CHAPTER ONE

INTRODUCTION

CHAPTER ONE

INTRODUTION

1.1 PREFACE

The Long Term Evolution project was initiated in 2004.The motivation for LTE included the desire for a reduction in the cost per bit, addition of lower cost services with better user experience, the flexible use of new and existing frequency bands, a simplified and lower cost network with open interfaces, and a reduction in terminal complexity with an allowance for reasonable power consumption LTE advanced achieve high throughput environment for key facilities with large numbers of users. And achieve high network capacity for areas with high traffic demand and large number of users [1].

The main objective of the Radio Resource Management (RRM) algorithms could simply maximize the system fairness factor; minimize the outage probability and delay. To this end, scheduling, routing, bit loading and adaptive modulation constitute some of the tools that are commonly employed in RRM solutions.

Scheduling is a key for (RRM) mechanism to realizing Quality of Service (QOS) requirements and optimizing system performance of Long Term Evolution (LTE) network.

Scheduling is the process of dynamically allocating physical resources to User's Equipment. (UEs) based on scheduling algorithms implemented at the LTE.

The choice of scheduling algorithms critically impacts resource utilization and the overall performance of LTE network.Using the Best CQI to determine performance metrics, such as: throughput, QOS satisfaction and delay.

In this project a comparison between three algorithms will be done and results will be obtained [2].

1.2 PROBLEM STATEMENT

LTE advanced has the capability to support high transmission rates and QOS for different applications. Due to the limited resources in this network, efforts to improve resource utilization are vital issues. In order to effectively support the heterogeneous traffics expected in this network, great challenges are anticipated in the radio resource management entity.

1.3PROPOSED SOLUTION

The resource management including QoS and fairness aware scheduling are used in order to realize an efficient and optimum network performance.

1.4 OBJECTIVES

1- Investigate and study three of the downlink scheduling algorithms used in LTE system, best-CQI, Round Robin and proportional fair.

2- Find the best algorithm to schedule the users, according to:

- Throughput.
- Delay.

3- Simulate the measures of the scheduling algorithms, and compare between them, using LTE- system level link simulator.

4- The project investigates different algorithms that assign resources to users in terms of fairness, throughput and delay.

1.5 METHODOLOGY

In order for this project to be finalized and complete steps had to be followed, all of them being five procedures.

The first phase being the overview of the LTE and LTE advanced along with their features and their accompanying technologies.

The second phase being also the overview of QoS aware and radio resource management.

Thirdly research is conducted on the algorithms related to scheduling, these algorithms being Proportional fair, round robin and Best CQI , where compared them to each other in the performance metrics.

The Fourth phase an operation was carried out being the code use MATLAB with the results in the end performing a discussion on the matter.

1.6 Thesis Outlines

The thesis is divided into five chapters with an appendix as follows:

First chapter The Introduction that was includes problem statement, proposed solution and objective.

Literature review the second chapter presents some basic background on LTE, LTE-Advanced, radio resource management and describes the main features and technology.

QoS aware radio resource included all the details such as algorithms, blocks diagram and mathematical Equation of performance metrics that was the third chapter.

The fourth chapter Results and discussion: design simulation code by using MATLAB language and provides results form a performance metrics and discussion.

The last chapter Conclusions and Recommendations was explain the result can be achieved and remained future works.

CHAPTER TWO

LITERATURE REVIEW

CHAPTER TWO

LITERATURE REVIEW

This chapter represents a general description of the most important topics related to this work. Overview of LTE and LTE_Advanced are first it described including architecture, feature and technology. Second it described Information of Radio Resource Management.And third described Related Work.

2.1 LTE Overview

LTE is the new standard recently specified by the 3GPP on the way towards fourth-generation mobile. The first Release LTE standard (Release 8) was deployed by the 3GPP, and it has already been finalized with Release 9 as its final version.

2.1.1 Feature of LTE

LTE use scalable bandwidth 1.25 MHZ to currently 20 MHZ , provide data rate up to 100Mbps in DL and 50 Mbps in UL, Spectral efficiency associated to data rate shown above : 5bit/sec/Hz in DL and 2.5 bit/sec/Hz in UL, Latency smaller than 5msec and Source e NB can use the x2 interface to forward DL Packet [3].

2.1.2 Technology of LTE

LTE based on SC-FDMA in uplink and OFDMA in downlink to support Internet service in high mobility and wide range of multimedia. OFDM which provide higher spectral efficiency and more robustness against multipath and fading OFDM divides the data over a number of subcarrier the spacing between two subcarrier is fixed at 15KHZ .In Figure 2.1 A resource blocks (small unit in time and frequency) is consisting of 12 subcarrier in frequency and 14 continuous symbols in Time. To make one resource blocks to span 180 kHz.

Each sub-frame consist of 6 or 7 OFDM symbol, this sub-frame is also minimum transmission time interval (TTI), this choice of short TTI helps to achieve the requirement of Low latency [4].

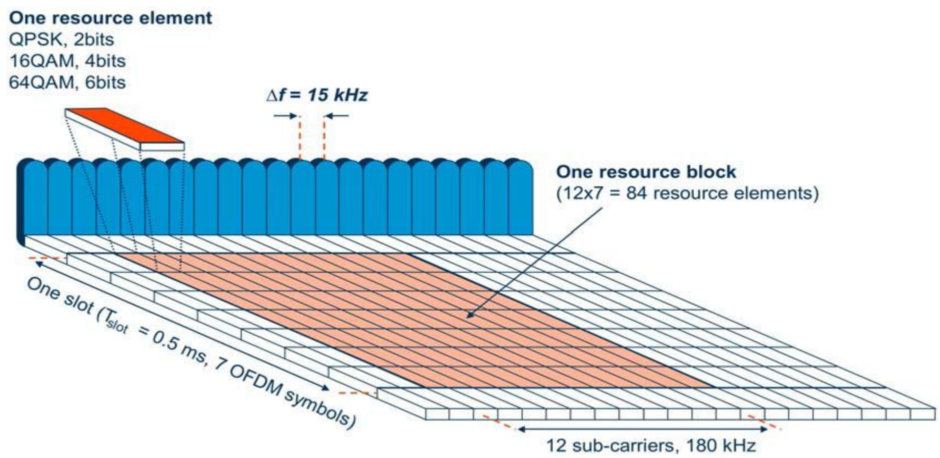

Figure 2.1: OFDM framing

- LTE system support in both FDD&TDD duplex methods.
- Adaptive modulation and coding using QPSK/16 QAM/64 QAM and HARQ in both UP&DL.
- 2 or 4 transmit antenna in DL.
- LTE to support QOS for real time packet data services like VoIP and video streaming [5].

2.2 LTE_Advanced Overview

During the last two decades, telecommunication industry has grown explosively. The huge popularity of Smartphone's has brought the need for mobile broadband networks. Apart from voice transmission, the current mobile networks can provide users with a variety of services, including web browsing, real time gaming, video live streaming.

Users and new applications need faster access speed as well as lower latency while operators need more capacity and higher efficiency. In order to fulfill these demands, the first Release LTE standard (Release 8) was deployed by the 3GPP, and it has already been finalized with Release 9 as its final version. However, the improvements offered by LTE are not enough to fulfill all the requirements for these potential demands. Furthermore, 3GPPkeeps working on further enhancements of LTE. The evolved versions of LTE under work(LTE Release 10 and beyond) are called LTE-Advanced, which is all about even higher-data-rates, higher base station densities and higher efficiencies. LTE - Advanced is able to fulfill the above mentioned requirements.

One of the main goals of this evolution is to reach or even exceed the International Mobile Telecommunications (IMT) Advanced requirements established by the ITU-R [6].

2.2.1 Feature of LTE Advanced

- Enhanced peak rates to support advanced services and applications (enable 100Mbps for high mobility and up to 1 Gbps for low mobility cases).
- A high degree of commonality of functionality world-wide while retaining the flexibility to support a wide range of services and applications in a cost-efficient manner.
- Compatibility of services within IMT and with fixed networks.
- Allow internetworking with other radio access systems.
- Enabling high-quality mobile devices.
- User equipment suitable for worldwide use.
- User-friendly applications, services, and equipment.
- Worldwide roaming capability [7].

2.2.2 System Architecture Evaluation

The figure 2.2 illustrate the architecture of advanced LTE network including several main blocks of the network

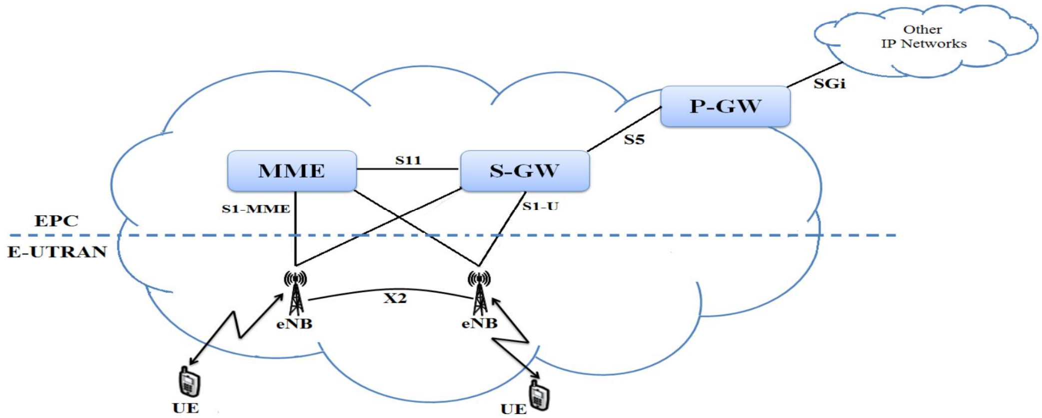

Figure 2.2: LTE-Advanced Network Architecture

2.2.2.1 Evolved-Universal Terrestrial Radio Access Network (E-UTRAN)

The core part in the E-UTRAN architecture is the enhanced Node B (eNodeB or eNB),the evolution of the eNodeB in a 3G system, which communicates with User Equipment(UEs) and it can serve one or several E-UTRAN cells at one time. The eNB nodes are directly connected to each other (this speeds up signaling procedures) through the calledX2 interface.

2.2.2.2 Evolved Packet Core Network

The EPC is an all-IP based core network specified to support the E-UTRAN through a reduction in the number of network elements, simpler functionality and most importantly allowing for connections and handover strategies to other fixed line and wireless access technologies, giving the providers the capacity to deliver a seamless mobility experience. The main components and functionalities of the EPC are as follows:

- . The Mobility Management Entity (MME) is a key control plane element. It is responsible for user mobility, intra-LTE handover as well as security functions (authentication, authorization, NAS signaling). The MME also selects the Serving Gateway (S-GW)and Packet Data

Network Gateway (PDN-GW) nodes. It is connected to the eNBs via the S1-MME interface.

- S-GW is termination node of the EPC. The main aim of the SGW is to route and forward user data packets among different LTE nodes and it also serves as a mobility point for both local inter-eNB handover and inter-3GPP mobility. It is connected to the E-UTRAN via the S1-U interface.
- . The Packet Data Network Gateway (P-GW) provides the UE with access to a Packet Data Network (PDN). The PGW accomplishes policy enforcement, packet filtering for each user or charging support among other functions [6]**.**

2.2.3 LTE Advanced Technology

A number of key technologies enable LTE-Advanced to support the high data rates demanded. OFDM, multiple input multiple output (MIMO), coordinated multi-point (COMP), carrier aggregation, heterogeneous networks, MIMO and Relay.

2.2.3.1 Orthogonal Frequency Division Multiplexing (OFDM)

OFDMA forms the basis of the radio bearer. The radio link uses OFDMA in the downlink and SC-FDMA in the uplink. OFDMA works by modulating data on a number of closely spaced low-rate carriers. As shown in figure 2.3The data in OFDMA systems is spread across a number of sub-carriers. Each sub-carriers a low data rate, which provide resilience against multi-path effects, by allowing the data to be sampled only when it is stable. OFDMA requires precise timing and frequency synchronization. If synchronization is not achieved, then carriers are no longer orthogonal, and the error rate may increase.

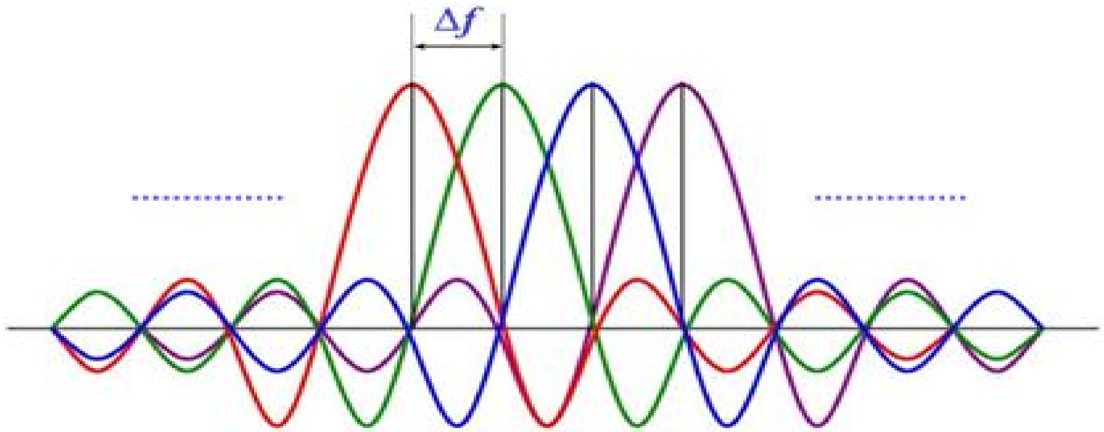

Figure 2.3: Subcarrier spacing in OFDM

2.2.3.2 Carrier Aggregation (CA)

In carrier aggregation, multiple carrier components are aggregated, to provide wider bandwidth for transmission purpose both in DL and UL and guaranteeing increase of network capacity as shown in figure 2.4. It allows the transmission bandwidth up to 100 MHz, by adding five component carriers of 20 MHz bandwidth. CA exploits the fragmented spectrum by aggregating noncontiguous component carriers [3].

Classification of carrier aggregation including : intra band aggregation is the separated to contiguous component carriers and non-contiguous component carriers, and inter band aggregation are shown.

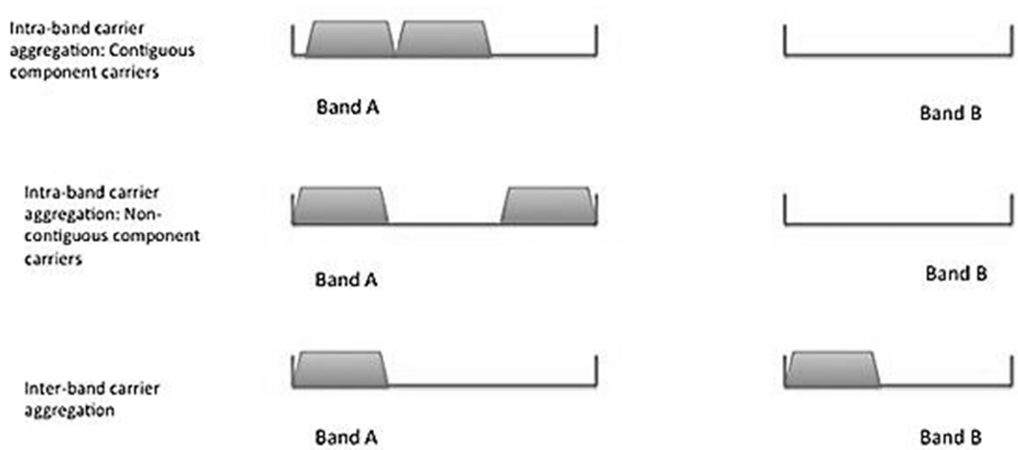

Figure 2.4: Carrier Aggregation

2.2.3.3 Extended MIMO

LTE-Advanced brings technological changes to multi-antenna transmission techniques. Systems with MIMO multiple antennas at the transmitter and at the receiver to provided simultaneous transmission of several data streams on a single radio link shown in figure 2.5, exploiting the so called spatial diversity gain, and increasing system capacity. It supports eight transmission layers in DL while up to four transmission layers in UL [8].

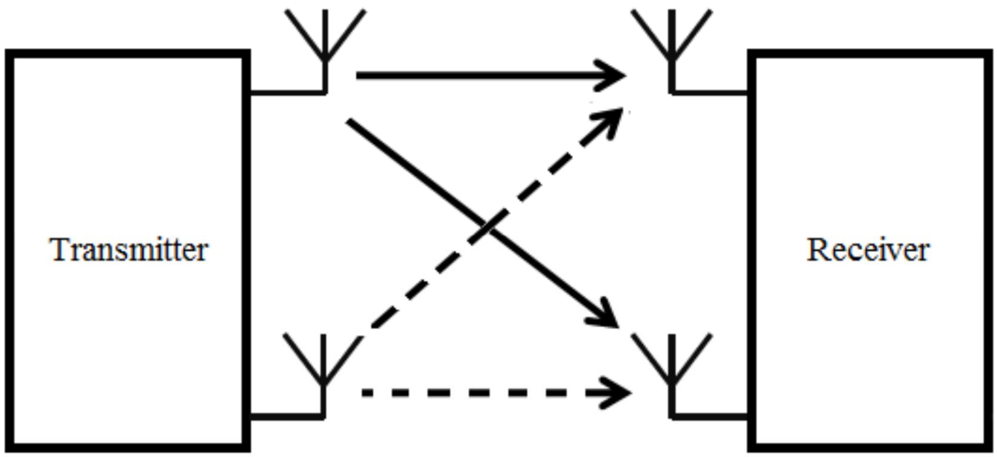

Figure 2.5: 2×2 antenna configuration

2.2.3.4 Coordinated Multi-Point (COMP)

In COMP transmission/reception, multiple geographically separated base station sites coordinate the transmission and reception, in order to achieve good system performance for high data rates, cell edge throughput and end-user service quality. COMP uses of coordination techniques namely as inter-cell scheduling coordination and joint transmission/reception. The former case deals with inter-cell interference between multiple base station sites. In later case, the transmitted/received data signals are jointly processed to enhanced the Transmission/reception performance [9].

2.2.3.5 Heterogeneous Network (HETNET)

LTE Advanced is support needs of heterogeneous networks that combine low power nodes (such as picocells,femtocells, repeaters, and RNs) within a macro cell.

It is a multi-layered network deployment scheme, comprising lower-power nodes, overlaid under the coverage area of a macro-cell. It aims to increase the network capacity as well as achieve peak data rates[10].

2.2.3.6 Relay Station

Relay transmission between a base station and mobile station to extend coverage to various locations. As shown in figure 2.6 There are two major advantages of using relays in cellular networks. The first and obvious advantages are to increase coverage. As relay stations are generally located closer to the cell-edge, they can provide cell-edge users with improved service, thereby increasing network coverage. The other advantage is that relays help improve the capacity of the network.

Relays are two main types. Amplify-and-forwards relays and Decode-and-forward relays [11].

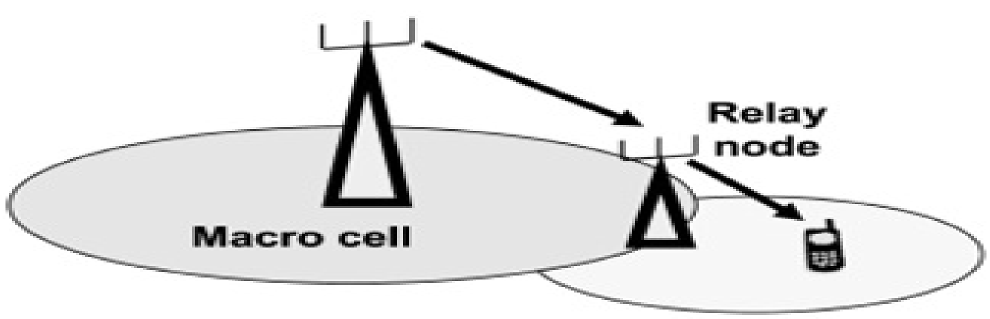

Figure 2.6: The use of relay node to extend cell coverage

2.3 Radio Resource Management

Radio Resource Management (RRM) is used in LTE-Advanced to assure that the available radio resources are utilized as efficiently as possible. In LTE-Advanced, a dynamic RRM is considered, meaning that the radio network parameters are adaptively adjusted to the traffic load, user positions and QOS requirements. For That purpose, Link Adaptation (LA) and other objects like the Packet Scheduling (PS) or Hybrid Automatic Repeat Request (HARQ) play such an important role [6].

2.3.1 Link Adaption

LA is a technique which adjusts dynamically some transmission parameters including modulation and coding schemes (MCS) to the radio channel conditions. However, in downlink transmissions, the eNB does not directly know the actual channel conditions of a certain UE, so it requires a Channel Quality Indicator (CQI) feedback from the UE in order to make a suitable selection of the MCS. This CQI value provides some knowledge about the channel in the latest TTIs, being the result of measurements based on signal to interference-plus-noise ratio (SINR) estimated by listening to some reference symbols. It allows matching the transmission parameters to the variations of that indicator.

Normally, a higher CQI indicates a higher SINR and, therefore, better channel conditions for an UE. The CQI feedback is periodically reported from the UE to the eNB. Apart from the CQI report, information about positive or negative acknowledgments from the HARQ can be involved in the LA operation [6].

2.3.2 Hybrid ARQ

In order to achieve continuous transmission and avoid wasting important time, eight independent S&W HARQ parallel processes can be active at the same time. Every time anew packet is transmitted to the UE, the eNB starts an HARQ process that will be active until the end of the transmission.

Furthermore, two sorts of HARQ schemes are defined. The first scheme is synchronous and non-adaptive, meaning that transmissions and retransmissions can only take place at predefined instants of time. In this case, the eNB knows exactly when and which HARQ process must be processed, avoiding signaling the HARQ process number and the transmission configuration. On the other hand, the second scheme employs asynchronous and adaptive retransmissions. They can occur at any instant of time, being necessary to signal the HARQ process number and transmission parameters [6].

2.3.3 Packet Scheduler

The packet scheduler is the entity responsible for allocation transmission and retransmission requests over the available resources. The PS between a Radio Access Network (RAN) and the users over the air-interface takes a very important role due to the fast changing nature of the channel and the diversity of the channel quality among users.

The overall scheduling decision can be taken simultaneously in time and frequency domain or be divided into two steps: a time-domain packet scheduling (TDPS) and a frequency domain packet scheduling (FDPS).

For the overall packet scheduler framework a simple two step algorithm is considered. First, the TD scheduler selects N users with the highest scheduling priority, being this set of users passed to the FD scheduler in each sub frame. The FD scheduler allocates the available resources to the N selected users. This framework is attractive from a complexity point of view, since the FD scheduler only needs to apply frequency multiplexing of a limited number of N users in each TTI. For this study, however, since the number of users per cell in the system will not be large enough compared with the set value of N during most of the time, the TDPS influence is reduced. Special attention is paid, therefore, on the FDPS.

A proper scheduling operation can be achieved if the packet scheduler is continuously fed with updated information about the link status and retransmissions as illustrated in Figure 2.6, so LTE places it within the eNB. Information regarding the status of the link is achieved by means of the LA functionality. Also, the packet scheduler is in charge of allocating retransmissions ordered by the HARQ processes [6].

2.4 Related Works

In the literature there are various studies about the LTE downlink scheduling and the methodologies that are used to implement it. However, to our best knowledge, we Have two studies that proposes three methods of scheduling; the Best CQI, the ROUND ROBIN and the proportional fair.

In the [12] study an LTE over view is presented, Then they introduced the scheduling methods: The Best CQI which serve users with the highest channel quality, The Round Robin which operates using the concept of first in first serve, and then they introduced another method which operates between the Best CQI and Round Robin which have an acceptable throughput and allow fairness between the users. Finally they proposed the performance of the LTE link level simulator in terms of throughput for different scenarios.

In the [13] study three algorithms are introduced: the Best CQI, the Max SNIR which assigns the PRBs to the UE with the highest Best CQI on that RB, and the Proportional Fair (PF) which tends to be fairer by prioritizing users with higher channel quality with lower throughput. Then they investigate the performance by simulators in terms of resource blocks and throughput.

In the [14] frame of thesis works, a detailed system model of LTE dedicated to radio resource management study has been developed, In a first phase, the study focuses on the dual time / frequency domain packet scheduling concept which allows a linear complexity, in a second phase he was used 10MHz system bandwidth and reduce the complexity of fast fading generation and SINR calculation by a factor of 50 while keeping the average interpolation error of the Effective SINR at 0.5 dB.

CHAPTER THREE

QOS AWARE RADIO RESOURCE MANAGEMENT

CHPTER THREE
QOS AWARE RADIO RESOURCE MANAGEMENT

3.1 Quality of Service (QOS)

In LTE Network QOS is implemented between UE and PDN Gateway and is applied to a set of bearers. 'Bearer' is basically a virtual concept and is a set of network configuration to provide special treatment to set of traffic e.g. VoIP packets are prioritized by network compared to web browser traffic.

3.2 QOS Performance Measures

To achieve a QOS for a certain application, the application requirements must be quantified in terms of parameters that identify the target performance level. Such a level is normally measured in terms of throughput, delay, jitter, and packet loss.

3.3 LTE identifies the following major quantitative parameters

3.3.1 Throughput: Characterized through the Guaranteed Bit Rate, Maximum Bit Rate and Aggregate Maximum Bit Rate.

3.3.1.1 The Guaranteed Bit Rate (GBR): Network resources allocated based on GBR is fixed and do not change after bearer establishment or modification. This is hence a guaranteed service data flow.

3.3.1.2 The Maximum Bit Rate (MBR): This parameter limits the bit rate that can be expected to be provided to GBR bearer, and is enforced by network shaper to restrict the traffic to its maximum bit rate agreement.

3.3.1.3 Aggregate Maximum Bit Rate (AMBR): This parameter is used for non-GBR Flows, and has two types, APN-AMBR and UEAMBR.

The APN-AMBR (Access Point Name-AMBR) is a subscription parameter stored at the HSS per APN. The HSS defines a QCI for each PDN (identifiable by an individual PDN identifier)and an APN-AMBR for each ARP. The APN-AMBR parameter refers to the maximum bit rate that can be consumed by all non- GBR bearers and all PDN connections of this APN. This parameter is enforced by P-GW in the downlink and by both UE and P-GW in the uplink. The UE-AMBR parameter, on the other hand, refers to the maximum bit rate allowed for all non-GBR bearer aggregates for the respective UE. The parameter is enforced in both the downlink and the uplink. Note that GBR and MBR are defined per bearer while the AMBR parameters are defined per a group of bearers. All throughput parameters have two Components, one for downlink and another for uplink.

3.3.2 Delay: Specified by the packet delay budget. LTE defines nine categories for Delay, with 50 ms being tightest and 300 ms being the slackest.

3.3.3Packet Loss: Defined as the Packet Error Loss Rate, and is similar to the Packet delay budget in having nine categories with 10−6being best and 10−2being the worst.

3.3.4 Priority: Specified by the Allocation/Retention Priority (ARP)parameter, this is used to indicate the priority of both allocation and retention of the service Data flow.

3.3.5 Jitter: The frequency of VCO control signal the variation is more rapid the instability of the clock is referred to as jitter.

Two type of jitter:

3.3.2.1 Random jitter: Random jitter (RJ) is unpredictable and has a Gaussian probability density function.

3.3.2.2 Deterministic jitter: Deterministic jitter (DJ) is predictable (assuming prior knowledge of the bit stream characteristics) and definite amplitude limits.

3.3.6 Wander: The frequency of VCO control signal varies slowly the variation is to as clock.

3.4 Scheduling Algorithms

Scheduling algorithms is employed to select different users in time domain and different Physical Resource Blocks (PRBs) in frequency domain depending on the available RB and bandwidth requirements of the user while ensuring fairness and minimum delay. Several scheduling algorithms have been designed for efficient scheduling based on the following three properties: outage probability, throughput and delay to optimize system performance.

It is a must when it comes to this particular network that the users are reinforced and provided with VOIP and video at any given moment in time. Been here work management for the resource blocks for each service through scheduling algorithms , comparison was made between Best CQI and Round Robin , Proportional Fair.

The comparison between in the performance metrics such as: the throughput, QOS satisfaction delay and clearly with simulation results for performance metrics by using code by MATLAB, this results shown in chapter four.

3.4.1 Best Channel Quality Indicator

In this algorithm the system started by measuring the CQIs of all of the users then detecting the user with the highest Channel Quality Indicator (CQI) is served. The CQI of channels can be calculated by the signal-to-noise ratio (SNR), the bit error rate (BER), the signal to interference plus noise ratio (SINR), and the packet loss rate (PLR) .the value of Channel Quality Indicator is took from 0 to 30.

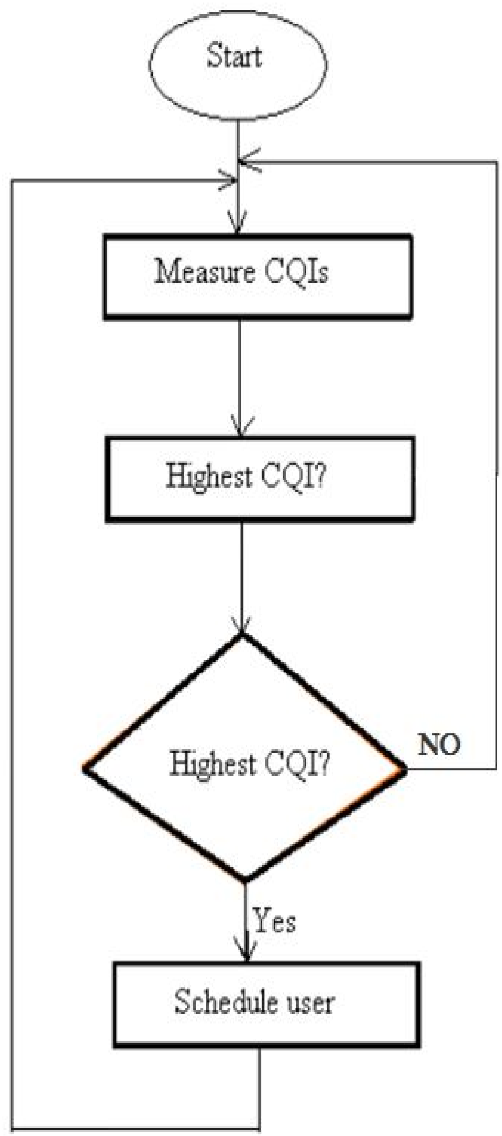

Figure 3.1: Best CQI flow Chart

3.4.2 Round Robin

In this algorithm after starting the system handle all the resources to the 1st user enter to the system and all resources are assigned to the 1stuser at a time and all other users have to be in the waiting queue until their turn comes and the resources get free and rescheduling all the users as First in First Out.

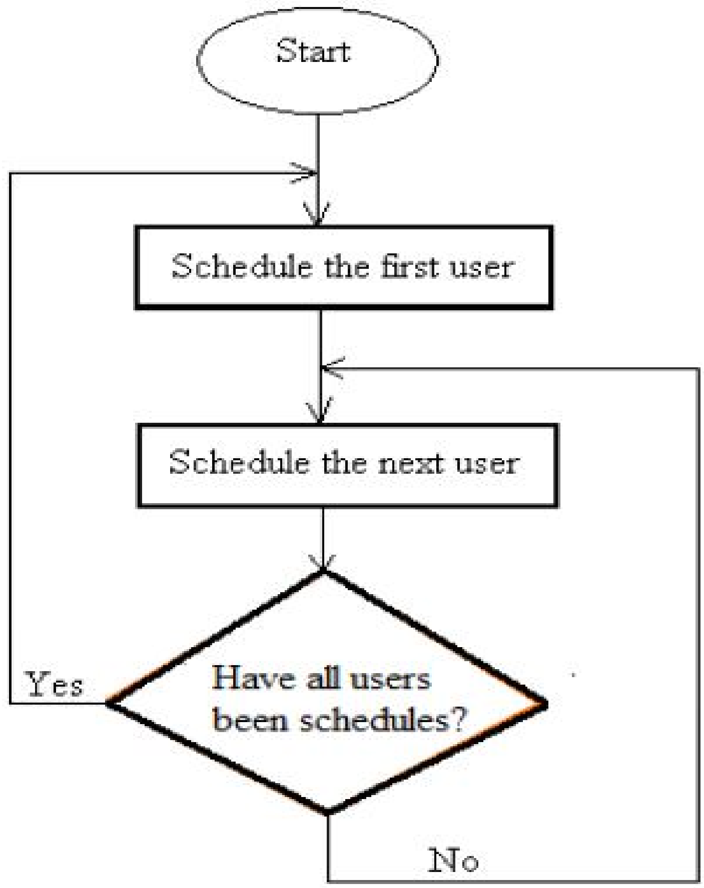

Figure 3.2: Round Robin flow chart

3.4.3 Proportional Fair

All users on the system the system started by measuring the CQIs of all of the users then detecting the user with the highest Channel Quality Indicator (CQI) is served A good trade-off between the overall system throughput and data-rate fairness among the users is achieved by assigning each data flow a data rate or a scheduling priority that is inversely proportional to its anticipated resource consumption.

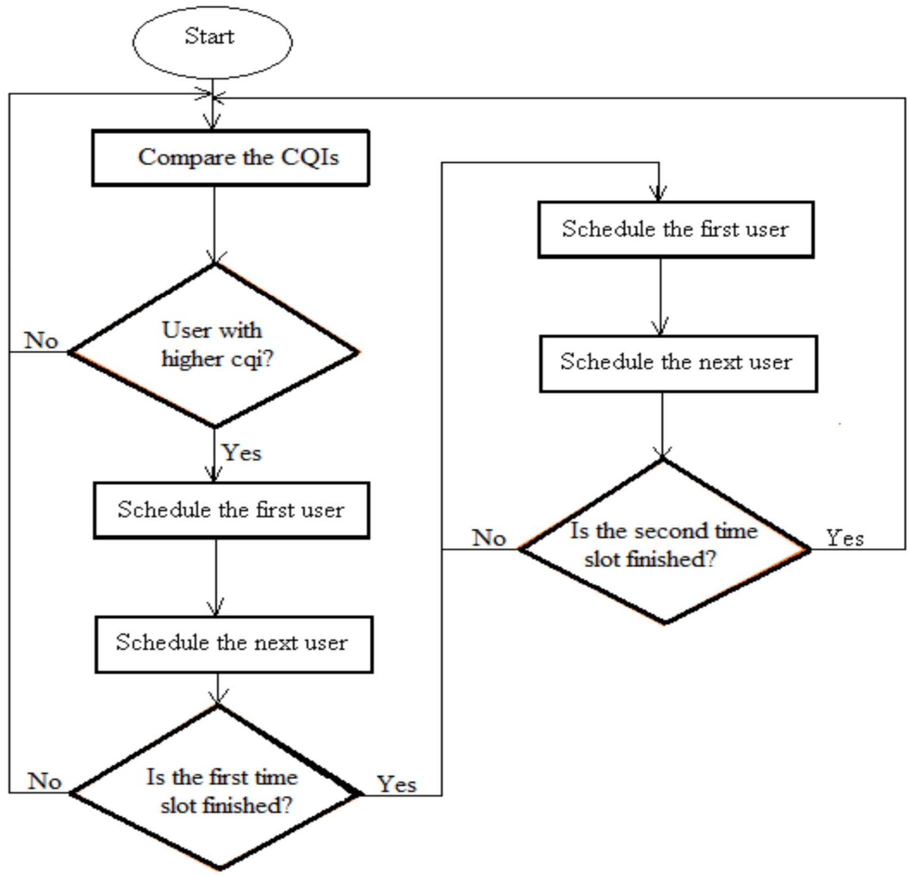

Figure 3.3: Proportional Fair flow chart

3.5 Performance metrics outage probability

The ratio of blocking services offer to active users in specific time period. The outage probability is calculated for any service by equations [3.1] depended on the traffic RB and the available RB equation.

$$Outage = 1 - \frac{Traffic\ RB}{Avilable\ RB} \qquad [3.1]$$

The traffic RB is calculated depend on number of users and the available RB for one user by the equation [3.2]

$$Traffic\ RB = (no._{UE}\ x\ RE_{UE}) \qquad [3.2]$$

The available RB for one user is calculated depends on the data rate and bit rate by equation [3.3]

$$RB_{UE} = \frac{Data\ Rate}{Bit\ rate} \qquad [3.3]$$

The bit rate is calculated depend on the modulation level, number of subcarrier, number of symbols and coding rate by equation [3.4].

$Bitrate = (M\_Level) x (no.\_subs) x (no\_Sym) x (Coding\_Rate \ldots$ [3.4]

Where

M_Level: Modulation Level

No_Subs: number of subcarriers

No._Sym: number of symbols

The datarate in the equation [3.3] of VOIP fixed either for the video; it depends on the bandwidth as shown in equation [3.5]

$Datarate = (Modulation\ Level) x (Bandwidth) \ldots\ldots\ldots$[3.5]

3.6 Throughput

Throughput is the rate of production or the rate at while processing and can be defined as the rate of successful message delivery over a communication channel. The data these messages belong to may be delivered over a physical or logical link or it can pass through a certain network node. Throughput is usually measured in bits per second (bit/s or bps), and sometimes in data packets per second or data packets per time slot.

The system throughput or aggregate throughput is the sum of the data rates that are delivered to all terminals in a network. Throughput is essentially synonymous to digital bandwidth consumption; it can be analyzed mathematically by applying the queuing theory, where the load in packets per time unit is denoted as the arrival rate (λ), and the throughput, in packets per time unit, is denoted as the departure rate (μ).

The throughput of a communication system may be affected by various factors, including the limitations of underlying analog physical medium, available processing power of the system components, and end-user behavior. When various protocol overheads are taken into account, useful rate of the transferred data can be significantly lower than the maximum achievable throughput; the useful part is usually referred to as good put. Equation 3.6

$$Throughput = \frac{Total\ Succesful\ Recived\ Bits}{Time\ in\ Seconds} \qquad [3.6]$$

3.7 Delay Time

Latency is a time interval between the stimulation and response, or, from a more general point of view, as a time delay between the cause and the effect of some physical change in the system being observed. Latency is physically a consequence of the limited velocity with which any physical interaction can propagate. This velocity is always lower than or equal to the speed of light. Therefore every physical system that has spatial dimensions different from zero will experience some sort of latency, regardless of the nature of stimulation that it has been exposed to.

The efficiency of the system can be determined by calculating the delay time through the following equation [3.7]

$$Delay = \frac{Processing\ Time}{Simulation\ Time} \qquad [3.7]$$

CHAPTER FOUR

RESULT AND DISCUSSION

CHAPTER FOUR

RESULT AND DISCUSSION

In this chapter simulation results is written in order to illustrate the three algorithms performance the results of Best channel quality indicator then round robin and finally the proportional fair.

4.1 Simulation parameters

In table (4.1) some of the main parameters was listed including the algorithms used and the number of subcarrier and OFDM sampler, these parameters were used to adjust and set the parameters of the simulation**.**

Table 4.1: Simulation Parameter

Parameters	Value
Scheduling algorithm	CQI and Round Robin, Proportional fair
Band width	1.4M
Number of subcarriers (N1)	12
Transmission scheme	SISO
Signal to noise ratio(SNR)	0,10,20,30,40,50
Number of OFDM samples	14
Total of CP time	6 ms
CQI Frame duration	10ms
Round Robin Frame duration	7ms

4.2 Simulation Scenarios

4.2.1 Best CQI Scenarios

In this scenario three users with different channel quality will be used in the simulation for transmitting their data over the channels one of the users configured as a video conference and the other two users become VOIP users.

The 16QAM modulation technique was used to modulate data of each user and the channel parameters is set as AWGN noise source with a range of 0 to 18 db.

Each channel will be examined with calculating the SNR to BER and the lowest error rate will be processed first.

The best CQI is to indicate the user that has the best channel characteristics by examining the received data on the channels.

Table 4.2: SNR Vs BER

No.	SNR	BER Simulation
1	0	0.555445
2	1	0.522478
3	2	0.466533
4	3	0.414585
5	4	0.367632
6	5	0.308691
7	6	0.26973
8	7	0.220779
9	8	0.162837
10	9	0.122877
11	10	0.084915
12	11	0.052947
13	12	0.032967
14	13	0.01998
15	14	0.00999
16	15	0.002997

4.2.1.1 Simulation Graphical Representation of SNR Vs BER

In figure (4.1) it was found that the increasing of signal to noise ratio decreases the bit error rate the signal to noise ratio starts in the graph from zero to 15 db. Using 64QAM and detecting the SNR to BER to detect the channel quality and as the above graph it was seen that the increasing of signal to noise ratio decrease the bit error rate.

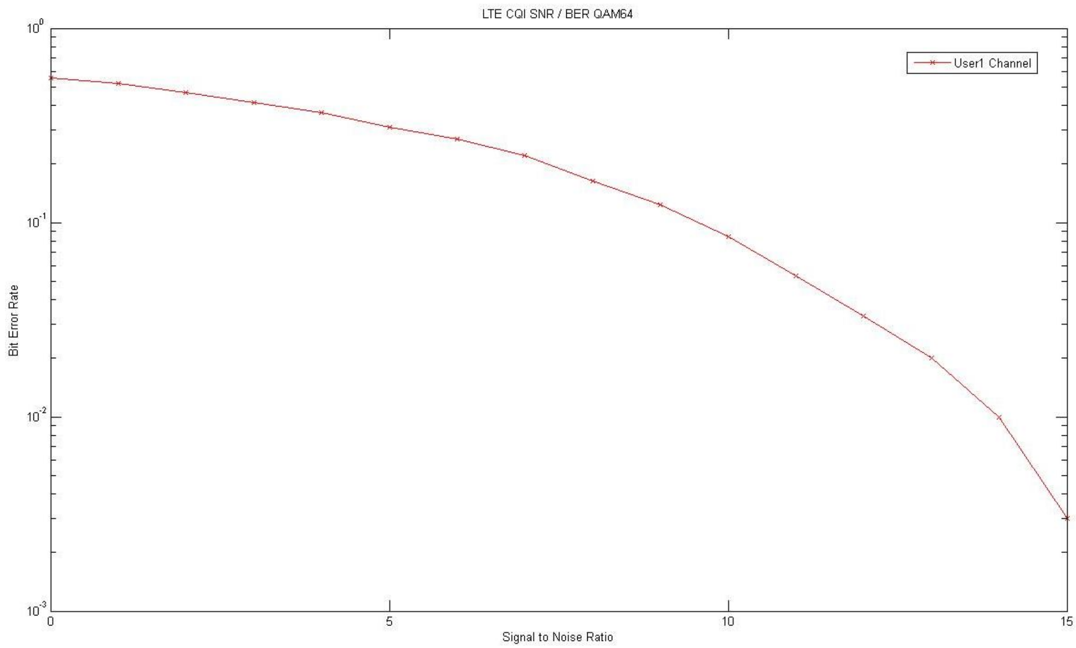

Figure 4.1: LTE CQI SNR Vs BER

Table 4.3: throughput Calculations

No.	SNR	Total Bits	Error Bits	Received Bits	Throughput
1	0	1001	556	445	0.445
2	1	1001	523	478	0.478
3	2	1001	467	534	0.534
4	3	1001	415	586	0.586
5	4	1001	368	633	0.633
6	5	1001	309	692	0.692
7	6	1001	270	731	0.731
8	7	1001	221	780	0.78
9	8	1001	163	838	0.838
10	9	1001	123	878	0.878
11	10	1001	85	916	0.916
12	11	1001	53	948	0.948
13	12	1001	33	968	0.968
14	13	1001	20	981	0.981
15	14	1001	10	991	0.991
16	15	1001	3	998	0.998

4.2.1.2 Simulation Graphical Representation of Throughput

In figure(4.2) it was found that the while the time of the simulation is running the throughput of the system increases to be stabilized on a linear way.

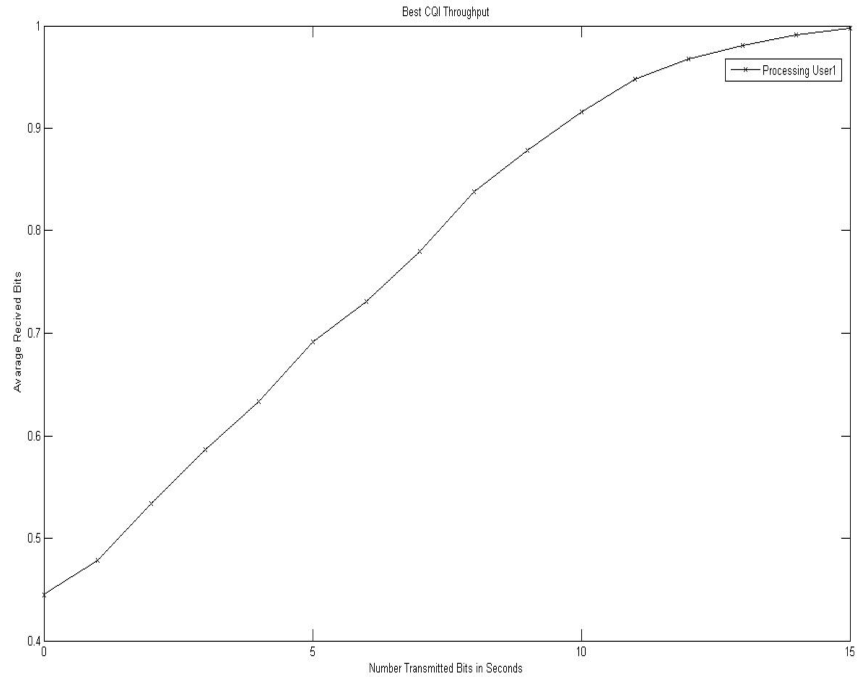

Figure 4.2: The throughput of the best CQI was detected that the increasing of SNR increases the throughput of the system.

Table 4.4: Delay Calculations

No.	SNR	Total Bits	Error Bits	Received Bits	Delay
1	0	1001	556	445	2.667333
2	1	1001	523	478	2.865135
3	2	1001	467	534	3.200799
4	3	1001	415	586	3.512488
5	4	1001	368	633	3.794206
6	5	1001	309	692	4.147852
7	6	1001	270	731	4.381618
8	7	1001	221	780	4.675325
9	8	1001	163	838	5.022977
10	9	1001	123	878	5.262737
11	10	1001	85	916	5.490509
12	11	1001	53	948	5.682318
13	12	1001	33	968	5.802198
14	13	1001	20	981	5.88012
15	14	1001	10	991	5.94006
16	15	1001	3	998	5.982018

4.2.1.3 Simulation Graphical presentation of Best Channel Quality Indicator Delay Time

In the figure (4.3) the user that permitted in the process has a linear delay time depend on the data rate that transmitted through the simulation time which increased due to the number of transmitted bits.

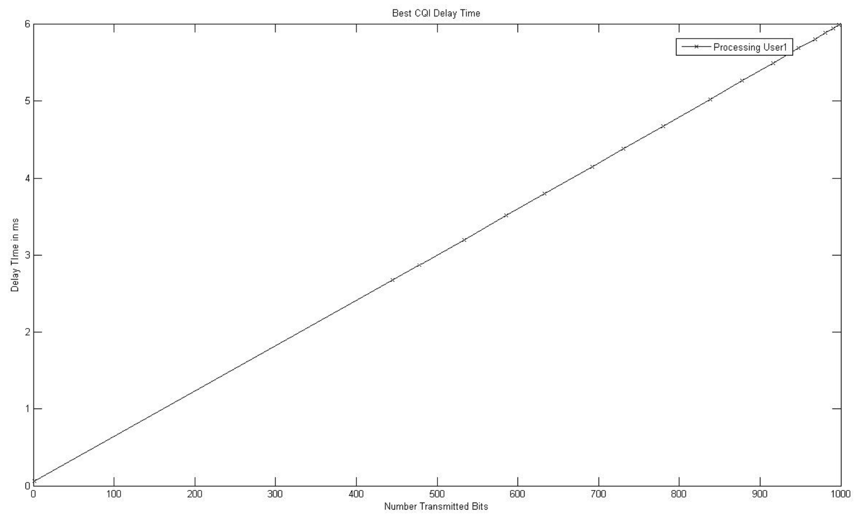

Figure 4.3: The delay time in CQI is discrete depend of user data size .

4.2.2 Scenario Round robin

In this scenario three users with different Loads will be used in the simulation for transmitting their data over the channels one of the users configured as a video conference and the other two users become a VOIP users.

The 16QAM modulation technique was used to modulate data of each user and the channel parameters is set as AWGN noise source with a range of 0 to 18 dB and calculating the time of entry of each user the system will pass the first in and the processing as FIFO.

In the round robin scenario the users are processed depends on the time of entering the system. The table (4.5) represents the delay time of each user on the system based on the traffic type.

Table 4.5: Users Delay Time

No.	SNR	Total Bits	Delay User1	Delay User2	Delay User3
1	0	1001	2.667333	2.934066	3.067433
2	1	1001	2.865135	3.151649	3.294905
3	2	1001	3.200799	3.520879	3.680919
4	3	1001	3.512488	3.863737	4.039361
5	4	1001	3.794206	4.173627	4.363337
6	5	1001	4.147852	4.562637	4.77003
7	6	1001	4.381618	4.81978	5.038861
8	7	1001	4.675325	5.142858	5.376624
9	8	1001	5.022977	5.525275	5.776424
10	9	1001	5.262737	5.789011	6.052148
11	10	1001	5.490509	6.03956	6.314085
12	11	1001	5.682318	6.25055	6.534666
13	12	1001	5.802198	6.382418	6.672528
14	13	1001	5.88012	6.468132	6.762138
15	14	1001	5.94006	6.534066	6.831069
16	15	1001	5.982018	6.58022	6.879321

4.2.2.1 Simulation Graphical Presentation of round robin delay time

In the Figure (4.4) all three users have the same delay time value due to the focusing of one user in each stage.

The transmitted bits in the x axis and the delay time in the y axis show that the delay are identical for all users and it starts from a various delay time because the users must wait to be quid by the system.

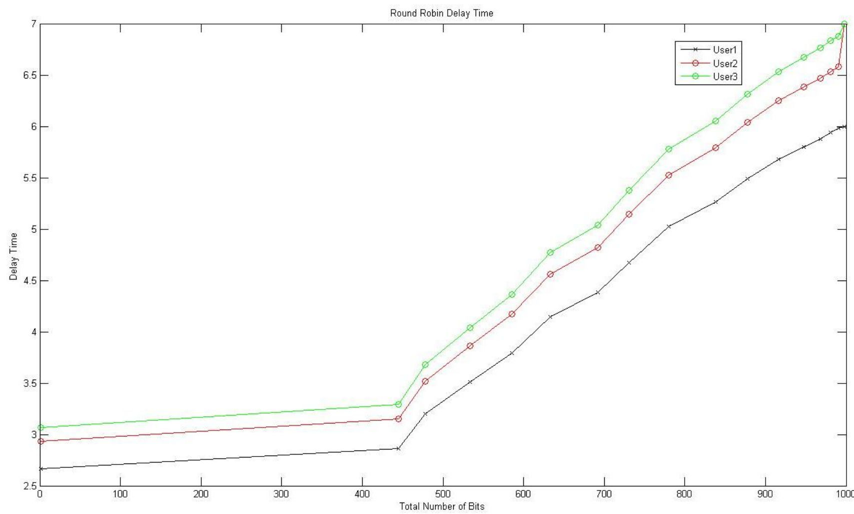

Figure 4.4: the delay time of round robin depends on the arrive time of the user and the delay is discrete.

Table 4.6: Throughput for round robin scenario

No.	SNR	Total Bits	Error Bits	Received Bits	Throughput User 1	Throughput User 2	Throughput User 3
1	0	1001	556	445	0.445	0.51175	0.534
2	1	1001	523	478	0.478	0.5497	0.5736
3	2	1001	467	534	0.534	0.6141	0.6408
4	3	1001	415	586	0.586	0.6739	0.7032
5	4	1001	368	633	0.633	0.72795	0.7596
6	5	1001	309	692	0.692	0.7958	0.8304
7	6	1001	270	731	0.731	0.84065	0.8772
8	7	1001	221	780	0.78	0.897	0.936
9	8	1001	163	838	0.838	0.9637	1.0056
10	9	1001	123	878	0.878	1.0097	1.0536
11	10	1001	85	916	0.916	1.0534	1.0992
12	11	1001	53	948	0.948	1.0902	1.1376
13	12	1001	33	968	0.968	1.1132	1.1616
14	13	1001	20	981	0.981	1.12815	1.1772
15	14	1001	10	991	0.991	1.13965	1.1892
16	15	1001	3	998	0.998	1.1477	1.1976

4.2.2.2 Simulation Graphical presentation of Throughput for round robin

In the Figure (4.5) the throughput is identical to all users because of the focusing of the system to process of each user alone the through put is linear and proportional to the bits transmitted on the system.

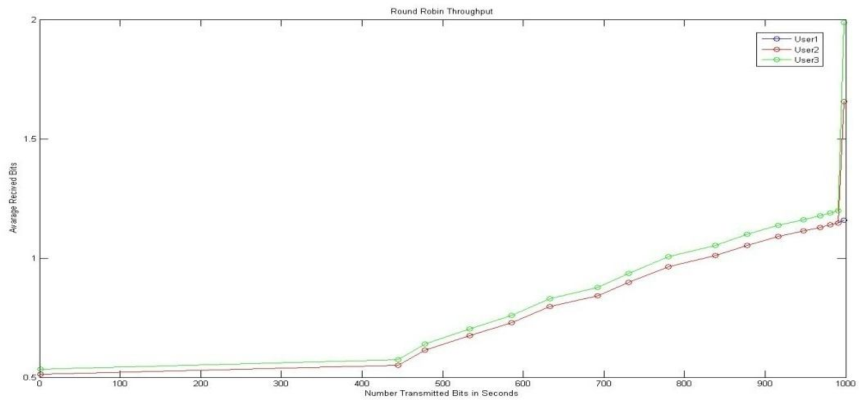

Figure 4.5: The throughput of round robin Users

4.2.3 Scenario Proportional Fair

In this scenario three users with different Loads will be used in the simulation for transmitting their data over the channels one of the users configured as a video conference and the other two users become a VOIP users.

The 16QAM modulation technique was used to modulate data of each user and the channel parameters is set as AWGN noise source with a range of 0 to 18 dB and calculating the time of entry of each user the system will pass the best CQI (Channel Quality Indicator) and the first in user with the best channel parameters.

The table (4.7) shows the delay time for the three users in the algorithm.

Table 4.7: delay time for three users

No.	SNR	Total Bits	Error Bits	Received Bits	Delay User1	Delay User2	Delay User3
1	0	1001	556	445	2.667333	2.727333	2.787333
2	1	1001	523	478	2.865135	2.925135	2.985135
3	2	1001	467	534	3.200799	3.260799	3.320799
4	3	1001	415	586	3.512488	3.572488	3.632488
5	4	1001	368	633	3.794206	3.854206	3.914206
6	5	1001	309	692	4.147852	4.207852	4.267852
7	6	1001	270	731	4.381618	4.441618	4.501618
8	7	1001	221	780	4.675325	4.735325	4.795325
9	8	1001	163	838	5.022977	5.082977	5.142977
10	9	1001	123	878	5.262737	5.322737	5.382737
11	10	1001	85	916	5.490509	5.550509	5.610509
12	11	1001	53	948	5.682318	5.742318	5.802318
13	12	1001	33	968	5.802198	5.862198	5.922198
14	13	1001	20	981	5.88012	5.94012	6.00012
15	14	1001	10	991	5.94006	6.00006	6.06006
16	15	1001	3	998	5.982018	6.042018	6.102018

4.2.3.1 Simulation Graphical presentation of three users delay time

It was found that every user in the system is processed in the same time with same delay time using parallel processing methodology so in the figure (4.6) each user is represented on a color , the transmitted bits increased so the delay time increased but in the same values for all of the users and in linear way.

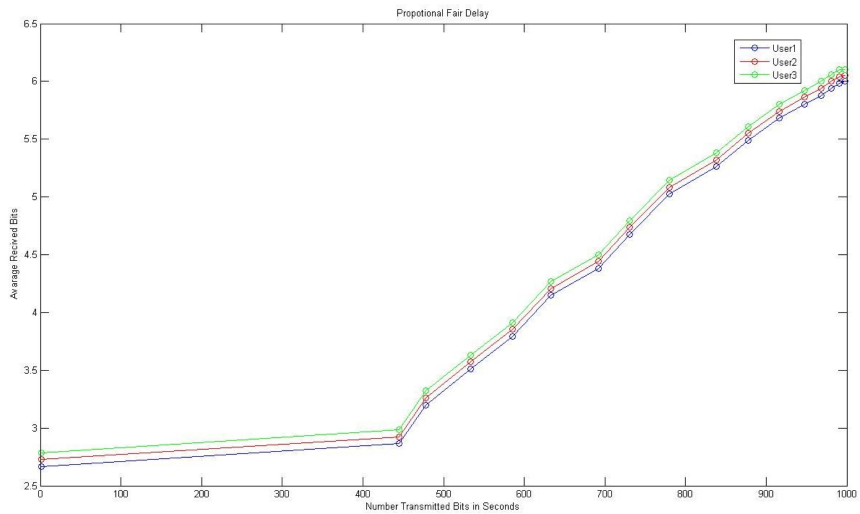

Figure 4.6: delay time for three users proportional fair algorithm

4.2.3.2 Throughput for proportional fair algorithm

The throughput of the system in the proportional fair is calculated for three users .The table (4.8) illustrates the throughput of each user on the system.

Table 4.8: Throughput proportional fair

No.	SNR	Total Bits	Error Bits	Received Bits	Throughput User 1	Throughput User 2	Throughput User 3
1	0	1001	556	445	0.445	0.2225	0.11125
2	1	1001	523	478	0.478	0.239	0.1195
3	2	1001	467	534	0.534	0.267	0.1335
4	3	1001	415	586	0.586	0.293	0.1465
5	4	1001	368	633	0.633	0.3165	0.15825
6	5	1001	309	692	0.692	0.346	0.173
7	6	1001	270	731	0.731	0.3655	0.18275
8	7	1001	221	780	0.78	0.39	0.195
9	8	1001	163	838	0.838	0.419	0.2095
10	9	1001	123	878	0.878	0.439	0.2195
11	10	1001	85	916	0.916	0.458	0.229
12	11	1001	53	948	0.948	0.474	0.237
13	12	1001	33	968	0.968	0.484	0.242
14	13	1001	20	981	0.981	0.4905	0.24525
15	14	1001	10	991	0.991	0.4955	0.24775
16	15	1001	3	998	0.998	0.499	0.2495

4.2.3.3 Simulation Graphical representation of throughput in proportional fair algorithm

Each user has a traffic parameters thus the throughput of each has a different graph, while comparing the graphs the users are not have an equal results.

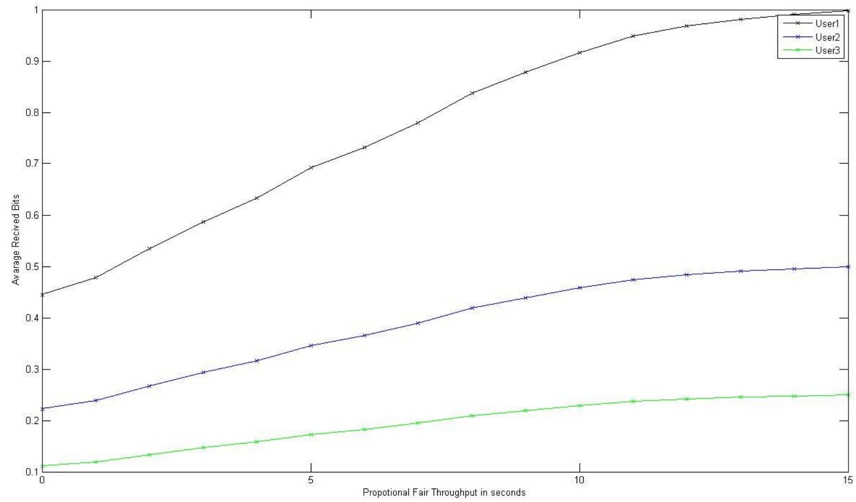

Figure 4.7: Three user throughput in the proportional fair

4.3 Comparison of Three Scheduling Algorithms

4.3.1 Delay time comparison

The comparison of the algorithms in delay time, the Figure (4.8) represent the delay time of each algorithm.

The black: best CQI, blue: round robin and red: proportional fair

It was found that the delay time is discrete linear just on the proportional fair algorithm and it starts from the less than 3 seconds compared with approximately 3 seconds on two other algorithms.

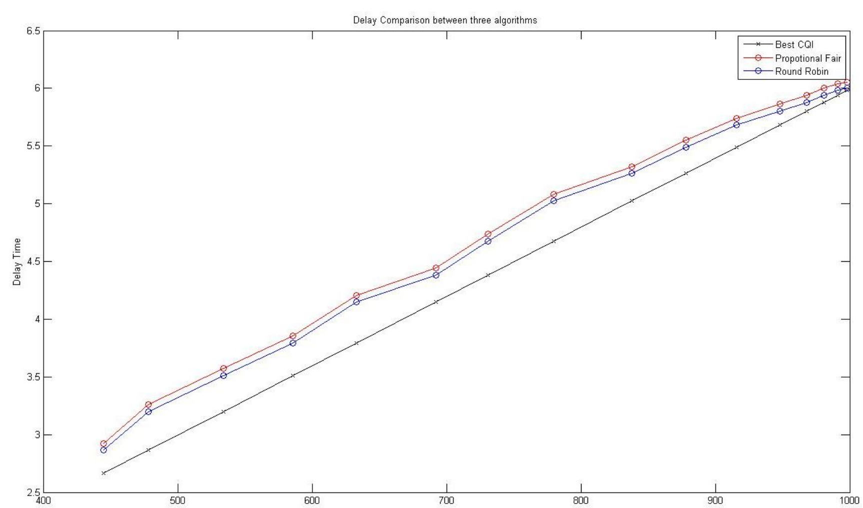

Figure 4.8: delay time comparison for three algorithms

4.3.2 Throughput comparison

The Figure (4.9) represents the Throughput of each algorithm

The red: Proportional fair, blue: round robin and red: Best CQI

While comparing the three algorithms in their throughput it was found that the round robin and best CQI has an increased through put than the proportional fair and it was found that the round robin and best CQI are identical through put results.

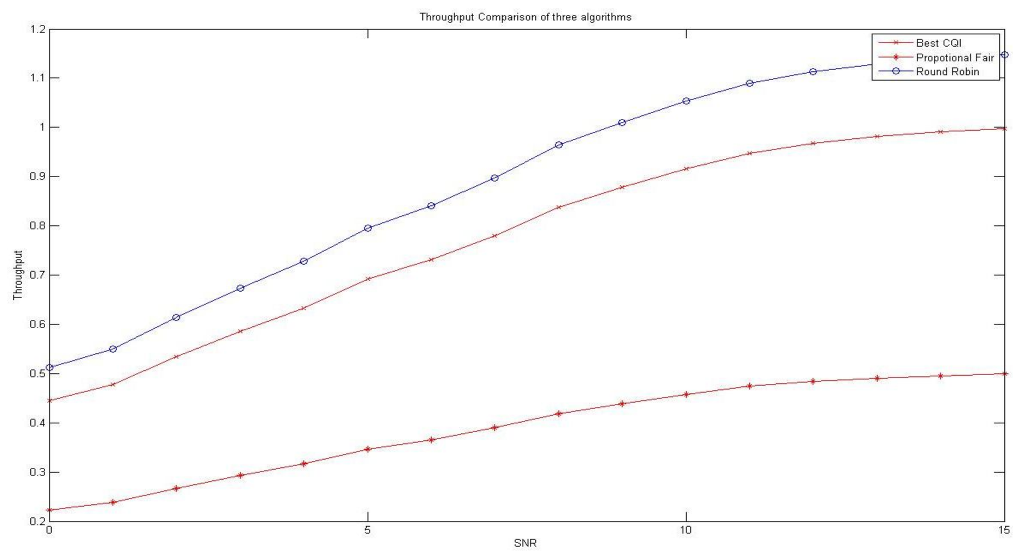

Figure 4.9: throughput comparison between three scheduling algorithm

4.3.3 Maximum Throughput for three algorithms

While examining the maximum throughput of the system for the three algorithms it was found that the Round robin and proportional fair has the same throughput and the Best CQI have the best throughput to the system because one user can access the system resources rather than the other algorithms.

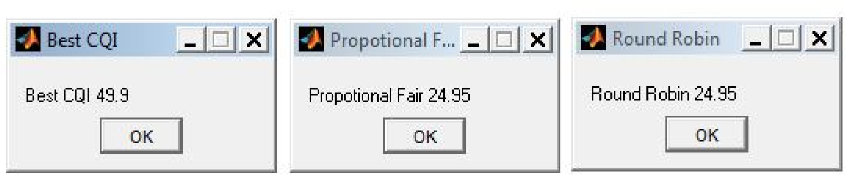

Figure 4.10: throughput percentage for three algorithms

4.3.4 Maximum Delay for three algorithms

While examining the maximum delay of the system for the three algorithms it was found that the Round robin and Best CQI has the same approximately delay time and the Best CQI have the best delay to the system because one user can access the system resources rather than the other algorithms.

Figure 4.11: Delay percentage for three algorithms

CHAPTER FIVE

CONCLUSION AND RECOMMEDATIONS

CHAPTER FIVE

CONCLUSION AND RECOMMENDATIONS

5.1 Conclusion

LTE is the evolution of (UMTS).It allows mobile users to access internet through their devices (mobile, laptop) , more than algorithm was used to handle users requests so three algorithms will be compared through a matlab code to analyze the performance and evaluate the algorithms to determine the best used algorithm. The aim of the project is to Investigate and study three of the downlink scheduling algorithms used in LTE system, best-CQI, Round Robin and proportional fair and Find the best algorithm to schedule the users, according to parameters Throughput and Delay and the comparison between them, using LTE-system level link simulator.

The project was split into phases in the first phase being the overview of the LTE and LTE advanced along with their features and their accompanying technologies and The second phase being also the overview of QOS aware and radio resource management while in third research must be conducted on the algorithms related to scheduling, these algorithms Proportional fair, round robin and Best CQI , where compared them to each other in the performance metrics. Moreover the performance metrics was also calculated in the fifth phase an operation was carried out being the code use MATLAB with the results in the end performing a discussion on the matter.

While running the simulation the system using the three downlink scheduling algorithm it was found that, the best CQI is the best with respect to QOS (Quality of service) due to the detection of best channel and the delay time and the throughput are discrete.

In round robin algorithm it was found that each user will spend a time on the system according the arriving time and the delay is proportional to the data size and the throughput is also discrete.

The proportional fair all users has the same throughput and delay time.

it was found that the Best CQI block a number of users that having a lower quality on their channel characteristics and the round robin is the default algorithm and based on the time of users, and the proportional fair is the best algorithm to handle the user requests because it evaluate the users using their time being entered to the system and the channel quality so a reduced delay time with almost the same throughput.

In the best channel quality indicator the total delay time is less for the best user and is 6 ms processing while in round robin is 7 ms delay time while in the proportional fair all users have the same delay time.

Throughput of the system was fixed for all the users 1000 mbps and it is not varying.

5.2 Recommendations

It is recommended to continue the investigation of scheduling algorithms:

- Using Opnet or other simulation programs to illustrate the downlink scheduling.
- Apply different modulation techniques such as BPSK, QPSK.
- Other new scheduling algorithms such as weight algorithms can be considered.
- Evaluate other performance metrics such as spectral efficiency, Fairness.

References

[1] Safjan, K., et al., Assessing 3GPP LTE-advanced as IMT-advanced technology: the WINNER+ evaluation group approach. Communication Magazine, IEEE, 2011.**49**(2):p.92-100.

[2] Mohamed Salem, A.A., Mahmudur Rahman, Halim Yanikomeroglu, David Falconer, and E.K Young-Doo Kim, and Yoon-Chae Cheong, An Overview of Radio Resource Management in Relay-Enhanced OFDMA-Based Networks.2010.

[3] Capozzi, F., et al., Downlink Packet scheduling in lte cellular networks: key design issues and a survey. Communications Surveys& Tutorials, IEEE, 2013. **15**(2):p. 678-700.

[4] Holma, H. and A. Toskala, LTE for UMTS-OFDMA and SC-FDMA based radio access.2009: John Wiley & Sons.

[5] Zahariev, N.K., Optimized service aware radio resource scheduler in lte. Master's thesis, University of Bremen, 2011.

[6] Grande, D.R., Performance Analysis of QoS in LTE-Advanced Heterogeneous Network, 2012-2013.

[7] Dahlman, E., S. Parkvall, and J. Skold, 4G: LTE/LTE-advanced for mobile broadband. 2013: Academic Press.

[8] Ratasuk, R., D. Toil, and A. Ghosh. Carrier aggregation in LTE-advanced. in Vehicular Technology Conference (VTC 2010- Spring), 2010 IEEE 7 1st. 2010. IEEE.

[9] Liu, L., et al., Downlink mimo in lte-advance: Su-mimo vs. mu-mimo. Communications Magazine, IEEE, 2012.**50**(2): P.140-147.

[10] Lee, D., et al., Coordinated multipoint transmission and reception in LTE-advanced: deployment scenarios and operational challenges. Communications Magazine. IEEE, 2012. **50**(2): P.148-155.

[11] Erik Dahlman, Stefan Parkvall, Johan Sköld "4G – LTE/LTE-Advanced for Mobile Broadband", Academic Press, 2011, ISBN 978-0-12-385489-6

[12] StefaniaSesia, IssamToufik, and Matthew Baker, "LTE – The UMTS Long Term Evolution – From Theory to Practice", Second Edition including Release 10 for LTE-Advanced, John Wiley & Sons, 2011, ISBN 978-0-470-66025-6

[13] Chris Johnson, "LTE in BULLETS", Create Space, 2010, ISBN 978-1-4528-3464-1

[14] Erik Dahlman, Stefan Parkvall, Johan Sköld, Per Beming, "3G Evolution – HSPA and LTE for Mobile Broadband", 2nd edition, Academic Press, 2008, ISBN 978-0-12-374538-5

[15] BorkoFurht, Syed A. Ahson, "Long Term Evolution: 3GPP LTE Radio And Cellular Technology", Crc Press, 2009, ISBN 978-1-4200-7210-5

[16] K. Fazel and S. Kaiser, Multi-Carrier and Spread Spectrum Systems: From OFDM and MC-CDMA to LTE and WiMAX, 2nd Edition, John Wiley & Sons, 2008, ISBN 978-0-470-99821-2

[17] Agilent Technologies, "LTE and the Evolution to 4G Wireless: Design and Measurement Challenges", John Wiley & Sons, 2009 ISBN 978-0-470-68261-6

[18] Sajal K. Das, John Wiley & Sons (April 2010): "Mobile Handset Design", ISBN 978-0-470-82467-2 .

[19] Beaver, Paul, "What is TD-LTE?",RF&MicrowaveDesign line, September 2011.

[20] HarriHolma, AnttiToskala, LTE for UMTS - OFDMA and SC-FDMA Based Radio Access, John Wiley & Sons 2009, ISBN 978-0-470-99401-6 : LTE Advanced for IMT-advanced.

[21] Moray Rumney (editor), LTE and the Evolution to 4G Wireless: Design and Measurement Challenges, Agilent Technologies Publication 2009, ISBN 978-0-470-68261-6,: Proving LTE Advanced.

[22] Christina Gessner (editor), Long Term Evolution: A concise introduction to LTE and its measurement requirements, Rohde & Schwarz Publication 2011, ISBN 978-3-939837-11-4,

Appendices

Appendix A

Simulation Code

```
clear all ;
clc ;
%% Defining the antenna arrays
Arrays = arrayparset ;
%% Defining the scenerio with BS , MS , Relays
BsAAIdxCell = {[1]; [2]} ;
MsAAIdx = [2 3];
L =3; % Number of links
S =12; % Identification number of the Scenario
layoutpar = layoutparset ( MsAAIdx , BsAAIdxCell , L, Arrays );
layoutpar .ScenarioVector = S* ones (1,L);
layoutpar . Stations (1) .Pos = [0;0;32]; % Position of BS1
layoutpar . Stations (2) .Pos = [250;250;32]; % Position of BS2
layoutpar . Stations (3) .Pos = [250;250;32]; % Position of MS1
layoutpar . Stations (4) .Pos = [0;500;1.5]; % Position of MS2
%% Defining the pairing of BS , MS , Relays
layoutpar . Pairing = [1 2 1;3 4 4];
%% Defining the initial conditions for the iteration
SER_tot_nc = zeros (11 ,1) ;
SER_tot_c = zeros (11 ,1) ;
SER_tot_wor = zeros (11 ,1) ;
for n =1:5000
wimpar = wimparset ; % Generation the Winner
parameters
```

```
[H, delays , out ]= wim(wimpar , layoutpar ) % Generation
of all the radio links according to the layout
% Processing the bits for Calculation
SP. FFTsize = 512; % The size of the FFT and IFFT
SP. CPsize = 20; % CP length
SP. SNR = [0:4:40]; % Simulated SNR range is from 0
dB to 40 dB with an increament of 4 dB.
SP. numRun = 10^4; % The number of simulation
iterations is 10^4
a= isnan( out . path_powers );
out .path_powers (a) =0;
% Calculation of path powers for Non - cooperative
h_1 = out .path_powers (1 ,:); % Relay Link
SP. channel_1 = h_1 ;
SP. channel_1 = h_1 / sqrt( sum ( h_1 .^2) );
h_2 = out .path_powers (2 ,:); % Access Link
SP. channel_2 = h_2 ;
SP. channel_2 = h_2 / sqrt( sum ( h_2 .^2) );
% Calculation of BER
SER_ofdm_nc = ofdm_wr_nc (SP);
SER_tot_nc = SER_tot_nc + SER_ofdm_nc ;
% Calculation of path powers for Cooperative
h_1 = out .path_powers (1 ,:); % Relay Link
SP. channel_1 = h_1 ;
SP. channel_1 = h_1 / sqrt( sum ( h_1 .^2) );
h_2 = out .path_powers (2 ,:); % Access Link
SP. channel_2 = h_2 ;
```

```
SP. channel_2 = h_2 / sqrt( sum ( h_2 .^2) );
h_3 = out .path_powers (3 ,:); % Direct Link
SP. channel_3 = h_3 ;
SP. channel_3 = h_3 / sqrt( sum ( h_3 .^2) );
h_4 = out .path_powers (2 ,:)+ out . path_powers (3 ,:);
SP. channel_4 = h_4 ;
SP. channel_4 = h_4 / sqrt( sum ( h_4 .^2) );
% Calculation of BER
SER_ofdm_c = ofdm_wr_c (SP);
SER_tot_c = SER_tot_c + SER_ofdm_c ;
% Calculation of path powers Without relay
h= out .path_powers (3 ,:); % Direct Link
SP. channel = h;
SP. channel = h/ sqrt( sum (h .^2) );
% Calculation of BER
SER_ofdm_wor = ofdm (SP);
SER_tot_wor = SER_tot_wor + SER_ofdm_wor ;
end
%% Taking Average of the data
SER_wor = SER_tot_wor /n; % SER for without relay
environment
SER_nc = SER_tot_nc /n; % SER for non co - operative
environment
SER_c = SER_tot_c /n; % SER for co - operative environment (
with one relay )
%% Defining the MS and BS positions for the co - operative
environments with 2 relays
```

```
BsAAIdxCell = {[1];[2];[2]};
MsAAIdx = [2 2 3];
L =5;
layoutpar = layoutparset ( MsAAIdx , BsAAIdxCell , L, Arrays );
layoutpar .ScenarioVector = S* ones (1,L);
layoutpar . Stations (1) .Pos = [0;250;32]; % BS1
layoutpar . Stations (2) .Pos = [500;500;25]; % BS2
layoutpar . Stations (3) .Pos = [500;0;25]; % BS3
layoutpar . Stations (4) .Pos = [500;500;25]; % MS1
layoutpar . Stations (5) .Pos = [500;0;25]; % MS2
layoutpar . Stations (6) .Pos = [1000;250;1.5]; % MS3
layoutpar . Pairing = [1 1 2 1 3 ; 6 4 6 5 6];
% Defining the initial condictions for the iteration
SER_tot_c_two = zeros (11 ,1) ;
for n =1:5000
  % Generation the Winner parameters
wimpar = wimparset ;
[H2 , delays2 , out2 ]= wim(wimpar , layoutpar )
% Processing the bits for Calculation
SP. FFTsize = 512;
SP. CPsize = 20;
SP. SNR = [0:4:40];
SP. numRun = 10^4;
a= isnan( out2 . path_powers );
out2 .path_powers (a) =0;
% Calculation of path powers for Cooperative
environment with two relays
```

```
h_1 = out2 .path_powers (1 ,:); % direct link
SP. channel_1 = h_1 ;
SP. channel_1 = h_1 / sqrt( sum ( h_1 .^2) );
h_2 = out2 .path_powers (2 ,:); % relay link 1
SP. channel_2 = h_2 ;
SP. channel_2 = h_2 / sqrt( sum ( h_2 .^2) );
h_3 = out2 .path_powers (3 ,:); % access link 1
SP. channel_3 = h_3 ;
SP. channel_3 = h_3 / sqrt( sum ( h_3 .^2) );
h_4 = out2 .path_powers (4 ,:); % relay link 2
SP. channel_4 = h_4 ;
SP. channel_4 = h_4 / sqrt( sum ( h_4 .^2) );
h_5 = out2 .path_powers (5 ,:); % access link 2
SP. channel_5 = h_5 ;
SP. channel_5 = h_5 / sqrt( sum ( h_5 .^2) );
h_6 = out2 .path_powers (1 ,:)+ out2 . path_powers (3 ,:)+out2 .
path_powers (5 ,:); % combined link in reciever
SP. channel_6 = h_6 ;
SP. channel_6 = h_6 / sqrt( sum ( h_6 .^2) );
% Calculation of SER
SER_ofdm_c_two = ofdm_wr_c_two (SP);
SER_tot_c_two = SER_tot_c_two + SER_ofdm_c_two ;
end
%% Taking Average of the data
SER_c_two = SER_tot_c_two /n; % SER for co - operative
environment with 2 relays
% saving data
```

```
save results / ofdm_data ;
% Creating figures
figure
semilogy (SP.SNR , SER_wor ,'bh -');
hold on
semilogy (SP.SNR , SER_nc ,'mo -');
semilogy (SP.SNR ,SER_c ,'rd -');
semilogy (SP.SNR , SER_c_two ,'kx -');
grid on
legend (' Without Relay Environment ', 'Non - Cooperative Environment
', ' Cooperative Environment ', ' Cooperative Environment (2 relay ) ');
xlabel (' Signal to Noise Ratio ( SNR ), dB ');
ylabel (' Symbol Error Rate ( SER ) ');
title (' SER for WINNER -II ');
```

Appendix B

Throughput Results

```
clearall;
closeall;
clc

figure
x=[0    1    2    3    4    5    6    7    8    9    10
11  12  13  14  15];
y1=[0.445      0.478     0.534     0.586     0.633
0.692    0.731    0.78     0.838    0.878    0.916
0.948   0.968   0.981   0.991   0.998];

plot(x,y1,'color','r','marker','x');

holdon
x=[0    1    2    3    4    5    6    7    8    9    10
11  12  13  14  15];
y2=[0.2225 0.239 0.267 0.293 0.3165 0.346 0.3655
0.39 0.419 0.439 0.458 0.474 0.484 0.4905 0.4955
0.499];
plot(x,y2,'color','r','marker','*');
holdon
x=[0    1    2    3    4    5    6    7    8    9    10
11  12  13  14  15];
```

```
y3=[0.51175 0.5497 0.6141 0.6739 0.72795 0.7958
0.84065 0.897 0.9637 1.0097 1.0534 1.0902 1.1132
1.12815 1.13965 1.1477 ];
plot(x,y3,'color','b','marker','o');
title ('Throughput Comparison of three
algorithms');
legend('Best CQI','PropotionalFair','Round
Robin');
xlabel ('SNR');
ylabel('Throughput');
p1=max(y1*100/2);
p2=max(y2*100/2);
p3=max(y3*100/2);
disp(p1);
disp(p2);
disp(p3);
msgbox(['Best CQI ' num2str(p1)], 'Best CQI');
msgbox(['Propotional Fair ' num2str(p2)],
'Propotional Fair');
msgbox(['Round Robin ' num2str(p3)], 'Round
Robin');
```

Appendix C

Delay Time Results

```
clearall;
closeall;
clc
figure
x=[ 445  478 534 586 633 692 731 780 838 878 916
948 968 981 991 998 ];
y1=[ 2.667333 2.865135     3.200799     3.512488
3.794206      4.147852      4.381618      4.675325
5.022977      5.262737      5.490509      5.682318
5.802198    5.88012 5.94006 5.982018];
plot(x,y1,'color','k','marker','x');
holdon
x=[ 445  478 534 586 633 692 731 780 838 878 916
948 968 981 991 998 ];
y2=[   2.925135   3.260799   3.572488   3.854206
4.207852  4.441618  4.735325  5.082977  5.322737
5.550509   5.742318   5.862198   5.94012   6.00006
6.042018 6.052018];
plot(x,y2,'color','r','marker','o');
holdon
x=[ 445  478 534 586 633 692 731 780 838 878 916
948 968 981 991 998 ];
y3=[  2.865135         3.200799         3.512488
3.794206      4.147852      4.381618      4.675325
5.022977      5.262737      5.490509      5.682318
5.802198      5.88012       5.94006       5.982018
6];
```

```
plot(x,y3,'color','b','marker','o');
title('Delay Comparison between three algorithms');
legend('Best CQI','PropotionalFair','Round Robin');
xlabel('Received Bits');
ylabel('Delay Time');
p1=max(y1*100/7);
p2=max(y2*100/7);
p3=max(y3*100/7);
disp(p1);
disp(p2);
disp(p3);
msgbox(['Best CQI ' num2str(p1)], 'Best CQI');
msgbox(['Propotional Fair ' num2str(p2)], 'Propotional Fair');
msgbox(['Round Robin ' num2str(p3)], 'Round Robin');
```