First (M-LWDF) [20], have been widely accepted for use in their proposed wireless systems. And other advance packet scheduling algorithms such as an intelligent scheduling [6], but most of the packet scheduling follows and depends on one of the following; dynamic scheduling, persistent scheduling or semi-persistent scheduling. The idea behind persistent scheduling is to pre-allocate a sequence of frequency-time resources with a fixed MCS to a VoIP user at the beginning of a specified period. This allocation remains valid until the user receives another allocation due to a change in channel quality or an expiration of a timer. The main disadvantage of such a scheme is the lack of flexibility in the time domain. This disadvantage has led to semi-persistent scheduling which represents a compromise between rigid persistent scheduling on the one hand, and fully flexible dynamic scheduling on the other. In semi-persistent scheduling, initial transmissions are persistently scheduled so as to reduce signalling overhead and retransmissions are dynamically scheduled so as to provide adaptability. In fact, the signaling overhead is reduced with respect to the dynamic scheduling, but it is slightly higher than the one obtained with persistent solutions.

There are many criteria are taken to classify the scheduling algorithms. In general, scheduling can be divided into two classes: channel-independent scheduling and channel-dependent scheduling. Channel conditions do not be taken into account when using Channel-independent scheduling. Thereby the performance of this kind of scheduling can never be optimal. On the other side, channel-dependent scheduling can achieve better performance by allocating resources based on channel conditions with optimal algorithms. This thesis focuses on channel-dependent scheduling. As mentioned above the performance of scheduling algorithms highly relies on the type of incoming traffic.

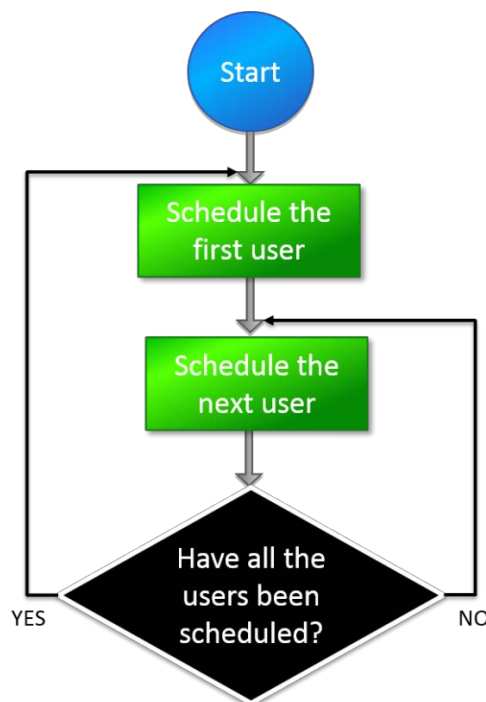
### **2.2.6. Scheduling Schemes**

In this section, brief descriptions about the most famous scheduling schemes are provided.

- **Round Robin Scheduling Scheme**

Round Robin is one of the simplest resources scheduling algorithm. Radio resources are allocated to users in a round-robin fashion. The whole frequency spectrum are assigned to the first reached user for a specific time period and all other users have to be in the waiting queue until their turn comes and these assigned resources are revoked back and assigned to the next user for another time period. The previously served user is placed at the end of the waiting queue so that it can be served with radio resources in next round. The new arriving requests are also placed at the tail of the waiting queue. This scheduling continues in the same manner. This scheme offers a great fairness among the users in radio resource assignment but it yields lower throughput, since the users in bad channel conditions need more resources to carry out the same rate. And because only one user is served at a time thus the overall system performance will be degraded frequently. And so it is not practical in Long Term Evolution technology. Round Robin scheme can be concluded as the shared resources are assigned in turn to the users. Thus every user is equally scheduled without taking the CQI into account. Since Round Robin doesn't take the channel quality information into account, it will result in low user throughput.

The flowchart of the Round Robin scheduling is shown in Figure 2.8.

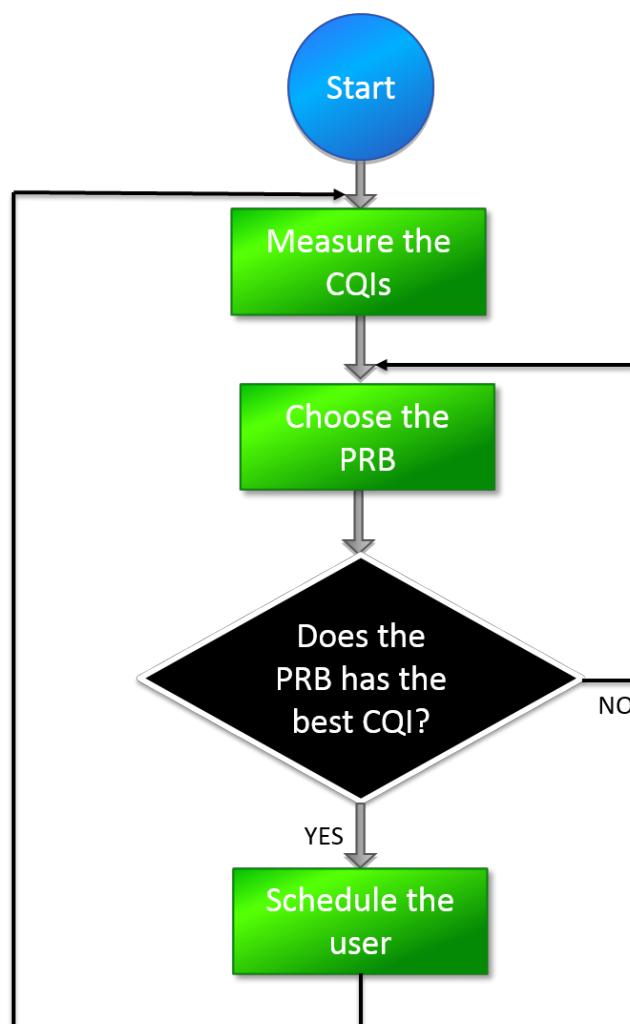


**Figure 2.8: Flowchart of Round Robin scheduling scheme.**

- **Best CQI Scheduling Scheme**

In this scheduling scheme the resource blocks are assigned to the user who has the best radio link conditions. First the CQI is sent to the eNodeB by the terminals. Basically in the downlink, the eNodeB transmits reference signal (downlink pilot) to terminals. These reference signals are used by UEs for the measurements of the CQI. A higher CQI value means better channel condition. Then terminals with higher CQI will be assigned to the resource blocks. The cell throughput can be increased by Best CQI scheduling scheme but at the expense of the fairness [21]. The disadvantage of this scheduling scheme, terminals located far from the eNodeB (i.e. cell-edge users) are unlikely to be scheduled.

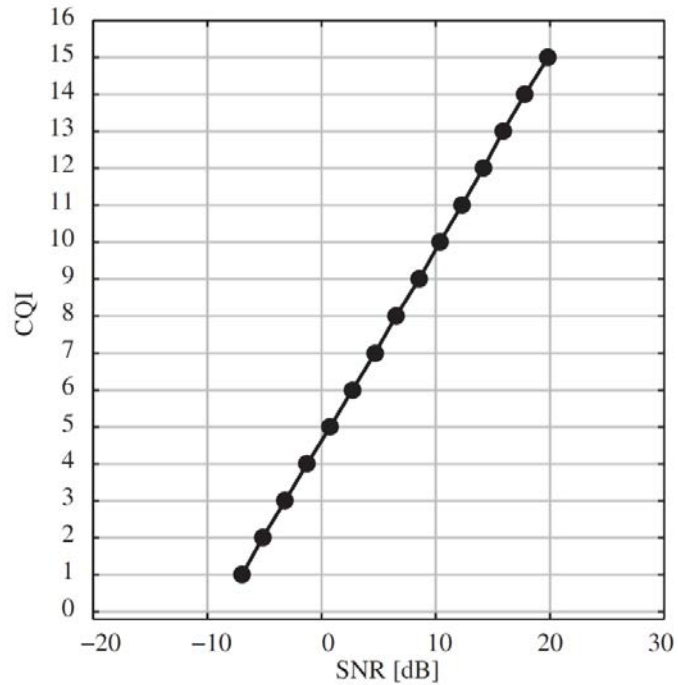
The Best CQI scheduling flowchart is illustrated in Figure 2.9.



**Figure 2.9: Flowchart of the Best CQI scheduling scheme.**



The measured CQI is reported to the eNodeB by mapping the measured SNR according to Figure 2.10. In the LTE simulator, the mapping of the SNR to the CQI is approximated through a linear function as shown in the Figure 2.10.



**Figure 2.10: SNR-CQI mapping model [22].**

**Table 2.5: CQI table [23].**

CQI index	Modulation	Coding rate	Efficiency [b/s/Hz]
0		Out of range	
1	QPSK	78/1024	0.1523
2	QPSK	120/1024	0.2344
3	QPSK	193/1024	0.3770
4	QPSK	308/1024	0.6016
5	QPSK	449/1024	0.8770
6	QPSK	602/1024	1.1758
7	16 QAM	378/1024	1.4766
8	16 QAM	490/1024	1.9141
9	16 QAM	616/1024	2.4063
10	64 QAM	466/1024	2.7305
11	64 QAM	567/1024	3.3223
12	64 QAM	666/1024	3.9023
13	64 QAM	772/1024	4.5234
14	64 QAM	873/1024	5.1152
15	64 QAM	948/1024	5.5547

The scheduler determines the modulation scheme and channel coding rate according to the CQI received value, The Table 2.5 contains the CQI index. A CQI index is defined in terms of a channel coding rate value and modulation scheme (QPSK, 16-QAM, 64-QAM) as given in the above table.

- **Proportional Fair Scheduling Scheme**

Proportional fair is a channel-state based scheduling algorithm that relies on the concept of exploiting user diversity. It is employed at the base station to schedule downlink traffic among different users. PF is a compromise between Best CQI scheme and Round Robin scheme. It's based upon maintaining a balance between two competing interests: Trying to maximize overall system throughput while at the same time assuring that none of users are starving. In PF scheduling algorithm for OFDMA, the priority for each user at each resource block is calculated firstly and then the user with the highest priority is assigned the RB and the scheduling process continues to assign the RB to the user with next highest priority. This process continues until all RBs are assigned or all users have been served with RBs. PF provides the proportional fairness among users by prioritizing the users using each user's ratio of the current channel rate to the average allocated rate. The average throughput for each user is updated, after each resource block is allocated.

### **2.3. Literature Review**

Since the 3GPP standardized the LTE as a new generation radio access network technology, many algorithms are proposed to enhance network delivery. Considering solution for providing scheduling algorithms in different services conditions, these algorithms are looking to provide a proper scheduling and prioritization of traffic, in order to increase system capacity and to get a satisfied QoS as well as to maximize the BE throughput. While Conventional scheduling algorithms such as Round Robin (RR), Maximum Carrier to Interference (MAX C/I) and Proportional Fairness (PF) improve system level performance in terms of fairness, system throughput and a trade-off between system throughput and user fairness, respectively. These algorithms however cannot support QoS support to RT and NRT traffic. For example MAX C/I scheduler is proposed in [24] and PF scheduler is provided in [17]. The max C/I scheduler always

chooses the user whose channel rate is the largest at each scheduling instance. Therefore it achieves the maximum system throughput, but many users whose channel states are not good may starve. PF scheduler uses each user's ratio of the current channel rate to the average allocated rate. It provides the proportional fairness among users. These two schedulers don't support specific QoS parameters like maximum allowable delay and minimum throughput.

The authors in [5] evaluate the performance of dynamic packet scheduling of 3GPP Universal Terrestrial Radio Access Network (UTRAN) LTE Downlink. They did not mention the service differentiation and dynamic control channel limitations. According to [16], the proposed scheduling enhanced the system performance in terms of throughput and fairness, but the performance depends on the accuracy of the CQI reports. Authors of [25] are analyzing the packet scheduling of mixed traffic in LTE downlink. They concluded that it is very important to perform service differentiation and prioritization of delay sensitive services such as VoIP service, especially when delivered in combination with delay-insensitive services like Hypertext Transfer Protocol/Transmission Control Protocol (HTTP)/(TCP) web surfing or File Transfer Protocol/Transmission Control Protocol (FTP)/(TCP) file download. Additionally, they show that prioritization of such a service as VoIP allows more efficient radio resource utilization, but causes slightly quality degradation of other services. According to [26], packet bundling can provide up to 80% gain to VoIP capacity together with LA.

Recently, the authors of [27] presented a study of VoIP performance in DL with realistic PDCCH including different packet scheduling schemes and CQI reporting resolution. The results show that the dynamic packet scheduling provides better VoIP capacity than semi-persistent scheduling with packet bundling. Dynamic packet scheduling provides about 10% capacity gain with Link Adaptation (LA) and full band CQI. The gain from dynamic packet scheduling comes from a combination of LA and packet bundling, but the capacity of this scheduling scheme is limited by the Control Channel (CCH) overhead, because the PDCCH saturates before the Physical Downlink Shared Channel (PDSCH) bandwidth is completely used.

In [28], urgency and efficiency based packet scheduling (UEPS) was proposed to support real-time (RT) and non-real-time (NRT) traffic. UEPS serves NRT packets until RT packets approach their deadlines, then RT packets are scheduled with higher priority during their marginal scheduling time interval. It tries to maximize the throughput of NRT traffic with satisfying the QoS of RT traffic. However it is not always an effective way that NRT packets have high priority over RT packets that have some time before their deadlines.

Furthermore, there have been many schedulers proposed that support specific QoS parameters in wireless environments. For example, Modified Largest Waited Delay First (MLWDF) scheduler [20] and exponential rule scheduler [29] consider both maximum allowable delay and instantaneous channel rate, respectively. It was proven that the two schedulers are throughput-optimal and keep all queues stable. MLWDF uses the head-of-line packet's waiting time in the queue or the total queue length as scheduling metric.

## Chapter 3

### THE PROPOSED PACKET SCHEDULING ALGORITHM ARCHITECTURE

#### 3.1. The Proposed Packet Scheduling

The packet switching networks is emerged to serve data services, consequently the main goal of IP based networks are to provide data traffic. These networks did not designed to guarantee QoS to BE traffic or provide prioritization to any service; therefore, the throughput depends on channel situation and quality, and on the network traffic load. As a result there will be unsatisfied users. But LTE is designed to support all services in packet switching scheme. So both data and voice services can be provided by LTE technology. Consequently, it is hard to provide QoS for VoIP traffic in LTE, because the packets delay or packets loss has bad impact on VoIP service. Thus, an algorithm will be provided, that has the ability to provide the required QoS and achieves users satisfaction with high probability.

If we give a strict prioritization of VoIP packets over other BE packets, we will get satisfied QoS of VoIP users, but the overall system performance will be degraded. This can simply be concluded, because when the VoIP packets have a strict prioritization, the BE packets may have not any Physical Resource Block (PRB) to use.

In this work; the BE throughput will be maximized while the QoS of VoIP traffic is supported. Also negative impact of having a mixed traffic on the overall system performance will be minimized. To achieve this, we have to differentiate the traffic and prioritize the services in a good manner, using a well-designed packet scheduler.

The PS algorithms are the most crucial functions in the LTE communication network systems. A scheduler assigns the available resource blocks (time and frequency) among users terminals.

The packet scheduling can be defined as:

"The process of assigning users' packets to appropriate shared resource to achieve some performance guarantee" [30].

### **3.1.1. General Introduction**

In this section, a general introduction to the proposed packet scheduling scheme, system model and simulation environment is presented. It is necessary for the understanding of the proposed schemes and our modeling choices.

In this study; an efficient well-designed packet scheduler is introduced, so that the NRT traffic is maximized while the QoS of VoIP traffic is provided, and the system overall performance is improved in term of system throughput and user fairness among the users. Users prioritized depend on both the QoS and CQI after the classifier sort the users into two independent groups based on the services type. Every Transmission Time Interval (TTI) the priority metrics are updated, so that multiuser diversity is exploited in the TD and FD. And the LA can choose a higher MCS to improve the system overall throughput.

As we concern in a mixed traffic over LTE system. For the efficiency of PS it is necessary for a classifier to differentiate the traffic, and to set two separate queues based on the services type. Then each queue is given a certain priority, thus, each service can be handled independently.

We consider that both RT e.g., VoIP and NRT e.g., web surfing services are concurrently running at a user terminal in the LTE networks, so it is very important to differentiate the traffic and prioritize the services of delay sensitive. Since it enhance the radio resource utilization.

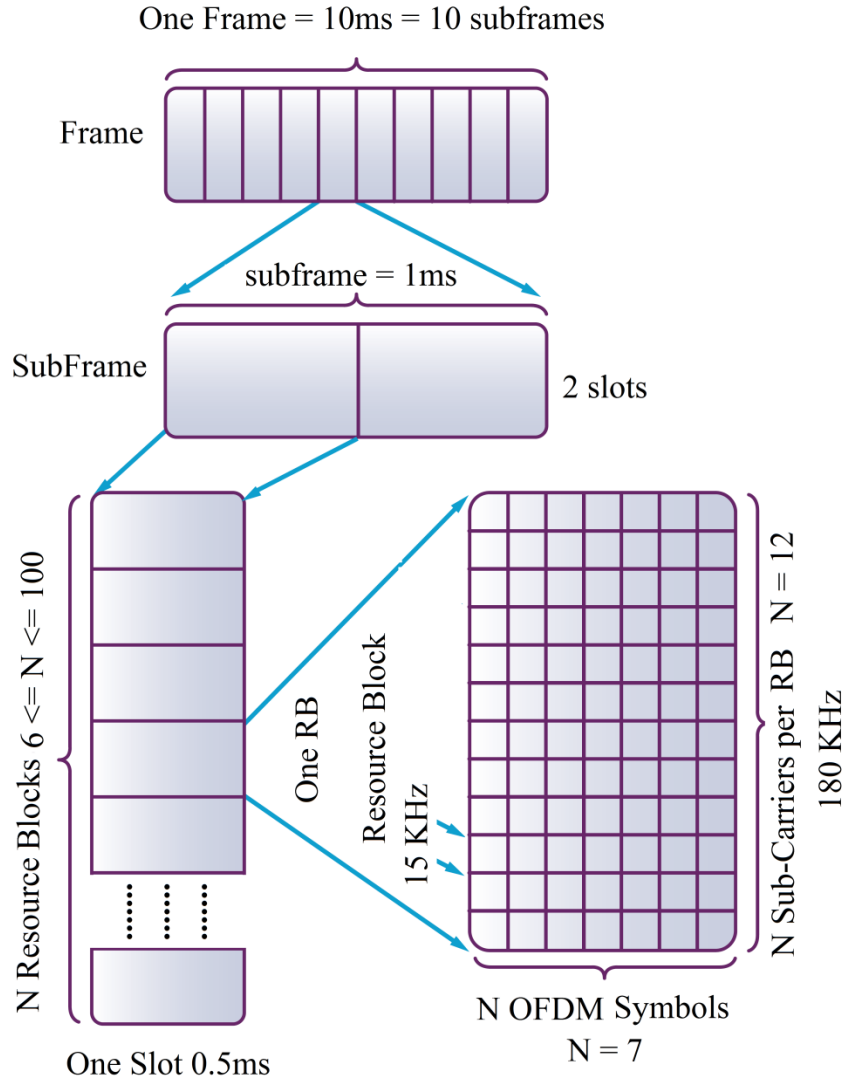
The accepted delay of VoIP, according to [31], the maximum acceptable mouth-to-ear delay for voice is around 250 ms; assuming that approximately 100 ms delay for CN, the tolerable delay for RLC and MAC buffering, scheduling and detection should be strictly lower than 150 ms. Hence, assuming that both end users are LTE users, tolerable delay for buffering and scheduling is lower than 80 ms. A delay bound of 50

ms (for delay from eNB to UE) has been chosen for the 3GPP performance evaluations to better account for variability in network end-to-end delays.

- The system capacity is defined as the number of users in the cell when satisfying more than 95% of the users.
- A VoIP user is satisfied if more than 98% of its voice frames are delivered successfully within 250 ms.

### **3.1.2 System Model**

Long Term Evolution downlink transmission which based at OFDMA is used, PRB is the transmission resource in OFDMA based wireless systems; The PRB is the smallest allocation transmission unit. The standard recommendations in [32], those 12 sub-carriers and 2 time slots each of 0.5ms are contained in one PRB. Since the subframe duration is 1ms. The TTI is 1ms duration. The PRB has a grid of frequency and time, the frequency side consists of 12 sub-carriers each sub-carrier spacing is 15 kHz, and the time side is one TTI of 1ms. Each subframe contains 7 OFDM symbol since we use normal cyclic prefix. The LTE frame structure consists of 10 subframes and its duration is 10ms. The frame structure is illustrated in Figure 3.1.



**Figure 3.1: The LTE frame structure.**

The system consists of  $K$  mobile users and  $M$  PRBs. The homogeneous power allocation is used as many papers assumed, as in [7] [33] [34], the same power is allocated on all sub channels  $P_m(t) = P/M$ .

Where,  $P$  is the total transmit power of eNodeB and  $M$  is the total number of sub channels. The CQI is known by the eNodeB at each scheduling drop.

The user achievable throughput is calculated by used (1).

$$\text{Achievable Throughput (Mb/s)} = \frac{TB\_size}{TTI} \times \frac{feedback\_ACK}{1e6} \quad (1)$$



where,  $TB\_size$  is the Transport Block size,  $feedback\_ACK$  is  $ACK/NACK$  UE feedback and  $TTI$  is the time in seconds.

### 3.2. Related Work

Many previous studies in packet scheduling are reviewed in section 2.3. Literature Review. However, the discussed schemes were talking about packet scheduling in general, but in this section more specific related work is reviewed. Especially those algorithms that were designed to schedule a multi-service mixed traffic environment such as LTE and LTE-A systems.

The current work related to packet scheduling in OFDMA system such as LTE and LTE-A takes into account both the CQI and Queue State Information (QSI), which help improving the support of QoS provision to RT and NRT traffic types and effectively use available radio resource by exploiting multiuser diversity both in time domain and frequency domain.

In [34], a service differentiation packet scheduling architecture is presented which classify mixed traffic into different service classes and grants different scheduling priorities to them. Such as, strict priority and fair queuing type per-queue scheduling algorithms have been applied. Two types of traffic are used, VoIP and BE are considered and the results show an improvement in RT QoS at the cost of system spectral efficiency, when the RT queue is granted the highest priority. They concluded that:

- Prioritizing of VoIP traffic is needed to keep user satisfaction under the defined constraints.
- The scheduling of VoIP with strict priority however reduces the system spectral efficiency into about 76% of pure BE spectral efficiency due to scheduling of smaller VoIP packets.
- Fair queuing provides better BE user throughput but at the cost of VoIP user satisfaction.
- Queue-specific sorting algorithms are needed to prioritize UEs within each queue.

Therefore, we can conclude that, this packet scheduling algorithm has the ability to improve the QoS of RT and NRT services, but it has not the ability to enhance the overall system performance in terms of overall system throughput and fairness among the users at the same time.

The authors in [35] have presented a QoS aware PS framework that is composed of three main units for the resource allocation in DL transmission for OFDMA-based networks. These units use different queue sorting, TD adaptive scheduling and FD scheduling algorithms to guarantee better QoS to different traffic types. System spectral efficiency is improved by optimizing the use of given radio resources and a certain degree of fairness among users is maintained at the same time. This is achieved by providing enough resources to RT traffic and distributing remaining resources efficiently to NRT services in adaptive way. The results show an improved QoS of RT traffic, since a delay dependent queue-sorting algorithm is used and users with relatively low channel conditions but more waiting time are scheduled to guarantee QoS of RT traffic. This leads to degradation in the system overall throughput. Also the user fairness is less than the PF algorithm which takes as a reference for fairness analysis.

According to [36], a Cross Layer Packet Scheduling Architecture (CLPSA) is proposed with service specific queue sorting and adaptive TD scheduling algorithms. The mixed traffic is differentiated into service specific queues, users are sorted into queues and available resources are adaptively reserved to RT and NRT traffic types. The QoS is improved in terms of average delay and Packet Drop Ratio (PDR) of RT traffic because the algorithm allocates just enough resources to each traffic type. As a result the average achieved throughput of NRT is degraded.

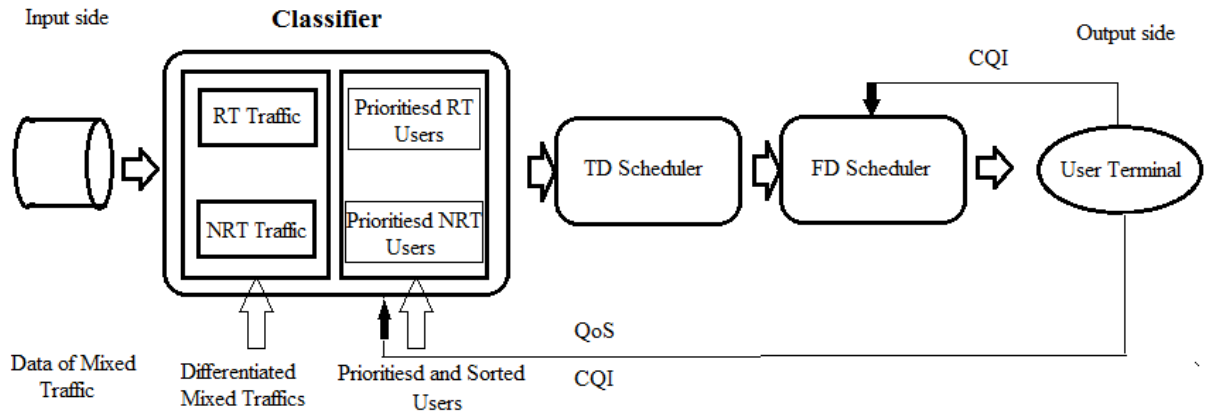
In [37], a Service Specific queue Sorting Algorithm (SSSA) is presented for RT, NRT streaming video services and BE traffic. The proposed SSSA is implemented in a QoS aware PSA for the LTE-A downlink. The SSSA packet scheduling enhanced the QoS by reducing the average delay, delay viability and PDR of RT traffic and satisfies minimum throughput requirements of NRT streaming video traffic however the achieved throughput of NRT streaming video traffic is lower than the reference throughput of NRT traffic, because SSSA allocates just enough resources to meet

throughput requirement. The throughput of BE traffic is decreased, because the proposed algorithm just allocate enough resources to RT and NRT users and the remaining of the resources are allocated to the BE users.

### 3.3. The Proposed Packet Scheduling Algorithm Architecture

The architecture of the proposed packet scheduling algorithm consists of three steps: Classifier, time domain scheduler and frequency domain scheduler. The classifier has two functions; the first function is to differentiate the mixed traffic into two groups, one for RT traffic such as VoIP users and the other group for NRT traffic such as FTP users, depending on the information taken from the application layer. The second function is to priorities the users at each group and sorts them in queues depending on reports of QoS and CQI taken from the physical layer, according to a certain equations in order to achieve the main algorithm goals. The users inside the queues are prioritized from the top to the bottom.

The proposed packet scheduling architecture is depicted in Figure 3.2.



**Figure 3.2: The proposed packet scheduler architecture.**

### 3.3.1. Queues Prioritizing and Sorting

- **RT traffic "VoIP users"**

First the users will be checked, the users have data packets to be sent are considered to be scheduled. The prioritizing metrics of the RT traffic depends on both QoS required for RT users and the CQI. The RT user waiting time is the time that the active RT user has not been allocated or delayed, and this should not exceed the upper bound of delay for RT traffic, the standard value in LTE networks is 40ms [7]. The waiting time and CQI are updated every TTI, as a result both the user delay and PDR will be reduced, the fairness and system overall throughput will be improved.

The priority of an RT user  $k$  can be calculated by:

$$\mathbf{Priority}_{voip} = T_k^{\text{delay\_time}} * CQI \quad (2)$$

where,  $T_k^{\text{delay\_time}}$  is the waiting time of VoIP user and  $CQI$  is the instantaneous channel quality indicator.

In each TTI, the user with the highest priority value is sorted at the front of the queue followed by users with priority value in descending order.

- **NRT traffic "FTP users"**

The average throughput and CQI are used to determine the priority metrics. The average throughput is used to maintain the fairness between the system users and the CQI to improve the system overall throughput. The priority order can be calculated by:

$$\mathbf{Priority}_{ftp} = CQI / R_k(t) \quad (3)$$

where,  $R_k(t)$  is average throughput of user  $k$ .

The average achieved throughput of the user  $k$  is updated using the moving average formula, as used in [38] [39] and many others papers, and can be calculated by:

$$R_k(t + 1) = (1 - 1/tc)R_k(t) + (1/tc) \sum_{k,m}^M r_{k,m}(t) \quad (4)$$

where  $tc$  is the length of time window to calculate the average throughput,  $1/tc$  is called the attenuation coefficient with the widely used value 0.001,  $r_{k,m}(t)$  is the acquired data rate of user  $k$  at PRB  $m$ , if  $m$  is allocated to  $k$  else it is zero [37].

### 3.3.2. Time Domain Scheduler (TDS)

The main function of the TDS is to pick a set of users with the highest priorities equal to available PRB to be allocated at Frequency domain scheduler. The decision of user selection must guarantee the QoS requirements of RT users first, then the NRT traffic. If there is a RT user in a deep fading and have bad CQI, this user will not be allocated and will be dropped and replaced with other user. The CQI will be updated and check every TTI.

### 3.3.3. Frequency Domain Scheduler (FDS)

The selected set of users at TDS will be passed to the FDS, and the actual resources allocation to these users is done by the FDS. For each user the FDS check the available PRBs and allocates the best PRB; which has the highest CQI value. And depending on the value of the reported CQI, the MCS will be determined and assigned to the scheduled user.

### 3.3.4. The Proposed Packet Scheduling Algorithm:

At a given time  $t$ , PRBs are allocated to users by the following steps:

**Step 1:** Differentiate the mixed traffic into two groups and initialize the number of PRBs.

**Step 2:** Priorities the users at each group and sorts them in queues depending on (2) and (3).

**Step 3:** Sort PRBs and determine PRBs group to each queue.

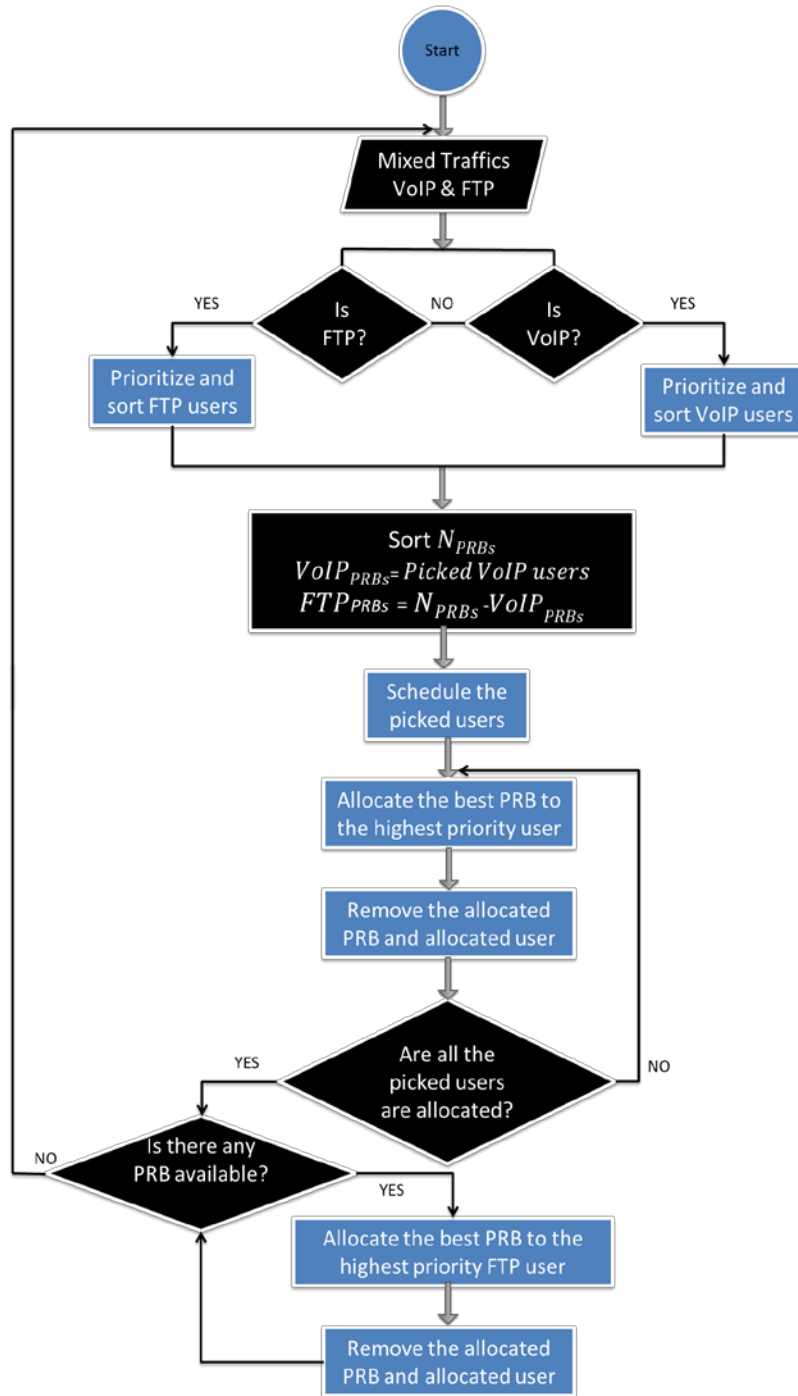
**Step 4:** Pick a set of prioritized users from the queues based on the resources reserved for VoIP and FTP traffic.

**Step 5:** Allocate the best PRB to the user with the highest priority.

**Step 6:** Remove the allocated PRB from the PRBs list and the allocated user from the users list.

**Step 7:** Go to step 5 if the PRBs set are not empty else go to next TTI.

Resource allocation is completed when all PRBs are allocated. Users with no data packets are not considered in FD scheduling. The flowchart of the proposed algorithm is depicted in Figure 3.3.



**Figure 3.3:** Flowchart of the proposed scheduling scheme.

### 3.4. Performance Metrics, Simulation Model and Results

#### 3.4.1. Performance Metrics

The proposed packet scheduling algorithm is evaluated under performance metrics of QoS of both traffic types; RT and NRT, overall system throughput and user fairness.

We used PDR and delay viability of RT such as VoIP and average achieved throughput of NRT such as FTP. The PDR can be calculated by: [6].

$$PDR_k = P_k^{dropped} / P_k^{total} \quad (5)$$

where,  $PDR_k$  is the packet drop ratio and it is the ratio of dropped packets to the total number of packets of a user  $k$ ,  $P_k^{dropped}$  is number of dropped packets of user  $k$  and  $P_k^{total}$  is the total number of packets generated by user  $k$ , and it used to measure the QoS for RT (VoIP) users. And the delay violation probability is given by: [6].

$$Delay\ viability = \max_{k \in RT} (PDR_k) \quad (6)$$

where,  $PDR_k$  is the PDR of RT user.

The minimal average throughput across all the NRT (FTP) traffic users is taken as the minimum throughput to measure the QoS for NRT users, and it can calculated by: [6].

$$r_{min} = \min_{k \in NRT} r_k \quad (7)$$

where,  $r_k$  is the achieved throughput by NRT traffic user  $k$ .

The overall system average throughput is the sum of average throughput achieved by all the system users.

The Raj Jain fairness index is used to measure the fairness between the system users, and it can be calculated by: [7].

$$\mathbf{Fairness} = [ \sum_{k=1}^k R_k ]^2 / \mathbf{K} \sum_{k=1}^k (R_k)^2 \quad (8)$$

Where,  $R_k$  is the time average throughput of user  $k$  and  $\mathbf{K}$  is the total number of the system users. The highest fairness among the users is 1, and this occurs when all the NRT users have the same data rate.

### 3.4.2. Simulation Model

The evaluation of the packet scheduling algorithm is based on the 3GPP UTRAN LTE downlink specifications and recommendations, so the simulation parameters and assumptions used for system level simulation are described in [1]. A lot of papers used these typical values. In this work, the values and the parameters are considered as one omnidirectional eNodeB in a single cell with 10 MHz total system bandwidth, the total system bandwidth is apportioned into 50 PRBs. Each PRB consist of 180 kHz in the frequency domain and 2 slots each of 0.5ms in time domain. The total eNodeB transmission power is 40w (46dBm), the carrier frequency of 2 GHz is used, a typical Urban Non Line of Sight (NLOS) wireless environment is considered, the path loss model is TS36942 is used, the number of users in the cell is constant and have random distribution and the number of VoIP users is considered to be equal to the number of FTP users as in [6]. The aforementioned specifications and parameters are listed in Table 3.1. Link adaptation selects the Modulation and Coding Scheme for each user based on CQI measurements.



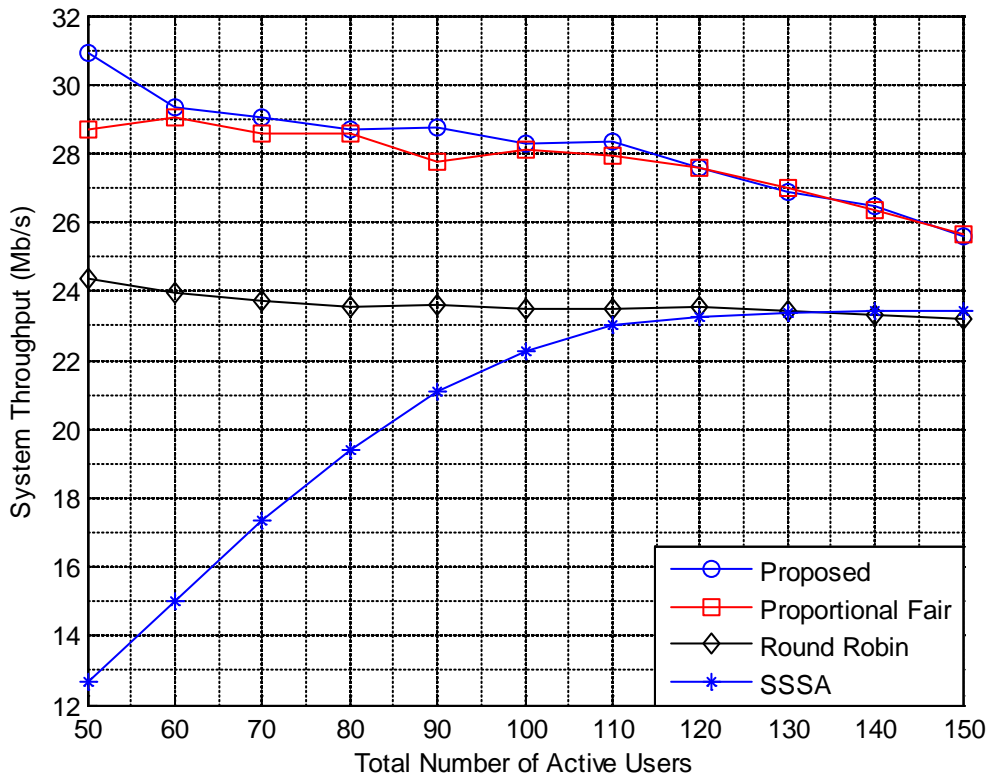
**Table 3.1: Simulation parameters values.**

Parameter	Value
Number of cell	One cell
Cell Radius	1000 m
Total system bandwidth	10 MHz
Number of PRB	50 (12 Sub-carriers / PRB)
PRB bandwidth	180 kHz
TTI Duration	1 ms (14 OFDM Symbols)
Carrier frequency	2 GHz
eNodeB total transmission power	40w (46dBm)
Users distribution	Random
User Velocity	3 kmph
Shadow fading standard deviation	8 dB
Smallest distance from UE to eNodeB	35 m
Path Loss model	TS36942

### 3.4.3. Simulation Results

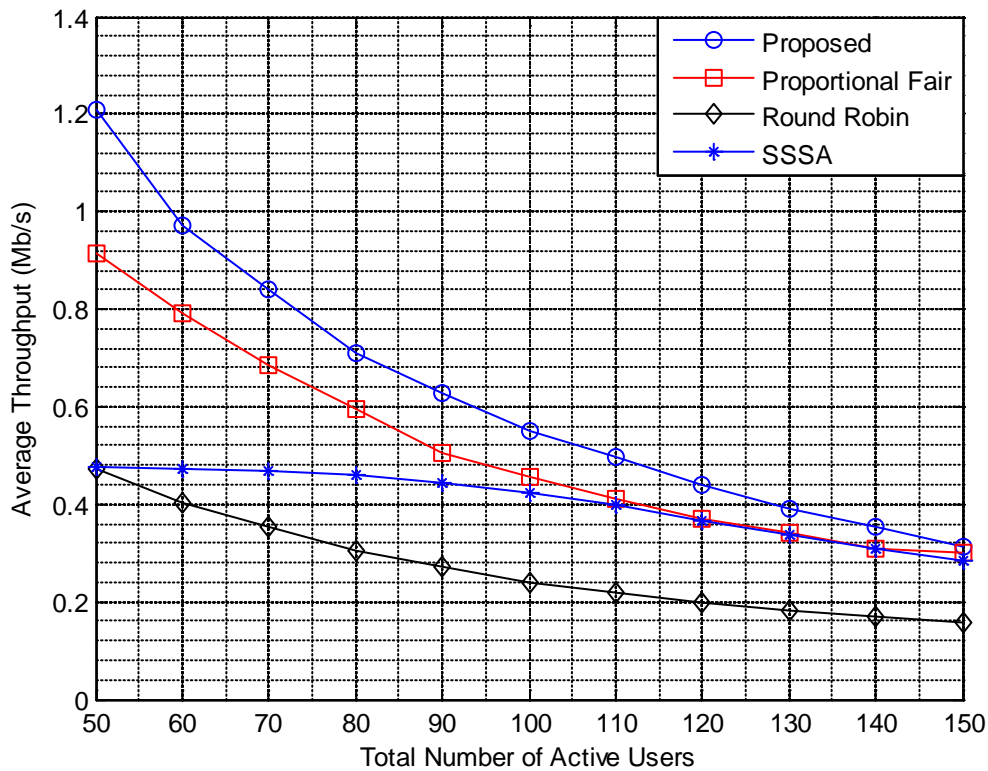
The performance of the proposed algorithm is evaluated by compared with RR and PF, those algorithms are explained in section 2.2.6., titled Scheduling Schemes, they are built in at the used simulator in reference [8], and with previous work called Service Specific queue Sorting and scheduling Algorithm (SSSA), for more details refer to reference [37].

The overall system throughput versus total number of users is shown in Figure 3.4. The proposed scheduling algorithm shows higher system throughput as compared with RR and SSSA. These good results are obtained because multiuser diversity is exploited by the proposed algorithm through updating priority metrics during each scheduling decision, by calculating the CQI during each TTI, and makes better resource utilization. Note that the performance of the PF is nearly, the same as the proposed scheduling algorithm because they use the same priority metrics. Additionally at higher system load the overall system throughput is slightly decreased because of the increase of number of the RT, “VoIP” users. Therefore they utilize more radio resources. For the SSSA scheduling algorithm at the lower system load it shows extremely degradation in the system performance, because this algorithm does not use the whole available free radio resources and it just assigns only one PRB to each user. In the lower system load the number of PRBs is greater than the number of users who have data to be sent. However, the overall system throughput of the RR scheduling is reasonable because it does not exploit the multiuser diversity and does not use any priority metrics; it just assigns the radio resources by turn, i.e. user by user.



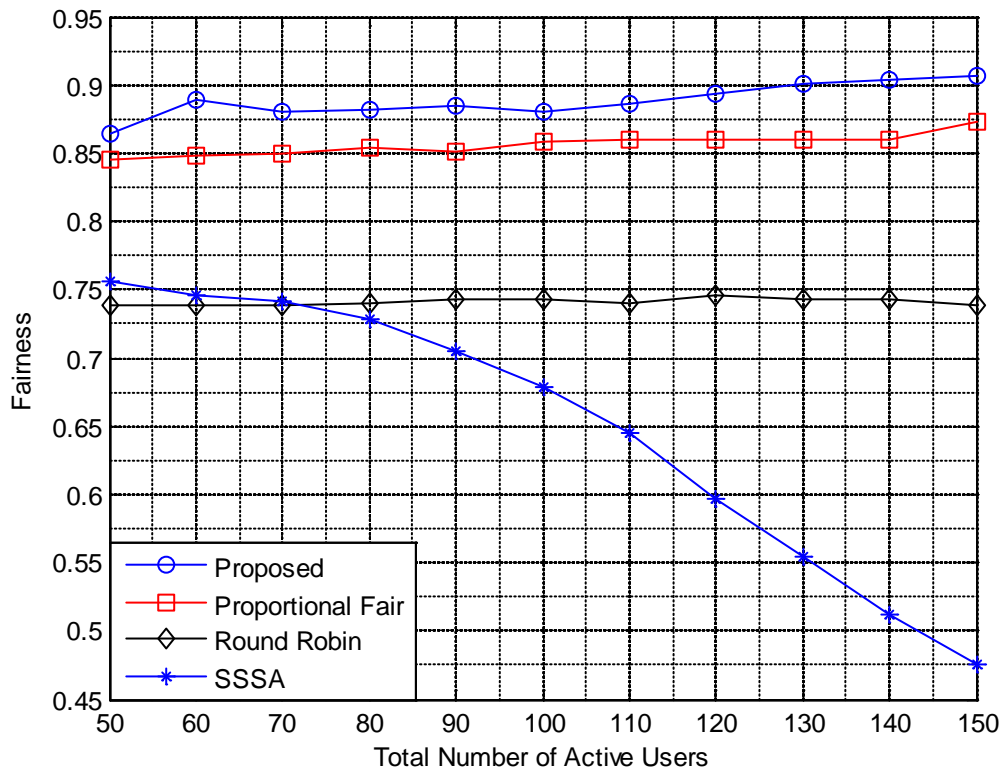
**Figure 3.4: System throughput.**

Figure 3.5 shows average achieved throughput by SSSA, RR, PF and the proposed scheduling algorithms for BE traffic “FTP traffic “ which does not have any QoS requirements. The proposed scheduling algorithm shows higher throughput because it assigns the best carriers among the total BW to the users at every TTI. It means that each TTI the proposed scheduling algorithm checks the total BW and sorts all the RBs and assigns the best RB to the user that has it. At lower system load, it is obviously, that the proposed scheduling algorithm is the superior to all the other scheduling algorithms, and that is because in addition to the above explanation, the algorithm assigns more RBs to the FTP users because not all the VoIP users have data to be sent.



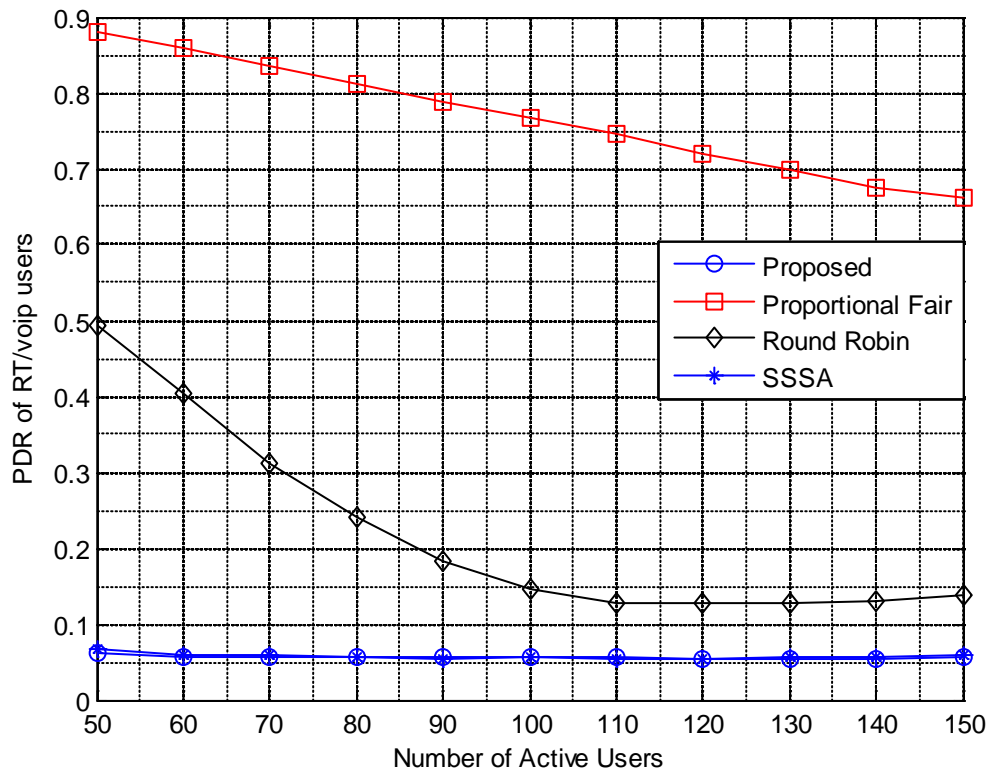
**Figure 3.5: Average achieved throughput of BE users.**

Fairness among users is calculated and has been shown in Figure 3.6. Fairness achieved by the proposed scheduling algorithm is improved significantly as compared to RR and SSSA. Because the proposed scheduling algorithm takes into account a fair share of resources among users. However, the proposed scheduling algorithm is updating the average achieved throughput and CQI of all users during each scheduling decision, allocates a fair share of radio resources among all users and improves overall system throughput at the same time. Also we can note that the proposed scheduling algorithm and the PF algorithm achieve nearly the same performance because both of them take into account a fair share of resources among users.



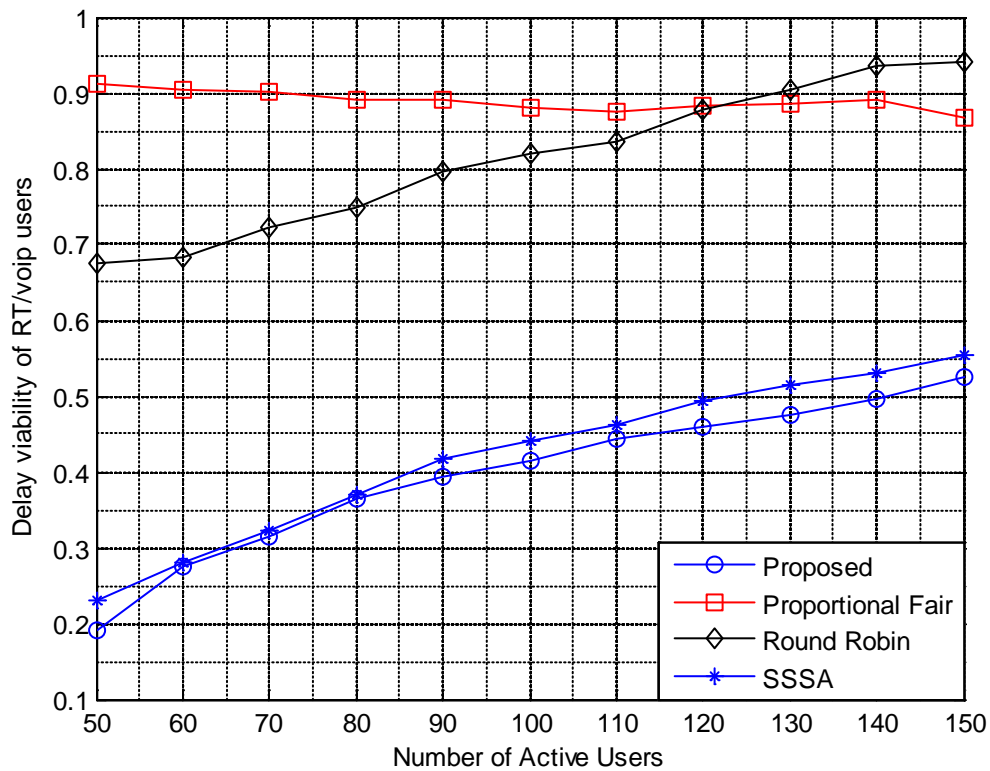
**Figure 3.6: Fairness among users.**

The averaged PDR of RT traffic versus total number of active users is shown in Figure 3.7. Both the proposed and the SSSA scheduling algorithms have the same PDR. This is because the proposed scheduling algorithm takes into account each user's updated waiting time during each TTI to take scheduling decisions, in the same manner the SSSA scheduling algorithm takes into account each user's updated queue length and the waiting time during each TTI to take scheduling decisions. This results in bringing packets with the longest delay at the front of queue thus reducing PDR due to time out. The proposed scheduling algorithm calculates the waiting time each TTI and uses the waiting time in calculating the priorities of the users inside the queues. So the user with the longest waiting time will be put in the top of the queue and will be allocated the radio resource firstly. In the other side both RR and PF take neither waiting time nor the queue length in the consideration when calculating the users' priorities. Therefore, the PDR is very large.



**Figure 3.7: Packer drop rate of RT traffic.**

Figure 3.8 shows that the delay viability of RT users increases with total number of users for both the proposed, RR and SSSA scheduling algorithms. However the proposed algorithm gives the best performance at higher system load. This is because it prioritizes packets with longer delays reducing PDR of RT users due to time out. It is done by updating each user's waiting time during queue sorting at each TTI. Note that the delay viability is completely depends on the PDR.



**Figure 3.8: Delay viability of RT users.**

We can summarize that, the obtained results show that the proposed packet scheduling algorithm is a well-designed packet scheduler. It proves that it is capable to improve the overall system performance. Since it enhances the throughput of NRT traffic such as FTP service, and it achieves a good QoS of RT traffic such as VoIP service. It maintains the fairness among the users and maximizes the overall system throughput. The proposed packet scheduling algorithm deals with mixed traffic that is supported by the LTE networks. The design of the proposed algorithm consists of three stages.

Classifier is the first stage, where the different types of the mixed traffic are differentiated firstly, and then sorted in a good manner depending on a special metrics. In order to improve the QoS of VoIP users we take into account both the waiting time of the users and CQI during the prioritizing the users inside the queue. To enhance the performance of FTP users, we take into account both the average achieved throughput of the user and the CQI during the prioritizing the users inside the queue. By using these

strategies we achieved a good degree of the fairness between the users and the throughput of the FTP traffic is maximized as well as the overall system throughput.

A set of users that have the highest priorities are picked by the TDS, where the TDS is the second stage of the proposed scheduler. The TDS selects the set of users in order to meet the QoS requirements and then these users are passed to the last stage, where the last stage in the proposed algorithm is the FDS. The FDS has the responsibility to assign the available radio resources to the selected users. The resources allocation process is occurred in a way that guarantees and achieves the best utilization of the radio resources, in order to obtain the best performance of the system; this happen when the best PRB with the highest CQI value is allocated to the user with the highest priority, and so on.

Finally, the main reasons that led us to obtain these great results are as follows:

- The use of the classifier that differentiates and sorts the different types of the mixed traffic.
- The good mechanism of the FDS, where the FDS sorts all the PRBs with respect to each user then the best PRB among all the PRBs is assigned to the user with highest priority.
- The metrics that are used to prioritize the users, where they depend on the waiting time of the RT user and CQI, and the averaged achieved throughput of NRT users.
- The exploiting of the users diversity in both FD and TD.
- The use of the different modulation and coding schemes.

In the comparison, the achieved results by the proposed algorithm are better than those of the SSSA algorithm, because of the good mechanism of the FDS, where all the PRBs are sorted with respect to each user first, and then the highest priority user is allocated the best PRB. Also the proposed algorithm is superior to the PF in QoS of RT users in terms of PDR and delay viability, because the PF lacks to the traffic differentiation function and does not take the waiting time of the VoIP packets in



consideration when prioritizing the RT traffic. Finally, the performance of the proposed algorithm is better than the RR algorithm, because the RR uses neither traffic differentiation nor services prioritization.

Also, the proposed algorithm is simpler than SSSA algorithm, because the proposed algorithm uses only the waiting time of VoIP packets and CQI when prioritizing the RT users, while the SSSA algorithm uses the product of normalized waiting time of each user and its channel conditions and the product is added into the queue length of the user.

Finally, some differences are noted between some of the results of the SSSA algorithm in the reference [37] and the results obtained when we built this algorithm. Especially, the PDR and fairness graphs. These differences are occurred because different assumptions are used. The reason behind the difference in the graphs related to PDR is, the authors of the paper in reference [37] used the traffic model is full buffer and we used standards for traffic model of VoIP, where the packets are generated only every 20ms. So related to the used number of the VoIP users in the simulation, there will be available resources all the time for VoIP users to be used. As a result the PDR will be minimized. Also, the difference in the graphs of the fairness because the authors of reference paper used fair scheduling at picking prioritized users from the queue while we used strict priority. In fair scheduling one user is picked from each queue at a time, starting from top queue and in strict priority queues are emptied completely one by one. In our work, VoIP queue is completely emptied first and then the FTP queue.

## **Chapter 4**

### **CONCLUSION AND FUTURE WORK**

#### **4.1. Conclusion**

The PS algorithms are the most crucial functions in the LTE communication network systems, since it enhances the QoS of both RT and NRT traffic. A priority packet scheduling algorithm is proposed, which has the ability to schedule the mixed traffic at the same time. The algorithm aims to maximize the BE throughput while achieves the satisfied QoS requirements of RT throughput. It decreases the negative impact of packets prioritization on the overall system throughput. Differentiating the traffic and prioritizing the services are very important when delivering VoIP with BE traffic like FTP. The proposed algorithm achieved a good results that guarantee a good end to end performance for both voice and data services. The Matlab program is used to build the scheduling algorithm. Using a Matlab based simulator, we analyzed its impact and compared it with other scheduling algorithms in the literature. It was found that the proposed scheduling scheme improves the overall system performance compared with existing scheduling schemes based on throughput and PDR. Also the results show an improved QoS of RT traffic and a better trade-off between user fairness and system overall throughput.

#### **4.2. Future Work**

There are three issues may be addressed to be done in the future work. And they are as follow; first, different mixed services should be used, secondly, adaptive TDS may be implemented and the third issue, the performance of proposed scheduling algorithm should be tested when applying MIMO technique. So that, Future work intends to investigate the performance of the proposed packet scheduling algorithm while applying more than one service as the BE, so we can apply a different mixed group of these services VoIP, FTP, HTTP, video streaming and gaming. Also, the TDS can select the users from the queues in adaptive way, the selection process may depend

on PDR, so at the moment the PDR is small the TDS can select more users from BE queue, and vice versa. In this manner further improvement of QoS of different traffic types will be achieved as well as the overall system performance in terms of system throughput and fairness among users.

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