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Performance Evaluation of Call Admission Control Strategies for LTE

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قال الله تعالى:

﴿اللَّهُ نُورُ السَّمَاوَاتِ وَالأَرْضِ مَثَلُ نُورِهِ كَمِشْكَاةٍ فِيهَا مِصْبَاحٌ الْمِصْبَاحُ فِي زُجَاجَةٍ الزُّجَاجَةُ كَأَنَّهَا كَوْكَبٌ دُرِّيٌّ يُوقَدُ مِن شَجَرَةٍ مُّبَارَكَةٍ زَيْتُونَةٍ لّا شَرْقِيَّةٍ وَلا غَرْبِيَّةٍ يَكَادُ زَيْتُهَا يُضِيءُ وَلَوْ لَمْ تَمْسَسْهُ نَارٌ نُّورٌ عَلَى نُورٍ يَهْدِي اللَّهُ لِنُورِهِ مَن يَشَاء وَيَضْرِبُ اللَّهُ الأَمْثَالَ لِلنَّاسِ وَاللَّهُ بِكُلِّ شَيْءٍ عَلِيمٌ﴾

صدق الله العظيم

سورة النور الآيه 35

Dedication

To our families…

For each one we love

To everyone who helped us through this year

To each of we spent with them the most beautiful moments

(Our Colleagues..)

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First of all we want to thank Allah for blessing us and giving us the power and complete this thesis. We would like to thank our supervisor **Dr. Ibrahim Elkhdir** for his advices and support during the writing of this thesis. His knowledge and dedication and opinion were useful in completing this research. We also want to thank everyone who supported us academically regardless of that support.

Abstract

The coexistence of different cellular networks in the same area necessitates joint radio resource management for enhanced QoS provisioning and efficient radio resource utilization. It has been proposed adaptive bandwidth management and joint call admission control (JCAC) scheme for heterogeneous cellular networks. The main goal of this research is to evaluate a proposed admission control algorithms in providing QoS . Two call admission controls are compared AJCAC and NAJCAC and we investigates performance metrics new call blocking probability, handoff call dropping probability, delay and throughput. MATLAB is used for the evaluation of algorithms. Simulation results shows higher connection admission rate with AJCAC. The results shows that as the number of user increase the rejection probability in both algorithms. Moreover, the result shows that the connection level of QoS can be significantly improved by using the AJCAC scheme.

المستخلص

وجود مجموعة من الشبكات الخلوية في نفس المنطقة الجغرافية يتطلب ادارة مشتركة لمصادر الراديو لتحسين جودة الخدمات المقدمة واستغلال المصادر بكفاءة عالية وتم اقتراح خوارزمية التحكم في قبول المكالمات التكيفي للشبكات الخلوية غير المتجانسة.

الهدف الرئيسي من هذا البحث هو تقييم خوارزميتين في التحكم في الدخول لتحسين جودة الخدمة وتمت مقارنة ادائية الخوارزميات المقترحة وفقا لاحتمالية الغاء والتاخير والخرج في الشبكات اللاسلكية المتجانسة.الخوارزمية الاولى تسمى بـ NAJCAC والثانية هي AJCAC, نتائج المحاكاة اظهرت ادائية عاليه في الاداء في الخوارزمية الثانية عند مقارنتها مع الاولى ,تم تطبيقها في الماتلاب لتقييم الخوارزميات واظهرت نتايج المحاكاة اعلي معدل لقبول الاتصال باستحدام خوارزمية التحكم في قبول المكالمات التكيفي .

واظهرت النتائج ان زيادة عدد المستخدمين تزيد من نسبة احتمالية الرفض في كلتا الخوارزميتين وان مستوي الوصول لجودة الخدمات قد تحسن بصورة واضحة في خوارزمية التحكم في قبول المكالمات التكيفية المشتركة.

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List of abbreviations

2G	Second Generation
3GPP	Third Generation Partnership Project
LTE	Long Term Evolution
CAC	Call Admission Control
RAN	Radio Access Network
eUTRAN	Evolved UMTS Terrestrial Radio Access Network
OFDMA	Orthogonal Frequency Division Multiple Access
SC-FDMA	Single Carrier frequency Division Multiple Access
FTP	File Transfer Protocol
QOS	Quality of Service
GoS	Grade of services
NC	New Call
HC	Handoff call
RB	Resource Block
WLAN	Wireless Local Area Network
Wi-MAX	Worldwide Interoperability for Microwave Access
NGWN	New Generation Wireless Network
SIR	Signal to Interference Ratio
MCDM	Multi Criteria Decision Making
PRB	Physical Resource Block
GC	Guard Channel

HQ	Handoff queening
SFR	Signal Frequency Reuse
SINR	Signal to Interference and Noise Ratio
NCB	New Call Bounding
FGC	Fractional Guard channel
RS	Relay Station
CWN	Cellular wireless Network
JCAC	Joint Call Admission Control
UPC	Uplink Power Control
RAT	Radio Access Technology
UE	User Equipment
EnodeB	Evolved NodeB
UMTS	Universal Mobile Telecommunication System
NAJCAC	Non Adaptive Joint CAC
AJCAC	Adaptive Joint CAC
NCBP	New Call Blocking Probability
HCDP	Handoff Call Dropping Probability
BBU	Basic Bandwidth Unit

List of Symbols

$t_{i,j}$	the total number of bbu available for handoff class-i calls in RAT-j.
$t_{0,j}$	the total number of bbu available for all new calls in RAT-j.
B_j	the total number of bbu available in RAT.
K	The heterogeneous network supports Classes of calls.
S	the state space of all admissible states of the group of co-located cells.
s	combination of the numbers of users In each class that can be supported simultaneously in the group of co-located cells.
$m_{i,j}$	the number of ongoing new class-i calls.
$n_{i,j}$	the number of handoff class-i calls
c	call of class-i.
b_i	to the new call blocking probability.
P(s)	the steady-state probability that the system is in.
P_{di}	The hand off call dropping probability (HCDP) for a class-I call.
P_{bi}	The new call blocking probability (HCDP) for a class-I call.
D	Delay.
N	Packet size.
R	Data rate.

BW	Bandwidth.
R_{vo}	Voice Data Rate.
P_{vo}	Packet Size Voice.
T	Simulation Time.

Chapter One

Introduction

1. Chapter one

1.1 Preface

Long-Term Evolution (LTE), commonly marketed as 4G LTE is a standard for wireless communication of high-speed data for mobile phones and data terminals. It is based on the Global System for Mobile Communication (GSM) Enhanced data GSM evolution (EDGE) and Universal of Mobile Telecommunication System (UMTS)/ High Speed Packet Access (HSPA) network technologies, increasing the capacity and speed using a different radio interface together with core network improvements.

LTE is considered by many to be the obvious successor to the current generation of UMTS 3G technologies, which is based upon Wideband Code Division Multiple Access (WCDMA), High Speed Downlink Packet Access (HSDPA), High Speed Uplink Packet Access (HSUPA), and HSPA. LTE is not a replacement for UMTS in the way that UMTS was a replacement for GSM, but rather an update to the UMTS technology that will enable it to provide significantly faster data rates for both uploading and downloading. Verizon Wireless was the first U.S. carrier to widely deploy LTE, though Metro PCS and AT&T have also done so, and Sprint and T-Mobile USA both have plans for LTE. In fact, Sprint is phasing out its WiMAX network in favor of LTE. Verizon Wireless and AT&T currently have incompatible LTE networks, even though they both make use of 700MHz spectrum. AT&T and Verizon Wireless LTE customers often see download speeds that exceed 15Mbps, and upload speeds in the 10Mbps range.

LTE is a field of interest throughout the world due to the demand of using data in mobile device in terms of streaming of media, for instance, internet TV, video conferencing, single or multiplayer online gaming as well as communicating through mobile video blogging. Is LTE is promising radio access network technology standardized in Third Generation Partnership Project (3GPP) in release 8. It is a system towards the 4G technology promising to be increased in data rates and more improve performance. Wireless networks are heading to their third phase. Where the first phase was concerned about voice traffic for voice calling, the second phase emphasize on data traffic.

The goal of LTE was to increase the capacity and speed of wireless data networks using new Digital Signal Processing (DSP) techniques and modulations that were developed around the turn of the millennium. A further goal was the redesign and simplification of the network architecture to an IP-based system with significantly reduced transfer latency compared to the 3G architecture. The LTE wireless interface is incompatible with 2G and 3G networks, so that it must be operated on a separate radio spectrum the Call Admission Control (CAC) optimize the use of allocated channels against offered maintaining the required Quality of Services (QoS) provisioning QoS to user at cell-edge is a challenge where there is limitation in cell resources due to Inter-Cell Interference (ICI).

1.2 Problem Statement:

Call admission control prevents oversubscription of Voice over Internet Protocol (VoIP) networks. 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 through test different parameters.

1.3 Proposed Solution

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

1.4 Objectives:

The aim of this research is to investigate different algorithms to choose optimal and acceptable algorithm of call admission control. Performance evaluation of call admission control is done through the sub objectives

- To compare new call blocking probability.
- To compare handoff call dropping probability.
- To compare throughput.
- To compare delay.

1.5 Methodology:

In this thesis two algorithms of admission control are evaluated, one of them is adaptive joint CAC and the other one is non-adaptive joint CAC, the performance metrics such as new call blocking probability, and handoff call dropping blocking probability, system throughput, system delay are used and simulated in MATLAB with the aid of mathematical equations to show the simulation results in order to evaluate those two algorithms.

1.6 Thesis Outline:

In this thesis, introduction and related work in chapter 2, call admission Control Call admission techniques in chapter 3. In chapter 4 the simulation results is described whereas, conclusion and recommendation are described in chapter5.

Chapter Two

Literature review

2. Chapter two

2.1 Background

The history of mobile telephony can be traced back to the1980's when the first Networks were (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 wire fourth-generation (4G).

The next evolution of the Radio Access Network (RAN) is LTE. This is also known as Evolved Universal Terrestrial Radio Access Network (eUTRAN). 3GPP LTE targets to support increase data rates and high efficiency, increased signal range with better user response time, interoperability with circuit-switched legacy networks compared to systems of today. LTE supports a wide range of bandwidth such as 1.4MHz, 3.0MHz, 5MHz, 10MHz, 15MHz and 20MHz bandwidths. LTE uses Orthogonal Frequency Multiple Access (OFDMA) for downlink and uplink Single Carrier Frequency Division Multiple Access (SC-FDMA). LTE is specified to provide downlink peak rates over 150Mbps, RAN round trip time less than 30ms and three times higher spectral efficiency than HSPA in 3GPP.

As shown in Table 2.1. LTE is demonstrates to be a high data rate and low latency system as the key performance. File Transfer Protocol (FTP), video streaming, VoIP, online gaming, real time video, push- to-talk, push-to-view is expected to support different types of services including web browsing in E-UTRA. For transmission and reception both UE are expected to be 20MHz. This gives an opportunity to the service provider to make changes to the amount of available spectrum. The spectrum for extra capacity starts with the limited amount of spectrum for lower upfront cost and growth. Most CAC algorithms work by regulating the total utilized bandwidth, the total number of calls, or the total number of packets or data bits passing a specific point per unit time. If a defined limit is reached or exceeded, a new call may be prohibited from entering the network until at least one current call terminates. Alternatively, a graceful degradation methodology can be implemented. This means that the audio quality of individual calls can deteriorate to a certain extent before new calls are denied entry. Another method involves the regulation of calls according to defined characteristics such as priority descriptors. Still another method prevents new calls from entering the network only if the resources of the central processing unit (CPU) of a particular computer or server would be overburdened by such calls. Call admission control can be tricky, because the volume of traffic in communications networks is inherently chaotic or "bursty," and traffic bursts are virtually impossible to predict. Another problem is that the actual content of a call may not conform to its descriptor.

Table 2-1: LTE performance requirement

Requirements	Measured
Downlink: 100 Mbps, Uplink: 50 Mbps For 20MHz spectrum	Packet data rates
Up to 500 km/h but performed for low speeds from 0 to 15 km/h	Mobility Supports
Less than 100ms both in idle and active.	Control plane latency (Transition time to active state)
Less than 5ms	User plane latency
More than 200 users per cell for 5MHz spectrum.	Control plane capability
5-100km with minor degradation following 30km.	Cell size (Coverage)
1.4, 3, 5, 10, 15 and 20MHz.	Range flexibility

2.2 Related work

2.2.1 Erlang B and Erlang C formula:

There are two types of trunked systems which are commonly used. The first type offers no queuing for call requests. That is, for every user who requests service, it is assumed there is no setup time and the user is given immediate access to a channel if one is available. If no channels are available, the requesting user is blocked without access and is free to try again later. This type of trunking is called blocked calls cleared and assumes that calls arrive as determined by a Poisson distribution.

Furthermore, it is assumed that there are an infinite number of users as well. as the following: (a) there are memoryless arrivals of requests, implying that all users, including blocked users, may request a channel at any time; (b) the probability of a user occupying a channel is exponentially distributed, so that longer calls are less likely to occur as described by an exponential distribution; and (c) there are a finite number of channels available in the trunking pool. This is known as an M/M/m/m queue, and leads to the derivation of the Erlang B formula (also known as the blocked calls cleared formula). The Erlang B formula determines the probability that a call is blocked and is a measure of the GOS for a trunked system which provides no queuing for blocked calls.

It is possible to model trunked systems with finite users, the resulting expressions are much more complicated than the Erlang B result, and the added complexity is not warranted for typical trunked systems which have users that outnumber available channels by orders of magnitude. Furthermore, the Erlang B formula provides a conservative estimate of the GOS, as the finite user results always predict a smaller likelihood of blocking.

The second kind of trunked system is one in which a queue is provided to hold calls which are blocked. If a channel is not available immediately, the call request may be delayed until a channel becomes available. This type of trunking is called Blocked Calls Delayed, and its measure of GOS is defined as the probability that a call is blocked after waiting a specific length of time in the queue. To find the Grade of Service (GOS), it is first necessary to find the likelihood that a call is initially denied access to the system. The likelihood of a call not having immediate access to a channel is determined by the Erlang C formula.

2.2.2 Quality of Service (QoS)

In the field of telephony, quality of service was defined by the International Telecommunication Union (ITU) in 1994. Quality of service comprises requirements on all the aspects of a connection, such as service response time, loss, signal-to-noise ratio, crosstalk, echo, interrupts, frequency response, loudness levels, and so on. A subset of telephony QoS is grade of service GoS requirements, which comprises aspects of a connection relating to capacity and coverage of a network, for example guaranteed maximum blocking probability and outage probability.

In wireless/mobile cellular networks, a mobile user's QoS requirements can be objectively expressed in terms of probabilistic connection-level QoS parameters related to connection establishment and management. In fact, the two types of call blocking had been distinguished: new-call (NC) blocking and handoff call (HC) dropping. The first type refers to the failure of the initial call-connection establishment, whereas the second type refers to the blocking of in-service calls when the mobiles move from one cell to another. From the user's point of view, it is more frustrating to lose a call that has already begun than to be prevented from establishing a new call. Therefore, QoS provisioning and mobility management are two key challenging issues that must be addressed in wireless multiservice mobile networks. In fact, wireless networks for multimedia services must incorporate an efficient connection-admission control strategy to provide QoS control and to reduce the blocking of new connections and the dropping of handoff connections as far as possible.

Hence, the goal of this article is to design a CAC scheme to reduce the handoff dropping in LTE networks. This probability is reduced by the processes:

Resource Block (RB) reservation and giving the priority to the handoff requests in admission. First, a novel was proposed handover procedure based on load balancing principle. In the second part, an admission control scheme is also adopted to achieve lower dropping probability and better resource utilization.

In this subsection, the modifications will be required of such a handover procedure in that a load balancing technique is proposed for initial neighboring cell selection and handover decision making. The LTE utilizes a network-controlled and the user equipment (UE) assisted handover procedure for mobility in connected mode: UE measures the power of signal strength Received Signal Strength (RSS) and sends the measurement report to the serving eNodeB. The serving eNodeB then makes the decision of handover based on the received measurement reports.

At present, dissimilar wireless access networks including G , 3G , Bluetooth, WLAN and Wi-Max coexist in the mobile computing environment, where each of these Radio access technologies offer complementary characteristics and features in terms of its coverage area, data rate, resource utilization , power consumption etc.. With all these there are constant improvements in the existing technologies offering better performance at lesser cost. This is beneficial in both the end users and service provider’s perspective. The idea of benefiting from integrating the different technologies has led to the concept of beyond International mobile telephony 2000(IMT-2000) wireless networks known as the next generation wireless networks (NGWN). In this heterogeneous environment, the end user is expected to be able to connect to any of the different available access networks. The end user will also be able to roam seamlessly within these access networks through vertical handover mechanisms. The global roaming is supplemented by the existence of IP networks as the backbone which makes the mobile computing environment to grow leaps and bounds and can effectively address the issue with regard to converge limitations is concerned. In this multifaceted wireless radio environment the radio resource management plays major role. The effective utilization of the limited available resources is the challenge. The admission control is one such challenge a network service provider face to achieve better system utilization face in handling this complex scenario to provide the best QoS to the users of the network.

The UTRAN LTE system is based on a decentralized architecture with different radio resource management functionalities, for example, admission control (AC), mobility control, etc., embedded in evolved Node-B (eNB). The AC for LTE uplink is located in the eNB at Layer 3. It will utilize the local cell load information to make the AC decision. UTRAN LTE is targeted to efficiently guarantee the QoS of services such as audio/video streaming, gaming and VoIP. The QoS classes supported in LTE are different from fixed networks. Two types of services for 3G mobiles can be defined: real-time services (e.g. video conferencing), and non-real-time services (e.g. database applications). In this context, the 3GPP defined four distinct traffic classes for LTE: (a) conversational and streaming for real-time classes and (b) interactive, and background for non-real-time classes. The main distinguishing factor between the four QoS classes is how sensitive the traffic to delay and delay variation. Conversational class is meant for traffic that is very delay sensitive while the background class is the most delay non-sensitive traffic class. As one of key techniques of LTE, OFDMA can offer two-dimensional time-frequency resources for more flexible allocation. In time domain, the duration of one frame is 10 ms. One frame is divided into 20 slots, and each slot contains 7 consecutive OFDMA symbols (including 1 control and 6 data symbols) and lasts 0.5 ms. In the frequency domain, the system bandwidth B is divided into N sub-channels each of which contains 12 consecutive subcarriers, and the spacing between the subcarriers, f, is 15 kHz. in addition, the OFDM symbol duration time is $1/f$ + *cyclic prefix*. The cyclic prefix is used to maintain orthogonality between the subcarriers, even for a time-dispersive radio channel. Data symbols are independently modulated and transmitted over a high number of closely spaced orthogonal subcarriers. In EUTRAN, downlink modulation coding schemes quadrature phase shift keying, 16QAM, and 64QAM are available (e.g. with 64QAM, each resource element carries six bits). The OFDM symbols are grouped into RBs. The basic resource unit is RB, which has a total size of 180 KHz in the frequency domain and 0.5 ms in the time domain. The resource block size is the same for all bandwidths; therefore, the number of available physical resource blocks depends on the bandwidth. All the available RBs in the LTE system are shared by all users that distribute uniformly.

There are good amount of work reported for homogenous wireless networks and single service wireless networks and few works in the heterogeneous wireless networks. The Call admission control in Heterogeneous wireless networks is a real challenge. The varied QoS requirements of multimedia applications and the coexistence of different RATs, facade major challenges in designing CAC algorithms for next generation heterogeneous wireless networks. The challenges are heterogeneous networking, multiple service classes, flexibility in bandwidth allocation and cross layer issues based design.

1-Heterogeneous networking:

4G networks will have different types of RATs different from each other by air interface technology, cell size, services, price, access method, coverage, so CAC schemes must be able to handle new type of handoff called vertical handoff.

2-Multiple service classes:

The 3G networks should be able to accommodate the applications and user with different QoS requirements, so the CAC algorithms should be designed to handle different classes of service to meet the QoS needs of all types of applications.

3- Flexible in bandwidth allocation:

The diversity is in multimedia applications and mobile users QoS requirements in NGWN. The resource utilization and QoS performance can be improved by adaptive bandwidth allocation. This clearly indicates that the CAC should be designed taking into consideration the flexible bandwidth allocation, where, more resources can be allocated when the there is less traffic and the allocated bandwidth can be revoked when there is congestion.

4-Cross layer issues based design*:*

The traditional CAC schemes were based on the call level QoS only and few of them have considered the physical layer QoS like SIR as QoS criteria. Unlike the tradition voice oriented circuit switched network, the Next generation network predicted to be pure packet based network and the QoS needs to be addressed both at call level as well as at packet level. This mandates that the new call has to be admitted only if both call level QoS metrics like call blocking and dropping probabilities and the packet level QoS measures like packet transmission delay and packet dropping probability are maintained at some desired levels. The other important solution for the decision making of call admission control is by Multi Criteria Decision Making (MCDM). This is an optimization technique used to analyze the contradicting decision making parameters. The MCDM based decision making systems are generally used in the fields of reliability, financial analysis, social and political related analysis and environmental impact analysis etc. The NGWN has different RATs coexisting which are with different capabilities and they should cater the varied QoS requirements of multimedia applications admission control with single criteria may be too trivial, in this prevailing scenario the admission control decision should be based on Multi criteria such that the optimization user satisfaction and selection of optimal RAT is achieved. There are several algorithms proposed on handling the admission control decision making using MCDM in heterogeneous wireless networks. There are Different admission control algorithms based on multiple criteria decision making. They are categorized as:

Utility- function based CAC and computation Intelligence CAC. In the Utility -function based CAC the incoming calls are admitted based on some utility or cost function based on multiple criteria. These algorithms are very optimal algorithms and in most of the case are complex in nature and pose high computational overhead. The computation-Intelligence-based CAC use evolutionary approaches like Genetic Algorithm (GA), fuzzy logic and Artificial Neural Networks (ANN).Majority of the computational-intelligence-based CAC algorithms incorporate fuzzy logic fuzzy neural and fuzzy MCDM methods. There are very few works reported on the usage of Artificial Neural Networks in CAC.

2.2.3 Classification of call admission control:

In our framework, a CAC algorithm was proposed for the LTE network, which provides a resource block allocation policy that takes into account the separation between incoming traffic for each class and prioritizes handoff calls over new calls. Moreover, based on the adaptive multimedia applications, we address an adaptive resource allocation mechanism that allocates connection resources for incoming calls to meet continuous QoS requirements under given network conditions. The objective of the adaptive resource allocation mechanism is to improve resource utilization while keeping blocking and dropping probability as low as possible. Quality of service profiles. Depending on the required data rate, each UE can be assigned one or more resource blocks in each transmission time interval of 1 ms. before going into details of the RB reservation algorithm, it should be noted that the UE communicates with the network. Connection-admission control determines the condition for accepting or rejecting a new call based on the availability of sufficient network resources to guarantee the QoS parameters without affecting the existing calls. There are several call admission control methods for new and handoff requests these can be grouped into the prioritized schemes and the non-prioritized schemes.

2.2.3.1 Non prioritized schemes:

The non-prioritized scheme is employed by typical radio technologies that have been proposed for personal communications services. In the non-prioritized scheme, the eNodeB handles handoff calls in exactly the same way as new calls (i.e., handoff call is blocked immediately if no resources are available). They proposed a CAC algorithm for LTE utilizing the Fractional Power Control formula agreed on in 3GPP. The admission criterion for the new user is that the sum of the required number of physical resource blocks (PRBs) per time transmission interval 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, all available resources in the eNodeB are shared by handoff and new calls. However, the drawback of this scheme is that it is difficult to guarantee the required dropping probability of handoff calls. Hence, the authors do not take the prioritization between the call while basing the type of call and their QoS requirements. Moreover, the thresholds management is static; it does not depend on the type of call. a Greedy Choice with bandwidth Availability aware Defragmentation

(GCAD-CAC). algorithm was proposed. It is able to guarantee respect for data flow delay constraints defined by three different traffic classes. To achieve good results, the algorithm tries to accept all the new requests, but when a higher priority request is received, a lower priority admitted request is preempted.

This preemption can leave some small gaps that are not sufficient for new connection admission; these gaps can be collected by the GCAD algorithm by activating bandwidth availability -based defragmentation process. The proposed CAC scheme for LTE systems with heterogeneous services introduces a transmission guard interval that gives high priority to the real time (RT) service packets approaching the delay deadline. However, no distinction is made between originating and handoff requests.

Furthermore, to reduce the call dropping probability, few other CAC algorithms that take into consideration neighboring cells information have been proposed. a method called the LA algorithm was proposed by the author because the call is made after ‘looking around’ at the neighborhood cells. A weighted sum of the number of ongoing calls in the region can be used to define the cell load or admission control that can be performed separately for each cell. In the latter case, an arriving call must pass the admittance test of all cells in the region to be accepted. Therefore, the call is accepted if the maximum effective load for the adjacent cells is below the threshold. The drawback of the conventional methods is that they cause an increase in the number of dropped calls in the adjacent cells. Furthermore, those algorithms only support users with fixed bandwidth requirements.

2.2.3.2 Prioritized schemes

Out of QoS and financial (increase profit) reasons, it is preferable for a cellular operator to give higher priority to some call types. These include already ongoing calls and calls carrying certain services. The CAC method should give prioritized admission to handoff requests because from the user's point of view, disruptions during handoffs are considered more objectionable than new call blocking. There exist several different procedures to achieve the prioritization, such as guard channel (GC) and handover queuing schemes. The basic idea of GC-based admission control strategies is to reserve resources in each cell prior to deal with handoff requests. To provide user's equipment with continuous connectivity, system reserves backup channels referred to as 'guard channels' to offer preferential treatment to priority calls and handoff calls. In such a system, call requests with lower priority are rejected if the number of available resource is less than a certain threshold. GC strategies differ in the number of guard channels to be chosen by a base station. They are called fixed guard channel and dynamic guard channel. The fixed guard channel schemes reserve a fixed number of channels for handoff calls. In this article, only one traffic class was considered. The advantage of this scheme lies in the simplicity of deployment, because there is no need to exchange control information between the base stations. However, with a small portion of handoff call, GC schemes results not only in increased blocking, but also in inefficient resource utilization, because only a few handoff calls are able to use the reserved channels exclusively. On the other hand, with a large number of handoff calls, it is difficult to guarantee the service requirements of handoff call. All these schemes proposed above are static because such GC schemes cannot adapt to quick variation of the traffic pattern. Dynamic GC schemes, improve the system efficiency while providing the QoS guarantees to priority calls. These schemes adaptively reserve the actual resources needed for priority calls and, therefore, accept more lower-priority calls compared with a fixed scheme. a distributed adaptive guard channel reservation scheme is proposed to give priority to handoff calls. This scheme is built upon the concept of guard channels and it uses an adaptive algorithm to search automatically the optimal number of guard channels to be reserved at each base station. Prioritization scheme based on the introduction of a handoff queuing (HQ) method. They have considered buffering handoff calls as a way of reducing their force termination probability. The buffer size would be adjusted depending on the input traffic rate. HQ based methods follow the principle: when resources become available, one of the calls in the handoff queue is served. If there are no available resources, call requests are queued until resources are available again. Handoff calls will only be blocked when the buffer is full, while new calls are blocked when no channel is idle. The HQ scheme needs a lot of buffers to deal with real-time multimedia traffic. Moreover, a sophisticated scheduling mechanism is needed to meet the QoS requirement for delay sensitive calls to guarantee that the queued data will not expire before they are transmitted. Therefore, to support real-time applications in wireless IP networks, a measurement-based admission control with priority criteria and service classes was proposed recently, several CAC algorithms and bandwidth adaptation algorithms have been proposed for wireless networks. An adaptive call admission control algorithm was proposed. This proposed scheme encompasses the bandwidth allocation/reallocation policy and the bandwidth adaptation algorithm. These algorithms are needed to reduce the requested or already connected call bandwidth allocation. We argue that with this CAC, we can reduce the handoff dropping probability. However, the computation and deployment of bandwidth reallocation consumes an amount of time capable of increasing the handoff latency.

2.2.3.3 Local and collaborative call admission control

Call admission control schemes can be divided into two Categories, local and collaborative schemes. Local schemes use local information alone (e.g. local cell load) when taking the admission decision. Examples of these schemes are Collaborative schemes involve more than one cell in the admission process. The cells exchange information about the ongoing sessions and about their capabilities to support these sessions. The fundamental idea behind all collaborative admission control schemes is to consider not only local information but also information from other cells in the network. The local cell, where the new call has been requested, communicates with a set of cells that will participate in the admission process. This set of cells is usually referred to as a cluster. In general, the schemes differ from each other according to how the cluster is constructed, the type of information exchanged and how this information is used In for example, the cluster is defined as the set of direct neighbors. The main idea is to make the decision of admission control in a decentralized manner.

2.2.4 Call admission schemes based on SFR:

Modern ICI mitigation schemes divide the cell into cell-edge and cell-core. In these schemes the users are divided according to their locations from the e-NodeB and the resources they can access to cell-edge and cell-core users. One of the most ICI mitigation schemes, which are used in LTE-Advanced network, is Soft Frequency Reuse (SFR). In SFR scheme, for each cell in the network; the cell is divided into two parts: cell-edge and cell core. In addition, the available Resource Blocks (RBs) (basic resource element in LTE networks) are divided into cell-edge RBs and cell-core RBs. All of users within each cell are also divided into two groups which based on the SINR: cell-edge users and cell-core users. It is called SFR as the frequency partition only applies to the cell edge users, the cell edge users are restricted to use this frequency sub-band only and all frequency bands are available to the cell-core users. So the effective frequency reuse factor is still close to one. Cell-edge performance is such a subject that faces challenge in modeling and analysis. What makes the modeling difficult is that, to achieve some sense of accuracy, one need to consider ICI impact as well as handover traffic effect from adjacent cells. Handoff priority-based CAC schemes have great impact on cell-edge users. First of all it occurs on cell-edges and secondly it depends on cell edge radio resources. Therefore we cannot ignore its effect in cell-edge performance. The handover process depends on the policy that control handover access to the cell which in turn depends on the associated CAC scheme adopted. Various handoff priority-based CAC schemes have been proposed. One of these schemes depends on reserving a portion of channel for handoff calls; whenever a channel is released, it is returned to the common poll of channels. This scheme is called the cutoff priority scheme. On the other hand, the fractional guard channels schemes which depend on admitting a new call with certain probability. it is shown to be more general than the cutoff priority scheme. A CAC scheme named New Call Bounding (NCB) scheme smoothly throttles the admission rates of calls according to their priorities as well as it aims to provide multiple prioritized traffic with a desired QoS. In the rigid division-based CAC scheme, all channels allocated to a cell are divided into two groups: one to be used by all calls and the other for handover calls only Four schemes are addressed in this work Cutoff priority scheme, Uniform Fractional Guard Channel, Limited Fractional Guard Channel and New Call Bounding scheme The cutoff priority technique keeps a certain amount of channels to handover calls only while the rest of the channels can be shared by both new calls and handover calls. Hence, handover calls are given higher priority over new calls, and as a result the reduction in the handover probability comes at the expense of higher blocking rate.

In Fractional Guard Channel (FGC) the new call is admitted by a certain probability which is a decreasing (or, more accurately, non-increasing) function of the number of occupied channels while a handover call is admitted as long as there is a free channel. Uniform FGC is special case of FGC where the acceptance probability has a constant probability that is independent of number of occupied channel. While Limited FGC is another type of FGC in which the acceptance probability varies between three values (1, βl, 0) according to channels occupation.

In New Call Bounding a threshold is used to limit the number of new calls in the cells. Handover calls are only blocked if all channels are occupied. The scheme works as follows: if the number of new calls in a cell exceeds a threshold when a new call arrives, the new call will be blocked; otherwise it will be admitted. The handover call is rejected only when all channels in the cell are used up. The idea behind this scheme is that we would rather accept fewer customers than drop the ongoing calls in the future, because customers are more sensitive to call dropping than to call blocking.

The focus has been on the single cell, where cell-center mobile platforms occupying channels from the center band or the edge band are distinguished in our analytical model. On the other hand, the channel condition of different ongoing cell-center mobile platforms may be different, and thus, in a real implementation, the termination of an ongoing cell center mobile platform should take account of QoS criteria.

2.2.5 Power Control

With the purpose of enhancing system capacity by setting the transmission power levels, Power control also substantially inputs the exhaustion rate of power as well as coverage and quality. The power that is received by the control unit is essentially increased while ensuring minimum interference to the process.

2.2.3.4 Uplink Power Control:

One of the mechanisms that LTE uses is Uplink Power Control (UPC). Received signals stability of the expect cell is controlled by the mechanism as well As ensuring control interference in connect cells. One of the principle characteristics of the mechanism is that fractional path-lose compensation which can be supported by eventually leads to less interference and power transmission to neighbor cells.

2.2.3.5 Downlink Power Control:

Transmission bandwidth consists of transmission power located in the Down link inter cell. The downlink coordination facilitates the relative narrow band transmission power indicator where a cell can transmit information to the neighboring cells. Dictated by these neighboring cells, which upon receiving the indication can schedule its downlink transmission, it contributes to the overall reduction of the output of the spectrum. A reuse is possible on its fullest frequency in neighboring cells within the core part of the inter-cell interference coordination scheme in LTE.

Chapter Three

Performance evaluation of Call Admission Control in LTE

3. Chapter three

Call Admission control (CAC) becomes important for the estimation of the traffic or the traffic prioritization of an incoming voice call. The admission control process does some math and activities another bearer if resources are available based on traffic activity .when a user generates data, a bearer is required to transmit data. Hence a UE send a bearer establishment request to enodeB, enodeB then executes an admission control algorithm to decide to admit the bearer or not.

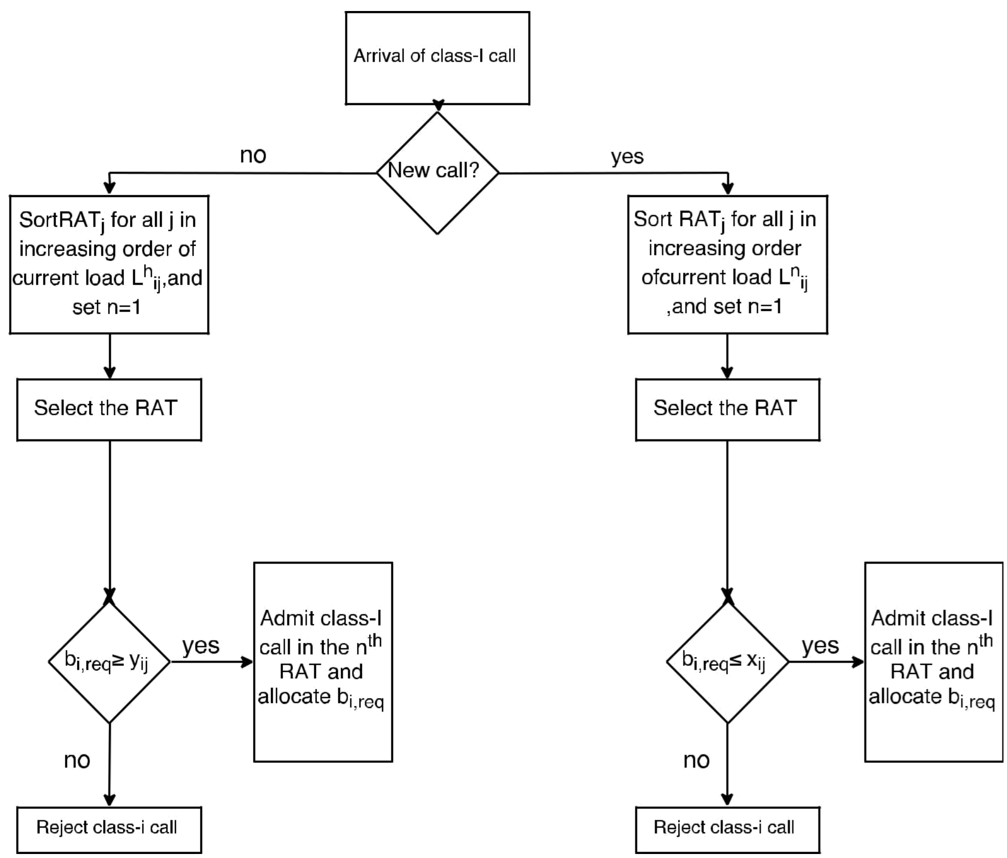
3.1 Call Admission Control Algorithms

In this project two algorithms were discussed these are Non adaptive joint call admission control (NAJCAC) and adaptive joint call admission control (AJCAC).

3.1.1 Non adaptive JCAC algorithm

The normal call admission control algorithms does not provide solution to fit heterogeneous wireless network ,therefore there is a need to develop RAT selection algorithm in addition to call admission control ,this term called joint call admission control. The algorithm show in figure 3.1 decide whether an incoming call can be accepted or not and also decide which of the available radio access networks is best suited to accommodate the incoming call, manage individual services and technologies and ensure that the (QOS) requirements of all admitted call are satisfied while at the same time making the best use of the total resources available in the heterogeneous network. NAJCAC algorithms admit calls into a particular RAT based on the class of service, such as voice, video streaming, real-time video, web browsing, etc. .This approach is based on the fact that different RATs are optimized to support different classes of service. For example, GSM is designed for voice services whereas EV-DO is optimized for data services. Therefore, the algorithm admits an incoming call into a RAT that can best support the service class of the call. Service- NAJCAC algorithms have the advantage of high packet-level QoS. However, they may lead to highly unbalanced network load.

NAJCAC algorithm tries to admit an incoming call of a specific class into a particular RAT, if the preferred RAT for this call cannot accommodate the call, probably because there is no enough radio resource, other RATs are not acceptable. Therefore the call is blocked



. Figure 3-1: Non adaptive CAC flow chart

3.1.2 Adaptive (JCAC) Algorithm

AJCAC which is consists of three components joint call admission controller, threshold based bandwidth reservation unit and bandwidth adaptation (BA) controller. Show in figure 3.2and in this scheme if the available bbu in the selected RAT is less than $b_{i,max}$ but greater than or equal $b_{i,request}$ the call will be assigned a bandwidth between those values, if the available bbu is less than $b_{i,request}$ but greater than or equal $b_{i,min}$the call will be assigned bandwidth between $b_{i,min}$ and $b_{i,request}$t and if the available bbu in all RATs is less than $b_{i,min}$ BA algorithm (BAA) will be invoked to reduce the bandwidth of some ongoing call (s) randomly in the chosen RAT ,and if the available bbu is still less than $b_{i,min}$ for all available RATs the call will be rejected.

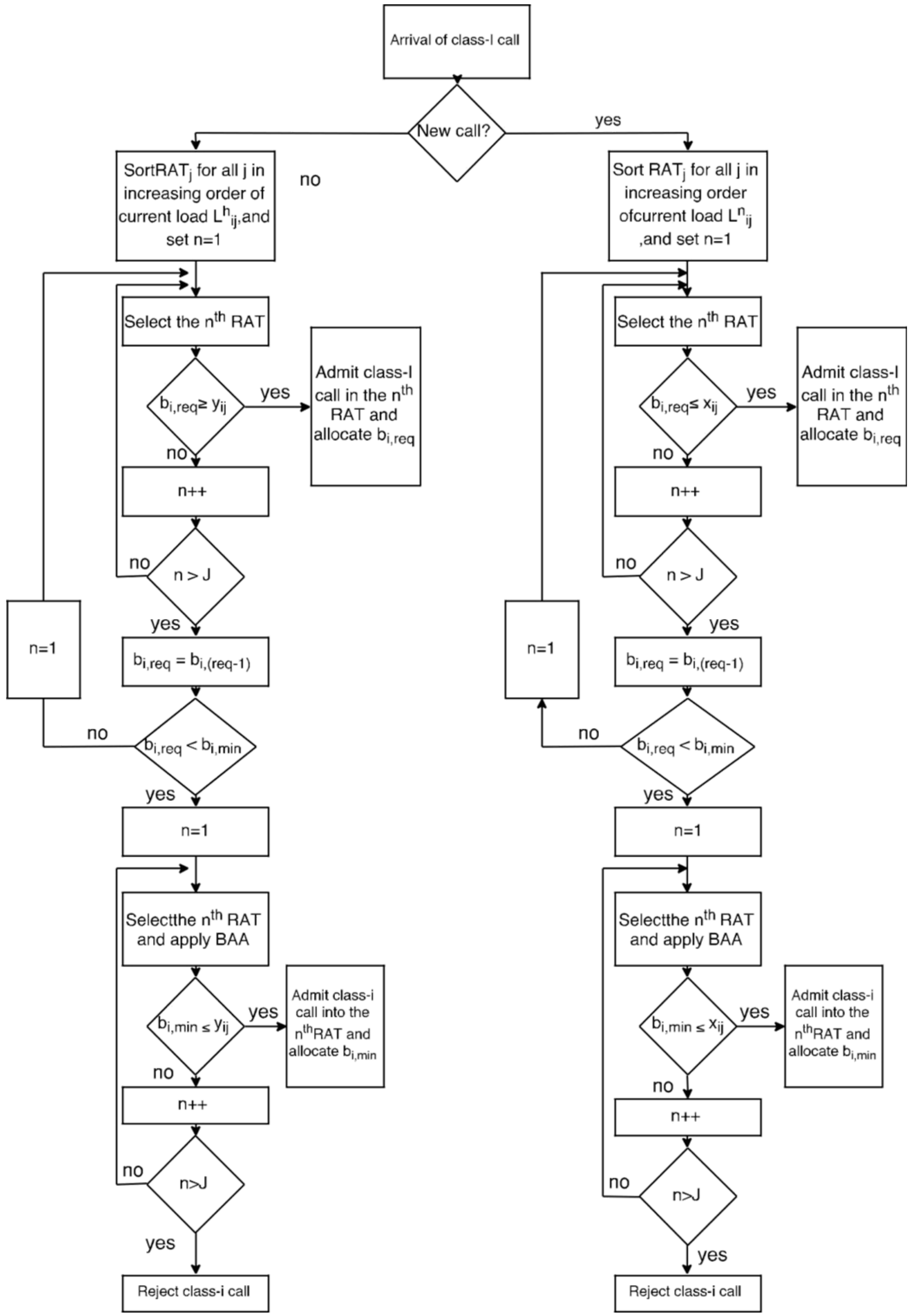


Figure 3-2: Adaptive CAC flow chart

3.2 Performance Evaluation Metrics for CAC Algorithms

3.2.1 New Call Blocking Probability

A new class-*i* call is blocked in the group of collocated cells if none of the available RATs has enough bbu to accommodate the new call with the minimum bandwidth requirement after degrading the ongoing new calls. Let $Sb_i \subset S$ denote the set of states in which a new class-*i* call is blocked in the group of collocated cells. It follows that

$$S_{b_i} = \{s \in S \,;\, (b_{i,min} + \sum_{x+1}^{k} m_{x,j} b_{x,min} > t_{0,j}^{n} \vee b_{i,min} + \sum_{x+1}^{k} m_{x,j} b_{x,min} + \sum_{x=1}^{k} \sum_{c=1}^{n_{x,j}} b_{x,assigned} > b_j) \forall_j\} \quad (3.1)$$

Thus the new call blocking probability (NCBP), p_{b_i}

$$p_{b_i} = \sum_{s \in S_{b_i}} P(s) \quad (3.2)$$

3.2.2 Handoff call dropping probability

A handoff class-*i* call is dropped in the group of collocated cells if none of the available RATs has enough bbu to accommodate the handoff call with the minimum bandwidth requirement after degrading the ongoing new calls and handoff calls. Let $Sd_i \subset S$ denotes the set of states in which a hand off class-*i*call is dropped in the group of collocated cells. It follows that

$$S_{d_i} = \{s \in S : ((1 + n_{i,j})\, b_{i,min} > t_{i,j}^{h} \vee b_{i,min} + \sum_{x+1}^{k} (m_{x,j} + n_{x,j}) b_{x,min} > B_j) \forall j\} \quad (3.3)$$

Thus the handoff call dropping probability (HCDP) for a class-*i* call, p_{d_i}in the group of co-located cells is given by

$$p_{d_i} = \sum_{s \in S_{d_i}} P(S) \tag{3.4}$$

3.2.3 Delay

The Voice is real time system, thus the Delay is effect for , it increase when the data rate is decrease as point in equation.

$$D = \frac{\mathrm{N}}{\mathrm{R}} \tag{3.5}$$

3.2.4 Throughout

throughput of system is defined as the summation of packets transmitted in a simulation time from eNB to all UE.

$$\text{system throughput} = \frac{1}{T}\sum_{i=1}^{k}\sum_{i=1}^{T} Psize \tag{3.6}$$

Chapter Four

Simulation and Discussion.

4. Chapter Four

4.1 Simulation Parameter

In this chapter, a program is written using MATLAB to calculate the performance metrics of the CAC such as new call blocking probability, handoff call dropping probability, average system utilization, system throughput and system delay. The simulation is performed by using the simulation parameters given in Table 4.1.

Table 4-1: Simulation parameters

PARAMETERS	SYMBOL	VALUE
Bandwidth	BW	5 MHZ
Voice data rate	Rvo	12.5Kbps
Packet size voice	Pvo	64 byte
Time simulation	T	100

4.2 Simulation Results

In this section, four different scenarios were simulated.

4.2.1 Scenario one: New call blocking probability VS Call Arrival Rate

In this scenario, two type of voice call are considered these are new call and handoff call, simulation is performed in total time of 100s and each second four calls were arrived for both call admission control mechanism; call admission with Adaptive joint and Non adaptive joint.

NCBP is increase with call arrival rate Figure 4.1 shows the NCBP calculated for Adaptive scheme is less than that for Non adaptive, because when the total bbu allocated to new call is being fully utilized, incoming new calls are rejected by Non adaptive joint CAC whereas Adaptive joint CAC adapts (degrades) the bandwidth of some of the ongoing adaptive calls to free just enough bbu to accommodate the new calls

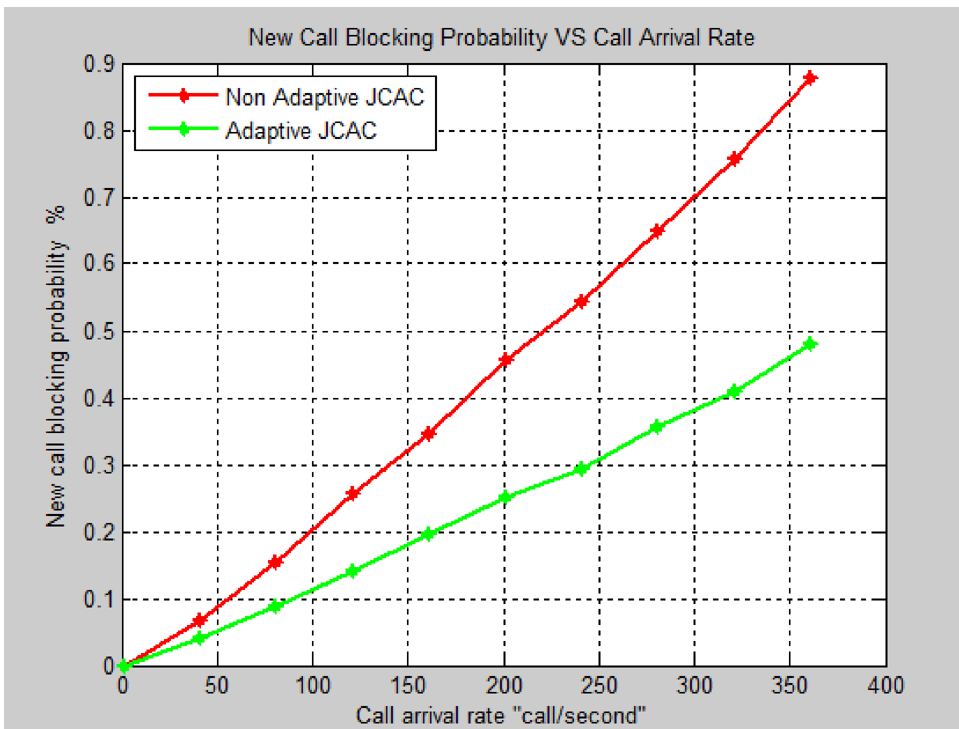


Figure 4-1: NCBP vs. call arrival rate

4.2.2 Scenario Two: Handoff call dropping probability vs Call Arrival Rate

In this scenario, two type of voice call are considered these are new call and handoff call, simulation is performed in total time of 100s and each second four calls were arrived for both call admission control mechanism; call admission with Adaptive joint and Non adaptive joint.

HCDP is increased with call arrival rate for both strategies Figure 4.2 shows the HCDP calculated for Adaptive joint CAC is less than that for Non adaptive joint CAC ,the reason of why HCDP for Adaptive joint CAC is less than Non adaptive joint CAC is as follows ,when the system is being fully utilized ,incoming handoff calls are rejected by Non adaptive joint CAC whereas Adaptive joint CAC adapts (degrades) the bandwidth of some of the ongoing adaptive calls to free just enough bbu to accommodate the incoming handoff calls .consequently the HCDP of Adaptive joint CAC is less than that for Non adaptive joint CAC.

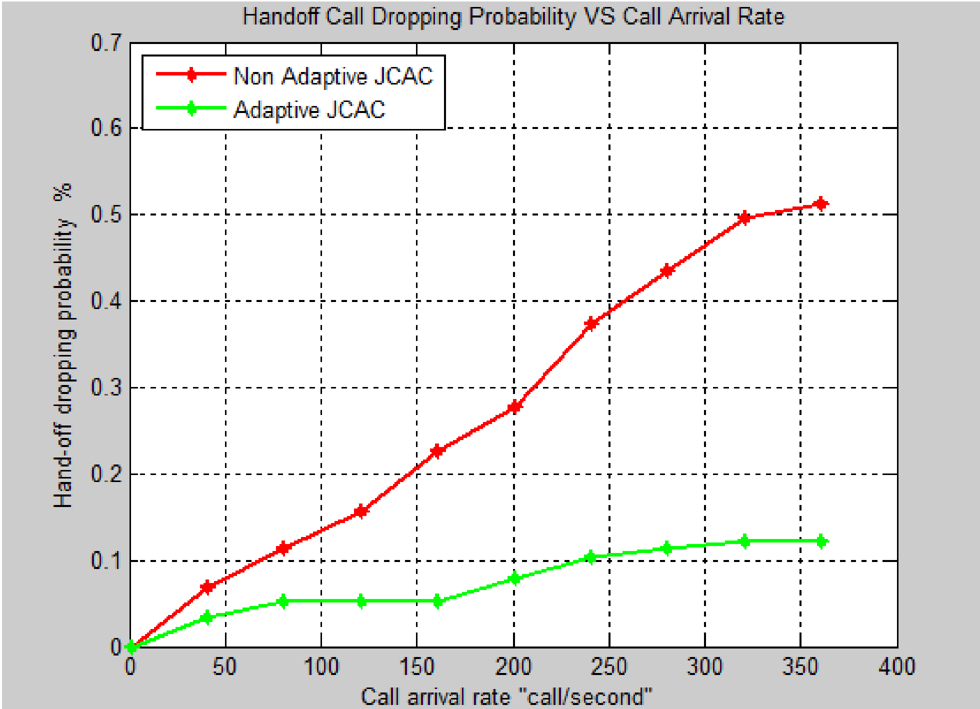


Figure 4-2: HCDP vs. call arrival rate

4.2.3 Scenario three throughput vs Call Arrival Rate

In this scenario, two type of voice call are considered these are new call and handoff call, simulation is performed in total time of 100s and each second four calls were arrived for both call admission control mechanism; call admission with Adaptive joint and Non adaptive joint.

Figure 4.3 shows the throughput calculated for voice, in Adaptive joint CAC, it is increased exponentially with the increase of voice users. But, in Non adaptive joint CAC the throughput is developed with a rate that is less than that of the Adaptive joint CAC because less VOIP users are admitted. From the Figure 4.1, it can be clearly seen that the throughput of Adaptive joint CAC is higher than for Non adaptive joint CAC due to more VOIP users admitted.

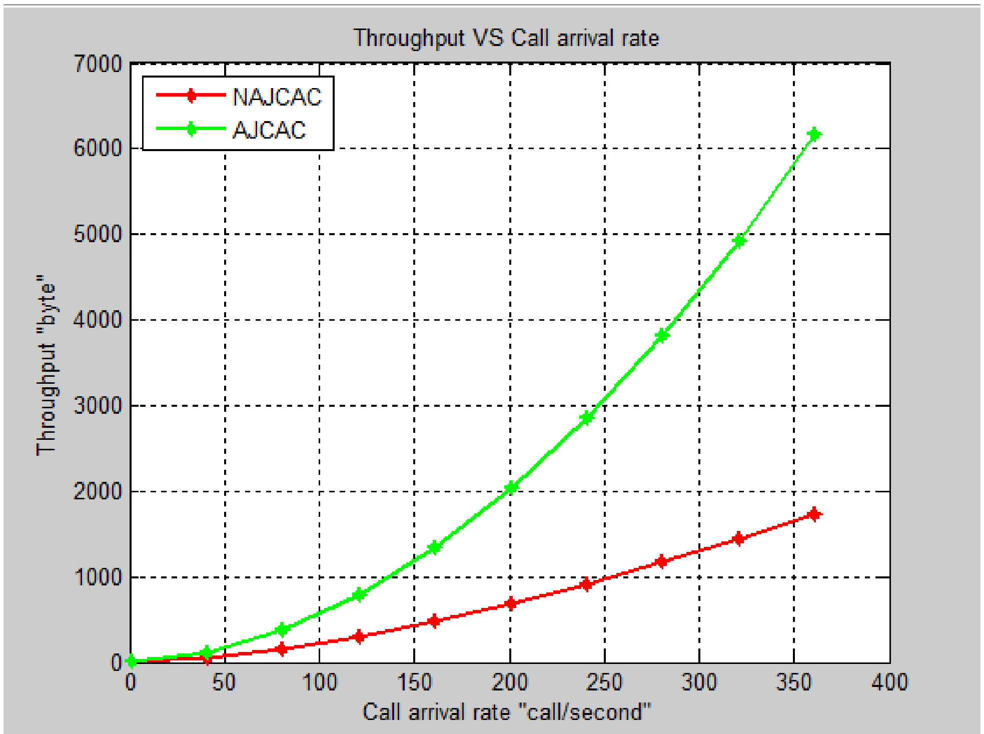


Figure 4-3: Throughput vs. call arrival rate

4.2.4 Scenario four: delay vs. call arrival rate

In this scenario, two type of voice call are considered these are new call and handoff call, simulation is performed in total time of 100s and each second four calls were arrived for both call admission control mechanism; call admission with Adaptive joint and Non adaptive joint**.**

Figure 4.4 the delay is calculated considering voice call, the delay increasing with the number of calls in both call admission control algorithm, but as shown in figure4.4 the delay in Adaptive joint CAC is higher than Non adaptive joint CAC.

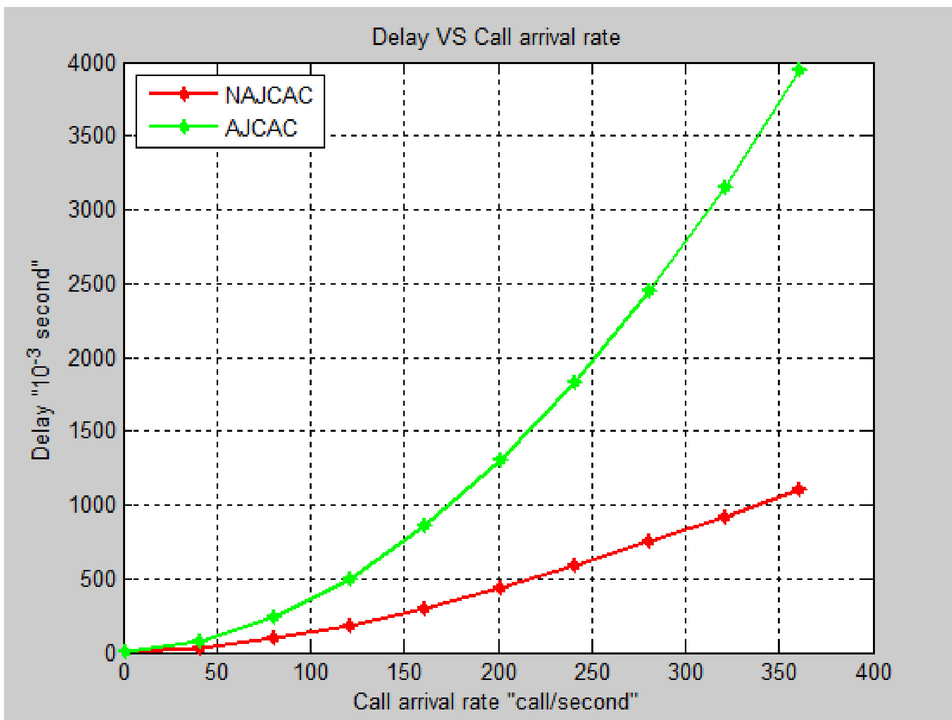


Figure 4-4: Delay vs. call arrival rate

Chapter 5

Conclusion and Recommendation

5. Chapter Five

5. 1Conclusion

This thesis was focused on evaluating two algorithms ;the first is NAJCAC which control and decide to accept or reject call based on available bbu in the available RAT without any additional work if there no enough bandwidth in available RATs and reject new incoming calls and the second algorithm is AJCAC which acts as improvement of the first one thus it does all its work in addition to features that if no enough bandwidth in selected RAT and new calls are coming it try to make room to admit new call by reducing the bandwidth of some ongoing calls chosen randomly, after that checks the condition of the bandwidth if still not enough finally it reject the new calls. The adaptive JCAC scheme improve average system utilization by adapting the bandwidth of calls based on current traffic condition and by uniformly distribute traffic load among the available RATs.

Adaptive JCAC guarantees the Qos requirements of all accepted calls and reduce (NCBP) and (HCDP) also it increase system throughput and delay in the heterogeneous wireless networks.

5.2 Recommendations

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

- ✓ Further study of algorithms could lead to the extension features and to create algorithm combined the specifications of those two algorithms with additional functions.
- ✓ You can also look at other parameters such as cost.
- ✓ Other services such as video call can be considered.

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Appendix A

```
clc,clear,close all

T=100; % simulation time
no=zeros(1,T); % log for the number of accepted calls from the start to certain second (for NAJCAC)
no2=zeros(1,T); % log for the number of accepted calls from the start to certain second (for AJCAC)
band_av=zeros(1,6*T); % log for the avilable banwidth (for NAJCAC)
band_av(1,1:6)=[5 7 5 7 5 7]; % intial value of bandwidth (for NAJCAC)
h_flag=ones(1,6*T); % flag used to know if the bandwidth reserved for the hand-off calls are used (for NAJCAC)
band_av2=zeros(1,6*T); % log for the avilable banwidth (for AJCAC)
band_av2(1,1:6)=[5 7 5 7 5 7]; % intial value of bandwidth (for AJCAC)
res_band2=zeros(1,6*T); % reserved from hand-off calls (for AJCAC)
resn_band2=zeros(1,6*T); % reserved from new calls (for AJCAC)
h_flag2=ones(1,6*T); % flag used to know if the bandwidth reserved for the hand-off calls are used (for AJCAC)

drop=0; % counter for the droped calls (for AJCAC)
block=0; % counter for the blocked calls (for AJCAC)
drop_p=zeros(1,T); % log for the droped calls from the start to certain second (for AJCAC)
```

```
     block_p=zeros(1,T); % log for the blocked calls from
the start to certain second (for AJCAC)

     drop2=0; % counter for the droped calls (for NAJCAC)
     block2=0; % counter for the blocked calls (for NAJCAC)
     drop2_p=zeros(1,T); % log for the droped calls from
the start to certain second (for NAJCAC)
     block2_p=zeros(1,T); % log for the blocked calls from
the start to certain second (for NAJCAC)
     numu=0; % number of new calls (total)
     numuh=0; % number of hand-off calls (total)
     % display the avilable bandwidth
     disp(['avilable bandwidth = ' num2str(band_av(1:6))])
     disp(' ')
     % simulation time
     for ii=1:T
         % value of bandwidth for the new second
     if ii>=2
        band_av(1,(ii-1)*6+1:ii*6)=band_av(1,(ii-
2)*6+1:(ii-
               1)*6)+band_av(1,(ii-1)*6+1:ii*6);
        band_av2(1,(ii-1)*6+1:ii*6)=band_av2(1,(ii-
2)*6+1:(ii-
                  1)*6)+band_av2(1,(ii-1)*6+1:ii*6);
        disp(' ')
        disp(['avilable    bandwidth    (NAJCAC)    =    '
num2str(band_av(1,(ii-2)*6+1:(ii-1)*6))])
        disp(['avilable    bandwidth    (    AJCAC)    =    '
num2str(band_av2(1,(ii-2)*6+1:(ii-1)*6))])
        disp(' ')
      end
     numc=4; % number of calls (new + hand-off)
```

```
    disp(['calls  in  sec('  num2str(ii)  ')  =  '
num2str(numc)])
        % to display the number of call in the certain
second
    for iii=1:numc
        req(iii)=2+round(2*rand(1)); % the values (random)
of requested bandwidth
        dur(iii)=1+round(14*rand(1))+round(0.55*rand(1));
% the values (random) of call duration
        disp(['[' num2str(iii) ']  bandwidth req = '
num2str(req(iii)) ' ,total time = ' num2str(dur(iii))])
        h(iii)=round(0.67*rand(1)); % the call is new(75%)
or hand-off(25%)
        req2(iii)=req(iii);
         if h(iii)==1
            disp('     --hand-off call')
            numuh=numuh+1;
            else
            disp('     --new call')
            numu=numu+1;
           end
         end
       for iii=1:numc
        if h(iii)==1
         for iiii=(ii-1)*6+1:ii*6
              % the value of the minimmum bandwidth
           if iiii<=(ii-1)*6+3
             m=2;
            elseif req(iii)<=2
             m=2;
              else
               m=2;
```

```
               end
                 % can the requested bandwidth be satisfied
?
        if req(iii)<=band_av(iiii)
           band_av(1,iiii)=band_av(1,iiii)-req(iii);
           band_av(1,iiii+6*dur(iii))=req(iii);
           if req(iii)~=0
             no(1,ii:T)=no(ii)+1;
             h_flag(iiii:6:iiii+6*dur(iii)-1)=0;
             h_flag(iiii+6*dur(iii))=1;
           end
               req(iii)=0;
             elseif rem(iiii,6)==0
                  drop=drop+1;
                  drop_p(ii:end)=drop;
             disp(['**no bandwidth avilible (NAJCAC)**
                 call number (' num2str(iii) ') has been
                        rejected'])
             end
                 % can the requested bandwidth be satisfied
?
     if req2(iii)<=band_av2(iiii)
        band_av2(1,iiii)=band_av2(1,iiii)-req2(iii);
        band_av2(1,iiii+6*dur(iii))=req2(iii);

        if req2(iii)~=0
           no2(ii:end)=no2(ii)+1;
           res_band2(iiii:6:iiii+6*dur(iii)-1)=req2(iii)-
m+resn_band2(iiii);
           h_flag2(iiii:6:iiii+6*dur(iii)-1)=0;
           h_flag2(iiii+6*dur(iii))=1;
          end
```

```
              req2(iii)=0;
               elseif rem(iiii,6)==0
               % can the requested bandwidth be satisfied
after
               % the banwidth adaption for the first RAT?
          for iiiii=(ii-1)*6+1:ii*6-3

band_av2(iiiii)=band_av2(iiiii)+res_band2(iiiii);
             res_band2(iiiii:6:60)=0;
            if req2(iii)<=band_av2(iiiii)
               band_av2(1,iiiii)=band_av2(1,iiiii)-req2(iii);
               band_av2(1,iiiii+6*dur(iii))=req2(iii);
                if req2(iii)~=0

res_band2(iiiii:6:iiiii+6*dur(iii)1)=req2(iii)-
                        m+resn_band2(iiiii);
                   h_flag2(iiii:6:iiii+6*dur(iii)-1)=0;
                   h_flag2(iiii+6*dur(iii))=1;
                end
                req2(iii)=0;
                   elseif rem(iiiii,3)==0
                        % can the requested bandwidth be
satisfied
                        % after the banwidth adaption for
the second RAT?
     for                                      iiiiii=ii*6-2:ii*6
band_av2(iiiiii)=band_av2(iiiiii)+res_band2(iiiiii);
     res_band2(iiiiii:6:60)=0;
           if req2(iii)<=band_av2(iiiiii)
              band_av2(1,iiiiii)=band_av2(1,iiiiii)-
req2(iii);
              band_av2(1,iiiiii+6*dur(iii))=req2(iii);
```

```
        if req2(iii)~=0
           no2(ii:end)=no2(ii)+1;
           res_band2(iiiiii:6:iiiiii+6*dur(iii))=req2(iii)-
           m+resn_band2(iiiiii);
           h_flag2(iiii:6:iiii+6*dur(iii)-1)=0;
           h_flag2(iiii+6*dur(iii))=1;
             end
              req2(iii)=0;
             elseif rem(iiiii,6)==0
                    drop2=drop2+1;
                    drop2_p(ii:end)=drop2;
                    disp(['**no  bandwidth  avilible  (AJCAC)**
   call    number (' num2str(iii) ') has been rejected'])
                        end
                      end
                    end
                   end
                  end
                 end
                end
              end
            for iii=1:numc
                if h(iii)==0
                    for iiii=(ii-1)*6+1:ii*6
                        if iiii<=(ii-1)*6+3
                           m=2;
                        elseif req(iii)==2
                               m=2;
                         else
                               m=2;
                        end
            if req(iii)<=band_av(iiii)-2*h_flag(iiii)
```

```
                        band_av(iiii)=band_av(iiii)-
req(iii);
            band_av(iiii+6*dur(iii))=req(iii);
            req(iii)=0;
             elseif rem(iiii,6)==0
                    block=block+1;
                    block_p(ii:end)=block;
     disp(['**no bandwidth avilible for new call (NAJCAC)**
   call number (' num2str(iii) ') has been    rejected'])
              end
     if req2(iii)<=band_av2(iiii)-2*h_flag2(iiii)
                         band_av2(iiii)=band_av2(iiii)-
req2(iii);
        band_av2(iiii+6*dur(iii))=req2(iii);
          if req2(iii)~=0
             no2(ii:end)=no2(ii)+1;
             resn_band2(iiii:6:iiii+6*dur(iii))=req2(iii)-
2;
                end
                  req2(iii)=0;
                     elseif rem(iiii,6)==0
                         for iiiii=(ii-1)*6+1:ii*6-3
                         band_av2(iiiii)=band_av2(iiiii)+resn_
                       band2(iiiii); % bandwidth adaptation
                             % but only using the bandwidth
reseved for new
                              calls
                              resn_band2(iiiii:6:60)=0;
     if          req2(iii)<=band_av2(iiiii)-2*h_flag2(iiiii)
band_av2(1,iiiii)=band_av2(1,iiiii)-req2(iii);

band_av2(1,iiiii+6*dur(iii))=req2(iii);
```

```
        if req2(iii)~=0
           no2(ii:end)=no2(ii)+1;
     resn_band2(iiiii:6:iiiii+6*dur(iii))=req2(iii)-m;
           end
           req2(iii)=0;
          elseif rem(iiiii,3)==0
      for iiiiii=ii*6-2:ii*6

band_av2(iiiiii)=band_av2(iiiiii)+resn_band2(iiiiii);
          resn_band2(iiiiii:6:60)=0;
        if req2(iii)<=band_av2(iiiiii)-2*h_flag2(iiiiii)
           band_av2(1,iiiiii)=band_av2(1,iiiiii)-req2(iii);
           band_av2(1,iiiiii+6*dur(iii))=req2(iii);
          if req2(iii)~=0
             no2(ii:end)=no2(ii)+1;

resn_band2(iiiiii:6:iiiiii+6*dur(iii))=req2(iii)-m;
                end
             req2(iii)=0;

     elseif rem(iiiiii,6)==0
            block2=block2+1;
            block2_p(ii:end)=block2;
            disp(['**no   bandwidth   avilible   for   new
call(AJCAC)**
             call  number  ('  num2str(iii)  ')  has  been
rejected'])
                    end
                 end
               end
             end
           end
```

```
          end
        end
      end
    end
    % display the last state of the bandwidth
    disp(' ')
    disp(['avilable bandwidth (NAJCAC) = ' num2str(band_av(1,55:60))])
    disp(['avilable bandwidth ( AJCAC) = ' num2str(band_av2(1,55:60))])
    disp(' ')
    % calculate the total rejected calls
    rej=drop+block;
    rej2=drop2+block2;
    disp(' ')
    disp(['The total number of calls been dropped (NAJCAC)= ' num2str(rej) ' call/s'])
    disp(['The total number of calls been dropped ( AJCAC)= ' num2str(rej2) ' call/s'])
    % calcuclate the probability of blocking and droping for AJCAC and NAJCAC
    for ii=1:1:T
        dp(ii)=drop_p(ii)/numuh;
        bp(ii)=block_p(ii)/numu;
        dp2(ii)=drop2_p(ii)/numuh;
        bp2(ii)=block2_p(ii)/numu;
    end
    % calculate the throughput for AJCAC and NAJCAC in certain second
    th=64*no/100;th2=64*no2/100;
    % calculate the throughput for AJCAC and NAJCAC from the start to a certain second
```

```
for ii=T:-1:2
    th(ii)=sum(th(ii:-1:1));
    th2(ii)=sum(th2(ii:-1:1));
end
% calculate the delay for AJCAC and NAJCAC
d=th*8/12.5;d2=th2*8/12.5;
% plot the figuers
plot(1:4*T/10:4*T,dp(1:T/10:T),'r-
*','linewidth',1.5,'MarkerSize',6),hold on
plot(1:4*T/10:4*T,dp2(1:T/10:T),'g*','linewidth',1.5,'
MarkerSize',6),legend('Non Adaptive
JCAC','AdaptiveJCAC','Location','NorthWest'),grid,ylabel('H
and-off dropping probability  %'),xlabel('Call arrival rate
"call/second" '),title(' Handoff Call Dropping Probability
VS Call Arrival Rate')
figure
plot(1:4*T/10:4*T,bp(1:T/10:T),'r*','linewidth',1.5,'M
arkerSize',6),hold on
plot(1:4*T/10:4*T,bp2(1:T/10:T),'g*','linewidth',1.5,'
MarkerSize',6),legend('Non Adaptive JCAC','Adaptive
JCAC','Location','NorthWest'),grid,ylabel('New call
blocking probability %'),xlabel('Call arrival rate
"call/second"'),title('New Call Blocking Probability VS
Call Arrival Rate')
figure
plot(1:4*T/10:4*T,th(1:T/10:T),'r*','linewidth',1.5,'M
arkerSize',6),hold on plot(1:4*T/10:4*T,th2(1:T/10:T),'g-
*','linewidth',1.5,'MarkerSize',6),legend('NAJCAC','AJCAC',
'Location','NorthWest'),grid,ylabel('Throughput
"byte"'),xlabel('Call arrival rate
"call/second"'),title('Throughput VS Call arrival rate')
figure
```

```
plot(1:4*T/10:4*T,d(1:T/10:T),'r*','linewidth',1.5,'MarkerSize',6),hold on plot(1:4*T/10:4*T,d2(1:T/10:T),'g-*','linewidth',1.5,'MarkerSize',6),legend('NAJCAC','AJCAC','Location','NorthWest'),grid,ylabel('Delay "10^-^3 second"'),xlabel('Call arrival rate "call/second"'),title('Delay VS Call arrival rate')
```