USMANU DANFODIYO UNIVERSITY, SOKOTO
(POSTGRADUATE SCHOOL)

ENHANCED ADAPTIVE CALL ADMISSION CONTROL (EA-CAC) SCHEME WITH BANDWIDTH RESERVATION FOR LTE NETWORKS.

A Dissertation
Submitted to the
Postgraduate School

USMANU DANFODIYO UNIVERSITY, SOKOTO, NIGERIA
IN PARTIAL FULFILLMENT OF THE REQUIREMENTS
FOR THE AWARD OF DEGREE OF
MASTER OF SCIENCE (COMPUTER SCIENCE)

BY

MALAMI, Maniru Umar
(16210310001)

Department of Mathematics

March, 2021

DEDICATION

This research work is dedicated to almighty Allah Subhanahu Wa Ta'ala who gave me the strength and wisdom to carry out the research work successfully.

CERTIFICATION

This dissertation by MALAMI, Maniru Umar (16210310001) has met the requirements for the award of Master of Science (Computer Science) of the Usmanu Danfodiyo University, Sokoto, and is approved for its contribution to knowledge.

______________________ External Supervisor	________________ Date
______________________ Prof. A. Mohammed Major Supervisor	________________ Date
______________________ Dr. A. Roko Co-Supervisor I	________________ Date
______________________ Dr. A.Y. Tambuwal Co-Supervisor II	________________ Date
______________________ Dr. I. Garba Head of Department	________________ Date

ACKNOWLEDGEMENTS

All thanks and glories to Allah (SWT) for making it possible for me to complete this research work successfully. I am forever indebted to my major supervisor, Professor Aminu Mohammed and co-supervisors Dr. Abubakar Roko and Dr. Ahmed Yusuf Tambuwal for their support, encouragement, guidance throughout this research work. This research work would not have been possible without their guidance and support.

The researcher would like to thank all lecturers of the Department of Mathematics for their guidance and inputs towards improving this dissertation especially during departmental seminar presentations.

The researcher is grateful to the entire family of Professor Malami Umar Tambawal (Zarumman Tambuwal) whom has always encourage and support me throughout this course, may almighty Allah reward them abundantly. I am always and will forever be proud of them. Also my sincere gratitude and appreciation goes to my loving and supporting wife for her support and relentless encouragement throughout my studies. My son Usman will also not be forgotten to be acknowledged.

I would also like to acknowledge and appreciate the tireless efforts made by my fellow research group colleagues; Abdulhakeem Abdulazeez and Solomon Yese. They played a vital and significant role and making sure that this research work take it shape. I will also like to appreciate my friends Ziya'u Bello and Sadiq Aliyu Ahmad and Asma'u Shehu for their support and encouragement.

Finally, the researcher sincerely appreciates the Jim-Ovia-NCS scholarship for financially supporting this research work.

TABLE OF CONTENTS

CHAPTER FOUR: RESULTS AND DISCUSSIONS

CHAPTER FIVE: SUMMARY, CONCLUSION AND RECOMMENDATIONS

LIST OF FIGURES

LIST OF TABLES

ABSTRACT

Call admission control (CAC) is one of the radio resource management (RRM) techniques that regulates and provide resources for new call requests or active call requests in the network. The existing call admission control schemes waste bandwidth due to its failure to check whether the degraded bandwidth will be enough to admit the new call requests. It also increases the call dropping probability (CDP) and calling blocking probability (CBP) of real time calls as a result of the delay incurred when bandwidth is degraded from the admitted real time (RT) calls. In this study, an Enhanced Adaptive Call Admission Control (EA-CAC) scheme with bandwidth reservation was proposed. The scheme proposed a prior-check mechanism which ensures bandwidth to be degraded will be enough to admit the new call request. It further incorporates an adaptive degradation mechanism which will degrade non-real time (NRT) calls first before degrading the RT calls, this also ensure that all admitted calls are not degraded below their minimum bandwidth requirement. The performance of the proposed scheme was evaluated against the benchmark scheme using different performance metrics. The EA-CAC increases the throughput of RT calls by 25% and also reduces the CBP and CDP by 12.2% and 15.2% respectively. The scheme performed better than the benchmark scheme in terms of throughput, CBP and CDP of RT calls without sacrificing the performance of the NRT calls.

CHAPTER ONE

GENERAL INTRODUCTION

1.1 Introduction

Today, wireless broadband (WiBB) technologies are fast evolving to satisfy the present and future demand of users for efficient transmission of multimedia applications. Long Term Evolution (LTE) is one of such WiBB technologies designed by the Third Generation Partnership Project (3GPP) for efficient transmission of multimedia applications. The LTE standard is focused on delivering high data rates for bandwidth-demanding applications, improving flexibility and spectral efficiency. These features make LTE an attractive solution for both users and mobile operators (Angelos, Elli, Luis and Christos, 2011).

The fundamental objective of LTE is to guarantee quality of service (QoS) requirements and minimize network congestion for different users (Mamman, Zurina, Azizol and Abdullah, 2018). This can be achieved through the radio resource management (RRM) techniques. Radio resource management techniques are employed by wireless networks to improve the utilization of radio resources. Radio resources are utilized using various schemes that can are categorized into three major groups (Mohamed, 2005). The first group represents frequency or time resource allocation schemes which include channel allocation, scheduling, transmission rate control and, bandwidth reservation schemes. The second group represents power allocation and control schemes which include transmitter power of the terminals and base stations. The third group represents access port connection schemes which include call admission control, base station assignment and handoff control algorithms. An efficient RRM technique that handles the network resources efficiently is required, this is because in most cases the network resources are scarce and therefore need to be efficiently handled (Daniel, Edem and Enoch, 2014). Specifically, an efficient Call admission control (CAC) scheme which regulates and provides resources for new call requests or active calls is needed.

Several CAC schemes have been proposed for LTE with the aim of reducing call blocking probability (CBP), call dropping probability (CDP), guaranteeing QoS requirements and utilization of network resources (Lei, Yu, Zhao, Chang and Yang, 2008; Ali, Fauzi and Lotfi, 2010; Chadchan and Akki, 2011; Senkapa and Franklin, 2012; Khabazian, Kubbar and Hassanein, 2012; Ramraj, Habibi and Ahmad, 2014; Belghith, Turki, Cousin and Obaidat, 2016a; Belghith, Turki, Cousin and Obaidat, 2016b; AlQahtani, 2017). An adaptive call admission control scheme with bandwidth reservation was proposed by Maharazu, Zurina, Azizol, and Abdullah (2017) to provide efficient resource utilization and prevent BE traffic starvation. The scheme deals with Real-Time (RT) and Non-Real Time (NRT) services. The scheme degrades bandwidth from admitted RT calls when a call arrives and there is no sufficient bandwidth to admit the call. It ensures that all the admitted calls at least retain their minimum bandwidth requirement to avoid call drop. The scheme increases the throughput of BE traffic and reduces both CBP and CDP for BE traffic. However, the scheme causes bandwidth wastage because it fails to check if the bandwidth to be degraded will be enough to admit the new requested call. The scheme also increases the delay of already admitted RT calls which consequently leads to an increase in CBP and CDP of RT calls.

To address the aforementioned problems, this study proposed an Enhanced Adaptive Call Admission Control (EA-CAC) scheme with Bandwidth Reservation which reduced the wastage of bandwidth by ensuring that the bandwidth to be degraded will be enough to admit the new call request. The EA-CAC scheme also reduced the CBP and CDP of RT calls without sacrificing the performance of NRT calls.

1.2 Problem Statement

CAC schemes in LTE generally focus on either reducing the call blocking and call dropping probabilities of calls or guaranteeing the QoS requirements of users or increasing resource utilization.

The schemes proposed by Ali *et al.* (2010); Senkapa and Franklin (2012); and Ramraj *et al*. (2014) which focused on reducing call blocking and call dropping probabilities for both new and handoff calls but suffers from starvation of lower priority call requests and poor network resource utilization. While the schemes by Chadchan and Akki (2011); Khabazian *et al.* (2012); Belghith *et al*. (2016a); AlQahtani (2017) were concerned with guaranteeing QoS of different users also relatively increases the CBP and CDP of the lower priority call request.

Recently, Maharazu *et al.* (2017) proposed an adaptive call admission control scheme with bandwidth reservation to provide efficient resource utilization and prevent BE traffic starvation. The scheme increases the throughput of BE traffic and also reduces both CBP and CDP of BE traffic. However, the scheme wastes bandwidth due to its failure to check whether the resources to be degraded from the already admitted RT calls will be enough to admit the new call request. In most cases where network resources are scarce, utilization of the limited available resources is very important. Also, the scheme increases the CBP and CDP of RT calls as a result of the delay incurred when bandwidth is degraded from all admitted RT calls.

1.3 Aim and Objectives of the Study

This work aimed to propose a Call Admission Control scheme that will improve the performance of LTE networks. This was achieved through the following objectives:

1. To propose a prior-check mechanism that will make sure that the bandwidth to be degraded will be enough to admit the new request.
2. To incorporate an adaptive degradation mechanism into the CAC procedure which will reduce the delay incurred by ongoing RT calls, thus decreasing CBP and CDP of RT and NRT calls.
3. To evaluate the performance of the benchmark scheme against the proposed EA-CAC scheme in terms of throughput, CBP and CDP of RT and NRT calls.

1.4 Motivation

LTE is a wireless standard developed by the 3GPP which focused on delivering high data rates for bandwidth-demanding applications and improving flexibility and spectral efficiency (Angelos *et al*, 2011). The 3GPP standard for LTE does not define any standard for Call Admission Control schemes, therefore it is left open for vendors and network operators to decide on how CAC schemes are developed. Users are always demanding for a better service, therefore there is always the need for an effective radio resources management technique such as call admission control.

1.5 Scope of the Study

This research work mainly concentrates on the radio resource management (RRM) technique that ensures effective resource utilization and guarantees QoS for users with diverse applications in LTE. It focuses on call admission control (CAC) which controls the numbers of connections or requests admitted in a network and also maintains the QoS of admitted/active connections or users.

1.6 Significance of the Study

The attraction of wireless technologies is increasing almost daily because of its flexibility and simplicity. Users on the network are always demanding a better service i.e. they need their QoS requirements to be satisfied. LTE is one of the latest fourth-generation (4G) wireless technology which has high speed and it's quicker than the third-generation (3G) wireless technology. LTE is compatible with previous mobile technologies such as GSM, EDGE, HSPA, etc. It also has better technology for the power consumption of mobile terminals. Satisfying the QoS requirements for different users requires efficient RRM strategies/techniques such as a CAC scheme. An efficient CAC scheme will control the number of users or request to be admitted in a network as well as improve the utilization of network resources. It will also improve the overall system throughput. An efficient CAC scheme will as well improve the revenue generated by service providers i.e. (in the case of revenue-based CAC).

1.7 Organization of the Dissertation

The rest of this dissertation is organized as follows: Chapter two presents an overview of the LTE network by explaining some of the core components of its architecture. Radio resource management (RRM) in the LTE network was also highlighted as well as call admission control (CAC) procedure in LTE were also discussed in the chapter. The chapter further discusses some of the related existing CAC schemes in LTE by highlighting the operations, strengths, and weaknesses of each scheme. Finally, the chapter concludes by summarizing the schemes reviewed in the literature.

In chapter 3, the performance evaluation technique used for this research work was presented. It also presented the description of the Vienna LTE system Level Simulator which was used to evaluate the performance of the proposed scheme against the benchmark scheme. The chapter also presented the research framework for this research work. It further described the proposed EA-CAC scheme by showing its diagrammatic representation and the pseudo-code. Performance evaluation metrics used to evaluate the performance of the proposed EA-CAC scheme against the benchmark scheme were also presented in the chapter.

Chapter 4 presented the simulation topology that was used for the simulation experiment. The chapter further presents the simulation parameters used for the simulation experiment. The simulation experiment results obtained were presented using graphs. The results were presented and discussed for all the performance metrics which are throughput, CBP and CDP of both RT and NRT calls. The chapter concludes by summarizing the discussion of the simulation experiment results that were obtained.

Finally, Chapter 5 presents the summary of this research work by highlighting the problems of the benchmark scheme which the proposed EA-CAC scheme addressed. It also presents how the proposed scheme was implemented and the results that were achieved after the simulation experiments. The chapter also presented a conclusion by highlighting what was achieved by the research and recommends some future work that can help to improve the research. It also presented the articles/papers that were published in the course of this research work.

CHAPTER TWO

LITERATURE REVIEW

2.1 Introduction

Quality of Service (QoS) provisioning is a critical issue in wireless broadband networks. LTE is one of the broadband technologies that ensure that the QoS of users is guaranteed by supporting different radio resource management techniques (RRM) such as scheduling, power saving, congestion control, call admission control, etc. This chapter presents an overview of long term evolution (LTE) network, radio resource management (RRM) in LTE and call admission control in LTE. The chapter also presented a review of some related or works on CAC schemes in LTE.

2.2 Long Term Evolution (LTE) Network

Long Term Evolution (LTE) is an evolving wireless standard developed by the 3rd Generation Partnership Project (3GPP) which, along with 3GPP HSPA+, 3GPP EDGE Evolution and Mobile WiMAX (IEEE 802.16e), opens the road to 4G technologies. The LTE standard is focused on delivering high data rates for bandwidth-demanding applications and improving flexibility and spectral efficiency, thus constituting an attractive solution for both users and mobile operators. LTE network was designed to deliver a peak data rate of 100Mbps in the downlink and 50Mbps in the uplink. This requirement was exceeded in the eventual system which delivers peak data rates of 300Mbps and 75Mbps for the downlink and uplink respectively (Ashish, Ankit, and Lalit, 2013).

The LTE architecture is also known as Evolved Packet System (EPS) comprises of two main components which are: Evolved Radio Access Network (E-UTRAN) and Evolved Packet Core (EPC) (3GPP, 2010). The E-UTRAN consists of a network of enhanced base stations referred to evolved NodeB (eNBs) whose main function is to manage the available radio resources and mobility in the cell to optimize the communication among all User Equipment (UEs). The EPC, on the other hand, is the core network that controls the activities of the user equipment (UEs). It comprises of Mobility Management Entity (MME), Serving Gateway (S-GW), Home Subscriber System (HSS) and Packet Data Network Gateway (P-GW). The MME controls the high-level operation of the mobile by sending it signaling messages about issues such as security and management of data streams that are unrelated to radio communication. On the other hand, HSS is a component that contains subscription data of the UE. The HSS stores user authentication data and subscription status. S-GW handles the data and packet routing within the LTE while P-GW handles data and packets routing towards non-3GPP data networks (3GPP, 2013). The system architecture of LTE is shown in figure 2.1.

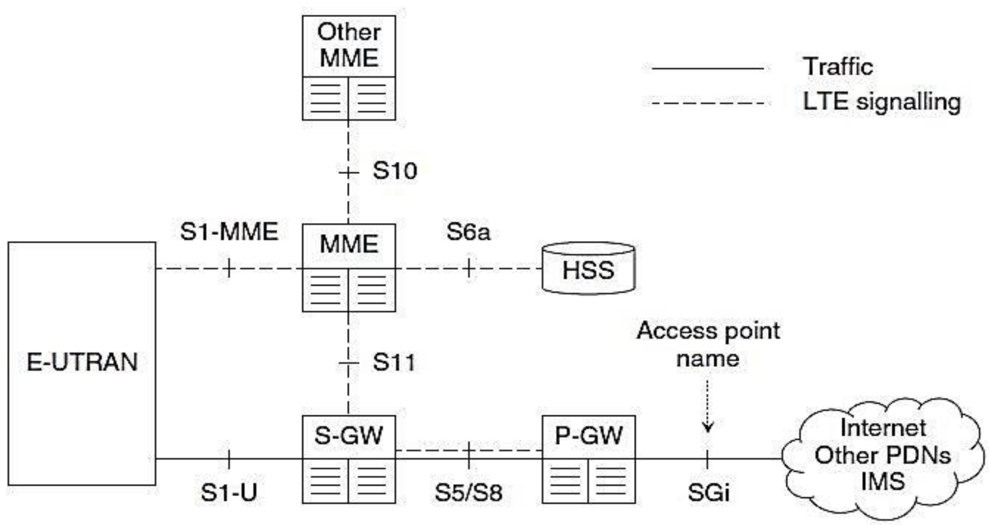

Figure 2.1: LTE System Architecture (3GPP, 2013).

The LTE radio interface supports three multiple access techniques which are Orthogonal Frequency Division Multiple Access (OFDMA), Multiple Inputs Multiple

Outputs (MIMO) and Single Carrier Frequency Division Multiple Access (SC-FDMA) techniques (Navita, and Amandeep, 2016). OFDMA is a multiple access technique used at the downlink channel in LTE system which supports high Quality of Service (QoS) to the accessing points. MIMO, on the other hand, uses multiple transmitters and receivers to transfer more user data at the same time. The MIMO supports high coverage, high data rate and better robustness, low bit error rate, and better spectral efficiency. It is also used as a downlink channel. SC-FDMA is a multiple access technique that is used in the uplink channel of the LTE system.

2.3 Radio Resource Management (RRM) in LTE

Radio resource management (RRM) is the system-level control of co-channel interference and radio transmission characteristics in a wireless communication system. RRM techniques are used to improve the utilization of radio resources of the wireless network. Radio resource management in divided into two phases (Kandaraj, Adlen, Jean and Cesar, 2011):

1. **Radio resource configuration:** It is responsible for allocating the proper resources to any incoming request into the system, this is done to ensure that the new request will not cause the network to be overloaded or congested thus compromising the quality of service (QoS) and stability of the network. Nevertheless, congestion might occur, thus affecting the QoS of users due to mobility.
2. **Radio resource re-configuration:** It is responsible for the re-allocation of network resources within the network when the traffic intensity is increasing or congestion starts to occur to maintain the QoS of different users throughout the network. It should

change the overloaded system back to the target system by rearranging the resource between various user applications on the same network.

To ensure efficient use of radio resources, several techniques are used as part of RRM to provide the users with a service following the configured QoS parameters. The main RRM techniques in LTE are; packet scheduling, call admission control, power control and handoff control (Kandaraj *et al*, 2011).

Packet scheduling is one of the radio resource management techniques which deals with the distribution of available network resources among users by implementing some set of rules. These rules are referred to as a decision-making process that chooses users for allocating the radio resource to fulfill their QoS requirements and also their scheduling priorities. The main responsibility of the scheduler is to schedule different users which require various services to achieve the desired QoS for robust system performance. Each user has different QoS needs according to the service it demands and is scheduled accordingly (Ayesha and Mohsin, 2017).

Power control is also one of the most important radio resource management techniques in the LTE system which ensures that each user equipment (UE) is provided with sufficient power and maximize the battery life of the UE. Power control is very important in uplink transmission. Power control is mainly divided into two main types which are: the open-loop power control and the closed-loop power control. (Ayesha and Mohsin, 2017).

Handoff control is another RRM technique that transfers an active call in one cell to one of its neighboring cells with a subscriber's movement. The main objectives of handoff control mechanisms are to ensure the stability of active calls with required QoS, ensures load balancing in a wireless network and minimize the interference level of the whole wireless system. Handoff control is divided into four main types which are: Intra system handoff, Inter-system handoff, hard handoff, and soft handoff. (Ayesha and Mohsin, 2017).

Call admission control (CAC) is one of the RRM techniques which handles the requests for new EPS bearers in the corresponding cell. The decision to admit a new user is made by considering several admission criteria, such as resource availability, QoS requirements of the new bearer, priority levels, and provided QoS to the current bearers served, etc (Maniru, Aminu, Abubakar, Ahmed and Abdulhakeem, 2019).

This research work focuses on CAC which is one of the RRM techniques in LTE that ensures QoS of different users is guaranteed and also ensures effective utilization of available network resources.

2.4 Call Admission Control (CAC) in LTE

Call admission control (CAC) is a process of accepting new calls or handoff calls in a network while regulating the QoS of existing or active calls without degrading any call drop (Mamman *et al.* 2018). CAC is an RRM technique and has a direct impact on QoS for individual connection and the overall system efficiency (Raymond, Rob, Riccrado and Mitsuhiro, 2010). Call admission control is located at layer 3 i.e. network layer in the evolved Node B (eNB) and is used for setup of both new user and handoff users (Al-Qahtani, 2017). Call requests are normally classified as New Call (NC) and Handoff Call (HC). NC is a type of call that is requesting for a new connection or requesting to be connected into the network while HC is an ongoing or active call that needs to be transferred from one cell to another and still maintain its connection (Maniru, Aminu, Abubakar, Ahmed and Abdulhakeem, 2021).

The major objective of CAC is to ensure efficient resource allocation and to monitor the resource utilization in the high volume of traffic. CAC determines the condition for accepting or rejecting an NC or HC into the network based on pre-defined criteria such as availability of network resources, network channel condition, etc. to guarantee the QoS parameters without affecting the existing calls (Faouzi, Khitem, Mohammed and Lotfi, 2012). CAC process is always performed when a UE starts communication with the eNodeB either through a new call or a handoff call or a new service request by the UE (Ayaz, Chowdhry, Baloch, and Pathan, 2006). When the UE wants to establish a connection with the eNodeB, it sends a request for resource allocation, admission control at eNodeB handles the request. For RT call requests, if connection causes excessive interference to the system, the request will be denied. Otherwise, resources will be allocated for that connection. For the NRT connection request, the optimum scheduling of the packets must be determined after the admission of the call.

The 3GPP standard does not define any standard for call admission control, it has been left open for vendors and network operators to decide on how the CAC schemes are developed.

Basic Call Admission control (BCAC) is a static call admission control scheme (Belghith, Turki, Cousin and Obaidat, 2016a). The decision for the acceptance and rejection of a call request depends only on the availability of network resources. Call requests are only admitted into the network when the requested resources are less than or equal to the available network resources, otherwise, the call request is rejected. Therefore, in the BCAC scheme, the admission criteria always depend on the availability of network resources. Figure 2.2 describes the operation of the BCAC scheme.

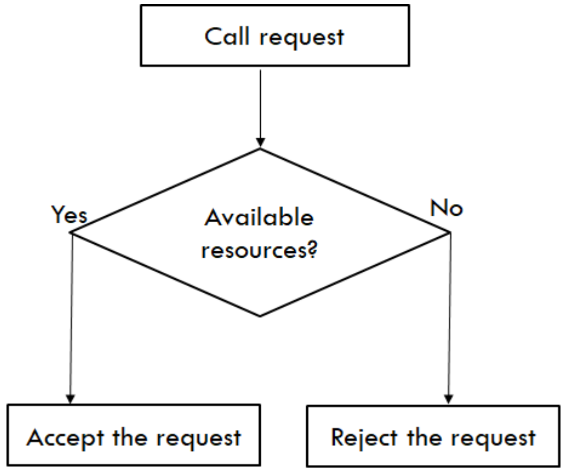

Figure 2.2 Description of Basic CAC scheme

The multi-service call admission control (MSCAC) was proposed for 3G/4G networks as an improvement to the BCAC scheme. MSCAC supports two types of services; RT and NRT where RT are for conversational and streaming calls while NRT are for BE calls (Belghith *et al*, 2016a). The scheme divides radio resources into two parts, one part for the RT calls and the other part for NRT calls.

The design of a CAC scheme depends on some parameters such as availability of resources, quality of network parameters, quality policies, call prioritization, mobility management, and optimization methodologies, etc. (Mohammed, 2005).

2.5 Review of Related Works

In this section, some related call admission control schemes in the LTE network are reviewed by highlighting the operation, strength(s) and weakness(s) of each scheme:

Lei *et al* (2008) presented an adaptive call admission control (CAC) scheme for LTE systems with heterogeneous services. The scheme adaptively determines the threshold for each service class based on the traffic condition. It further employs a transmission guard interval strategy which gives higher priority to RT services that are close to their delay deadline. The scheme accepts RT calls by employing the QoS indicator but unconditionally accepts NRT calls in the presence of available resources. Furthermore, the scheme degrades admitted NRT calls to admit handover traffics when the network is overloaded. The scheme achieves low blocking probability under low traffic but starves lower priority traffic and increases their dropping probability due to degradation strategy used.

Manli et al (2009) presented a novel admission control scheme for multiclass services in LTE system to reduce the blocking and dropping probabilities users. The scheme combines complete sharing (CS), virtual partitioning (VP) and service degradation strategies. It groups users into three groups: group 1 are services whose resources can be preempted, group 2 are services whose resources cannot be preempted and group 3 are services that can prempt resources from group 1. The scheme admits a new call request of service group 1 if the available bandwidth in group 1 is greater than or equal to the requested bandwidth, otherwise the call is rejected. Similarly, it accept a new call of service group 2 if the available bandwidth in group 2 and 3 are greater than or equal to the requested bandwidth, otherwise bandwidth is degraded from admitted calls in the group. If the degraded bandwidth is enough to admit the new call, then the call is accepted otherwise the call is rejected. Furthermore, the scheme accept a new call of service group 3 if the available bandwidth in service group 2 and 3 is greater than the requested bandwidth, otherwise the call is rejected. The scheme reduces the CBP of users and also gurantee the QoS of some service types. However, the QoS of lower priority users is not guranteed due to the fact that they are degraded whenever resources are not suffiient. It also increases in the call dropping probability of these users.

Ali *et al.* (2010) proposed a CAC and Resource Block (RB) reservation scheme to reduce dropping probability of calls for handoff calls. The scheme separates incoming traffic according to their priority and assigns higher priority to handoff calls. The RB's strategy allocates a maximum number of RB's when resources are sufficient but assigns lower than the required resources when resources are insufficient. Also, the scheme employs a degradation procedure to degrade RB's of the lowest priority to admit NC that has not exceeded its latency. The scheme reduces handoff dropping probability of handoff calls and maintain low blocking probability. However, the scheme starves lower priority class due to its degradation procedure and as such their QoS is not guaranteed. It also increases the call blocking probability of new calls.

Chadchan and Akki (2011) presented a CAC with a Priority-Scaled (PS) preemption scheme for LTE networks to guarantee Quality of Service (QoS) of different users. The PS scheme computes two parameters on arrival of a request; R_{Total} and R_{Min} where R_{Total} is the number of resources that can be obtained by total preemption of all Lower Priority Preemptable Active Bearers (LP PABs) while R_{Min} is the number of resources that can be obtained by reconfiguring all LP PABs to their minimum QoS level. The scheme blocks a new request if R_{Total} is not sufficient to satisfy its QoS needs else if R_{Min} is sufficient to service the new request, then the Priority-Scaled Minimum QoS Preemption Algorithm (PS-MQPA) is used. The PS-MQPA preempts more resources from the lower priority bearers than from higher priority bearers to ensure better QoS for higher priority bearers. Furthermore, If R_{Min} is sufficient to service a new request but the new request requirements are less than R_{Total} then the Total Preemption Algorithm (TPA) is employed. The TPA employs a total preemption strategy by dropping all LP PABs with the lowest priorities and highest resources. The scheme guarantees QoS for LP PABs but in the presence of a large number of higher priority requests, LP PABs experiences a higher dropping rate due to the total preemption strategy used by the scheme.

Faouzi, Khitem, Mohammed & Lotfi (2012) proposed an adaptive call admission control scheme to reduce the dropping probability of handoff calls. The scheme employs a resource block (RB) reservation strategy which gives higher priority to handoff calls and reserves a certain amount of resource for the handoff calls. It employs a load balancing mechanism which adjust the number of resources to be reserved for handoff calls. The scheme accepts a new call request when the requested resources are less than or equal to the available resources otherwise the call is rejected. It admits a handoff call when if the requested resources are less than or equal to the available and reserved resources for handoff calls otherwise the call is queued into a waiting queue. It served the queued calls based on their latency. The scheme reduces the handoff dropping probability of handoff calls because they are given higher priority but increase the call blocking probability of new call requests. It also reduces the utilization of network resource because of the reservation strategy employed by the scheme.

Senkapa and Franklin (2012) proposed an Extensive Dynamic Bandwidth Adaptation Call Admission Control (DB-CAC) scheme to reduce call dropping probability and to increases the utilization of network resources. The DB-CAC scheme takes into account the separation between incoming traffic for each class of service and then prioritizes HC over NC using a load balancing strategy. It also employs a prediction technique which helps in advance resource reservation whenever a call is detected based on the user's past behavior. The scheme operates in two stages: arrival and departure. At the arrival stage, the algorithm gets as many resources as required to service the queued HCs and NCs by degrading the active NRT service. At the departure stage, more resources are assigned to the RT service calls to increase system utilization. The scheme arranges all the NRT calls in descending order and degrades them to service RT calls when resources are not sufficient, but when the resources are sufficient, RT calls are serviced else they are discarded. The scheme reduces new call blocking probability due to the prediction strategy employed by the scheme. It also improves network resource utilization because resources are restored when a call is over. However, the NRT calls are not treated fairly due to the degradation strategy employed by the scheme.

Khabazian *et al.* (2012) presented a CAC scheme with resource reservation to avoid call QoS degradation. The scheme takes into consideration two types of traffics which are: narrow-band and wide-band services. The CAC scheme accepts a narrow-band service call when there are enough unused resources to provide the data rate of admission and during its call holding time, otherwise, the call is blocked. It accepts a wide-band service call if there are enough resources that can provide the requested data rate at admission time and when a narrow-band service call is terminated or leaves the cell, otherwise the call is blocked. Furthermore, the scheme reserves a constant amount of extra resources to a service during admission to eliminate QoS degradation. The scheme reduces call's QoS degradation under heavy traffic but it increases call blocking probability for the wide- band service calls.

Jie & Yangfan (2013) proposed a call admission control scheme for LTE femtocell networks to support multimedia services with diverse traffic classes and different bandwidth requirement. The scheme operates in two stages: Subscriber authentication stage and admission control stage. On arrival of an E-UTRAN Radio Access Bearer (E-RAB) request, the scheme checks if the threshold based on subscriber authentication is not exceeded and then checks if there are available PRBs in the system. The request is accepted if the conditions are satisfied otherwise, the request is queued. The scheme accepts any queued request if it satisfies the predefined admission criteria and then leave the remaining requests in the queue. It rejects the remaining queued request when they reach their queue timeout. The scheme reduces CBP for each class of traffic and also increases resource utilization. However, it increases CDP of some users when the queued requests reaches queue timeout.

Ramraj *et al.* (2014) proposed a CAC scheme for high-speed vehicular communications to reduce new call blocking and handoff call dropping probability for RT and NRT traffic. The scheme was based on Resource Blocks (RBs) reservations which reserves resources for active calls and new calls. It also estimates the Bit Error Rate (BER) based on the Rayleigh fading model in high vehicular speed. The scheme accepts a new call request when the requested RBs are less than or equal to the available resources. Otherwise, if the RBs are not sufficient, then the remaining RBs will be reserved for future or expected incoming calls. The scheme further accepts a future or expected an incoming call when the required resources are equal or less than the available resources i.e. reserved resources and available network resources otherwise, the call is rejected. The scheme reduces CBP and CDP of calls but fails to utilize network resources efficiently because the resources reserved may not be fully utilized by the future calls or expected incoming calls.

Fatima, Doan & Iain (2014) presented a fair intelligent admission control for to provide fair resource allocation and guarantee maximum resource utilization for different service types. The scheme combines a complete sharing (CS) and virtual partitioning (VP) resource allocation techniques. It uses CS for multiclass users to share available network resources. The scheme further uses VP to different among multi service users when the network resources are scare. It classifies call requests and categorizes the requests as GBR and MBR based on their service types. The scheme then give higher priority to GBR. It accepts a higher priority request by applying a step-wise degradation approach which degrades resources allocated to lower priority bearers when there is no sufficient resources to admit the request. It admits a lower priority calls when there is enough resources otherwise, the call re rejected. The scheme reduces call blocking probability for higher priority calls and guarantees fair resource sharing among service types. However, the scheme increases call blocking and call dropping probability for lower priority calls.

Belghith *et al.* (2016a) presented a Flexible Call Admission (FCAC) scheme to increase reduce the CBP of new calls and increases the utilization of network resources. The scheme classifies requests into RT and NRT and also estimates the channel quality based on received signal strength (RSS) to identify a new and handoff call request. The scheme accepts RT request with the bad channel if the Occupation Ratio of the Bandwidth (OR_BW) is lower than the threshold set for RT calls (th_RT). It further classifies RT requests as RT_NC and RT_HC. It accepts RT_HC automatically when the channel condition is good because it has the highest priority. It also accepts RT_NC requests with the lowest blocking rate probability (BRnc_rt) and rejects a request with the highest BRnc_rt. The scheme accepts NRT requests if the OR_BW is lower than a threshold set for NRT requests (th_NRT), otherwise, the request is rejected if the total number of available PRBs is not sufficient to service the request. The scheme further employs a preemption strategy to preempt resources from admitted NRT calls that have been fully or partially served to service RT requests. The scheme reduces CDP for RT calls due to higher priority given to RT requests but increases CBP and CDP of NRT call requests due to the preemption strategy applied to them.

Belghith *et al.* (2016b) proposed an Efficient Bandwidth Call Admission Control (EB_CAC) to reduce CBP and satisfy the QoS for RT and NRT calls in LTE networks. The scheme classifies service types as RT and NRT and also classifies call requests as NC and HC. The scheme also estimates channel quality based on received signal strength (RSS) to determine good and bad channels. It applies a congestion thresholds as well as a blocking probability for each call type. The scheme classifies RT call type as either RT_HC or RT_NC and admits an RT_HC request if there are sufficient PRBs without considering the channel condition and BOR. It rejects RT_NC having a bad channel if the bandwidth occupational ratio (BOR) exceeds the set threshold, else the request is accepted. RT_NC with a good channel is also rejected if the BOR exceeds the threshold set. The scheme rejects NRT requests if there are no sufficient PRBs on the system. Furthermore, it further classifies NRT requests into NRT_HC and NRT_NC and NRT_HC are admitted independently of their channel quality with a blocking probability ratio. NRT_NC having bad channels are accepted with a blocking probability Ratio and NRT_NC having good channels are also accepted with a blocking probability Ratio. The scheme guarantees QoS for different service classes and increases the total system throughput. It also increases the number of accepted RT_HC calls but NRT request experience high dropping rate due to priority given to RT requests.

AlQahtani (2017) presented a Delay Aware and Users' categorizing based call admission control scheme with adaptive resource reservations to guarantee QoS and increase resource utilization. The scheme categorizes users as golden and silver users and further classifies service types of each user as RT and NRT. It virtually reserves a set of physical resource blocks (PRBs) for each service type. The scheme admits a request when there are available PRBs to service the request else, all the requests are admitted into a waiting queue provided the queue is not filled up otherwise the request is rejected. It drops a queued request if it exceeds its predefined queuing time limit. The scheme further determines the Adaptive Priority (AP) of all non-empty queues using the total number of PRBs currently used by all users, number of virtual reserved PRBs, Maximum tolerable delay and Current latency. It gives the highest priority to the queue with the minimum AP and the queue is served first. The scheme guarantees QoS and efficiently utilizes resources because of the virtual resources reservation strategy used. However, requests with the lowest priority which are the NRT and

BE traffics experience a high blocking rate and sometimes even starved due to priority given to higher priority requests.

Maharazu *et al.* (2017) proposed an Adaptive Call Admission Control with Bandwidth Reservation scheme to provide efficient resource utilization and prevent BE traffic starvation. The scheme classifies incoming call requests as RT and NRT. It allocates maximum required bandwidth to RT calls and minimum required bandwidth to NRT calls at the point of admission. The scheme degrades bandwidth from admitted RT calls when a call arrives and there is no enough bandwidth to admit the call. It ensures that all the admitted calls at least retain their minimum bandwidth requirement to avoid call drop. The scheme further introduces a threshold value that changes the reserved bandwidth using various traffic intensity for handoff calls. The scheme increases the throughput and reduces the CBP and CDP for the BE traffics. However, the scheme causes bandwidth wastage due to its failure to check whether the bandwidth to be degraded will be sufficient to admit the new request. It also increases the delay of admitted RT calls which consequently leads to an increase in CBP and CDP of the RT calls.

Therefore, there is a need for a CAC scheme that will reduce the wastage of network resources and also reduce the delay experienced by RT traffic to reduce the CBP and CDP of the RT calls.

2.6 Summary of Reviewed Literature

Table 2.1 presents the summary of reviewed related CAC schemes highlighting the strength (s) and weaknesses (s) of each.

Table 2.1: Summary of Reviewed Related Literature

S/N	Scheme	Strength(s)	Weakness(s)
1	Adaptive call admission control (CAC) scheme. (2008)	Achieves low blocking probability under low traffic.	Starves lower priority traffic and increases their dropping probability due to degradation strategy used.
2	Novel admission control scheme for multiclass services in LTE. (2009)	Reduces the CBP of users and also gurantee the QoS of some service types.	QoS of lower priority users is not guranteed due to the fact that they are degraded whenever resources are not suffiient. It also increases in the call dropping probability of these users.
3	CAC and Resource Block (RB) reservation scheme (2010).	Reduces handoff dropping probability and maintain low blocking probability.	Starves lower priority class due to its degradation procedure and as such their QoS is not guaranteed.
4	CAC with Priority - Scaled (PS) preemption scheme (2011).	Guarantees QoS for LP PABs but in the presence of a large number of higher priority requests.	LP PABs experiences higher dropping rate due to total preemption strategy used by the scheme
5	Adaptive call admission control scheme (2012)	The scheme reduces the handoff dropping probability of handoff calls because they are given higher priority but increase the call blocking probability of new call requests.	It also reduces the utilization of network resource because of the reservation strategy employed by the scheme
6	Extensive Dynamic Bandwidth Adaptation Call Admission Control scheme (DB-CAC) (2012).	Reduces new call blocking probability due to the prediction strategy employed by the scheme. It also improves resource utilization because resources are restored when a call is over.	NRT calls are not treated fairly due to degradation strategy employed by the scheme
7	CAC scheme with resource reservation (2012)	Reduces call's QoS degradation under heavy traffic but it.	Increases call blocking probability for the wide-band service calls
8	CAC scheme for LTE femtocell networks (2013).	The scheme reduces CBP for each class of traffic and also increases resource	It increases CDP of some users when the queued requests reaches queue

		utilization.	timeout.
9	CAC scheme for high-speed vehicular communications (2014).	Reduces CBP and CDP of calls.	Fails to utilize network resources efficiently because the reserved resources may not be fully utilized by the calls.
10	Fair intelligent admission control (2014).	The scheme reduces call blocking probability for higher priority calls and guarantees fair resource sharing among service types.	The scheme increases call blocking and call dropping probability for lower priority calls.
11	Flexible Call Admission (FCAC) scheme (2016).	Reduces CDP for RT calls due to higher priority given to RT requests.	Increases CBP for NRT requests due to the preemption strategy employed.
12	Efficient Bandwidth Call Admission Control (EB_CAC) scheme (2016).	Guarantees QoS for different service classes and increases the total system throughput. It also increases the number of accepted RT_HC calls.	NRT request experience high dropping rate due to priority given to RT requests.
12	Delay Aware and Users' categorizing based Call Admission Control scheme with adaptive Resource Reservation (DA-UC-ARR) (2017).	Guarantees QoS and efficiently utilizes resources because of the virtual resources reservation strategy used	Requests with the lowest priority which are the NRT and BE traffics experiences a high blocking rate and sometimes even starved due to priority given to higher priority requests
14	Adaptive Call Admission Control with Bandwidth Reservation scheme (2017).	Increases the throughput of BE traffic and reduces both CBP and CDP for BE traffic.	Causes bandwidth wastage due to its failure to check whether the bandwidth to be degraded will be sufficient to admit the new request. It also increases the delay of already admitted RT calls which leads to an increase in CBP and CDP of calls

CHAPTER THREE

METHODOLOGY

3.1 Introduction

This section presents the performance evaluation technique that was used to evaluate the performance of the proposed EA-CAC scheme against the benchmark scheme. The description of the Vienna LTE system level simulator which was used to perform the simulation experiments were also presented. The Research framework and the description of the proposed EA-CAC scheme were also discussed in this section. Finally, performance metrics used to evaluate the performance of the EA-CAC scheme against the benchmark scheme were also presented.

3.2 Performance Evaluation Techniques

Performance evaluation is important in wireless networks to measure the efficiency and effectiveness of the network. It involves certain techniques such as direct measurements using testbed, analytical or simulation modeling.

Network testbeds are designed to offer environments to researchers and practitioners in which experimental systems, configurations, and protocols can be carefully tested and evaluated. Testbed produces results are closer to reality than other evaluation techniques. However, researchers find it difficult to use due to its high cost of implementation and hardware limitation (Charles, *et al.* 2012).

An analytical model is primarily quantitative or computational nature and represents the system in terms of a set of mathematical equations that specify parametric relationships and their associated parameter values as a function of time, space, and/or other system parameters. Analytical modeling is less expensive as compared to other techniques.

However, it requires a lot of simplified assumptions and mathematical computations to obtain a tractable result (Sanford, Alan and Rick 2015).

On the other hand, simulation is used to predict the performance of a wireless network's architecture, protocol, device, topology, etc. It is a cost-effective and flexible technique for performance evaluation of wireless systems. In simulations, network scenarios are easily created and modified, and data can be easily collected. More importantly, simulations can model large network topologies which would be very expensive to experiment using testbed (Obaidat and Green, 2003).

In this study, the Vienna LTE System-level simulator was used for the implementation and evaluation of the proposed EA-CAC scheme against the benchmark scheme. The simulator is an open-source released free for academic and non-commercial purposes (Taranetz *et al*, 2015).

3.3 Vienna LTE System Level Simulator

The Vienna LTE system level simulator is developed at Vienna's University Institute of Telecommunications. It was developed and implemented using the Object-Oriented Programming (OOP) concept of MATLAB. The simulator follows the schematic block diagram as shown in Figure 3.1. Similar to other system-level simulators, the core parts of the simulator consists of two main models which are: link measurement and link performance. The link measurement model abstracts the measured link quality used for link adaptation and resource allocation. While the link performance model determines the link block error ratio (BLER) at reduced complexity (Josep, Martin, and Markus, 2010).

Implementation-wise, the simulator flows follow the normal execution format of an object-oriented programming concept i.e., it executes the first line, followed by the second line, third line as so on. The simulation is performed by defining a Region of Interest (ROI) in which the eNodeB's and UE's are positioned and the simulation length in Transmission Time Intervals (TTIs).

The simulator has a rich set of features and easy adaptability which led to numerous publications from researchers all over the globe including studies in energy-efficient cell-coordination schemes, call admission control schemes, handover algorithms in self-optimizing networks and resource allocation techniques for femtocell networks as well as machine-to-machine communications. Currently, the simulator counts more than 30,000 downloads and undergoes permanent peer-review from a substantially large online community (Martin *et al.*, 2015).

The Vienna LTE system level simulator was used to compare the performance of the proposed EA-CAC scheme against that of the benchmark scheme. It was chosen because it is open-source and the license is free for academic and non-commercial work. Similarly, the simulator was used by the authors of the benchmark scheme to evaluate the performance of their scheme which makes it easier for this work to be compared with the benchmark scheme.

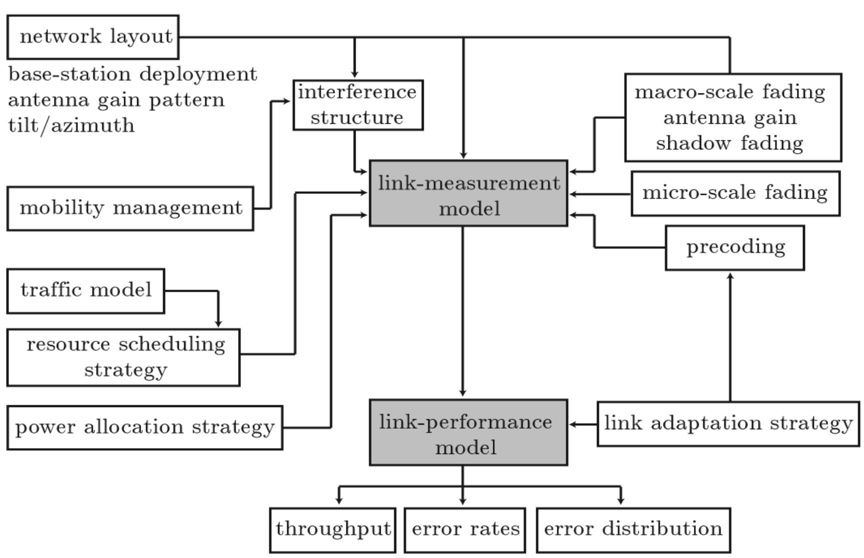

Figure 3.1: Overview of the Vienna LTE system level simulator

In the course of implementing the proposed EA-CAC scheme, new modules were incorporated into the resource scheduling strategy component of the simulator. The modules incorporated were; Calls generation module, calls classification and prioritization module, CAC procedure module and the CAC degradation mechanism module.

The call generation module generates several calls that are transmitted to the eNodeB by the user equipment for the admission process. The calls generated are of different QoS classes and have some features such as call type, size, arrival time, service time, deadline and latency, etc. Generated calls are then passed to the calls classification and prioritization module for further process. These calls are generated before simulation experiment is started.

The generated calls are classified into different QoS classes by the calls classification and prioritization module. The module classifies the calls into real-time (RT) and non-real time (NRT) calls. Each call is then prioritized as either a new call (NC) and handoff call (HC) i.e. we have a real time-new call (RT-NC), real time-handoff call (RT-HC), non-real time-new call (NRT-NC) and non-real time-handoff call (NRT-HC). These prioritized calls are then forwarded to the CAC procedure module.

The CAC procedure module is the module that is responsible for the admission process. On arrival of each call, the modules check some parameters to take the admission decision. These parameters are arrival time, QoS class, call type, the bandwidth required and available bandwidth in the system. After checking these parameters, it then checks the admission criteria for the class and type of call. For instance, if the class is RT and the type is NC, it checks whether the admission criteria specified for the RT-NC call is satisfied. If the criteria are satisfied, the call is accepted else it is rejected or passed to the degradation mechanism module as the case may be.

In the situation where the amount of bandwidth requested by a particular call is not enough to admit the call, the call is passed to the degradation mechanism module which will then degraded appropriate admitted calls if the degradation criteria are satisfied. The degradation criteria are defined in this module and the degradation process is performed by the module.

Apart from the above-mentioned modules, all other modules in the simulator were adopted and used for the simulation experiment. This is because, the modules suite the need for the implementation of the proposed EA-CAC scheme.

3.4 The Research Framework

The research framework for this research is shown in figure 3.2. This framework consists of different stages starting from the analysis and review of previous CAC schemes, problem formulation, proposed EA-CAC scheme, simulation, performance evaluation and comparison with benchmark scheme i.e. an adaptive call admission control scheme with bandwidth reservation for downlink LTE networks.

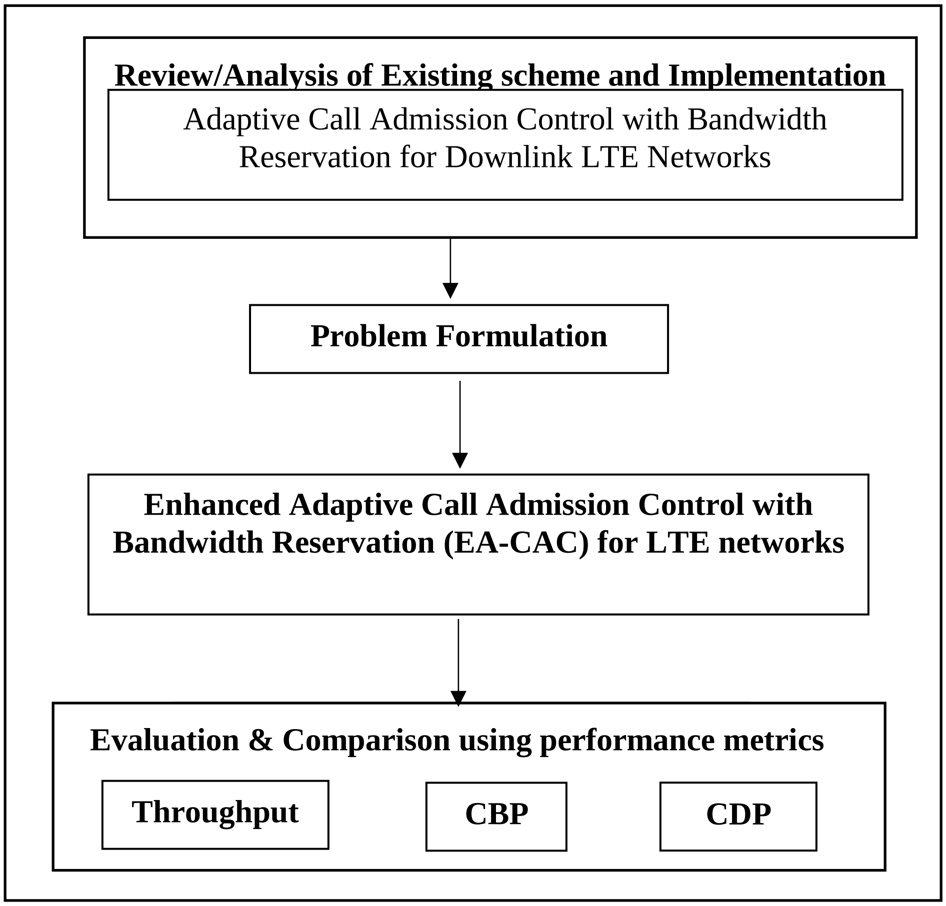

Figure 3.2: Research Framework

3.5 Enhanced Adaptive Call Admission Control (EA-CAC) Scheme with Bandwidth Reservation for LTE networks.

This section presents the description of the proposed EA-CAC scheme which improved the performance of the benchmark scheme proposed by Maharazu *et al* (2017). First, the description of the benchmark scheme will be given. The benchmark scheme allocates maximum and minimum bandwidth requirements to RT and NRT calls respectively at the point of admission. It accepts an RT call when the requested bandwidth is less than or equal to the available bandwidth otherwise the call is rejected. The scheme admits an NRT call request if the requested bandwidth is less than or equal to the available bandwidth, otherwise a degradation procedure is applied to all admitted RT calls since they were assigned their maximum at the point of admission. All admitted RT calls are degraded to their minimum and then if the degraded bandwidth is less than or equal to requested bandwidth, the call is admitted otherwise rejected. However, the scheme wastes bandwidth and also increases the CBP and CDP of RT calls. The scheme is diagrammatically shown in figure 3.3

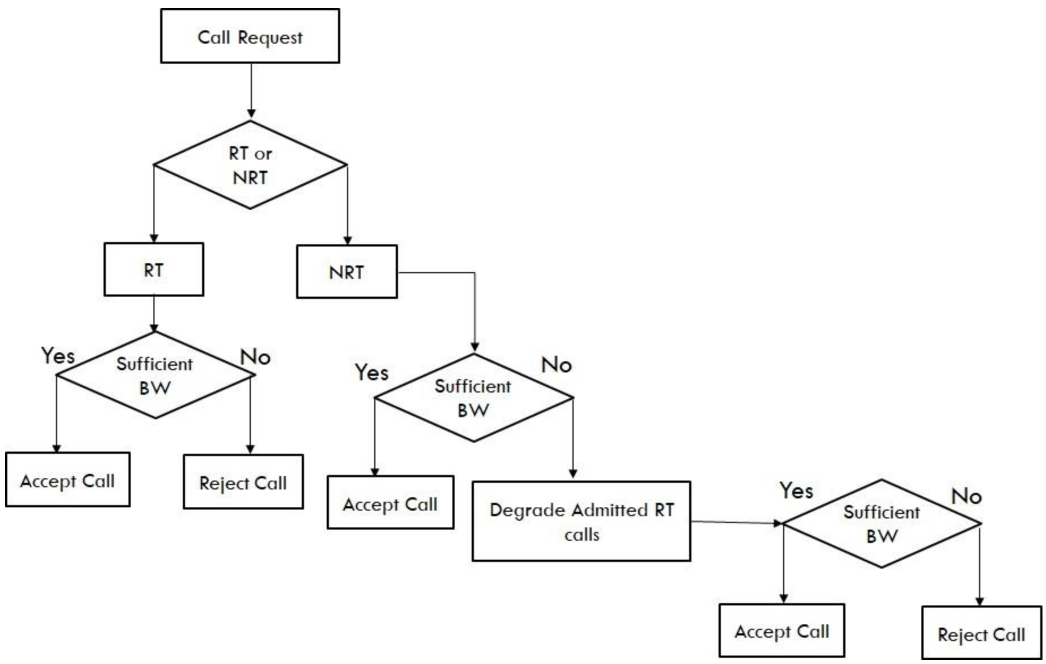

Figure 3.3: Diagrammatic Description of the Benchmark Scheme

The proposed EA-CAC scheme was developed to address the shortcomings of the benchmark algorithm. The EA-CAC is described below:

The proposed EA-CAC scheme allocates maximum bandwidth requirements to both RT and NRT at the point of admission. For RT call requests, the maximum bandwidth requirement is described as:

$$Call_{RT} = BW_{max} \tag{3.1}$$

Where $Call_{RT}$ denotes an RT call and BW_{max} represent the maximum bandwidth for an RT call. Similarly, for NRT call requests, the maximum bandwidth requirement is denoted as:

$$Call_{NRT} = BW_{max} \tag{3.2}$$

Where $Call_{NRT}$ denotes an NRT call and BW_{max} represents the maximum bandwidth for an NRT call. Furthermore, a new call request is admitted into the network, if there is sufficient bandwidth i.e. if the requested bandwidth is less than or equal to the total available bandwidth as described in equation 3.3.

$$NC_{accept} = BW_{req} \leq \ BW_{avail} \qquad (3.3)$$

Where NC_{accept} is a new call to be accepted, BW_{req} is the requested bandwidth and BW_{avail} is the total available bandwidth. Similarly, a new handoff request is accepted into the network if there is sufficient bandwidth i.e. the requested bandwidth is less than or equal to the total available bandwidth and total reserved bandwidth as shown in equation 3.4.

$$HC_{accept} = BW_{req} \leq \ BW_{avail} \qquad (3.4)$$

Where HC_{accept} is the HC to be accepted, BW_{req} is the requested bandwidth, BW_{avail} is the total available bandwidth.

If there is insufficient bandwidth to admit a new call request, then a degradation mechanism is applied. The degradation is applied in two stages. At the first stage, degradation is applied to all admitted NRT calls. The degradable bandwidth for a call can be computed as shown in equation 3.5

$$BWC_{deg} = BW_{max} - \ BW_{min} \qquad (3.5)$$

Where BW_{deg} is the degradable bandwidth for an admitted call, BW_{max} is the maximum bandwidth requirement for a call and BWC_{min} is the minimum bandwidth requirement for a call.

After the first degradation stage, then the total degraded bandwidth is added up to the available bandwidth as shown in equation 3.6 and then the requested call is admitted if the bandwidth is enough. Calls admitted after degradation are admitted by allocating their minimum bandwidth requirement to them.

$$\sum NRT\_BW_{deg} + \ BW_{avail} \qquad (3.6)$$

Where $\sum NRT\_BW_{deg}$ is the sum of degraded bandwidth from admitted NRT calls and BW_{avail} is the total available bandwidth of the system.

If $\sum NRT\_BW_{deg}$ is not sufficient to admit the new call request, then the second stage of degradation is employed on all admitted RT calls. But before the degradation is done, a prior-check mechanism is first used to check whether the degradable bandwidth from admitted RT calls and the available bandwidth will be enough to admit the new call as described below:

$$\sum RT\_BW_{deg} + \ BW_{avail} \geq \ BW_{req} \qquad (3.7)$$

Where $\sum RT\_BW_{deg}$ is the sum of degradable bandwidth from admitted RT calls, BW_{avail} is the bandwidth and BW_{req} is the requested bandwidth.

If equation 3.7 is satisfied then the second stage degradation is performed otherwise the degradation is not performed and the call request is rejected. This will ensure that the bandwidth to be degraded will be utilized i.e. will be enough to admit the new call request. Thus, this will reduce the bandwidth wastage thereby improving the utilization of bandwidth.

Finally, the EA-CAC scheme adopts the adaptive reservation procedure that was used in the benchmark scheme. The adaptive reservation will prevent fixed reservation of bandwidth for handoff calls. It will ensure that a limited amount of bandwidth is reserved if there are few arrivals of handoff calls, likewise, a large amount will be reserved if there is frequent arrival of handoff calls. This will prevent resource wastage compared to what happened in the reservation-based schemes that reserved a fixed amount of bandwidth for handoff calls. The reserved bandwidth is thereby wasted when there is few or no handoff call. Figure 3.4 shows the diagrammatic description of the proposed EA-CAC scheme. The pseudo-code of the proposed EA-CAC is shown in algorithm 3.1, 3.2 and 3.3.

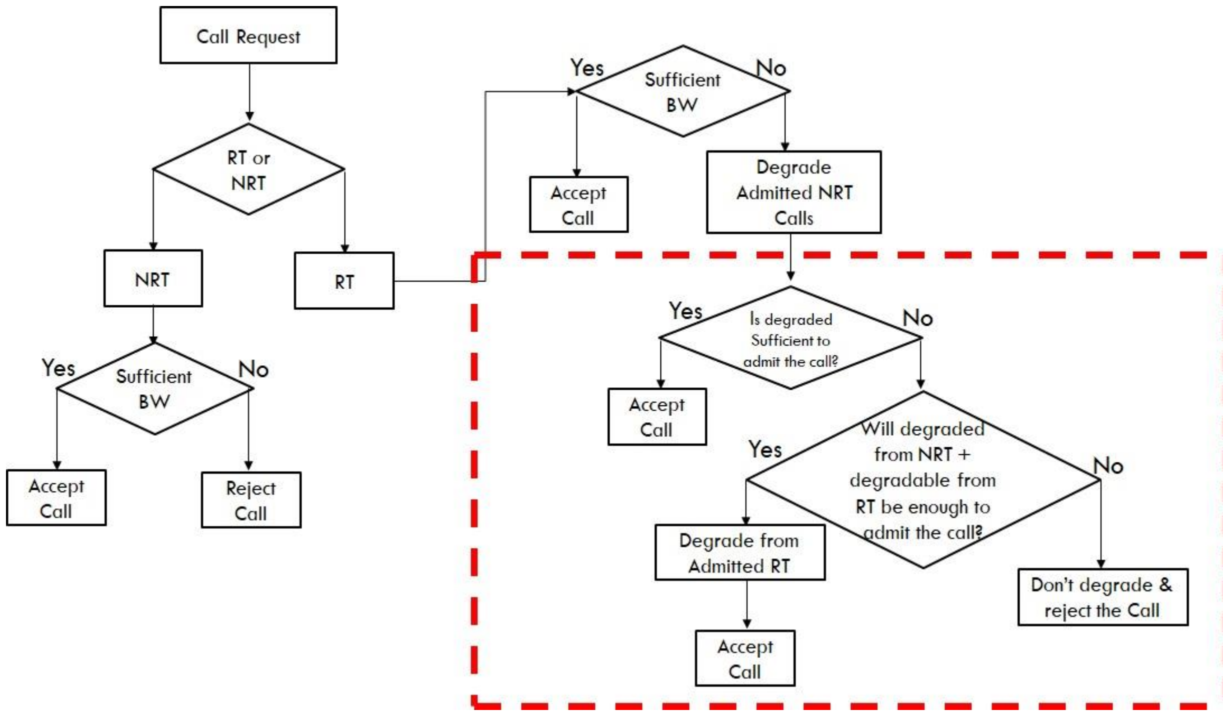

Figure 3.4: Diagrammatic Description of the Proposed EA-CAC Scheme

Algorithm 3.1 represents the pseudo-code for the prior-check mechanism that is employed in the EA-CAC scheme.

Algorithm 3.1: EA-CAC Pre-check Mechanism Algorithm

Input:
AD_RT_{calls}: Admitted RT calls
$\sum RT\_BW_{deg}$: Sum of degradable bandwidth from admitted RT calls
BW_NRT_{deg}: Degraded bandwidth from admitted NRT calls
Initializations
while $BW\_NRT_{deg} \leq BW_{req}$ **then check**
    if $BW\_NRT_{deg} + \sum RT_{BW\,deg} \geq BW_{req}$
        degrade AD_RT_{calls} **then**
            accept call
    else
        don't degrade **then**
            reject call
    end if
end while

Algorithm 3.1 is employed by the EA-CAC scheme before the second stage of degradation is applied. This is to ensure that the degradable bandwidth from admitted RT calls together with the available bandwidth will be available to admit the new call request. By this, the EA-CAC scheme will reduce the bandwidth wastage because the second stage of degradation will not be performed if the conditions in algorithm 3.1 are not satisfied.

Algorithm 3.2 presents the pseudo-code of the adaptive degradation mechanism employed by the proposed EA-CAC scheme.

Algorithm 3.2: EA-CAC Adaptive degradation mechanism.

Input:
BW_{avail}: Available bandwidth
BW_{req}: Requested bandwidth
AD_NRT_{calls}: Admitted NRT calls
AD_RT_{calls}: Admitted RT calls
$\sum BW\_NRT_{deg}$: Sum degraded bandwidth from admitted NRT calls
Initialization
if $BW_{req} \leq BW_{avail}$ **then**
    degrade AD_NRT_{calls}
else if
    $\sum BW\_NRT_{deg} + BW_{avail} \geq BW_{req}$ **then**
        degrade AD_RT_{calls}
else
    don't degrade
        reject call
end if

Algorithm 3.2 is used by the EA-CAC scheme to ensure that the degradation of admitted calls is done in a stepwise manner. The algorithm first degrades admitted NRT calls and then if the resources are not sufficient, it degrades admitted RT calls but after using algorithm 3.1 to reduce the wastage of bandwidth.

Algorithm 3.3 presents the pseudo-code for the Enhanced Adaptive Call Admission Control (EA-CAC) Scheme with bandwidth Reservation for LTE networks.

Algorithm 3.3: Enhanced Adaptive Call Admission Control (EA-CAC) Scheme with bandwidth Reservation for LTE networks algorithm

---

```
1.  Input:
2.  NC: New call
3.  HC: Handoff call
4.  RT: Real-Time traffic
5.  NRT: Non-real time traffic
6.  SMT: Simulation time
7.  Initializations
8.  while TTI is within SMT do
9.      for NC
10.         compute NC according to equation (3.3)
11.             if equation (3.3) holds then
12.                 accept NC
13.             else
14.                 degrade admitted NRT according to equation (3.6)
15.             end if
16.     end for
17.     if equation (3.6) holds then
18.         accept NC
19.     else if algorithm 3.1 holds
20.         else if algorithm 3.2 holds
21.             accept NC
22.     else
23.         reject NC
24.     end if
25.     for HC
26.         compute HC according to equation (3.4)
27.             if equation (3.7) holds then
28.                 accept HC
29.             else
30.                 execute step 17 to 24
31.             end if
32.     end for
33. end while
```

Algorithm 3.3 shows the pseudo-code on how the proposed EA-CAC scheme works. It comprises of the pre-check and the adaptive degradation that is shown in algorithm 3.1 and 3.2.

3.6 Performance Metrics

Evaluation of the proposed scheme was being carried out using three performance metrics that were adopted from the Adaptive call admission control scheme proposed by Maharazu *et al.* (2017). These metrics are as follows:

i. **Throughput:** This is the total number of calls admitted into the network (both new calls and handoff calls) over a particular simulation time.

$$Throughput = \frac{AD_{calls}}{S_T} \tag{3.8}$$

Where AD_{calls} is the total number of admitted calls and S_T is the simulation time.

ii. **Call blocking probability:** This is the total number of new calls blocked i.e. calls that were not admitted into the network over the total number of new connection requests.

$$CBP = \frac{BL_{calls}}{NC_{requests}} \tag{3.9}$$

Where BL_{calls} is the total number of blocked new calls and $NC_{requests}$ is the total number of new connection requests.

iii. **Call dropping probability:** This is the total number of dropped calls i.e. these calls that were dropped while in progress or have not successfully being handed over from one cell to another over the total number of handoff call requests.

$$CDP = \frac{DP_{calls}}{HC_{requests}} \tag{3.10}$$

Where DP_{calls} is the total number of dropped handoff calls and $HC_{requests}$ is the total number of handoff requests.

CHAPTER FOUR

RESULTS AND DISCUSSIONS

4.1 Introduction

This section presents the simulation topology that was used to implement the proposed EA-CAC scheme. It also presents the results of the performance of the benchmark scheme and that of the proposed scheme. The results of the performance were presented based on the various performance metrics used which are throughput, CDP and CBP of both RT and NRT traffic. Finally, the summary of all results and discussions are also presented.

4.2 Simulation Topology

The simulation topology used to evaluate the performance of the two schemes i.e. the benchmark and proposed EA-CAC is shown in figure 4.1. The topology consists of one eNodeB, one application server and several UEs connected to the eNodeB for different simulation experiments. The server generates two traffics each from a different application. The two types of traffic are RT and NRT calls. An example of an RT call can be a live video streaming while an NRT can be an email. A call request can either be an RT or NRT while a call type can either be NC or HC.

Table 4.1 Simulation Parameters

Parameter	Value
System Bandwidth	5MHz
Number of RBs	25
TTI	1ms
Call Arrival	Poisson Process
Simulation period	1000s
Transmission scheme	2x2 MIMO, OLSM
Cyclic prefix used	Normal cyclic prefix
UE distribution	Uniform

The total bandwidth used for the simulation is 5MHz with 25 resource blocks (RBs) per slot of 12 subcarrier spacing. The simulation time used is 1000s while the results were obtained by taking the average over several times of simulation experiments. The simulation parameters used were adopted from Maharazu *et al* (2017) as shown in table 4.1. Different simulation experiments were conducted for 20, 40, 60, 80, 100 and 120 UEs. In each experiment, RT and NRT traffic are generated using a Poisson distribution. The experiment for both the benchmark and proposed EA-CAC was conducted using the same traffic parameters.

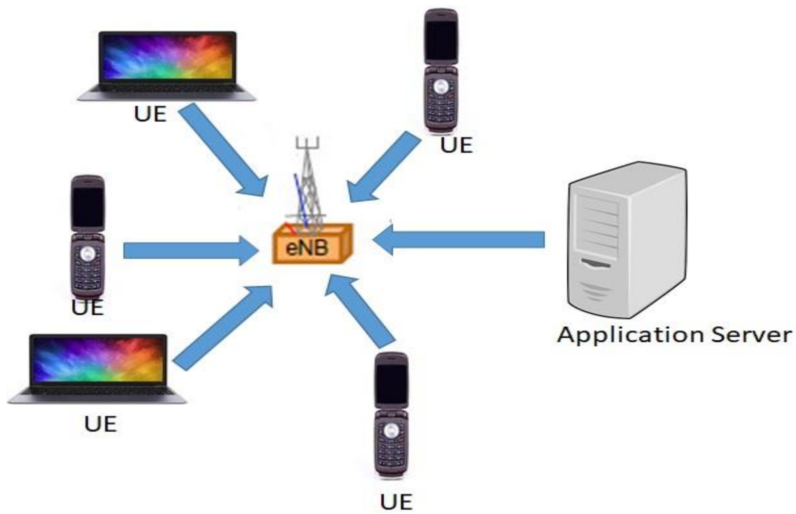

Figure 4.1: Simulation Experiment Topology

4.3 Results and Discussions

In this section, results obtained from the simulation experiments are used compared to the performance of the proposed EA-CAC scheme and the benchmark scheme is presented and discussed. The results are discussed using the performance metrics discussed in section 3.6.

4.3.1 Results of Throughput Achieved by the Schemes

Figure 4.2 illustrates the throughput achieved by the two schemes for RT calls. The figure demonstrates that the EA-CAC scheme increases the throughput of RT traffic compared to the benchmark scheme by admitting more RT calls. It can be observed that when the traffic intensity is low, both schemes perform well by admitting a reasonable number of calls. But when the traffic intensity increases, the EA-CAC scheme admits more RT calls than the benchmark scheme. The improved performance can be traced to the maximum bandwidth requirements that are allocated to all RT calls at the point of admission. It is also as a result of the fact that degradation is not applied to admitted RT calls unless if all admitted NRT calls have been degraded. The EA-CAC scheme increases the throughput of RT calls by 25.0% compared to the benchmark scheme.

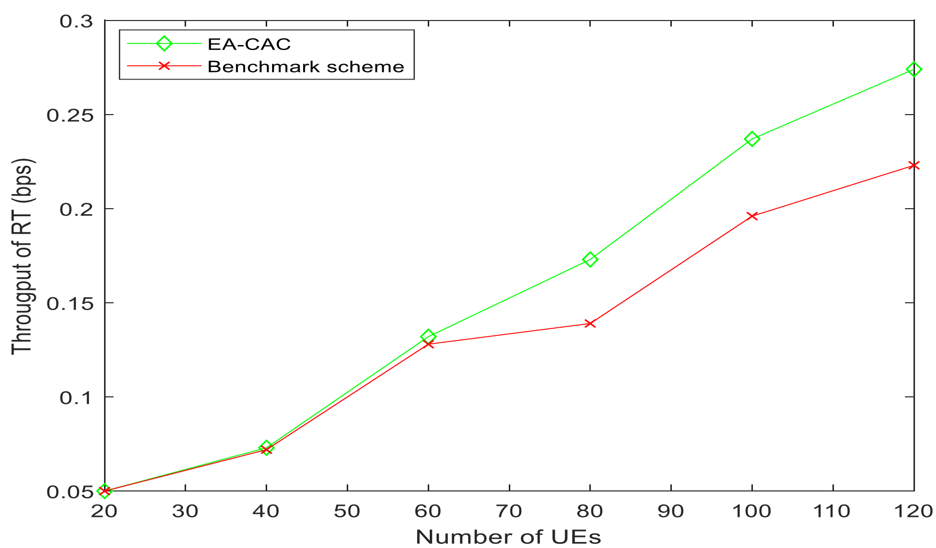

Figure 4.2: Throughput Achieved by the two Schemes for RT calls

Figure 4.3 demonstrates the throughput achieved by the two schemes for NRT calls. It can be seen that the benchmark scheme admits more calls when the traffic intensity is both low and high. The two schemes have almost the same throughput when the traffic intensity is low but the benchmark scheme admits more calls when the traffic intensity is high. This is because the benchmark scheme gave higher priority to NRT calls. The EA-CAC tries to maintain the throughput to avoid trading off the throughput of RT calls which are supposed to have higher priority than the NRT calls. Therefore, the difference of 2.7% exists between the benchmark scheme and EA-CAC.

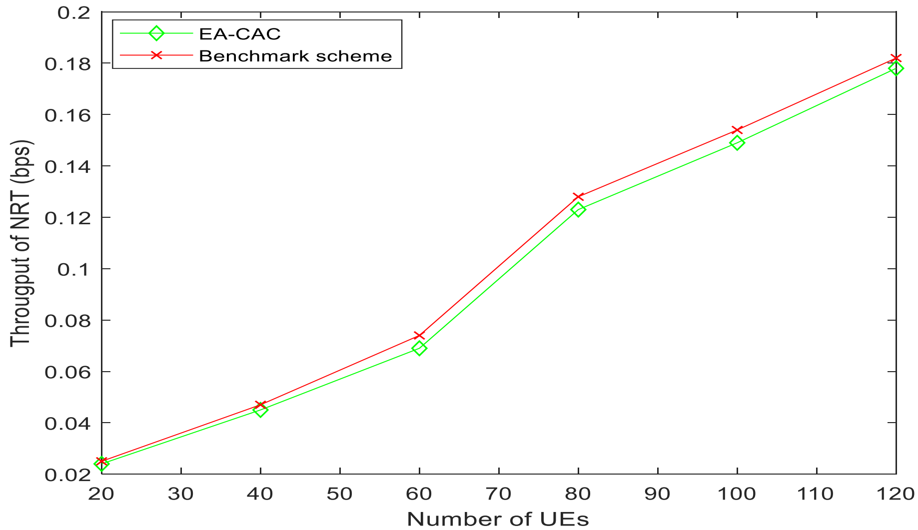

Figure 4.3: Throughput Achieved by the two Schemes for NRT calls

4.3.2 Results of CBP Achieved by the Schemes

Figure 4.4 shows the CBP achieved by the two schemes for RT calls. The figure shows that the EA-CAC scheme blocks fewer RT calls compared to the benchmark scheme. The two schemes have the same performance when the traffic intensity is low i.e. they both did not block any call. But when the traffic intensity is high, the EA-CAC scheme drops fewer RT calls compared to the benchmark scheme. This improvement is a result of the degradation procedure implemented in the EA-CAC scheme. Thus, EA-CAC reduces the dropping rate of RT calls in the benchmark scheme by 12.2%.

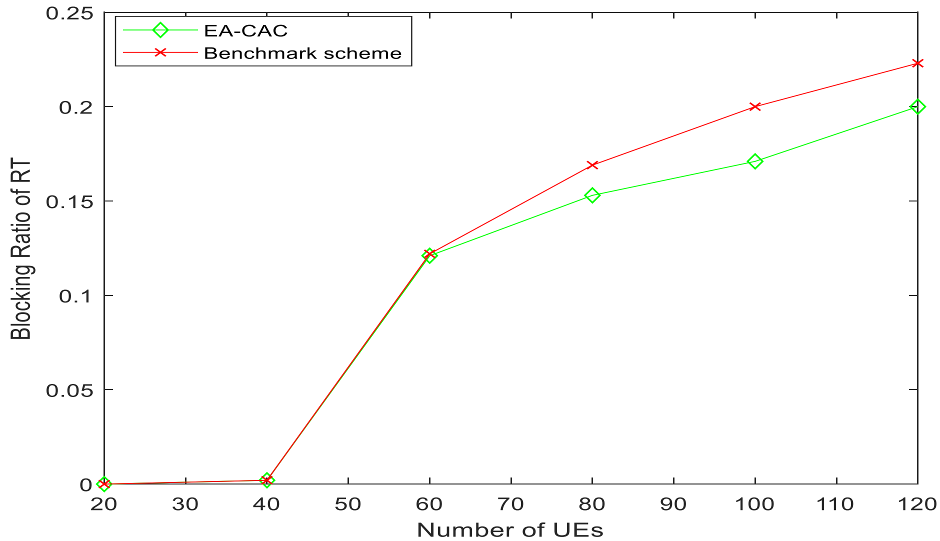

Figure 4.4: CBP Achieved by the two Schemes for RT calls.

Figure 4.5 demonstrates the CBP achieved by the two schemes for NRT calls. It can be seen that when the traffic intensity is low, the performance of the two schemes is the same i.e. no call is blocked. But when the traffic intensity is high, the EA-CAC blocks more NRT calls than the benchmark scheme. This is because the benchmark scheme admits more NRT calls than the EA-CAC, therefore the calls to be blocked will be lesser than that of the EA-CAC. The blocking ratio difference between the two schemes is 2.2% in favor of the benchmark scheme.

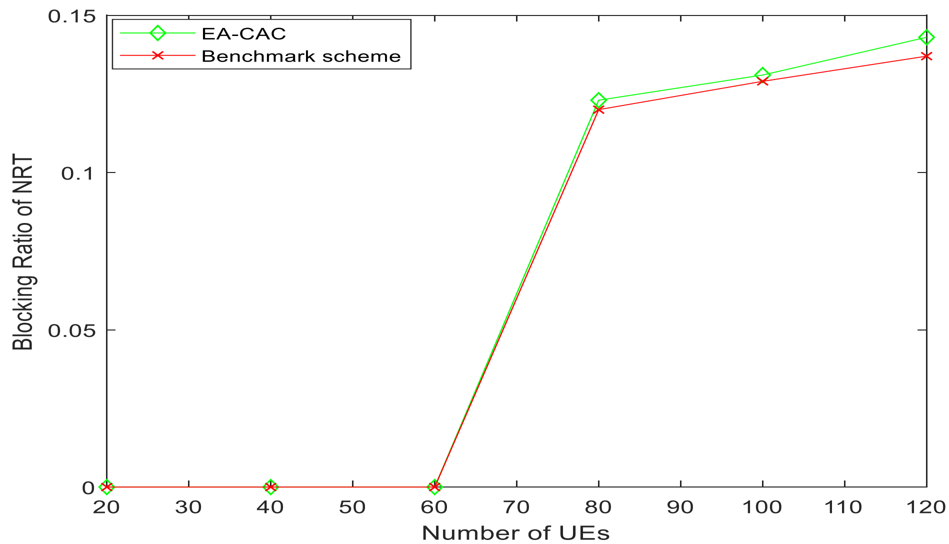

Figure 4.5 CBP Achieved by the two Schemes for NRT calls

4.3.3 Results of CDP Achieved by the Schemes.

Figure 4.6 illustrates the dropping ratio achieved by the two schemes for RT calls. The results reveal that when the traffic intensity is low, the performance of the two schemes is the same i.e. both schemes drop almost the same number of RT calls. But when the traffic intensity increases, the EA-CAC drops fewer calls than the benchmark scheme. This improvement is a result of the degradation approach applied to admitted NRT when there are insufficient resources to admit a new call request. The EA-CAC scheme reduces the dropping rate of RT calls by 15.2%

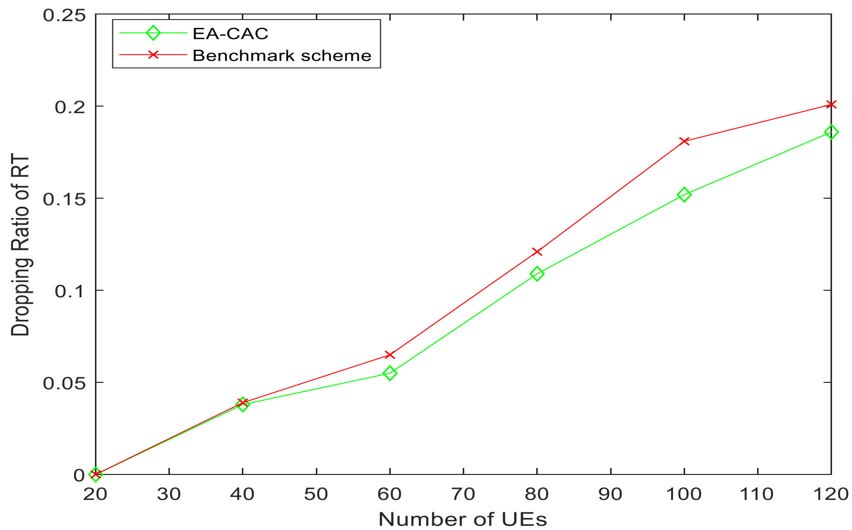

Figure 4.6: CDP Achieved by the two schemes for RT calls

Figure 4.7 shows the CDP achieved by the two schemes for NRT calls. The results show that when the traffic intensity is low, both schemes have the same performance. However, when the traffic intensity is high, the EA-CAC scheme drops more NRT calls than the benchmark scheme. This is as a result of NRT calls are delay tolerable, even when degraded, they can survive for a longer time in the system than the RT calls. The difference between the two schemes in terms of dropping ratio is 1.9% in favor of the benchmark scheme.

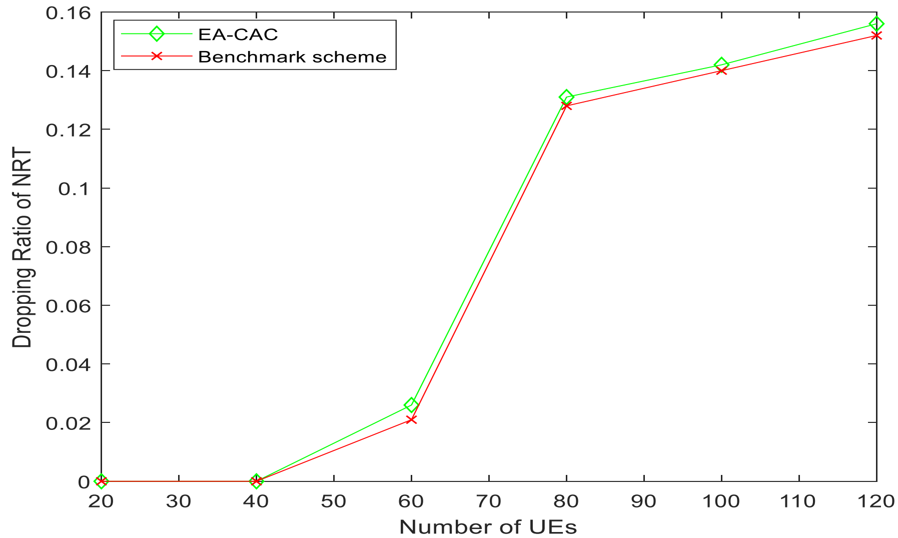

Figure 4.7: CDP Achieved by the two Schemes for NRT calls

4.4 Summary of Discussions

From the above discussion, it can be seen that the proposed EA-CAC scheme has a better performance in terms of throughput, CBP and CDP for real-time traffic. It also maintains the same throughput, CBP and CDP for the non-real time traffic.

The throughput of RT calls has been improved by 25.0% compared to the benchmark scheme, thereby admitting more RT calls. This improvement is as a result of the maximum bandwidth allocated to RT calls at the point of admission thereby reducing the delay of RT calls, once admitted with their maximum they will be serviced within their required time. The improvement can also be seen as a result that the EA-CAC scheme doesn't apply the degradation procedure on RT calls unless if the degraded bandwidth is not enough to admit the new call and the degradable bandwidth will be enough to admit the new call. It can also be seen that the EA-CAC scheme reduces both the dropping rate and blocking rate of RT calls compared to the benchmark scheme. This can be traced to the degradation procedure applied to the admitted NRT calls when there are insufficient resources to admit a new call.

This results in admitting more RT calls thereby reducing the number of blocked and dropped RT calls. The EA-CAC reduces both blocking and dropping rate of RT calls by 12.2% and 15.2% respectively.

The throughput of the benchmark scheme is higher than the throughput of EA-CAC by 2.7%. This can be traced to the fact that the benchmark scheme gave higher priority to NRT calls and applying the degradation procedure to RT traffic only when there are insufficient resources to admit a new call. The EA-CAC tries to maintain the same throughput to avoid falling back to the initial problem that was addressed by the benchmark scheme. The 2.7% difference can be regarded as an insignificant difference between the two schemes. Similarly, the benchmark schemes reduce the CBP and CDP of NRT calls by 2.2% and 1.9% respectively. This is also as a result of the higher priority given to NRT calls by the benchmark scheme. The 1.9% and 2.2% difference can be regarded as an insignificant difference between the two schemes.

Finally, it can be concluded that the EA-CAC performed better than the benchmark scheme because it increases the throughput and reduced the CBP and CDP of RT calls without sacrificing the performance of NRT calls.

CHAPTER FIVE

SUMMARY, CONCLUSION AND RECOMMENDATIONS

5.1 Introduction

This section presents summary of the entire research study by highlighting the key components of the work. Conclusion, recommendations and future works ware also presented in this section.

5.2 Summary

In this research study, we proposed a CAC scheme to improve resource utilization in LTE networks. The proposed EA-CAC scheme addresses some of the limitations of the existing CAC scheme. The benchmark scheme wastes bandwidth due to its failure to check whether the bandwidth to be degraded from admitted RT calls will be enough to admit the new call requests. It also increases the CBP and CDP of RT calls as a result of the delay incurred by the admitted RT calls when they are degraded to admit the new call request.

In this study proposed an Enhanced Adaptive Call Admission Control (EA-CAC) scheme with bandwidth reservation was proposed. The scheme introduces a prior-check mechanism that will ensure that the bandwidth to be degraded from admitted calls will be sufficient to admit the new call request. By this, the wastage of bandwidth is reduced as degradation is done after ensuring that the amount of bandwidth to be degraded will be enough to admit the new call requests. Further, the EA-CAC incorporates an adaptive degradation mechanism which reduces the delay incurred by admitted RT calls. It degradeds NRT calls first and then check if the amount of bandwidth to be degraded from RT calls will be sufficient to admit the requested calls, if it will be sufficient then it degrades the admitted

RT calls otherwise it will reject the calls. By this mechanism, the EA-CAC scheme reduces the delay incurred by RT calls.

The EA-CAC scheme was subjected to several simulation experiments that was performed with the help of the Vienna LTE system-level simulator. The performance of the EA-CAC and the benchmark scheme was evaluated in terms of throughput, call blocking and call dropping probabilities. The simulation results reveal that the EA-CAC performed better than the benchmark scheme in terms of throughput, CBP and CDP of both RT calls without sacrificing the performance of NRT calls.

5.3 Conclusion

In this research work, an EA-CAC scheme was proposed to improve the utilization of network resources, reduces the delay incurred by RT calls and also increase the CBP and CDP of both RT and NRT calls. The scheme admits both RT and NRT calls with their maximum requirements and then first degraded all admitted NRT calls when a call arrives and there are insufficient resources to admit the call. It then degraded all admitted RT after ensuring that the bandwidth to be degraded will be enough to admit the requested call.

The EA-CAC scheme allocates maximum bandwidth requirements to both RT and NRT at the point of admission. It accepts an NRT call request if the requested bandwidth is less than or equal to the available bandwidth in the system otherwise the call is rejected. The scheme accepts an RT call if the request if the requested bandwidth is less than or equal to the available bandwidth otherwise a degradation mechanism, is applied on all admitted calls. At the first stage of degradation, all admitted NRT calls are degraded to their minimum requirement. If the degraded bandwidth is enough then the requested calls are accepted otherwise the second stage of degradation is applied on all admitted RT calls. But before the degradation is applied, a pre-check mechanism is employed to check if the degradable bandwidth plus the available bandwidth will be enough to admit the new call, if yes then the degradation takes place otherwise the degradation will not occur and the call is rejected.

Simulation experiments were performed using the Vienna LTE system level simulator to evaluate the performance of the proposed scheme and the benchmark scheme. The simulation was conducted using different number of User Equipment for different experiments (20, 40, 60, 80, 100 and 120 UEs). The metrics; throughput, CBP and CDP were used to evaluate the performance of the proposed EA-CAC scheme against the benchmark scheme.

The simulation experiment results reveal that the proposed EA-CAC scheme demonstrates better performance in terms of the throughput, CBP and CDP of RT calls by 25.0%, 12.2% and 15.2% respectively. The improved performance was a result of the maximum bandwidth allocation to RT calls at the point of admission and also the EA-CAC scheme degrades NRT calls first before degrading RT calls. Therefore, the proposed EA-CAC scheme admits more RT calls into the network and as such less RT calls are blocked and dropped by the scheme. The proposed EA-CAC scheme improved the performance of RT calls without sacrificing the performance of NRT calls.

5.4 Recommendations

Based on the simulation experiment results obtained in this study, the following recommendations are made:

1. Telecommunication providers can implement the proposed EA-CAC scheme because the scheme considered the QoS classes of both real time and non-real time traffics without sacrificing the performance of one another.

2. To ensure QoS, the proposed EA-CAC can be implemented by different telecommunication providers for the scheme will improve the throughput and reduces both CBP and CDP of calls.

5.5 Future Works

The proposed EA-CAC scheme was designed to support both RT and NRT calls and also ensure that resources are effectively utilized. The possible ways the work can be extended include:

1. A bandwidth adaptation mechanism to be incorporated into the EA-CAC scheme. This will ensure the usage of all bandwidth that is released by calls that have been admitted and serviced by the system.
2. The EA-CAC should also be enhanced to consider channel quality as part of the admission criteria. Incorporating channel quality into the EA-CAC scheme will make the admission criteria more efficient than only considering available resources.

5.6 Publications from the work

This research work made some contributions to knowledge, one conference and two Journal articles have been extracted from this work. The articles are listed below:

1. S. O. Yese, A. Abdulazeez, A. Mohammed, ***M. M. Umar*** & Z. Y. Yeldu (2019): A Survey on Call Admission Control Schemes in LTE. ***International Journal of Computer Science & Engineering Survey, Vol. 10 (4/5), page 1-20.*** **DOI:10.5121/ijcses.2019.10501**

2. **M. M. Umar**, A. Mohammed, A. Roko, A.Y. Tambuwal & A. Abdulazeez (2019): QoS-Aware Call Admission Control (QA-CAC) scheme for LTE networks. ***Proceedings of the IEEE 15th International Conference on Electronics Computer and Computation (ICECCO 2019), page 1-5. DOI:*** 10.1109/ICECCO48375.2019.9043228**.**

3. **M. M. Umar,** A. Mohammed, A. Roko, A. Y. Tambuwal & A. Abdulazeez (2021): Enhanced Adaptive Call Admission scheme with bandwidth Reservation for LTE networks. ***International Journal of Mobile Computing and Multimedia Communications. Vol 12 (1), pp 23-42.*** *DOI:* 10.4018/IJMCMC.2021010102

References

3GPP (2010). Evolved Universal Terrestrial Radio Access (E-UTRA) and Evolved Universal Terrestrial Radio Access Network (E-UTRAN) (Release 10); Overall description stage 2, 3rd Generation Partnership Project (3GPP), TS 36.300, v.10.2.0.

3GPP (2013). Evolved Universal Terrestrial Radio Access Network (E-UTRAN); Architecture Description, Release 11, September 2013. 3GPP TS 36.401

Ali K.B, Fauzi Z and Lotfi K (2010). Reducing handoff dropping probability in 3GPP LTE Network. *International Conference on Communication and Networks (ComNet),* London, 1-8.

Al-Qahtani S (2017). Users' classification-based call admission control with adaptive resources reservation for LTE-A networks. *Journal of King Saud University-Computer and Information Science. 29* (1), 103-115

Angelos A, Elli K, Luis A and Christos V (2011). Dealing with VoIP Calls during "Busy Hour" in LTE. In Jia-Chin Lin, IntechOpen (Ed.). *Recent Advances in Wireless Communications and Networks.* 343-360

Ashish K, Ankit A and Lalit S (2013). 4G Wireless Technology: A Brief review. *International Journal of Engineering and Management Research, 3* (2), 35-43.

Ayaz A.S, Chowdhry B. S, Baloch, A. K and Pathan, A. H (2006). Radio Resource Management Strategies in 3G UMTS Network. *2nd conference on next generation internet design and engineering,* Valencia, 24-36

Ayesha H. A. and Mohsin N. (2017). Radio Resource Management with QoS guarantees for LTE-A system: a review focused on employing the multi-object optimization techniques. *Telecommunication System, Springer Science + Business media, 67* (2), 349-365.

Belghith A, Turki N, Cousin B and Obaidat M.O (2016a): Felxible Call Admission Control with Preemption in LTE networks. *IEEE ICC Mobile and Wireless Networking Symposium.* Kuala Lumpur, Malaysia, 4799-4806

Belghith A, Turki N, Cousin B and Obaidat M.O (2016b): Efficient Bandwidth Call Admission Control in 3GPP LTE networks. *IEEE Global Communications Conference (GLOBECOM),* Washington DC, 1251-1258.

Chadchan S.M and Akki C.B (2011): Priority-Scaled Preemption of Radio Resources for 3GPP LTE Networks. *International Journal of Computer Thoery & Enineering 3* (36) 1-7

Charles T, Joel S, Paul B, Dongchan K, Ananya D, Roberto S, and Mark C, (2012). A Passive Measurement System for Network Testbeds. In Korakis T., Zink M., Ott M. (eds). *Testbeds and Research Infrastructure. Development of Networks and Communities. Springer,* (pp. 1-16) Berlin, Heidelberg.

Daniel E. A, Edem E.W and Enoch O.N (2014): A survey of Call Admission Control Schemes in wireless cellular networks. *International Journal of Scientific & Engineering Research*, *5* (2), 111-120.

Faouzi Z, Khitem B. A, Mohammed S.O and Lotfi K (2012). Adaptive call admission control in 3GPP networks. *International Journal of Communication Systems*, *27* (10) 1522-1534.

Fatima F, Doan B.H, Iain B.C (2014). LTE-Advanced fair intelligent admission control LTE-FIAC. *IEEE International Symposium on a World of Wireless, Mobile and Multimedia Networks 2014,* Sydney, NSW, Australia, 1-4

Jie W. and Yangfan Q. (2013). A new call admission control strategy for LTE Femtocell Networks. *2nd International Conference on Advances in Computer Science and Engineering.* China, 334-338

Josep C.I, Martin W and Markus R (2010). System level simulation of LTE networks. *IEEE 71st Vehicular Conference,* Taiwan, 50-54.

Kandaraj P, Adlen K, Jean M. B, and Ceser V (2011). Radio resource management in emerging heterogeneous wireless networks. *Computer Communications*, *34* (2011), 1066-1076.

Khabazian M, Kubbar O, and Hassanein H (2012): Call Admission Control with Resource Reservation for Multi-service OFDM Networks. *IEEE International Conference on Computing, Networking and Communications, (ICNC' 12),* Maui, Hawaii 781–785.

Lei H, Yu M, Zhao A, Chang Y, and Yang D (2008): Adaptive Connection Admission Control Algorithm for LTE Systems. *IEEE Vehicular Technology Conference*, Canada, 2336–2340.

Maharazu M, Zurina M.H, Azizol A and Abdullah M (2017): An Adaptive Call Admission Control with Bandwidth Reservation for Downlink LTE Networks. *IEEE Access 5,* 10986-10994.

Mamman M, Zurina M.H, Azizol A and Abdullah M (2018). Call Admission Control for Real-Time and Non-real-time traffic for Vehicular LTE downlink networks. *Lecture Notes in Electrical Engineering. (Mobile and Wireless Technologies 2017)* 46-53.

Maniru, M. U, Aminu M, Abubakar R, Ahmed Y.T & Abdulhakeem A (2019). QoS-aware call admission control (QA-CAC) scheme for LTE networks. Paper presented at the *15th International Conference on Electronics Computer and Computation, Abuja,* Nigeria.

Maniru, M. U, Aminu M, Abubakar R, Ahmed Y.T & Abdulhakeem A (2021). Enhanced Adaptive Call Admission Control Scheme with Bandwidth Reservation for LTE networks. *International Journal of Mobile Computing and Multimedia Communications, 12(1), 23-43.*

Manli Q, Yi H, Jinglin S, Yao Y, Lin T and Eryk D (2009). A novel radio admission control scheme for multiclass services in LTE systems. *IEEE 2009 Global Telecommunications conference*. Honolulu, 1-6

Martin T, Thomas B, Thomas K, Martin K. M, Stefan S and Markus R (2015). Runtime Precoding: Enabling Multipoint Transmission in LTE-Advanced System-Level Simulations. *IEEE Access, 3*, 725-736.

Mohamed H. A (2005). Call Admission control in wireless networks: A comprehensive survey. *IEEE Communications Survey & Tutorials, 7* (1) 50-69.

Navita M and Amandeep M (2016). Performance analysis of OFDMA, MIMO and SC-FDMA technology in 4G LTE networks. *6th International Conference of Cloud System and Big Data Engineering,* Noida, 554-558.

Obaidat M.S and Green D.B (2003). Simulation of Wireless Networks. In: Obaidat M.S, Papadimitriou G.I (eds.) *Applied System Simulation*, *Springer Science, (pp 115-153)* Boston, MA.

Ramraj R, Habibi D and Ahmad I (2014): Call Admission Control in 3GPP LTE Systems at High Vehicular Communications. *International Journal of Scientific & Engineering Research 5* (3) 1146–1153.

Raymond K, Rob A, Riccrado T and Mitsuhiro K (2010). On Pre-emption and Congestion control for LTE Systems. *IEEE 72nd Vehicular Technology Conference,* Ottawa, 1-5.

Sanford F, Alan M and Rick S (2015). Integrating SysML into a system development environment. In *A practical guide to SysML*, 3rd (ed.), (pp. 507-541). Elsevier Inc.

Senpaka P.V and Franklin V.J (2012). Extensive DBA-CAC Mechanism for Maximizing Efficiency in 3GPP : LTE Networks. *International Conference on Recent Advances in Computing and Software Systems(RACSS 12)*, Cyprus, 233–237.

Taranetz M, Blazek T, Kropfreiter T, Muller M.K, Schwarz S and Rupp M (2015). Runtime Precoding: Enabling Multipoint Transmission in LTE-Advanced System-level simulations. *IEEE Access 3,* 725-736.

Appendices

Source code for the new modules incorporated into the Vienna LTE system-level simulator.

//CallS Generation module

```
classdef Packet
        properties
%       Instance variables
        packet_size=0;
        packet_type='';
        packet_allocated_bandwidth=0;
        packet_arrival_time=0;
        packet_service_time=0;
        packet_waiting_time=0;
        packet_burst_time=0;
        packet_deadline=0;
        packet_status='';
        packet_completed_time=0;

    end

    methods
        function obj = Packet(size,type,arrivalTime,deadLine)
            %UNTITLED Construct an instance of this Packet class
            %bandwidth,arrivalTime,serviceTime,waitingTme
            obj.packet_size =size;
            obj.packet_type =type;
            obj.packet_arrival_time=arrivalTime;
            obj.packet_deadline=deadLine;
            obj.packet_burst_time=0;
            obj.packet_waiting_time=0;
            obj. packet_service_time=0;
            obj.packet_allocated_bandwidth=0;
            obj.packet_status='on';
            obj.packet_completed_time=0;

        end

         function obj= set.packet_size(obj,pSize)
        % Assign new type to this packet
             obj.packet_size=pSize;
         end

          function obj= set.packet_status(obj,pstatus)
        % Assign new status to this packet
             obj.packet_status=pstatus;
          end

          function obj= set.packet_completed_time(obj,pCTime)
        % Assign new status to this packet
             obj.packet_completed_time=pCTime;
        end
```

```
function obj= set.packet_type(obj,pType)
% Assign new type to this packet
     obj.packet_type=pType;
end


function obj= set.packet_allocated_bandwidth(obj,pBandwidth)
% Assign new bandwidth to this packet
     obj.packet_allocated_bandwidth=pBandwidth;
end

 function obj= set.packet_arrival_time(obj,pArrivalTime)
% Assign new arrival time to this packet
     obj.packet_arrival_time=pArrivalTime;
 end

  function obj= set.packet_service_time(obj,pServiceTime)
% Assign new arrival time to this packet
     obj.packet_service_time=pServiceTime;
 end

  function obj= set.packet_waiting_time(obj,pWaitingTime)
% Assign new waiting time to this packet
     obj.packet_waiting_time=pWaitingTime;
  end

  function obj= set.packet_burst_time(obj,pBurstTime)
      % Assign new waiting time to this packet
      obj.packet_burst_time=pBurstTime;
  end

  function obj= set.packet_deadline(obj,pDeadline)
      % Assign new waiting time to this packet
      obj.packet_deadline=pDeadline;
  end

function obj= get.packet_size(obj)
% Get the size of this packetcl
     obj=obj.packet_size;
end

function obj= get.packet_type(obj)
% Get the size of this packetcl
     obj=obj.packet_type;
end

function obj= get.packet_allocated_bandwidth(obj)
% Get the size of this packetcl
     obj=obj.packet_allocated_bandwidth;
end

function obj= get.packet_arrival_time(obj)
% Get the size of this packetcl
     obj=obj.packet_arrival_time;
end

function obj= get.packet_service_time(obj)
```

```
        % Get the size of this packetcl
             obj=obj.packet_service_time;
        end

        function obj= get.packet_waiting_time(obj)
        % Get the size of this packetcl
             obj=obj.packet_waiting_time;
        end

        function obj= get.packet_burst_time(obj)
        % Get the size of this packetcl
             obj=obj.packet_burst_time;
        end

        function obj= get.packet_deadline(obj)
        % Get the size of this packetcl
             obj=obj.packet_deadline;
        end


         function obj= get.packet_status(obj)
        % Get the size of this packetcl
             obj=obj.packet_status;
        end

        function obj= get.packet_completed_time(obj)
        % Get the size of this packetcl
             obj=obj.packet_completed_time;
        end

    end
end
```

//CallS Classification and Prioritization module

```
classdef LTE_Traffic_Model

       methods (Static)
        function   generatedPackets   =   Generate_Traffic_Model(noUE,pRT,
   pNRT,simTime,rtType,nrtType)

          %LTE_TRAFFIC_MODEL Generate Traffice for simulation
            if (pRT + pNRT)==100
                exp_packs_simTime=noUE*0.02*simTime;
                numPack = exp_packs_simTime/4;

                r1=randi([1 2500],1,numPack);
                r2=randi([2501 5000],1,numPack);
                r3=randi([5001 7500],1,numPack);
                r4=randi([7501 10000],1,numPack);

                nRT=(pRT/100)*exp_packs_simTime;

                nNRT=(pNRT/100)*exp_packs_simTime;
```

```
            packetsAT=horzcat(r1,r2,r3,r4);
            numPackets=numel(packetsAT);

            for k=1:numPackets
                if mod(k,2)==0
                    if nRT>0

                    dLine=packetsAT(k)+100; %arrival plus latency

generatedPackets(k)=Packet(1054,'RT',packetsAT(k),dLine); %generate rt
packet
                    elseif nNRT>0

                    dLine=packetsAT(k)+100; %arrival plus latency

generatedPackets(k)=Packet(1054,'NRT',packetsAT(k),dLine);    %generate
nrt packet

                    end

                else
                    if nNRT>0

                    dLine=packetsAT(k)+100; %arrival plus latency

generatedPackets(k)=Packet(1054,'NRT',packetsAT(k),dLine);    %generate
nrt packet
                   elseif nRT>0

                    dLine=packetsAT(k)+100; %arrival plus latency

generatedPackets(k)=Packet(1054,'RT',packetsAT(k),dLine); %generate rt
packet

                    end
                end
            end

        else
            disp('Sum of percentages must be 100');
        end
    end

    function simPac=emptyQueue(simPackets)

        n=numel(simPackets);
        for c=1:n
            if numel(simPackets)>0
            simPackets(1)=[];
            end
        end

    simPac=simPackets;
    end

    function adaptedBw=adaptBw(simPackets,cSimTime)
        n=numel(simPackets);
```

```
            adBw=0;
            if n>0
            for k=1:n
                cStatus=simPackets(k).packet_status;
                cCompletedTime=simPackets(k).packet_completed_time;

                if cSimTime<= cCompletedTime && strcmp(cStatus,'on')==1
                        pbw=simPackets.packet_allocated_bandwidth;
                        adBw=adBw+pbw;

                        simPackets(k).packet_status='Compeleted';
                end

            end
            end
            adaptedBw=adBw;

        end
    end
end

function packk=Prioritize_AT(Packet)
     packk=Packet;

    n=numel(Packet);
    ind=n;

    for i=0:n-1
        m=numel(Packet);
        % disp(m);
         maxInd=1;

            for k=1:m
                a=Packet(maxInd).packet_arrival_time;
                b=Packet(k).packet_arrival_time;
                if a<b
                    maxInd=k;
                end

            end

       %disp(maxInd);
        packk(ind)=Packet(maxInd);
        ind=ind-1;
        Packet(maxInd).packet_arrival_time=-1;

    end
%      disp(numel(packk));

end

//CAC procedure module
%ECAC
for w=1:5

%     numberOfUEs=numUEs(w);
```

```
     dg=0;
     totalSimulationTime_E=10000; %in mili seconds
     availlableBandwidth_E=360000;

    admittedRT_Connections_E(1)=Packet(9,'D',4,6);
    admittedNRT_Connections_E(1)=Packet(9,'D',4,6);

    %blockedRT_Connections;
    %blockedNRT_Connection;

    droppedRT_Connections_E(1)=Packet(9,'D',4,6);
    droppedNRT_Connections_E(1)=Packet(9,'D',4,6);

    simTraffic_E=[];
    %Generate simulation traffics
    disp('Generating Simulation Traffic> > > > > > > > > > > > > > >');
%
   simTraffic=LTE_Traffic_Model.Generate_Traffic_Model(numberOfUEs,percen
   tageOfRT_Connections,percentateOfNRT_Connections,totalSimulationTime,'
   VoIP','FTP');
    if w==1
        simTraffic_E=simT1;
    elseif w==2
        simTraffic_E=simT2;
    elseif w==3
        simTraffic_E=simT3;
    elseif w==4
        simTraffic_E=simT4;
    else
        simTraffic_E=simT5;
    end

    disp(numel(simTraffic_E));
    % disp(numel(simTraffic));

    simTraffic_E=Prioritize_AT(simTraffic_E);
    disp(numel(simTraffic_E));

    % disp(simTraffic(5).packet_type);

    %start CAC procedure
    disp('Starting Simulation > > > > > > > > > > > > > > > >');
    admittedRT_Connections_E(1)=[];
    admittedNRT_Connections_E(1)=[];
    droppedRT_Connections_E(1)=[];
    droppedNRT_Connections_E(1)=[];

    for cSimTime_E=1:totalSimulationTime_E
% % %         adRTBW=0; adNRTBW=0;
%                                  %                                  %
   adRTBW=LTE_Traffic_Model.adaptBw(admittedRT_Connections_E,cSimTime_E);
%                                  %                                  %
   adNRTBW=LTE_Traffic_Model.adaptBw(admittedNRT_Connections_E,cSimTime_E
   );
% % %
% % %          sAdb=adRTBW+adNRTBW;
% % %          availlableBandwidth_E=availlableBandwidth_E+sAdb;
```

```
    if numel(simTraffic_E)>0
        nextConnection_E=simTraffic_E(1);
        pType_E=nextConnection_E.packet_type;
        arrTime_E=nextConnection_E.packet_arrival_time;

        if cSimTime_E>=arrTime_E

                  %re-try admission of RT
                  if dg==0
                      br=300;
                  else
                      br=200;
                  end
            simTraffic_E(1)=[];

            if strcmp(pType_E,'RT')==1 %Start of RT Admission
                if availlableBandwidth_E>=br
                    disp(sprintf('Allocated RT DBW= %d',br));
                   %Admit RT Connection
                   x_E=numel(admittedRT_Connections_E)+1;
                   nextConnection_E.packet_allocated_bandwidth=br;
                   psize=nextConnection_E.packet_size;
                   pBurstTime=psize/br;

nextConnection_E.packet_completed_time=pBurstTime+cSimTime_E;
                   admittedRT_Connections_E(x_E)=nextConnection_E;

                   availlableBandwidth_E=availlableBandwidth_E-br;
                   nextConnection_E(1)=[];

                else

                  % Degradation for RT
                  dgrRT=0; dgrNRT=0;  %allBW_E1=0; nn=0; mme=0;
                  nn=numel(admittedRT_Connections_E);
                  mme=numel(admittedNRT_Connections_E);
                  if nn>0
                      for e=1:nn

allBW_E1=admittedRT_Connections_E(e).packet_allocated_bandwidth;
                           if allBW_E1 > 200
                               dgrRT=dgrRT+ (allBW_E1-200);

admittedRT_Connections_E(e).packet_allocated_bandwidth=200;
                           end

                      end
                  end

                  if mme>0
                      allBW_E2=0;
                      for q=1:mme

allBW_E2=admittedNRT_Connections_E(q).packet_allocated_bandwidth;
                           if allBW_E2 > 200
                               dgrNRT=dgrNRT+ (allBW_E2-200);
```

```
admittedNRT_Connections_E(q).packet_allocated_bandwidth=200;
                             end

                        end

                    end

                    vb=0;
                   vb= dgrNRT+dgrRT;
                    availlableBandwidth_E=availlableBandwidth_E+vb;
                     disp('Degradation occured');
                     disp(vb);
                    dg=1;
                    %re-try admission of RT
                     if availlableBandwidth_E>=200

                             p=numel(admittedRT_Connections_E)+1;

nextConnection_E.packet_allocated_bandwidth=200;

                             psize=nextConnection_E.packet_size;
                             pBurstTime=psize/br;

nextConnection_E.packet_completed_time=pBurstTime+cSimTime_E;


admittedRT_Connections_E(p)=nextConnection_E;


availlableBandwidth_E=availlableBandwidth_E-200;
                              nextConnection_E(1)=[];
                     else
                             h=numel(droppedRT_Connections_E)+1;

droppedRT_Connections_E(h)=nextConnection_E;
                              nextConnection_E(1)=[];

                     end % end of degration



                 end % end of RT admission



             else % End of RT Admission
                  if dg==0
                        br2=300;
                    else
                        br2=200;
                    end
                 if availlableBandwidth_E>=br2
                     disp(sprintf('Allocated NRT DBW= %d',br2));
                     %Admit NRT Connection
                     y=numel(admittedNRT_Connections_E)+1;
                     nextConnection_E.packet_allocated_bandwidth=br2;
```

```
                     psize=nextConnection_E.packet_size;
                     pBurstTime=psize/br2;

nextConnection_E.packet_completed_time=pBurstTime+cSimTime_E;

                     admittedNRT_Connections_E(y)=nextConnection_E;

                     availlableBandwidth_E=availlableBandwidth_E-br2;
                      nextConnection_E(1)=[];
                 else  % Perform degradation proedure


                    dgrRT2=0; dgrNRT2=0;allBW_E1x=0; nn2=0; mm2=0;
                    nn2=numel(admittedRT_Connections_E);
                    mm2=numel(admittedNRT_Connections_E);
                    if nn2>0
                        for u=1:nn2

allBW_E1x=admittedRT_Connections_E(u).packet_allocated_bandwidth;
                             if allBW_E1x > 200
                                 dgrRT2=dgrRT2+ (allBW_E1x-200);

admittedRT_Connections_E(u).packet_allocated_bandwidth=200;
                             end

                        end
                    end


                    if mm2>0
                        allBW_E2x=0;
                        for q=1:mm2

allBW_E2x=admittedNRT_Connections_E(q).packet_allocated_bandwidth;
                             if allBW_E2x > 200
                                 dgrNRT2=dgrNRT2+ (allBW_E2x-200);

admittedNRT_Connections_E(q).packet_allocated_bandwidth=200;
                             end

                        end

                    end


availlableBandwidth_E=availlableBandwidth_E+(dgrNRT2+dgrRT2);
                    dg=1;
                    %re-try admission of NRT
                     if availlableBandwidth_E>=200
                             disp('Degradation occured');
                             y=numel(admittedNRT_Connections_E)+1;

nextConnection_E.packet_allocated_bandwidth=200;
                             psize=nextConnection_E.packet_size;
                             pBurstTime=psize/br;
```

```
   nextConnection_E.packet_completed_time=pBurstTime+cSimTime_E;


   admittedNRT_Connections_E(y)=nextConnection_E;


   availlableBandwidth_E=availlableBandwidth_E-200;
                              nextConnection_E(1)=[];
                     else
                             dn=numel(droppedNRT_Connections_E)+1;

   droppedNRT_Connections_E(dn)=nextConnection_E;
                              nextConnection_E(1)=[];
                     end % end of degration
                 end

             end
         end
     end
 end
arrayAdmittedRT_E(w)=numel(admittedRT_Connections_E);
arrayAdmittedNRT_E(w)=numel(admittedNRT_Connections_E);

arrayDroppedRT_E(w)=numel(droppedRT_Connections_E);
arrayDroppedNRT_E(w)=numel(droppedNRT_Connections_E);

disp(numel(arrayAdmittedRT_E));
disp(numel(arrayAdmittedNRT_E));
disp(numel(arrayDroppedRT_E));
disp(numel(arrayDroppedNRT_E));

disp(sprintf('%d ECAC---------------------------------------------',w));
disp(sprintf('Admitted RT=%d',numel(admittedRT_Connections_E)));
disp(sprintf('Admitted NRT=%d',numel(admittedNRT_Connections_E)));
disp(sprintf('Dropped RT=%d',numel(droppedRT_Connections_E)));
disp(sprintf('Dropped NRT=%d',numel(droppedNRT_Connections_E)));
disp(sprintf('Queued Packets=%d',numel(simTraffic_E)));

admittedRT_Connections_E=LTE_Traffic_Model.emptyQueue(admittedRT_Connecti
   ons_E);
admittedNRT_Connections_E=LTE_Traffic_Model.emptyQueue(admittedNRT_Connec
   tions_E);
droppedRT_Connections_E=LTE_Traffic_Model.emptyQueue(droppedRT_Connection
   s_E);
droppedNRT_Connections_E=LTE_Traffic_Model.emptyQueue(droppedNRT_Connecti
   ons_E);
 totalSimulationTime_E=0; %in mili seconds
    availlableBandwidth_E=0;
    br=0;
    dg=0;
end
```