Sudan University of Science and Technology

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CALL ADMISSION CONTROL IN LTE

التحكم في قبول المكالمة في التطور بعيد الامد

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Prepared By:

1. Alaa Mohammed Yousef Ahmed

2. Esraa Salih Mustafa Abdelghani

3. Razan Farouk Abdelrahman Babeker

4. Zeinab Adam Hassan Ali

Supervised By:

Dr. Fath Elrahman Ismael Khalifa

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الآيـــة

قال تعالى :

بسم الله الرحمن الرحيم

(قَالَ يَا قَوْمِ أَرَأَيْتُمْ إِنْ كُنْتُ عَلَى بَيِّنَةٍ مِنْ رَبِّي وَرَزَقَنِي مِنْهُ رِزْقًا حَسَنًا وَمَا أُرِيدُ أَنْ أُخَالِفَكُمْ إِلَى مَا أَنْهَاكُمْ عَنْهُ إِنْ أُرِيدُ إِلَّا الْإِصْلَاحَ مَا اسْتَطَعْتُ وَمَا تَوْفِيقِي إِلَّا بِاللَّهِ عَلَيْهِ تَوَكَّلْتُ وَإِلَيْهِ أُنِيبُ)

صدق الله العظيم

سورة هود الآية 88

Dedication

To Our parents…

To each of our loved…

To each of our support financially and morally…

To each of we spent with them the most beautiful moments (Our Colleagues)…

ACKNOWLEDGEMENT

First of all we would like to thank GOD for blessing us and giving us the power to work and complete this thesis.

We would like to thank our supervisor **Dr. Fath Elrhaman Ismael** for his advice and support during the writing of this thesis. His knowledge and dedication and opinion were useful in completing this research.

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ABSTRACT

In radio resource management (RRM) of Long Term Evolution (LTE) networks, the call admission control (CAC) techniques play an important role in providing Quality of service. To achieve this role must granted bit rate to services that affected by packet latency and packet dropping to satisfy the QoS requirement. The main goal of this research is to evaluate a proposed admission control algorithms in providing QoS. Two algorithms of admission control are compared, admission with priority and without priority are depends on quality of service class identifier (QCI) investigates the performance metrics blocking probability, available physical resource block, throughput and delay. MATLAB was used for the evaluation of algorithms. Simulation results show higher connection admission rate as compared with non-priority services. The results showed that as the number of user increase the blocking probability increased by 100% due to available limited resources. It was also noticed that as number of user increase available physical resource block decreased by 3.93% from total bandwidth. The simulation also showed that as the number of user increase the throughput increased by 33% and the Delay reduced by 80.17% in priority.

المستخلص

في اداره مصادر الراديو لشبكات التطور بعيدة المدى ان تقنيات التحكم في دخول المكالمات تلعب دورا مهما في التزويد بجوده الخدمة. جوده الخدمة ضرورية في كثير من الخدمات خصوصا الخدمات التي تتأثر بتأخير الحزمة و انقطاع الحزمة لذا يجب ضمان معدل البتات لايفاء متطلبات جوده الخدمة. الهدف الرئيسي من هذا البحث هو تقييم خوارزمتين في التحكم في الدخول للتزويد بجودة الخدمة , الاولى تسمح بالدخول باولوية و الثانية تسمح من دون اولويه اعتمادا على معرف صنف جودة الخدمة. تمت مقارنة ادائية الخوارزميات المقترحة وفقا للتاخير , المصادر الفيزيائيه المتاحه , احتماليه السحب و الخرج. تم تطبيقها في الماتلاب لتقييم الخوارزميات. اظهرت نتائج المحاكاة اعلى معدل لقبول الاتصال مقارنة مع من دون اولوية .

اظهرت النتائج ان زيادة عدد المستخدمين تزيد من نسبة احتمالية السحب ب 100% نظرا لمحدودية الموارد المتاحة , كما لوحظ ان زيادة عدد المستخدمين يقلل من المصادالفيزيائية المتاحة بنسبة 3.93% من عرض النطاق الترددي الكلي و زاد ايضا الخرج بنسبة 33% والتأخير قل بنسبة 80.17% في خوارزمية الاولوية .

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List of Abbreviations

3GPP	**Third Generation**
AC	**Admission Control**
BW	**Bandwidth**
CAC	**Call Admission Control**
eNB	**Evolved Node B**
ERABs	**Evolved Radio Access Bearers**
E-UTRAN	**Evolved UMTS Terrestrial Radio Access Network**
EPC	**Evolved Packet Core**
EPS	**Evolved Packet System**
FDD	**Frequency Division Duplexing**
GBR	**Guaranteed Bit Rate**
GSM	**Global System for Mobile Communication**
IP	**Internet protocol**
LTE	**Long Term Evolution**
MCS	**Modulation and Code Scheme**
Non-GBR	**Non Guaranteed bit Rate**
OFDMA	**Orthogonal Frequency Division Multiple Access**
PRBs	**Physical Resource Blocks**

PS	**Packet Switch**
PUCCH	**Physical Uplink Control Channel**
QCI	**QoS Class Identifier**
QoS	**Quality of Service**
RAC	**Radio Access Control**
RAN	**Radio Access Network**
RRM	**Radio Resource Management**
RSRQ	**Reference Signal Received Quality**
RT	**Real Time**
TB	**Transport block**
TDD	**Time Division Duplexing**
TFT	**Traffic Flow Template**
TDMA	**Time Division Multiple Access**
TTI	**Time Transmission Inte**
UE	**User Equipment**
UMTS	**Universal Mobile Telecommunication System**

CHAPTER ONE

INTRODUCTION

1.1 Preface

The area of 4G in cellular networks became a tangible reality in the technology life. Since the baby steps time of mobile networks, the ultimate goal was to fulfill the Quality of service level to end-user and provide network of high connectivity. Therefore the incremented demand for network-related services like VOIP, web data and video streaming, meanwhile ensuring delay limits and bandwidth requirements introduce new challenges within the design of the 4G and beyond generation of cellular networks[1]. In order to provide quality of service (QoS) in cellular networks, the role of radio resource management (RRM) is very important. The performance of RRM techniques not only has an impact on the performance of individual user, but also on the overall network performance. The important task of RRM includes – call admission control (CAC), the CAC decides the acceptance or rejection of service requests, to ensure the QoS of the ongoing calls, the preemption methods are used in case of limited resource conditions [2]. LTE defines two categories of traffic flows: guaranteed bit rate (GBR) and non-guaranteed bit rate (non-GBR), GBR traffic usually have strict delay and bandwidth demands. Therefore, they have to be prioritized over non-GBR traffic flows so as to meet their demands [3].

The QoS parameters associated with a bearer include QoS Class Identifier (QCI). QCI is a scalar value that refers to a set of access Node-specific parameters which determine packet forwarding treatment.

1.2 Problem Statement

The Quality of service is necessary in many services especially that services affected by packet latency and packet dropping.There for, the QoS requirements should be satisfied and must granted bit rate for these services.

1.3 Proposed Solution

The effect of call admission control in satisfying the QoS requirement for the Delay sensitive application is investigate in this research.

1.4 Aim and Objectives

The aim of this research is to investigate different algorithms to choose optimal and acceptable algorithm of call admission control depending on priority of GBR.

Performance evaluation of call admission control is done through the sub objectives:

-To compare the available physical resource block.

-To compare the blocking probability.

- To compare the throughput.

-To compare the delay.

1.5 Methodology

In this thesis two algorithms of admission control are evaluated , one of them admitted by priority depends on quality of service class identifier, and other admit without priority , the performance metrics such as delay, throughput, physical resource block and blocking probability are used and simulated in MATLAB with the aid of mathematical equations to show the simulation results in order to evaluate these two algorithms.

1.6 Thesis Outline:-

This research composed of five chapters their details are as following:

Chapter One: contains the introduction, problem statement, proposed solution, methodology and aim & objective.

Chapter Two: Presents Literature Review and background of CAC.

Chapter Three: Applies the equations and techniques that are used to solve the problem, get results and provides a description of the model simulations and tool.

Chapter Four: Presents the results obtained from the simulation model and discuss them.

Chapter Five: Summarizes the results of the thesis and give recommendations for future work.

CHAPTER TWO

LITRITURE REVIEW

2.1 Background

2.1.1 Evolution of Mobile Communication

The history of mobile telephony can be traced back to the 1980's when the first-generation (1G) voice-only analog networks were introduced. Replaced by the second-generation (2G) digital phones equipped with fax, data and messaging services. The third generation (3G) ushered in the era of multimedia computing and entertainment on mobile phones and today we are at the cusp of a wireless revolution with superior fourth-generation (4G) shown in table 2.1.

In wireless communication the long term evolution (LTE) evolved from an earlier 3GPP system known as the Universal Mobile Telecommunication System (UMTS), which in turn evolved from the Global System for Mobile Communications (GSM).

Table 2.1: Evolution of Mobile Communication Systems

Property	1G	2G	2.5G&3G	4G
Starting Time	1985	1992	2002	210-2012
Representative Standard	AMPS	GSM	IMT-2000	UWB
Radio Frequency (Hz)	400M-800M	800M-900M	1800M-2400M	2G-8G
Bandwidth(bps)	2.4K-3K	9.6K-14.4K	384K-2M	20M-100M
Multiple Access Technique	FDMA	TDMA, CDMA	WCDMA	OFDM
Switching Basis	Circuit	Circuit	Circuit,Packet	Packet
Cellular Coverage	Large area	Medium area	Small area	Mini area
Service Type	Voice	Voice, limited data	Voice, data, limited multimedia	Multimedia

2.1.2 Long Term Evolution (LTE) Network Architecture

Long Term Evolution (LTE) is a significant project of 3rd Generation Partnership Project (3GPP), initially proposed on the Toronto conference of 3GPP in 2004 and officially started as LTE work item in 2006. LTE, as a transition from the 3rd generation (3G) to the 4th generation (4G), has achieved great capacity and high speed of mobile telephone networks without doubt[4].

Long Term Evolution (LTE) it is divided into two main parts: radio access network which is called Evolved UMTS Terrestrial Radio Access Network (E-UTRAN) and packet core network which is called Evolved Packet Core (EPC).

The E-UTRAN has two essential components: the User Equipment (UE) and eNodeB which is the base station. UE is any device used directly by an end user to communicate with the base station. However eNodeB controls all radio access functions in addition each eNodeB is connected to the evolved packet core by means of the S1 interface, It allows to user equipment to communicate with the radio access network (EPC). eNodeB interconnected with each other's by means of the X2 interface and the main goal of this interface is to minimize packet loss due to user mobility. Note the eNodeB functionalities are: radio resource management radio bearer control, radio admission control, connection mobility control and scheduling in both uplink and downlink shown in figure 2.1 [5] [6].

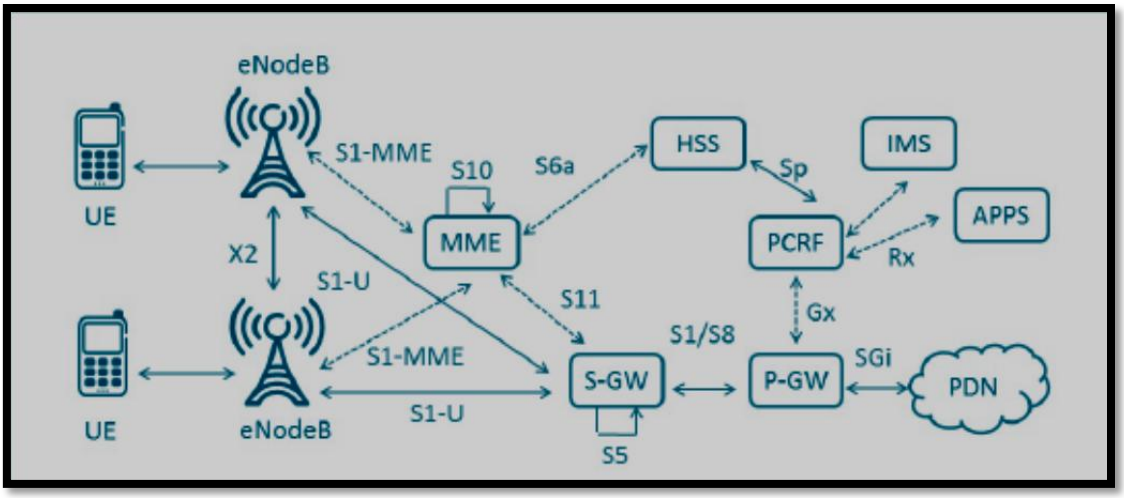

Figure 2.1: LTE System Architecture

The EPC is responsible for the overall control of the User Equipment (UE) as well as establishment of Bearers. The EPC is composed of several components: The Mobility Management Entity (MME), Serving Gateway (S-GW), Packet Data Network Gateway (PDN-GW) and the Policy and Charging Rules Function (PCRF) as well as in figure 2.2.

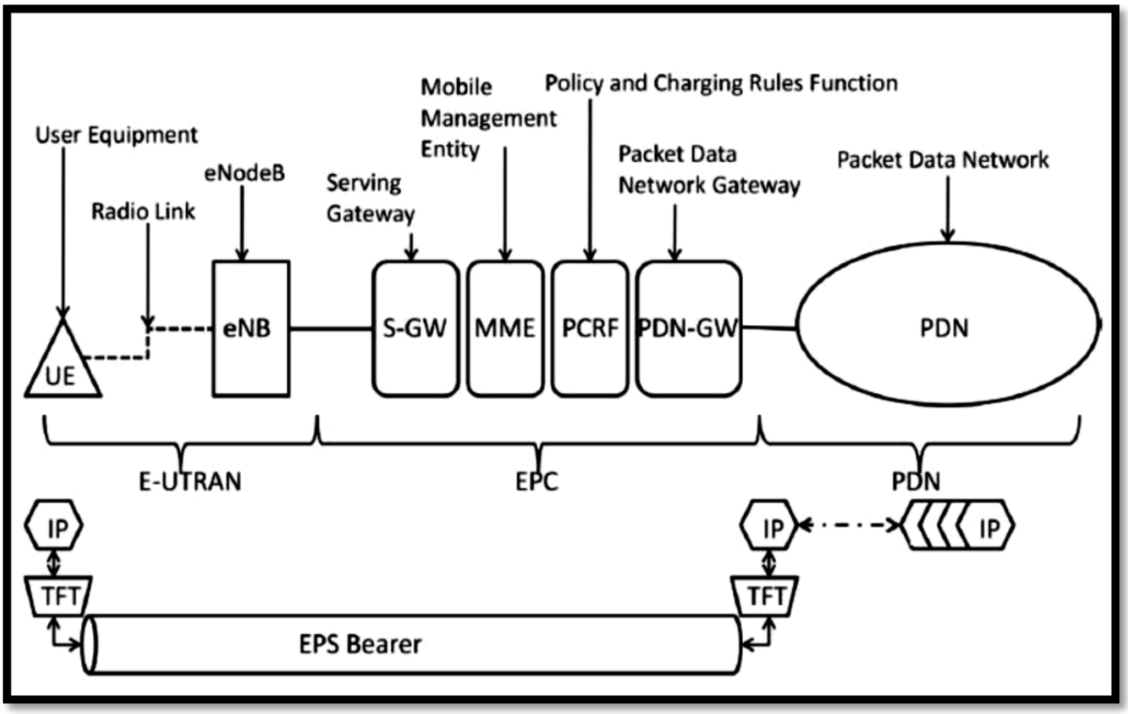

Figure 2.2: The main component of the LTE network

2.1.3 Overview of LTE Air interface

LTE is the 1’rst technology that supports both Time Division Duplexing (TDD) and Frequency Division Duplexing (FDD). Orthogonal Frequency Division Multiple Access (OFDMA) is selected as the air-interface for LTE downlink (i.e. from the base station to the mobile device). Key characteristics of OFDM are its resistance to interference and multi-path fading. It is also used in technologies such as Wi-Fi and Wi-Max, and is well suited for technologies that require high data rates.

In LTE six different channel bandwidths that include 1.4 MHz, 3 MHz, 5 MHz, 10 MHz, 15 MHz and 20 MHz. With larger bandwidth can obtain high channel capacity. One of the key requirements for LTE is to provide high data rates, and using OFDM it can be achieved. Also with a bandwidth of 20MHz, peak data rate of up to 150Mbps in the down-link can be achieved.

For the air interface of LTE up-link (i.e. from the mobile device to the base station), another form of OFDMA technology is been used which is known as Single Carrier Frequency Division Multiple Access (SC-FDMA).

The most important consideration for a UE is its battery life. Unlike the base stations, the UEs power supply is limited to a battery. Hence, utmost consideration must be provided to the transmission mechanisms that are adopted by the UE. The highest power consumption that can happen over the air interface of the UE is the transmission of data in the form of radio frequency signals by the UEs' antenna to the base station. Hence the signal transmission should be done effectively [7].

2.1.4 LTE Quality of Service (QoS)

In LTE QoS evolved packet system (EPS) bearers are a connection oriented transmission network which requires the establishment of a virtual connection between two end points in LTE between the user equipment and the PDN-GW. There are two types of bearers supported in LTE network namely default bearer and dedicated bearer , based on the nature of the Quality of Service(QoS).

In default bearer each user's equipment has at least one and it is initiated when connected to the network. The dedicated bearers are initiated at any time and each user's equipment can have different one, this dedicated bearer is used where the QoS requirement for some traffic is different than the one given by the default bearers shown in figure 2.3 [6].

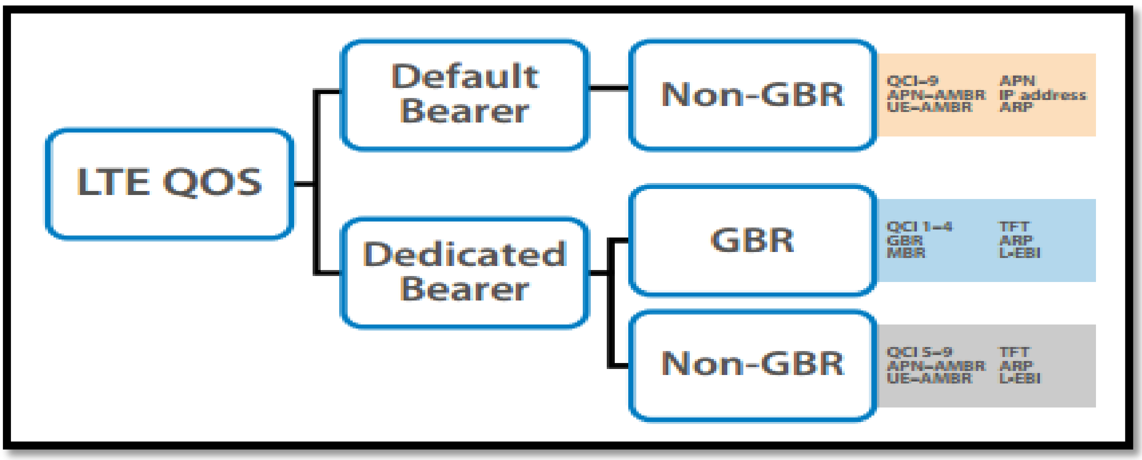

Figure 2.3: LTE Quality of service

Moreover dedicated bearers are divided into two types:Guaranteed Bit Rate(GBR) bearers and Non-Guaranteed Bit Rate(Non-GBR) bearers.

In GBR bearers there are no packet drops in case of congestion since the system insures that a minimum bit rate will be given to the GBR bearer. However in non-GBR bearers we may have packets loss in case of congestion since there are no guarantees that the amount bit rate needed will be given to the non-GBR bearers. Not to mention that GBR bearers are used when it is better to block a service to not have a problem in case it is accepted, for example in case of a real time voice call it is better to block this call if the system can insure a minimum bit rate than accept it because if the system accept it and can't insure a minimum bit rate to maintain it all time this will lead to an incomprehensible voice. Nevertheless Non-GBR bearers are used in other applications for example e-mail or web-browsing. Note that the importance of an application doesn't depend on the type of bearers (GBR bearers or non-GBR bearers) [6].

Finally the bearer mechanisms allow differentiating between the applications traffics and putting in the same bearer the traffics that have similar QoS. So when an EPS bearers are established a single user equipment can have multiple bearers related to him because the traffics that have similar QoS are grouped together and send it in an EPS bearers, and it's the role of Traffic Flow Template (TFT) to identify which IP packets should be assigned to which bearers by using IP header information.

2.1.5 Overview of QoS Class Identifier(QCI)

The eNB in the network use QCI to recognize how to treat the packets for each bearer. Briefly, the QCI parameter will let the node know how to handle the packets for each bearer or to arrange the resources between the packets following their importance, so QCI is related to the RAN in the network.

QCI parameter is described by a simple scalar value. So it specifies when traffic should be sent to or received from the mobile user equipment by considering the packet latency and the packet loss rate in both downlink and uplink. Briefly the QCI mechanism explicit QoS characteristics (packet latency or packet delay budget and packet loss rate) and this will lead to prioritization.

LTE supports nine types of QCI values that can be associated with the bearers. QCI 1 is for conversational voice. QCI 2 is associated with conversational video. QCI 3 is for real-time gaming. QCI 4 is for non-conversational video. QCI 5 is for IMS signaling. QCI 6, 8 & 9 are for video and TCP traffic, and QCI 7 is for voice, video and interactive gaming.

In the table below can see the QCI values that have been specified and this table include QCI priority level, packet delay, packet error loss rate denoted in table 2.2.

Table 2.2: Standardized QoS Class Identifier in LTE

QCI	Bearer Type	Priority	Packet Delay	Packet Loss	Example
1	GBR	2	100 ms	10	VOIP Call
2		4	150 ms	10	Video Call
3		3	50 ms		Online Gaming (Real Time)
4		5	300 ms	10	Video Streaming
5	Non-GBR	1	100 ms		IMS Signaling
6		6	300 ms		Video, TCP based services e.g. email, chat, ftp etc.
7		7	100 ms	10	Voice, Video, Interactive gaming
8		8	300 ms	10	Video, TCP based services e.g. email, chat, ftp etc.
9		9			

When a user generates packets of certain QCI value (for example, conversational video with QCI value 2) and a dedicated bearer has not been established for the packets then the RRM module in LTE requests for bearer admission. Then the QCI value for the bearer is passed to the Radio Admission control at the eNodeB [8].

2.1.6 Radio Resource Management (RRM) Overview

Radio Resource Management (RRM) is an eNB application level function figure 2.4 that ensures the efficient use of available radio resources. RRM manages the assignment, re-assignment and release of radio resources considering single and multi-cell aspects.

The primary goal of RRM is to control the use of radio resources in the system while also ensuring that the Quality of Service (QoS) requirements of the individual radio bearers are met and the overall usage of radio resources on the system level is minimized. The objective of RRM is to satisfy the service requirements at the smallest possible cost to the system, ensuring optimized use of spectrum.

RRM has various functions. key functions are: Radio Admission Control (RAC),Radio Bearer Control (RBC),Connection Mobility Control (CMC),Dynamic Allocation of resources to UE's in both uplink and downlink (DRA),Inter-Cell Interference co-ordination (ICIC),Load Balancing (LB).

Radio Admission Control (RAC) is a sub-function of RRM. The task of RAC is to admit or reject the establishment requests for new radio bearers. The establishment of a bearer is based on the outcome ofthe RAC Algorithms [9].

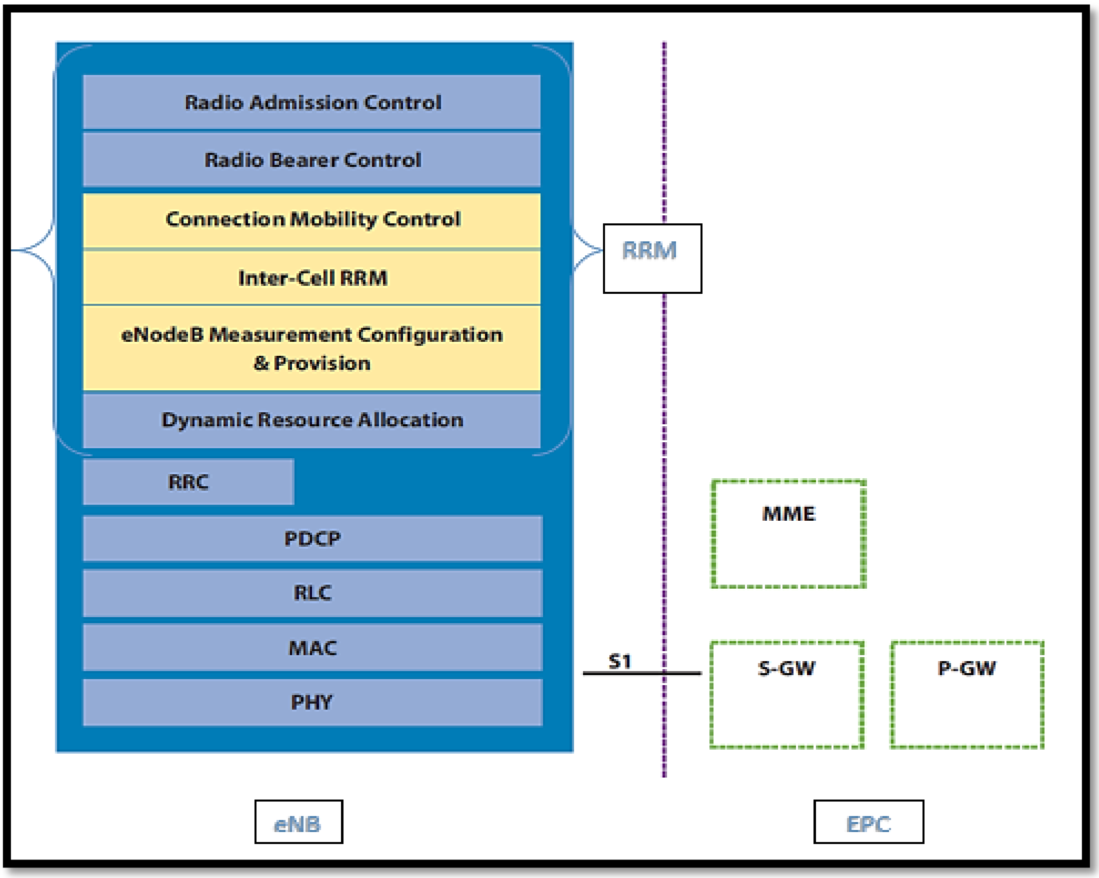

Figure 2.4: Radio Resource Management Functions in eNB Architecture

2.1.7 Radio Admission Control (RAC) framework

In E-UTRAN, RAC applies on radio bearers.The RAC functionality must evaluate if the system has enough resource available to support the incoming UE. For that purpose, Information about the state of the system is available. This encompass the different statistics or information that can be generated by the Packet Switch (PS)functionality and the Quality of Service (QoS) parameters of the UEs already present in the network with an active connection. Moreover, the RAC can use the incoming UE QoS parameters as well a channel quality indicator called Reference Signal Received Quality (RSRQ) that is transmitted on the Physical Uplink Control Channel (PUCCH). [10]

2.2 Related Work

To maintain the QoS of in-progress sessions in a cell it is important to admit a new radio bearer only if all the new bearer and the existing sessions can be guaranteed QoS according to their requirements. In [11] author propose an AC algorithm for LTE uplink utilizing the fractional power control formula agreed in 3GPP which provisions QoS. AC algorithm modifies itself with respect to load conditions variations without additional complexity. AC algorithm checks if the current resource allocation can be modified so as to admit the new user and satisfy the Guaranteed Bit Rate (GBR) requirements of all the active users and the new user. In this algorithm the closed loop adjustments of fractional power control are not taken into account.

The admission criterion for the new user is that the sum of the required number of Physical Resource Blocks (PRBs) per Time Transmission Interval (TTI) by a new user requesting admission and existing users is less than or equal to the threshold which is the total number of PRBs in the system bandwidth. In these schemes, the authors do not distinguish between the handoff call and the new call. Then, the authors do not take the prioritization between the call while basing the type of call and their QoS requirement. Also, the management of threshold is static; it does not depend on the type of call was proposed by authors in [12] .

Authors from the academia and the industry have proposed various call admission control algorithms. Bandwidth-based Call Admission Control (CAC) algorithm has been proposed in [13]. If there is sufficient bandwidth available to admit a connection, the connection gets admitted. Quality of service-aware call admission control algorithm is proposed in

[14]. When BS receives the admission control request, it shall queue it in its queue depending on the QoS parameters of the connection. The BS shall then scan through the queues to decide whether to admit the connection or not. The paper also lists the criteria to admit the connection [11]. Propose a resource allocation algorithm and a Connection Access Control scheme for LTE systems with heterogeneous services. The CAC scheme introduces a transmission guard interval which gives high priority to real-time (RT) service packets approaching the delay deadline. The Connection Admission Control scheme can adaptively adjust the threshold according to the network condition. It balances the ongoing connections of different classes of traffics but also be easy to reserve resource and support handover users potentially.

This scheme for LTE systems with heterogeneous services introduces a transmission guard interval which gives high priority to the real time (RT) service packets approaching the delay deadline. In this approach, the authors treat all the calls Handover Call and New Call (HC & NC) in the same way.

CHAPTER THREE

PERFORMANCE EVALUATION IN CALL ADMISSION CONTROL IN LTE

3.1 Introduction

In LTE QoS is enforced at the granularity level of the bearer. Each bearer is associated with a QoS class identifier (QCI). This thesis considered two type of QCI services VOIP and video.

Admission Control (AC) becomes important for the estimation of traffic or the prioritization of an incoming VOIP call or video. The admission control process does some math and activates another bearer if resources are available based on traffic activity. When a user generates data, a bearer is required to transmit the data. Hence the UE sends a bearer establishment request to eNodeB. eNodeB then executes an admission control algorithm to decide to admit the bearer or not. [10]

3.2 Call Admission Control Algorithms

In this project two algorithms were discussed, these are GBR with priority and without priority. The first algorithm is designed to allocate physical resource block according to the UE's generated GBR signal to achieve the QoS requirement.

3.2.1 Prioritized Admission Control Algorithm

Priority algorithm show in figure 3.1 start by getting the available BW for uplink and downlink in LTE system network [9]. Privileged services and schedules QCI are sorted in list with high priority. The design is initially mentioned that Guaranteed Bit Rate (GBR) QoS Class Identifier (QCI) 1,2 to carry service on the Radio Access Network (RAN). VOIP is high priority than video, if service is less priority then return to the sort list of QCI and select another one higher if found, then determine the bit rate value. if the Minimum Bandwidth(Min BW) configured Allocate Min BW to all ERABs, calculate the blocking probability of UE's if resource not available, then calculate accumulative throughput for active users. and return to the sort the list of QCI to get another service Else sort the ERAB first on priority next on GBR basis, if service last of list End else return to the sort list of QCI to get another service.

3.2.2 Non-Prioritized Admission Control Algorithm

The algorithm in Figure 3.2 without priority after getting available BW for uplink and downlink, QCI services VOIP and video are arrived randomly. GBR for actual service was examined. If Minimum BW configured allocate Minimum BW to all ERABs do equations of blocking probability, throughput and delay return to QCI. Else sort ERAB list if the list is not end return to QCI, else end.

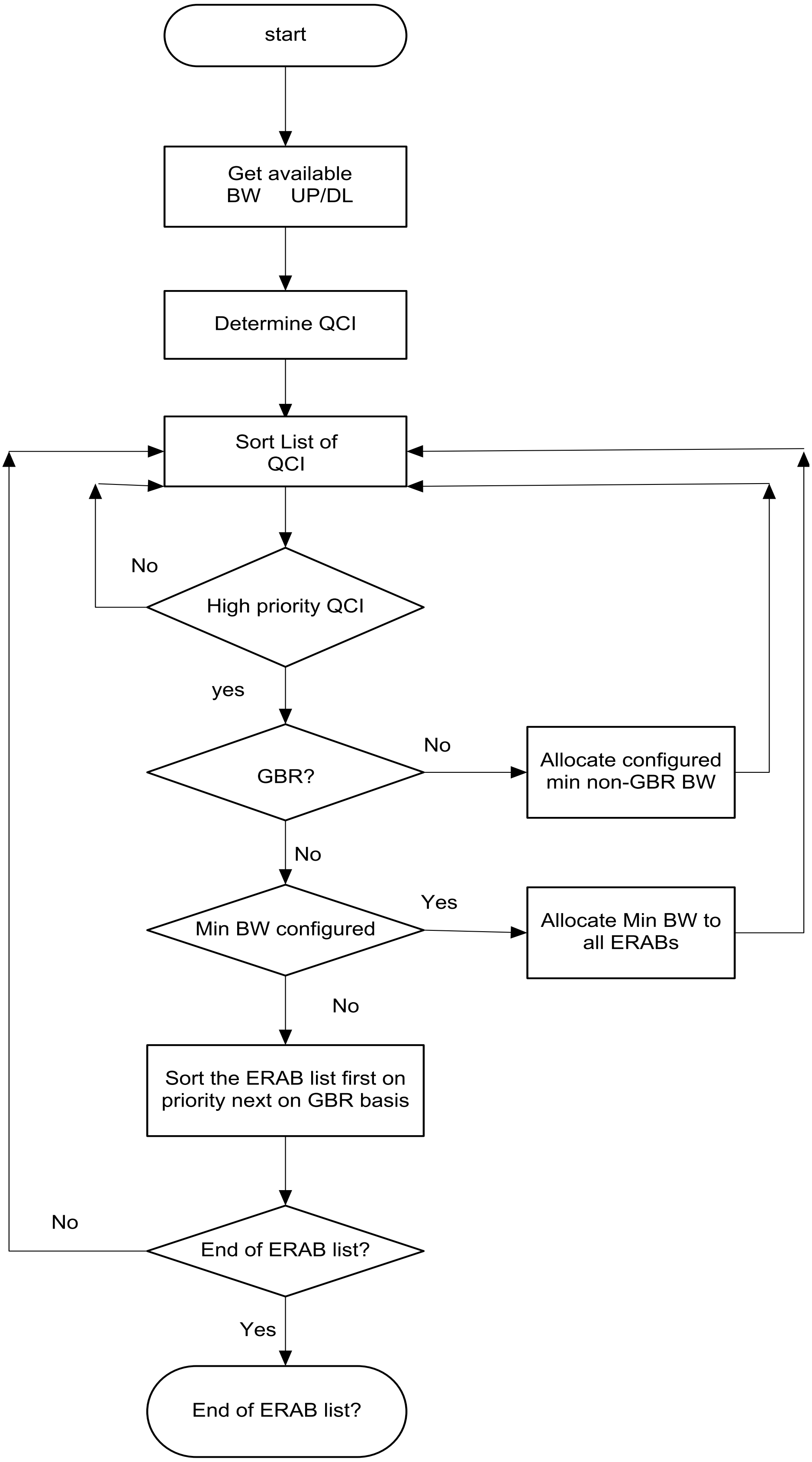

Figure1: Prioritized GBR based RAC Algorithm

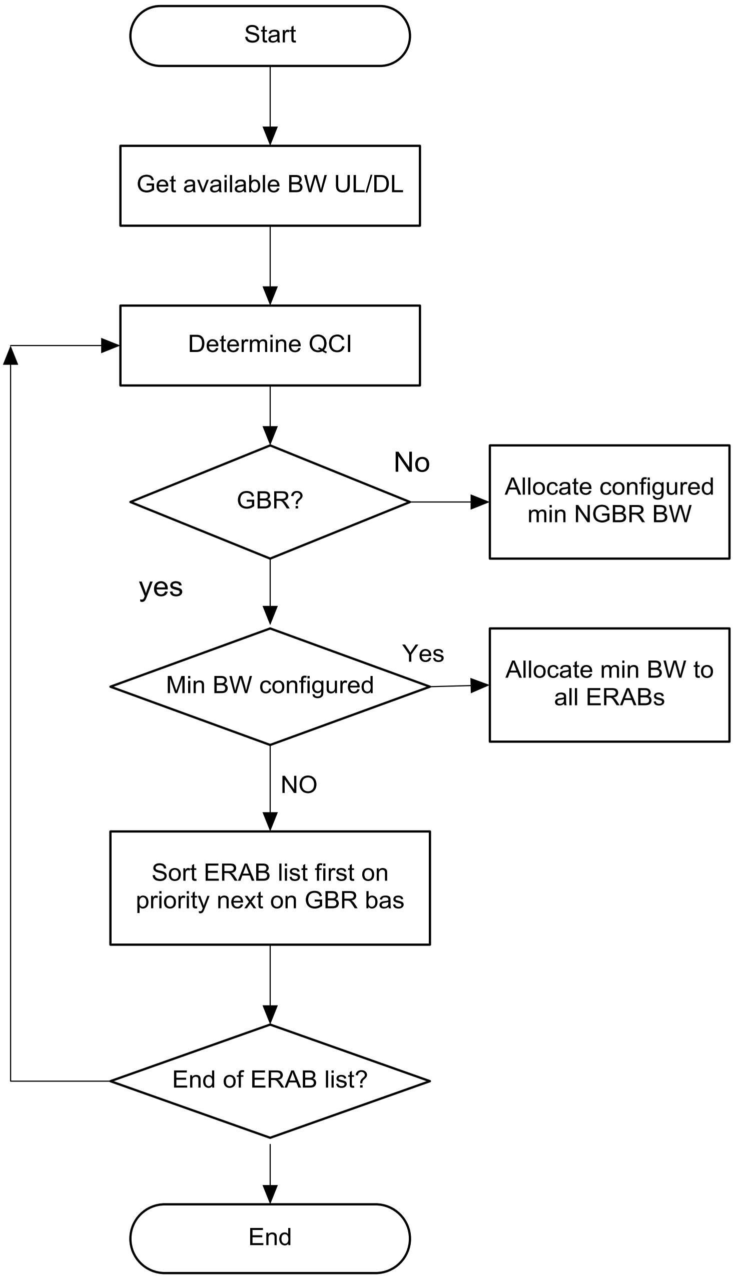

Figure 2 :Non Prioritized GBR based RAC Algorithm

3.3 Performance Evaluation Metrics for CAC Algorithms

The LTE standards promote a similar form of OFDMA, in which the end user is assigned a portion of the spectrum for a time sub-frame. Each sub-band in OFDMA is orthogonal and allocated in a TDMA fashion, along the time axis. Objective is to make the overall performance of the network under proposed admission access control scheme.

3.3.1 System Throughput

The aggregate throughput of system is defined as the summation of packets transmitted in a simulation time from eNB to all UE [15]. A certain amount of the total packets is the control portion and considered overhead. But for general throughput measurements, it increase when users are arrive as illustrative in equation 3.1 Consider the aggregate of all packets. In mathematical form, the summation throughput of the system is

$$\text{system Throughput} = \frac{1}{T}\sum_{i=1}^{K}\sum_{t=1}^{T} Psize \qquad (3.1)$$

WhereT represents the total time taken for simulation and p_{size} describes the packet size in bits transmitted from eNB to a particular user *i* aggregated for the simulation time. *K* is the total number of active users in the system.

3.3.2 Available Physical Resource Block (PRBavi)

The LTE network has bandwidth consist of PRB_i^{total} amount of PRBs allocated to service class i. PRBs occupied by PRB_i^{used} and the remind PRBs arePRB_i^{ava} it reduce whenever the number of users increase as illustrate in equation 3.2, the relationship between them:

$$PRB_i^{ava} = PRB^{tot} - PRB_i^{used} \qquad (3.2)$$

$$PRB_i^{used} = \sum_{u=1}^{k} \frac{R_i^u}{TB_i^{mcs}} \qquad (3.3)$$

R_i^u And TB_i^{mcs} represent the rate of user u and amount of transport block (TB) with modulation and code scheme(MCS) [16].

3.3.3 System Blocking Probability

Blocking probability occur when the number of users increase and not enough PRB^{avi} as illustrate in equation (3.4) .

$$P = \sum_{i=0}^{nu} \frac{blk}{nu} \qquad (3.4)$$

Where:

P: blocking probability for service i.

blk: blocked users .

nu: total number of users.

3.3.4 System Delay

The VOIP and video are real time system, thus the Delay is effect for them, it increase when the data rate is decrease as point in equation (3.5).

$$D = \frac{N}{R} \qquad (3.5)$$

The element D,N and R denoted to the delay of the service at the system , packet size and the data rate respectively.

CHAPTER FOUR

RESULT AND DISCUSSIONS

4.1 Simulation Parameter

In this chapter, a program was written using MATLAB to calculate the performance metrics of the CAC such as throughput, delay, PRB^{avi}, blocking probability. The simulation is performed by using the simulation parameters given in Table 4.1

Table 4.1: Simulation Parameter

PARAMETER	SYMBOL	VALUE
Bandwidth	BW	5 MHZ
Total data rate	TR	50 Mbps
Physical resource block	PRB	5 MHZ
Voice data rate	Rvo	12.5 Kbps
Video data rate	Rvi	180 kbps
Packet size voice	Pvo	64 Byte
Packet size video	Pvi	320 Byte
Time simulation	T	100
Transport block	TB	15

4.2 Simulation Results

In this section, four different scenarios were simulated. Two type of service were considered; service one is voice over IP(VOIP), and second is video.

4.2.1 Scenario One: Throughput vs. Number of users

In this scenario two services are considered, these are VOIP and video services. In which, twenty users are defined for both call admission mechanisms; admission with the service priority and without service priority. In this scenario, fifteen VOIP users are admitted out of twenty using the priority aware call admission control algorithm. However, for the call admission control algorithm without priority, the twenty VOIP and Video users are selected randomly based on their arrival.

Figure 4.1 shows the throughput calculated for voice, in priority based algorithm, it is increased exponentially with the increase of voice users. But, in non-priority call admission control algorithm, the throughput is developed with a rate that is less than that of the priority based call admission control because less VOIP users are admitted. From the Figure 4.1, it can be clearly seen that the throughput of priority based algorithm is higher with 33 % than for non-priority due to more VOIP users admitted.

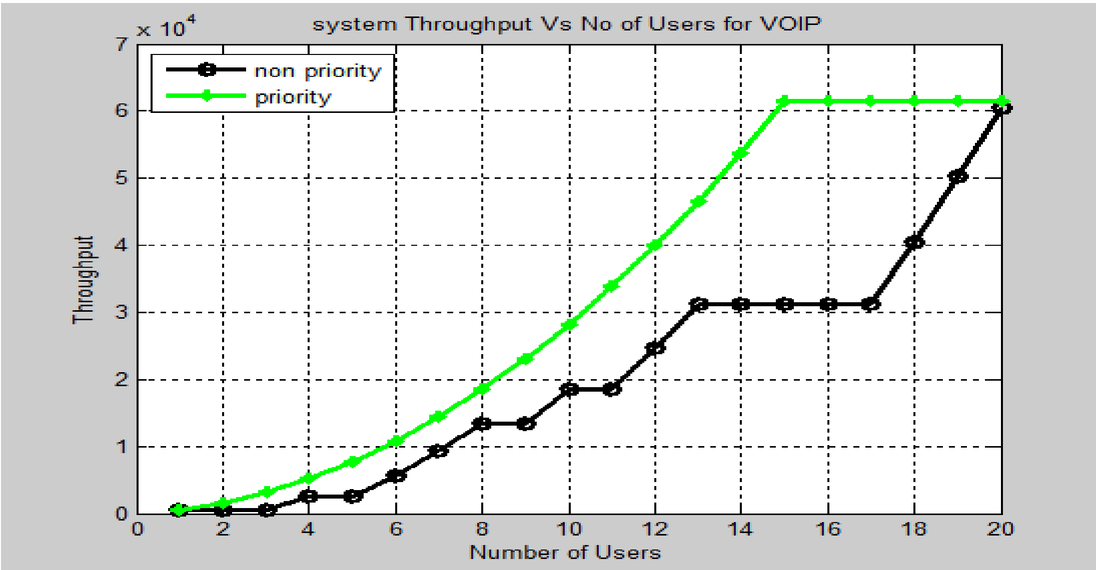

Figure4.1: Throughput vs. number of user

In Figure 4.2, the throughput is calculated considering the video service. In priority based CAC, the throughput of the Video service is increased linearly after fifteenth VOIP users are served because VOIP users are admitted first in this algorithm. On the other side, for non-priority CAC the video throughput is developed with the random arrival of Video users. It can be noted that, the Video throughput of the based algorithm achieved less throughput with 66% than non-priority CAC algorithm. Because the Video users have less priority and hence less Video users are admitted using priority based CAC.

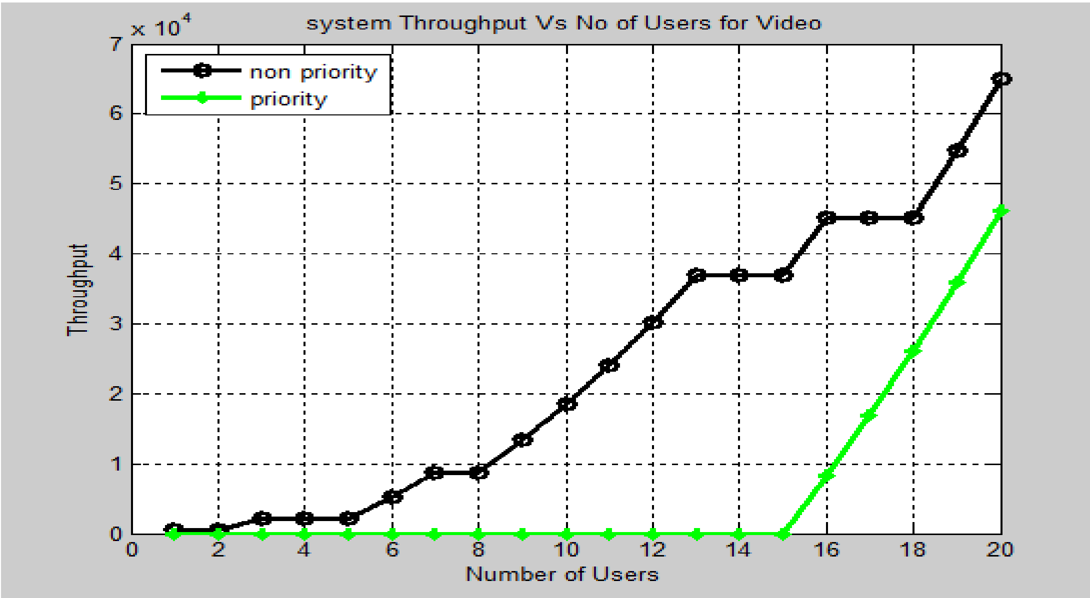

Figure4.2: Throughput vs. number of users

4.2.2 Scenario Two: Available physical Resource Block vs. Number of User

In this scenario also two services are considered (VOIP & Video). In which, twenty users are defined for both call admission mechanisms priority & Non-Priority, the PRB^{avi} is calculated for two mechanisms priority and without priority in both services .Figure 4.3 and 4.4 shows the relationship between the PRB^{avi} and the number of user in voice and video alternative, and found that the PRB^{avi} is decrease when the number of users is increase due to limited resource.

In figure 4.3 the PRB^{avi} is calculated considering the VOIP service. In priority based Admission Algorithm, PRB^{avi} is decreasing due to more VOIP users admitted. But in non-priority call admission control algorithm, the PRB^{avi} is decreased with a rate that is higher than that of the priority based call admission control because less VOIP users are admitted. it can be clearly seen that the PRB^{avi} of priority based algorithm is higher with 3.93%.

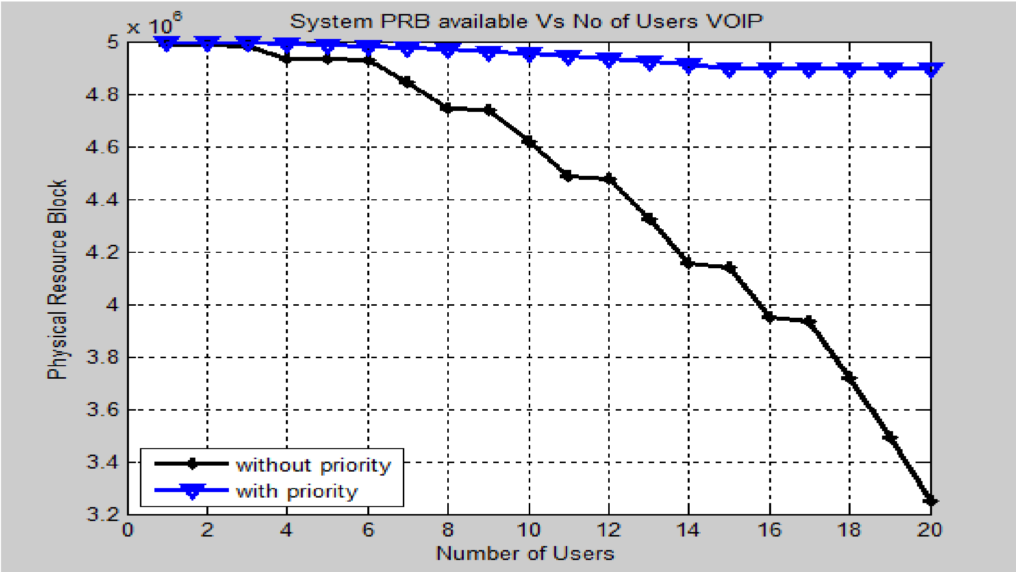

Figure 4.3: PRB^{avi} vs. number of user

In Figure 4.4, the PRB^{avi} is calculated considering the video service. In priority based CAC, the PRB^{avi} of the Video service is decreased after fifteenth VOIP users are served because VOIP users are admitted first in this algorithm. On the other side, for non-priority Admission Algorithm the video PRB^{avi} is decreased with the random arrival of Video users. It can be noted that, the Video PRB^{avi} of priority based algorithm achieved high PRB^{avi} with 10.80% than non-priority; because the Video users have less priority and hence less Video users are admitted using priority algorithm.

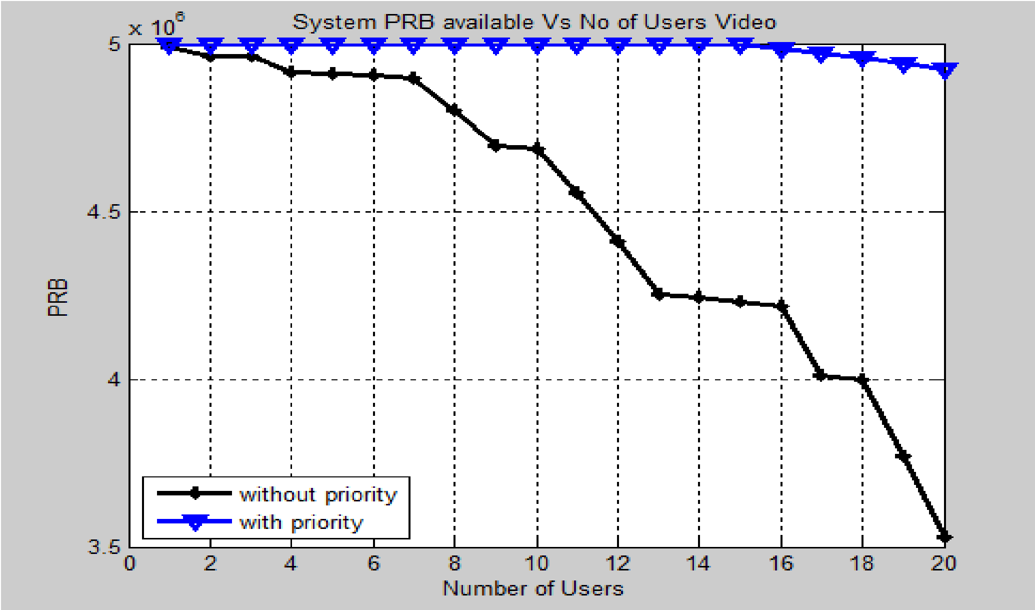

Figure4.4: PRB^{avi} vs. number of users

4.2.3 Scenario Three: Blocking Probability vs. Number of Users

In this scenario also two services are considered, these are VOIP and video services. In which, Thirty users are defined for both call admission mechanisms admission with the service priority and without service priority. In this scenario, Twenty five VOIP users are admitted out of thirty using the priority aware call admission control algorithm. However, for the call admission control algorithm without priority, the thirty VOIP and Video users are selected randomly based on their arrival.

Figure 4.5 shows the blocking probability calculated for voice, in priority based algorithm, it is equal zero that means no blocking occurred, of voice users. But in non-priority call admission control algorithm, increasing linearity when the available physical resource is decreased and number of VOIP users is increased, then the blocking probability the priority based algorithm is higher with 100% than non priority due to more VOIP users admitted.

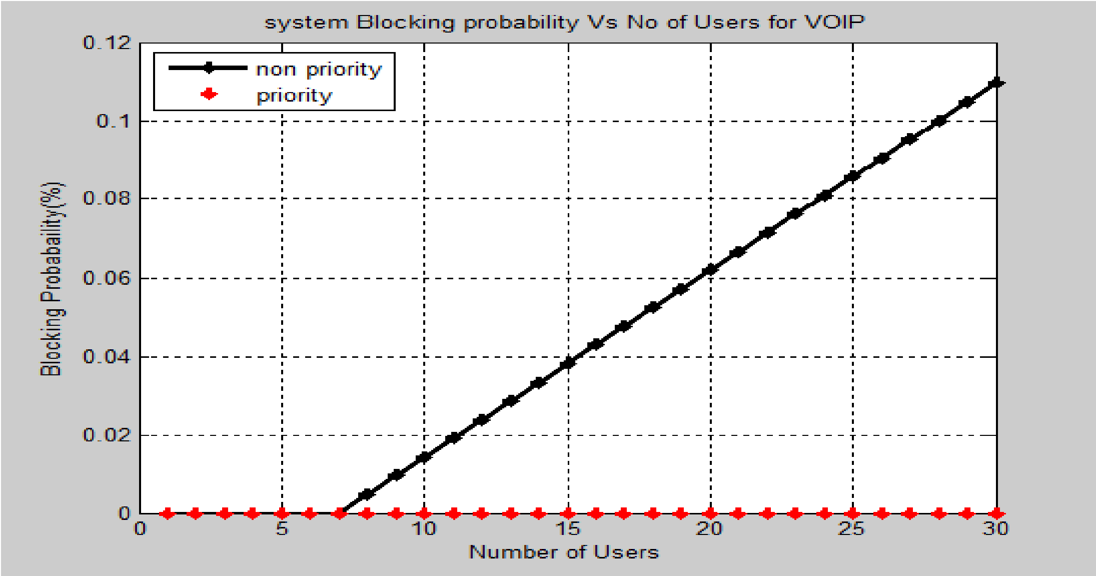

Figure4.5: Blocking Probability vs. Number of user

The figure 4.6 the blocking probability is calculated considering the video service. In priority based CAC, blocking probability of the Video service is increased after twenty five VOIP users are Admit because VOIP users are admitted first in this algorithm. On the other side, for non-priority CAC the video blocking probability accrue when PRB_{avi} is not enough. It can be noted that, the Video blocking probability of the based algorithm achieved less blocking probability in priority with 98.8% than non-priority CAC algorithm. Because the Video users have less priority and hence less Video users are admitted using priority based CAC.

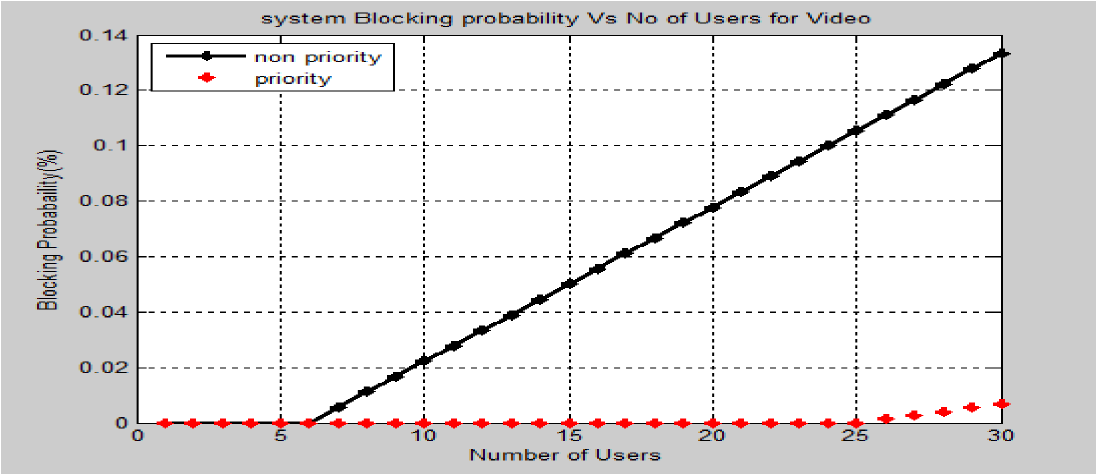

Figure4.6: Blocking Probability vs. Number of user

4.2.4 Scenario Four: Delay vs. Number of user

In this scenario two services are considered, these are VOIP and video services. In which, twenty users are defined for both call admission mechanisms admission with the service priority and without service priority. In this scenario, fifteen VOIP users are admitted out of twenty using the priority aware call admission control algorithm. However, for the call admission control algorithm without priority, the twenty VOIP and Video users are selected randomly based on their arrival.

Figure 4.7 shows the Delay calculated for VOIP, in priority based algorithm, it is increased linearity with the increase when fifteen of VOIP users are Admitted and being fixed value to video user. But in non-priority call admission control algorithm, the delay is developed with a rate that is less than that of the priority based call admission control because less VOIP users are admitted. From the Figure 4.7, it can be clearly seen that the delay of priority based algorithm is less with 80.17%. Than non-priority due to more VOIP users admitted.

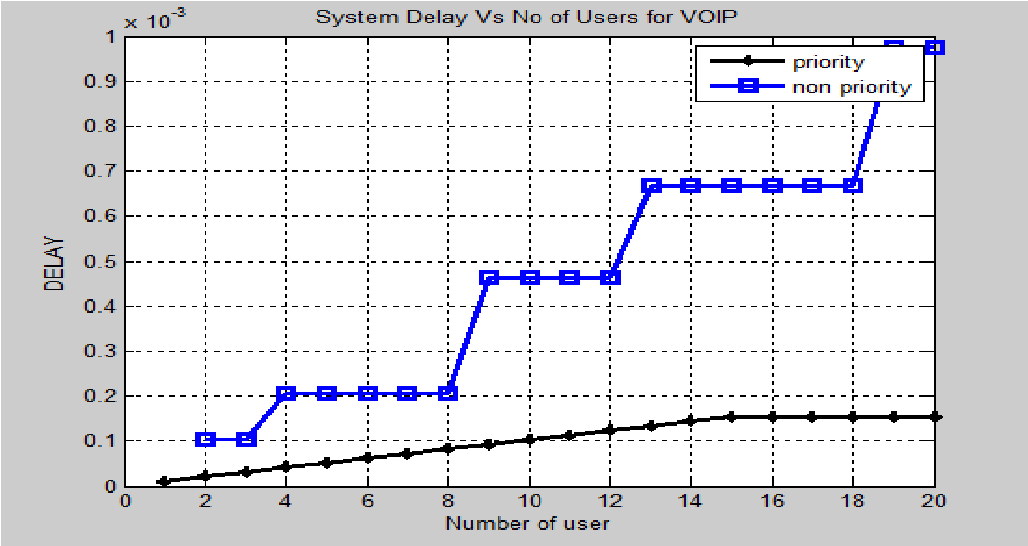

Figure 4.7: Delay vs. Number of user

Figure 4.8 the delay is calculated considering the video service. In priority based CAC, the Delay of the Video service is increased after fifteenth VOIP users are admit. On the other hand, for non-priority Admission Algorithm the video Delay is increased with the random arrival of video users. So the video delay of priority based algorithm achieved higher Delay with 99.7% than non-priority.

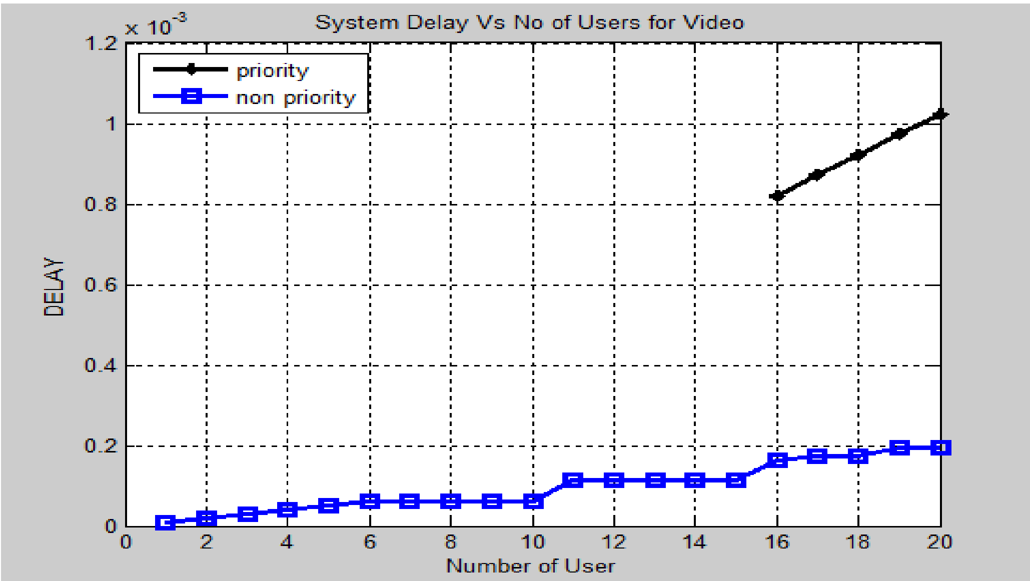

Figure 4.7: Delay vs. Number of user

CHAPTER FIVE

CONCLUSION AND RECOMMNDATION

5.1 Conclusion

This thesis was focused on evaluating two algorithms; the first priority algorithm aims to provide best QoS and best user satisfaction by using different parameters. The higher priority, guaranteed bit rate and minimum BW configured are used in decision making. Admission control algorithms generally take into consideration the QoS parameters like throughput, available physical resource block and delay, etc., during the admission control process. MATLAB was used for the evaluation of algorithms. Simulation results proved higher connection admission rate to show the percentage of Improvement for priority services compared to non-priority services. The VOIP results cleared that the throughput is higher with 33%, PRB^{avi} is less by 3.93%, the blocking probability is less by 100% and the Delay is less with 80.17%.

5.2 Recommendations

After finishing this research there is still some issues can be considered for future research such as:

- A further study of algorithms could lead to the extension of non-GBR in combination with the algorithm.
- You can also look at other parameter such as cost and evaluate the effect of the algorithm.
- In this simulator, two services are considered in the study, so in the future can be considered other services.

References

[1] Asadi, A., Wang, Q., & Mancuso, V. (2014). A survey on device-to-device communication in cellular networks. *Communications Surveys & Tutorials, IEEE*, *16*(4), 1801-1819.

[2] Chadchan, S. M., and C. B. Akki. "Priority-scaled preemption of radio resources for 3GPP LTE networks." *International Journal of Computer Theory and Engineering* 3.5 (2011).

[3] Wang, You-Chiun, and Song-Yun Hsieh. "QoS-provisioning Downlink Resource Management in 4G Cellular Systems."

[4] LTE, D., Uplink, O. F. D. M. A., SC-FDMA, M. I. M. O., & TDD, F. Liwen Zhang, lz6@ cec. wustl. edu (A project report written under the guidance of Prof. Raj Jain) Download.

[5] Chapter 4 Long Term Evolution (Professor Ramon Ferrus (UPC)).

[6]Policies for Public Safety Use of Commercial commercial WirelessNetworks by Ryan Hallahan and Jon M.Peha,Carnegie Mellon University,October2010;http://users.ece.cmu.edu/~peha/public_safety_priority_access.pdf

[7]https://into.aalto.fi/.../Swaminathan_Vasanth_Rajaraman*Accessed: 25/9/2015*

[8] ALGUR, Siddu Payappa, and Niharika KUMAR. "Novel, graded, priority oriented admission control in mobile networks."

[9]http://www.tcs.com/resources/white_papers/Pages/RadioResource-Management-Radio-Admission-Control-Bearer Control.aspx 25 Sep 2015 22:35:41 GMT.

[10] Nageshar, Nikesh. "Voice quality control in packet switched wireless
networks." PhD diss., Faculty of Engineering, University of the Witwatersrand, Johannesburg, 2012.

[11] M. Anas, C. Rosa, F. D.Calabrese, P. H. Michaelsen, K. I. Pedersen, and P. E. Mogensen (2008), “QoS-Aware Single Cell Admission Control for UTRAN LTE Uplink”, VTC Spring,IEEE.

[12]Haipeng LEI, Mingliang YU, Anjun ZHAO, Yongyu CHANG, and Dacheng YANG, (2008) , “Adaptive Connection Admission Control Algorithm for LTE Systems”, VTC Spring , IEEE.

[13]Ping Xia, Vikram Chandrasekhar and Jeffrey G. Andrews (2010) ", *Open vs. Closed Access Femtocells in the Uplink,* 15 feb, p. 1.

[14] Magdalena Nohrborg *the Mobile Broadband Standard, Available at:* http://www.qub.ac.uk *(Accessed: 3/3/2015).*

[15]*Ali, Salman, Muhammad Zeeshan, and Anjum Naveed. "A capacity and minimum guarantee-based service class-oriented scheduler for LTE networks." EURASIP Journal on Wireless Communications and Networking 2013.1 (2013): 1-15.*

[16]*Jie Wang, YangfanQiu(2013) ", A New Call Admission Control Strategy for LTE Femtocell Networks, , p. 1.*

APPENDIX

APPENDIX

```
clc,clear,close all
 PRB=5000000;                  %Total BW
pvo=64*8;           %packet size voice/ip
Pvi=320*8;          %packet size video/ip
Rvo=12.5*10^3;       %Data rate of voice
Rvi=180*10^3;       %Data rate of video
TR=50000000;         %Total Data Rate
T=100;              %Time sumilation
ser = [randi([0 1],1,20) ones(1,30)];    % random genration of the service
ser = ser(randperm(20));                %zero for voice and 1 for video
%%%%%%% Throughput of video%%%
disp(ser);
nov=0;  %number of user
 nov1=zeros;
 thro=zeros;
 sum=0;
for j=1:numel(ser)
if ser(j)==0
        fprintf('\n')
        disp('service is voice');
        nov=nov+1;
        nov1(j)=nov;
        thro(j)=sum/T;
else
        fprintf('\n')
       disp('service is video');
       nov=nov+1;
        nov1(j)=nov;
for n=1:nov
for k=1:T
```

```
        sum =sum+pvo;
end
end
         thro(j)=sum/T;
end
end
  Qvo=[1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 1 2 3 4 5 6 7 8 9 10 11 12 13 14
15];
  Qvi=[16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 16 17 18 19 20 21 22
23 24 25 26 27 28 29 30];
servi=linspace(1,20,20);
nov=0;  %number of user
 nov1=zeros;
 throp=zeros;
 sum=0;
for i=1:numel(servi)
if servi(i)==Qvo(i)
      disp('service is voice');
        nov=nov+1;
        nov1(i)=nov;
        throp(i)=sum/T;
else
        servi(i)=Qvi(i);
        fprintf('\n')
       disp('service is video');
       nov=nov+1;
        nov1(i)=nov;
for n=1:nov
for k=1:T
        sum =sum+pvo;
end
end
       throp(i)=sum/T;
```

```
end
end
%%%%%%%%%% throughput for voice%%%%%%%%
ser = [randi([0 1],1,20) ones(1,30)];    % random genration of the service
ser = ser(randperm(20));                 %zero for voice and 1 for video
disp(ser);
nov=0;  %number of user
 nov1=zeros;
 thro1=zeros;
 sum=0;
for j=1:numel(ser)
if ser(j)==0
        fprintf('\n')
        disp('service is voice');
        nov=nov+1;
        nov1(j)=nov;
for n=1:nov
for k=1:T
        sum =sum+pvo;
end
end
        thro1(j)=sum/T;
else
        fprintf('\n')
       disp('service is video');
       nov=nov+1;
        nov1(j)=nov;
    thro1(j)=sum/T;
end
end
  Qvo=[1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 1 2 3 4 5 6 7 8 9 10 11 12 13 14
15];
```

```
Qvi=[16  17  18  19  20  21  22  23  24  25  26  27  28  29  30  16  17  18  19  20  21  22
23 24 25 26 27 28 29 30];
servi=linspace(1,20,20);
nov=0;  %number of user
 nov1=zeros;
 throp1=zeros;
 sum=0;
for i=1:numel(servi)
if servi(i)==Qvo(i)
      disp('service is voice');
        nov=nov+1;
        nov1(i)=nov;
for n=1:nov
for k=1:T
        sum =sum+pvo;
end
end
        throp1(i)=sum/T;
else
        fprintf('\n')
       disp('service is video');
       nov=nov+1;
        nov1(i)=nov;
       throp1(i)=sum/T;
end
end
%%%%%%%%%%%%5 PRB for Video %%%%%%%%%%%%%%%%
TB=15;
ser = [randi([0 1],1,20) ones(1,30)];    % random genration of the service
ser = ser(randperm(20));                 %zero for voice and 1 for video
disp(ser);
no=0;
PRBavo=PRB;
```

```
pr=zeros;
nov1=zeros;
PRBuse=0;
for i=1:numel(ser)
if PRBavo<=0
       blk=blk+1;    %number of blocked users
else
if ser(i)==0
          fprintf('\n')
        disp('service is voice');
       no=no+1;
       nov1(i)=no;
for j=1:no
PRBuse=(Rvo/TB)+PRBuse; %PRBuse
end
PRBavo =PRB-PRBuse;
pr(i)=PRBavo;
else
     fprintf('\n')
       disp('service is video');
        no=no+1;
        nov1(i)=no;
for j=1:no
PRBuse=(Rvi/TB)+PRBuse; %PRBuse
end
PRBavi =PRB-PRBuse;
pr(i)=PRBavi;
end
end
end
Qvo=[1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 1 2 3 4 5 6 7 8 9 10 11 12 13 14
15];
```

```
Qvi=[16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 16 17 18 19 20 21 22
23 24 25 26 27 28 29 30];
servi=linspace(1,20,20);
no=0;
PRBavo=PRB;
PRBavi=PRB;
prt=zeros;
nov1=zeros;
PRBuse=0;
for i=1:numel(servi)
if PRBavo<=0 && PRBavi<=0
else
if servi(i)==Qvo(i)
          fprintf('\n')
        disp('service is voice')
       no=no+1;
PRBavo =PRB-PRBuse;
prt(i)=PRBavo;
nov1(i)=no;
else
    servi(i)=Qvi(i);
     fprintf('\n')
       disp('service is video')
        no=no+1;
for j=1:no
PRBuse=(Rvo/TB)+PRBuse; %PRBuse
end
PRBavi =PRB-PRBuse;
prt(i)=PRBavi;
nov1(i)=no;
end
end
end
```

```
%%%%%%%%%%%%5 PRB for Voice %%%%%%%%%%%%%%%%

TB=15;
ser = [randi([0 1],1,20) ones(1,30)];    % random genration of the service
ser = ser(randperm(20));                 %zero for voice and 1 for video
disp(ser);
no=0;
PRBavo=PRB;
pr1=zeros;
nov1=zeros;
PRBuse=0;
for i=1:numel(ser)
if PRBavo<=0
    blk=blk+1;   %number of blocked users
else
if ser(i)==0
      fprintf('\n')
     disp('service is voice');
    no=no+1;
    nov1(i)=no;
for j=1:no
PRBuse=(Rvo/TB)+PRBuse; %PRBuse
end
PRBavo =PRB-PRBuse;
pr1(i)=PRBavo;
else
   fprintf('\n')
    disp('service is video');
     no=no+1;
     nov1(i)=no;
for j=1:no
PRBuse=(Rvi/TB)+PRBuse; %PRBuse
end
```

```
PRBavi =PRB-PRBuse;
pr1(i)=PRBavi;
end
end
end
Qvo=[1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15];
Qvi=[16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 16 17 18 19 20 21 22 23 24 25 26 27 28 29 30];
servi=linspace(1,20,20);
no=0;
PRBavo=PRB;
PRBavi=PRB;
prt1=zeros;
nov1=zeros;
PRBuse=0;
sum=0;
for i=1:numel(servi)
if PRBavo<=0 && PRBavi<=0
else
if servi(i)==Qvo(i)
          fprintf('\n')
        disp('service is voice')
       no=no+1;
for j=1:no
PRBuse=(Rvo/TB)+PRBuse; %PRBuse
end
PRBavo =PRB-PRBuse;
prt1(i)=PRBavo;
nov1(i)=no;
else
    servi(i)=Qvi(i);
     fprintf('\n')
```

```
        disp('service is video')
         no=no+1;
PRBavi =PRB-PRBuse;
prt1(i)=PRBavi;
nov1(i)=no;
end
end
end
%%%%%%%%%%%% Blocking probability %%%%%%%%%%
ser = [randi([0 1],1,20) ones(1,30)];    % random genration of the service
ser = ser(randperm(30));                 %zero for voice and 1 for video
disp(ser)
PRBavi=PRB;
PRBuse=0;
blk=0;
nu=zeros;
nou1=zeros;
nou=0;  %number of user
p1=zeros;
for i=1:numel(ser)
if PRBavi<=0
         blk=blk+1;    %number of blocked users
else
if ser(i)==0
         fprintf('\n')
         disp('service is voice');
         nou=nou+numel(ser);
         nou1(i)=nou;
for h=1:nou
         PRBuse=(Rvo/TB)+PRBuse ;      %PRBuse
end
         PRBavi=PRB-PRBuse;
else
```

```
        fprintf('\n')
        disp('service is video');
        nou=nou+numel(ser);
        nou1(i)=nou;
for h=1:nou
        PRBuse=(Rvi/TB)+PRBuse ;        %PRBuse
end
        PRBavi=PRB-PRBuse;
end
end
    p=blk/nou;  % blocking probabaility
    p1(i)=p;
fprintf('\n%s %d','Number of Blocked Services:',blk);
nu(i)=i;
end
Qvo=[1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 1 2 3 4 5 6 7 8 9 10 11 12 13 14
15];
Qvi=[16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 16 17 18 19 20 21 22
23 24 25 26 27 28 29 30];
servi=linspace(1,30,30);
PRBavi=PRB;
PRBuse=0;
blk=0;
nou1=zeros;
nou=0;  %number of user
pt1=zeros;
 nu=zeros;
for i=1:numel(servi)
if PRBavi<=0
         blk=blk+1;    %number of blocked users
else
if servi(i)==Qvo(i)
         fprintf('\n')
```

```
        disp('service is voice');
        nou=nou+numel(servi);
for h=1:nou
        PRBuse=(Rvo/TB)+PRBuse ;      %PRBuse
end
        PRBavi=PRB-PRBuse;
        nou1(i)=nou;
else
        servi(i)=Qvi(i);
       fprintf('\n')
       disp('service is video');
     nou=nou+numel(servi);
       PRBavi=PRB-PRBuse;
       nou1(i)=nou;
end
end
    pt=blk/nou;  % blocking probabaility
 pt1(i)=pt;
   nu(i)=i;
fprintf('\n%s %d','Number of Blocked Services:',blk);
end
  ser = [randi([0 1],1,25) ones(1,30)];    % random genration of the service
ser = ser(randperm(30));                %zero for voice and 1 for video
disp(ser)
  PRBavi=PRB;
PRBuse=0;
blk=0;
nu=zeros;
nou1=zeros;
nou=0;  %number of user
pb=zeros;
p=0;
```

```
for i=1:numel(ser)
if PRBavi<=0
        blk=blk+1;    %number of blocked users
else
if ser(i)==0
        fprintf('\n')
        disp('service is voice');
        nou=nou+numel(ser);
        nou1(i)=nou;
for h=1:nou
        PRBuse=(Rvo/TB)+PRBuse ;      %PRBuse
end
        PRBavi=PRB-PRBuse;
else
       fprintf('\n')
       disp('service is video');
       nou=nou+numel(ser);
%nov=nov+1;
       nou1(i)=nou;
for h=1:nou
       PRBuse=(Rvi/TB)+PRBuse ;       %PRBuse
end
       PRBavi=PRB-PRBuse;
end
end
    p=blk/nou;  % blocking probabaility
    pb(i)=p;
fprintf('\n%s %d','Number of Blocked Services:',blk);
nu(i)=i;
end
Qvo=[1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 1 2 3 4 5 6 7 8 9 10 11 12 13 14
15];
```

```
Qvi=[16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 16 17 18 19 20 21 22
23 24 25 26 27 28 29 30];
servi=linspace(1,30,30);
PRBavi=PRB;
PRBuse=0;
blk=0;
nou1=zeros;
nou=0;  %number of user
bpt1=zeros;
 nu=zeros;
for i=1:numel(servi)
if PRBavi<=0
        blk=blk+1;    %number of blocked users
else
if servi(i)==Qvo(i)
        fprintf('\n')
        disp('service is voice');
        nou=nou+numel(servi);
        PRBavi=PRB-PRBuse;
        nov=nov+1;
        nou1(i)=nou;
else
        servi(i)=Qvi(i);
       fprintf('\n')
       disp('service is video');
     nou=nou+numel(servi);
for h=1:nou
        PRBuse=(Rvo/TB)+PRBuse ;      %PRBuse
end
       PRBavi=PRB-PRBuse;
        nou1(i)=nou;
end
end
```

```
   pt=blk/nou;  % blocking probabaility
 bpt1(i)=pt;
   nu(i)=i;
fprintf('\n%s %d','Number of Blocked Services:',blk);
end

%%%% The Delay %%%%%%%%%%%%
  del1=zeros;
nvo=0;
Rvo1=0;
nvo1=zeros;
R1=zeros;
ser = [randi([0 1],1,20) ones(1,30)];    % random genration of the service
ser = ser(randperm(20));                 %zero for voice and 1 for video
disp(ser)
R=zeros;
for i=1:numel(ser)
if ser(i)==0
        fprintf('\n')
        disp('service is voice')
        nvo=nvo+1;
        nvo1(i)=nvo;
        TRO=TR-Rvi;
        Rvo1=TRO/nvo;
        R(i)=Rvo1;
        del1(i)=Pvi/Rvo1;
else
         fprintf('\n')
        disp('service is Video')
         nvo=nvo+1;
        nvo1(i)=nvo;
       R1(i)=Rvo1;
         del1(i)=Pvi/Rvo1;
```

```
end
end
del=zeros;
 nvo=0;
nvo1=zeros;
R1=zeros;
Rvo1=0;
Qvo=[1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 1 2 3 4 5 6 7 8 9 10 11 12 13 14
15];   %no of voice services
Qvi=[16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 16 17 18 19 20 21 22
23 24 25 26 27 28 29 30]; %no of v services
servi=linspace(1,20,20);
disp(servi)
for i=1:numel(servi)
if servi(i)==Qvo(i)
        fprintf('\n')
        disp('service is voice')
        nvo=nvo+1;
        nvo1(i)=nvo;
        Rvo1=TR/nvo;
        R(i)=Rvo1;
        del(i)=pvo/Rvo1;
else
servi(i)=Qvi(i);
nvo=nvo+1;
        nvo1(i)=nvo;
        R1(i)=Rvo1;

       del(i)=pvo/Rvo1;
end
end

%%%%%%%%%%%%%%%%%%%%%%-------------------------
```

```
%%%%55 Video%%%%%%%%%%%%%%%%
 dl1=zeros;
nvo=0;
Rvo1=0;
nv1=zeros;
R1=zeros;
ser = [randi([0 1],1,20) ones(1,30)];    % random genration of the service
ser = ser(randperm(20));                 %zero for voice and 1 for video
disp(ser)
R=zeros;
for i=1:numel(ser)
if ser(i)==1
            fprintf('\n')
            disp('service is voice')

            nvo=nvo+1;
            nv1(i)=nvo;
            TRO=TR-Rvo;
            Rvo1=TRO/nvo;
            R(i)=Rvo1;
            dl1(i)=pvo/Rvo1;
else
             fprintf('\n')
            disp('service is Video')
             nvo=nvo+1;
            nv1(i)=nvo;
           R1(i)=Rvo1;
            dl1(i)=pvo/Rvo1;
end
end
m=zeros;
de=zeros;
nvo=0;
```

```
nv1=zeros;
R1=zeros;
Rvo1=0;
Qvo=[1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 1 2 3 4 5 6 7 8 9 10 11 12 13 14
15];
Qvi=[16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 16 17 18 19 20 21 22
23 24 25 26 27 28 29 30];
servi=linspace(1,20,20);
disp(servi)
for i=1:numel(servi)
if servi(i)==Qvi(i)
        fprintf('\n')
        disp('service is voice')
        nvo=nvo+1;
        nv1(i)=nvo;
        Rvo1=TR/nvo;
        R(i)=Rvo1;
        de(i)=Pvi/Rvo1;
else
servi(i)=Qvo(i);
nvo=nvo+1;
        nv1(i)=nvo;
        R1(i)=Rvo1;

       de(i)=Pvi/Rvo1;
end
end
plot(nv1,de,'-*K',nv1,dl1,'-s','linewidth',2.5)
hleg1=legend('priority','non priority');
  set(hleg1,'Location','Northwest')
title('System Delay Vs No of Users for Video')
hold on
grid on
```

```
xlabel('Number of user')
ylabel('DELAY')
figure
plot(nvo1,del,'-*K',nvo1,del1,'-s','linewidth',2.5)
hleg1=legend('priority','non priority');
  set(hleg1,'Location','Northeast')
title('System Delay Vs No of Users for VOIP')
hold on
grid on
xlabel('Number of user')
ylabel('DELAY')
figure
 plot(nu,pb,'-*k',nu,bpt1,'*r','linewidth',2.5)
hleg1=legend('non priority','priority');
  set(hleg1,'Location','NorthWest')
title('system Blocking probability Vs No of Users for Video')
  hold on
  grid on
  xlabel('Number of Users')
  ylabel('Blocking Probabaility')
 figure
 plot(nu,p1,'-*k',nu,pt1,'*r','linewidth',2.5)
hleg1=legend('non priority','priority');
  set(hleg1,'Location','NorthWest')
title('system Blocking probability Vs No of Users for VOIP')
  hold on
  grid on
  xlabel('Number of Users')
  ylabel('Blocking Probabaility')
figure
plot(nov1,pr,'-*k',nov1,prt,'-v','linewidth',2.5)
hleg1=legend('without priority','with priority');
set(hleg1,'Location','SouthWest')
```

```
title('System PRB available Vs No of Users Video')
xlabel('Number of services')
  ylabel('PRB')
  hold on
grid on
figure
plot(nov1,pr1,'-*k',nov1,prt1,'-v','linewidth',2.5)
hleg1=legend('without priority','with priority');
set(hleg1,'Location','SouthWest')
title('System PRB available Vs No of Users VOIP')
xlabel('Number of Users')
  ylabel('Physical Resource Block(%)')
  hold on
grid on
 figure
 plot(nov1,thro1,'-ko',nov1,throp1,'-*g','linewidth',2.5)
hleg1=legend('non priority','priority');
  set(hleg1,'Location','NorthWest')
title('system Throughput Vs No of Users for VOIP')
  hold on
 grid on
  xlabel('Number of Users ')
  ylabel('Throughput')
 figure
 plot(nov1,thro,'-ko',nov1,throp,'-*g','linewidth',2.5)
hleg1=legend('non priority','priority');
  set(hleg1,'Location','NorthWest')
title('system Throughput Vs No of Users for Video')
  hold on
 grid on
  xlabel('Number of Users ')
  ylabel('Throughput')
```