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UNIVERSITI TEKNOLOGI PETRONAS

IMPROVING QoS OF VoWLAN VIA CROSS-LAYER BASED ADAPTIVE APPROACH.

by

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SAMEHA AHMED ALI ALSHAKHSI

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DEDICATION

With deepest love and gratitude to my late father, may the mercy of Allah be upon him, my beloved mother, my sisters and my brothers for their endless love, care, motivation and encouragement

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ABSTRACT

Voice over Internet Protocol (VoIP) is a technology that allows the transmission of voice packets over Internet Protocol (IP). Recently, the integration of VoIP and Wireless Local Area Network (WLAN), and known as Voice over WLAN (VoWLAN), has become popular driven by the mobility requirements of users, as well as by factor of its tangible cost effectiveness. However, WLAN network architecture was primarily designed to support the transmission of data, and not for voice traffic, which makes it lack of providing the stringent Quality of Service (QoS) for VoIP applications. On the other hand, WLAN operates based on IEEE 802.11 standards that support Link Adaptive (LA) technique. However, LA leads to having a network with multi-rate transmissions that causes network bandwidth variation, which hence degrades the voice quality. Therefore, it is important to develop an algorithm that would be able to overcome the negative effect of the multi-rate issue on VoIP quality. Hence, the main goal of this research work is to develop an agent that utilizes IP protocols by applying a Cross-Layering approach to eliminate the above-mentioned negative effect. This could be expected from the interaction between Medium Access Control (MAC) layer and Application layer, where the proposed agent adapts the voice packet size at the Application layer according to the change of MAC transmission data rate to avoid network congestion from happening. The agent also monitors the quality of conversations from the periodically generated Real Time Control Protocol (RTCP) reports. If voice quality degradation is detected, then the agent performs further rate adaptation to improve the quality. The agent performance has been evaluated by carrying out an extensive series of simulation using OPNET Modeler. The obtained results of different performance parameters are presented, comparing the performance of VoWLAN that used the proposed agent to that of the standard network without agent. The results of all measured quality parameters have proved that the agent has improved the network performance as it changed the

network state from congested to uncongested. It is an evident that adapting the voice packet size to cope with the LA problem has enhanced the performance of VoIP. That is because the voice packet size plays a key role in changing the required network bandwidth for voice transmission, particularly the bandwidth of the overhead that is attached with each transmitted voice packet. Hence, this research work showed the importance of packet size adaptation scheme and how it can be utilized to address the multi-rate issue and enhance the VoIP QoS.

ABSTRAK

Voice over Internet Protocol (VoIP) adalah suatu teknologi generasi masa hadapan yang membenarkan penghantaran paket-paket suara melalui protokol Internet (IP). Baru-baru ini, penyatuan di antara VoIP dan rangakaian setempat tanpa wayar (WLAN), dan dikenali sebagai *Voice over WLAN* (VoWLAN), telah menjadi popular yang dipacu oleh keperluan pengguna untuk bergerak, dan juga disebabkan oleh faktor keberkesanan kos yang ketara daripadanya. Bagaimanapun, asalnya arkitektur WLAN telah direkabentuk untuk menyokong penghantaran data dan bukan untuk trafik suara, yang mana menjadikan ianya tidak berupaya untuk menyediakan *Quality of Service* (QoS) yang ketat untuk aplikasi-aplikasi VoIP. Di pihak yang lain, WLAN beroperasi berdasarkan kepada piawaian IEEE 802.11 yang menyokong teknik *Link Adapative* (LA). Bagaimanapun, teknik LA menuju ke arah di mana rangkaian akan mempunyai penghantaran pelbagai-kadar yang akan menyebabkan kualiti suara menurun. Oeh itu, adalah penting untuk membangunkan suatu algoritma yang membolehkan kesan negatif dari isu pelbagai-kadar ke atas kualiti VoIP diatasi. Justeru, matlamat utama kerja kajiselidik ini adalah untuk membangunkan suatu agen yang mempergunakan protokol-protokol IP dengan menggunakan pendekatan *Cross-Layering* untuk menghilangkan kesan negatif yang dinyatakan. Ini boleh di jangkakan daripada interakasi yang berlaku di antara lapisan *Medium Access Control* (MAC) dan lapisan *Application,* dimana agen yang dicadangkan akan menyesuaikan saiz paket suara pada lapisan *Appication* menuruti perubahan pada kadar penghantaran data MAC agar kesesakan rangkaian dapat dielakkan daripada terjadi. Juga, agen akan mengawasi kualiti perbualan daripada laporan berkala *Real Time Control Protocol* (RTCP) yang dijanakan. Jika kualiti suara dikesan menurun, agen kemudiannya akan melakukan penyesuaian kadar, yang seterusnya memperbaikki kualiti. Prestasi agen telah dinilaikan dengan melakukan suatu siri simulasi yang ektensif dengan menggunakan permodelan OPNET. Keputusan-keputusan yang dihasilkan daripada

beberapa parameter berbeza telah dibentangkan, membandingkan prestasi VoWLAN menggunakan agen yang dicadangkan dengan rangkaian piawai tanpa agen. Keputusan-keputusan dari kesemua parameter kualiti yang diukur telah menunjukkan bahawa agen telah memperbaikki prestasi rangkaian, kerana ianya telah menukar keadaan rangkaian daripada kesesakan kepada tiada-kesesakan. Ianya adalah suatu bukti bahawa dengan menyesuaikan saiz paket suara untuk menyelesaikan masalah LA telah meningkatkan prestasi VoIP. Ini adalah kerana saiz paket suara memainkan peranan penting untuk mengubah/menukar lebarjalur rangkaian yang diperlukan untuk penghantaran suara, terutamanya lebarjalur untuk *overhead* yang disertakan ke atas setiap paket suara yang dihantar. Oleh itu, kerja kajiselidik ini telah menunjukkan kepentingan skim penyesuaian saiz paket dan bagaimana ianya telah dipergunakan untuk menyelesaikan isu pelbagai-kadar dan meningkatkan QoS VoIP.

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LIST OF ABBREVIATIONS

AC	Access Categories
ACK	Acknowledgement
AP	Access Point
ARF	Auto Rate Fallback
ASNR	Adaptive source-network rate control scheme
BC	Back-off Counter
BER	Bit Error Rate
BSS	Basic Service Set
CAA	Codec Adaptation Algorithm
CAC	Call Admission Control
CCK	Complementary Code Keying
CLAA	Cross-Layering Adaptive Agent
CNAME	Canonical Name
CODEC	Coder/Decoder
CSMA/CA	Carrier Sense Multiple Access with Collision Avoidance
CTS	Clear to Send
Curr_Tx_R	Current Transmission Rate
CW	Contention Window
DBPSK	Differential Binary PSK
DCF	Distributed Coordination Function
DIFS	DCF Inter-Frame Spacing
DQPSK	Differentially-encoded Quadrature PSK
DSP	Digital Signal Processor
DSSS	Direct Sequence Spread Spectrum
EDCF	Enhanced DCF
ESS	Extended Service Set
ETSI	European Telecommunications Standard Institute
E2E delay	End to End delay
FEC	Forward Error Correction
FHSS	Frequency Hopping Spread Spectrum

FS	Frame Size
FSMs	Finite State Machines
H	Packet header size
HCF	Hybrid Coordination Function
HTTP	Hypertext Transfer Protocol
Hz	Hertz
IETF	Internet Engineering Task Force
IP	Internet Protocol
IR	Infrared
ISP	Internet Service Providers
ITU-T	International Telecommunication Union-Telecommunication
LACAM	Link Adaptation Codec Adaptation Mechanism
LA	Link Adaptive
MAC	Medium Access Control
MGW	Media gateway
MRV	Multi-Rate VoIP
MOS	Mean Opinion Score
N	Number of frames
NAV	Network Allocation Vector
NB	Network Bandwidth
NoB	Number of Beacon frames
OPNET	Optimized Network Engineering Tool
PCF	Point Coordination Function
PCM	Pulse Code Modulation
PHY	Physical
PLC	packet-loss concealment
PLCP	Physical Layer Convergence Protocol
PMD	Physical Medium Dependent
Prev_Tx_R	Previous Transmission Rate
PSK	Phase Shift Keying
PSTN	Public Switched Telephone Network
P2P	Peer to Peer
QoS	Quality of Service

Rc	Arrival time of current packet
Rp	Arrival time of previous packet
RFC	Request for Comments
RF	Radio Frequency
RR	Receiver Report
RTCP	Real Time Control Protocol
RTP	Real-time Transport Protocol
RTS	Request to Send
R-factor	Rating factor
R_th	Threshold value of R-factor
Sc	Timestamp of current packet
Sp	Timestamp of previous packet
SDES	Session Description
SDP	Session Description Protocol
SIFS	Short IFS
SIP	Session Initiation Protocol
SIP URI	SIP Uniform Resource Identifier
SMTP	Simple Mail Transfer Protocol
SNR	Signal to Noise Ratio
SNs	Station Nodes
SR	Sender Report
STDs	State Transition Diagrams
SSACC	Scalable Speech/Audio Coder Control
T	Frame delay
TCP	Transmission Control Protocol
TIA	International Telecommunications Industries Association
Tx	Transmitter/Transmission
UA	User Agent
UDP	User Datagram Protocol
VoIP	Voice over IP
VoWLAN	VoIP over WLAN
WLAN	Wireless Local Area Network
WNs	Workstation Nodes

CHAPTER 1

INTRODUCTION

This chapter is an introduction of this research work. It discusses the WLAN multi-rate issue that is addressed by this research work. The chapter also describes the research objectives to be achieved and the research approach to be followed, in order to meet the research objectives. Lastly, the chapter is concluded with the contributions and organization of the thesis.

1.1 Introduction

Voice is an analog signal that is used to be transported as an electrical wave signals in the old telephone networks. However, through the telecommunication revolutions experienced over the last decades, today voice can be transported in a digital form solving the problems analog system used to have, which are mainly caused by noise and not economic [1]. Initially, voice was digitally transmitted over circuit-switched connections in which voice travels over the same path from the sender to the receiver. In circuit-switched network, resources are dedicated to a call that no another call can use them at the same time, and voice packets are always sent during both active and silent periods. Therefore, it wastes network resources and consumes more bandwidth. On addressing this inefficiency issues, transmitting voice over packet-switched or IP-based networks (VoIP) has been introduced, which happens to be one of the telecommunications worldwide significant revolutions occurring over the recent years. In packet-switched networks, unlike circuit-switched networks, packets are routed independently from each other. Accordingly, voice packets of the same call session can be transported in different paths to be gathered and re-arranged at the destination end. Packet-switched networks also allow packets of different application type (data, voice, etc.) to share the network utilizing the available link capacity [2].

Basically, VoIP service is highly on demand nowadays and deployed widely by many people around the globe. This fast growth is due to the advantages that VoIP service offers over the traditional Public Switched Telephone Network (PSTN) systems and mainly due to the economical savings feature [3]. It offers flexibility, simplicity in implementation and maintenance, and more features with no extra charges such as voice mail and call forwarding [4, 5]. With VoIP services, calls can be made from any place in the world as long as Internet connection is available and free calls are gained particularly when the connection is from PC to PC [6, 7]. Apart from all these advantages of VoIP, the quality that VoIP can offer is all that matters to the users.

In IP-based networks, however, there is no guarantee made about the quality of VoIP communication and therefore, VoIP application cannot yet meet the level of quality that PSTN system provides. The challenges VoIP face primarily take place due to the fact that packet-switched networks are mainly designed for transmitting best effort data traffic and not for voice packets [8] , which unlike data traffic, require high level of Quality of Service (QoS). In fact, QoS is an important issue in VoIP application which is, as a real time application, sensitive to delay and packet loss. These two parameters, as well as jitter, and bit error rate (BER) are the major parameters that have impact on the QoS of VoIP [9]. By improving the values of these parameters not to exceed their acceptable thresholds, QoS of VoIP can be enhanced.

On the other hand, VoIP application has been deployed over different types of IP-based networks. However, the integration of VoIP and IEEE 802.11 WLAN technologies (VoWLAN) has become very popular recently and it is now gaining the attention not only from researchers, but also from Internet Service Providers (ISP) [10]. That is because of the fact that increasing LAN network users applying voice communication over the Internet [11] is leading to high demand on applying VoIP on wireless networks as well. It is also due to the characteristics of WLAN network and the features it can provide for VoIP application including mobility, simplicity, scalability, flexibility, and potential economical advantage [12, 13]. Besides, WLAN network eliminates the need of physical wires since wireless nodes communicate with each other via radio frequency (RF) or Infrared (IR). Moreover, WLAN provides ubiquitous communication in offices, campuses, hospitals, airports, factories, hotels, cafes, stock markets, shopping malls, and more of previously unconnected places. In view of that, people's demand, nowadays, is increasing to have the services of real time applications such as VoIP even while they are roaming in such network access.

Nevertheless, the characteristics of WLAN networks add further challenges that can degrade VoIP QoS such as packet header overhead, bandwidth limitation, channel access mechanism, channel condition variability, transmission rate variation, fading, interference, mobility, etc [14]. Hence, WLAN network still cannot afford to provide the stringent QoS that VoIP application requires. As a result, there is a need to address these challenges in order to meet the quality requirements of VoIP application to be as good as the transmission of voice over PSTN or even better.

1.2 Research Problem Statement

Since the channel condition of WLAN network varies, it was suggested to adapt the WLAN transmission data rate at MAC layer according to this variation in order to maintain its bit error rate [15]. This adaptation method of MAC transmission data rate is called Link Adaptive (LA) technique. Although IEEE 802.11 standard allows the transmission data rate to be adapted, it does not specify any particular way of when to adapt and select one of the available transmission data rates [16]. Consequently, several LA techniques have been proposed by manufacturers and research communities.

Generally, the LA techniques intend to change the transmission data rate when WLAN channel condition is degraded. The channel condition degradation can happen due to facing different factors, such as user moving away from the Access Point (AP) as shown in Figure 1.1. As a node moves away from the AP, it enters a low coverage area where the traveling signal strength decreases. Therefore, the LA changes the transmission data rate to a lower value to reduce the BER. LA unfortunately leads to having a multi-rate network where different nodes in the same WLAN network apply different transmission data rates. Despite the fact that the multi-rate feature provides flexibility to customers who demand to have both high and low rate in the devices

[17], it can negatively affect the network performance, particularly the service quality of VoIP. The multi-rate feature is an issue for the QoS of VoIP systems because it causes variation in the network bandwidth. The bandwidth variation is caused by the network nodes that apply different transmission data rates where the transmission at low data rate occupies the channel for long period and hence reduces the bandwidth availability. This network variation causes the transmission delay and packet loss to increase thus the VoIP network quality degrades [18]. Furthermore, the multi-rate feature causes network congestion particularly when the network capacity is full.

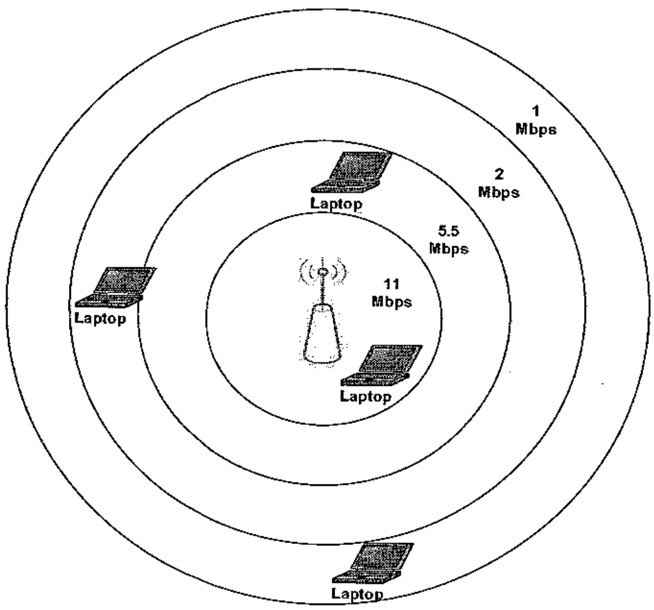


Figure 1.1: Multi-rate WLAN Network

Hence, when the transmission data rate of a mobile node changes in a VoWLAN network particularly when the network capacity has reached its limit, the quality of all active VoIP calls are negatively affected. As this quality degradation issue caused by the multi-rate feature cannot be solved by the standard protocols, a rate adaptive approach is proposed. The rate adaptive approach aims to address the multi-rate issue by adapting the rate of voice packets at the Application layer when the transmission data rate change happens. Most of the proposed studies so far have utilized adapting the CODEC at the Application layer as the key parameter of rate adaptive technique. Since different VoIP CODECs produce different bit rates due to applying different compression ratios, CODEC rate adaptive technique applies a different CODEC for a different transmission data rate. Therefore, applying this parameter requires the availability of more than one CODEC in VoIP system. Yet, most of CODECs are not free and must be purchased. This is adding extra charges to the user to apply such technique. Furthermore, CODECs result different QoS and the CODECs that use high compression ratio reduce the quality of VoIP. Thus, there is a need to develop a solution that overcomes the drawbacks of CODEC rate adaptation technique in order to improve the VoIP quality degradation happening in a multi-rate WLAN network.

1.3 Research Objectives

Since the rate adaptive technique can play key role in adapting the voice rate at the Application layer, this research work aims to develop and evaluate a new algorithm based on rate adaptive technique that addresses the issue of multi-rate feature of IEEE 802.11 WLAN network. The new algorithm should be able to minimize the effect of multi-rate on VoIP quality in order to achieve a performance enhancement of VoIP that provides a level of user satisfaction. Therefore, the research objectives can be highlighted in the following points:

- To investigate the impact of voice packet size parameter and MAC transmission data rate on VoIP over WLAN network performance.
- To develop an algorithm based on rate adaptive technique and Cross-Layering approach that is able to achieve an improvement in VoIP quality in a multi-rate WLAN network.
- To evaluate the new proposed algorithm enhancement on the performance of VoIP over WLAN network incorporating different QoS parameters.

1.4 Research scope

This research study intends to enhance the performance of VoIP over WLAN when the multi-rate issue is occurred causing degradation on VoIP quality by using voice rate adaptive technique. Therefore, this work focuses on only voice traffic in the network. It does not include other type of traffic such as TCP or video. Moreover, the network studied in this work is an infrastructure WLAN network that is based on the common standard IEEE 802.11b. The WLAN network is of only one Basic Service Set (BSS) that consists of several WLAN nodes that communicate with each other through an AP node. This WLAN network scenario is simulated based on the specifications of its standard. The VoIP system with its main components is also simulated according to its standard and the most common CODEC G.711 is applied for the encoding process as it provides the highest MOS value.

Furthermore, the proposed algorithm and its effectiveness are evaluated by measuring the QoS parameters of VoIP such as MOS, E2E delay, packet loss, jitter, and throughput.

1.5 Research Approach

In order to address the research problem and provide a suitable solution, this research work is divided into four main parts as shown in Figure 1.2. In the first part, the research starts with a theoretical study on rate adaptive technique and its importance for real-time applications. It also discusses Cross-Layering approach that is used in developing this technique. Besides, it studies the effect of voice packet size on VoIP quality. The second part intends to meet the main research objective and improve the QoS of VoIP over a multi-rate WLAN network by proposing an agent called Cross-Layering Adaptive Agent (CLAA). This part describes the agent model and its algorithm that is developed utilizing the rate adaptive technique and Cross-Layering approach. The agent operates by utilizing information available in MAC layer and Application layer. It also utilizes RTCP reports to monitor the network quality. In the third part, a simulation study is performed. This part of research approach is divided into two phases, the first phase implements a preliminary performance study of voice packet size parameter. The preliminary study is based on simulation that is performed to analyze the effect of voice packet size on the different transmission rates of MAC. This is to define an optimum range of voice packet sizes to be used by the algorithm of the proposed agent. In the second phase, the proposed CLAA agent is validated by performing a set of simulation scenarios. This part also describes the design and configuration setup of WLAN network topology and VoIP system operating on this network. Finally, the agent performance is evaluated in the fourth part of this research approach. To carry out this performance evaluation, the obtained simulation results of different measured quality parameters are analyzed and compared with similar simulation results obtained from a standard network.

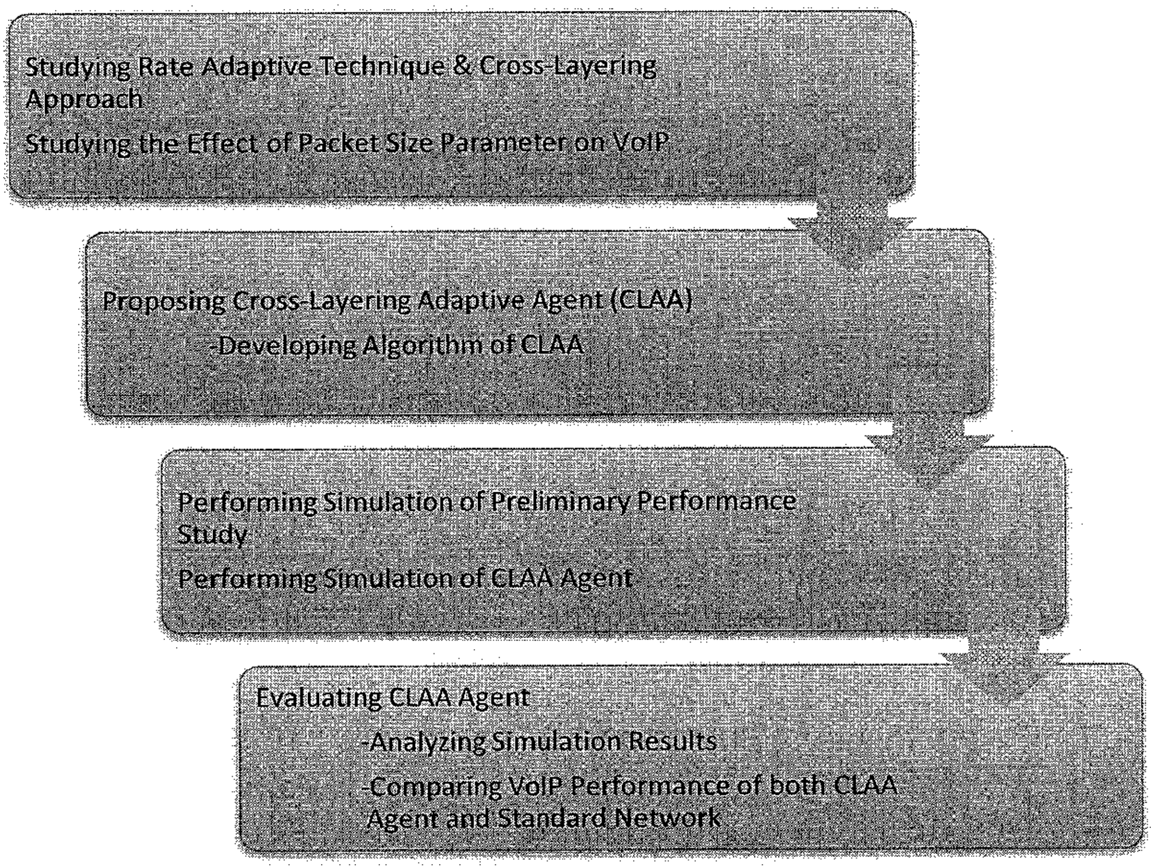


Figure 1.2: Research Approach Diagram

1.6 Research Contributions

As a main contribution of this research, the CLAA agent that is based on packet size rate adaptive technique is proposed. The agent intends to adapt the voice packet size parameter in real time according to the transmission data rate and network state variation happening in the WLAN network. The agent utilizes the packet size parameter due to the fact that it plays role in changing the network bandwidth requirements. Moreover, the CLAA agent employs Cross-Layering approach that enables the communication between the different layers of network. Therefore, it makes the rate adaptive technique implementation possible. This rate adaptive technique is the key technique this work depends on where it is applied by the agent to adapt the voice packet size. Generally, the research contributions can be summarized as follows:

1- Performing simulation study that analyzes the performance of VoIP over WLAN network with different MAC transmission data rates and voice packet sizes. The detailed analysis study investigates the relationship between the voice packet size and voice quality when applying different transmission data rates. This investigation illustrates the importance of utilizing packet size parameter to enhance VoIP performance in bandwidth variation network.

2- Proposing an adaptive algorithm for the developed Cross Layering Adaptive Agent (CLAA) that addresses the effect of multi-rate issue on VoIP quality. The proposed CLAA agent attempts to address this issue by applying rate adaptive technique that utilizes the packet size parameter. Since the packet size parameter can change the network bandwidth requirement, it is adapted according to the change of transmission rate at MAC Layer. In addition, the agent utilizes Cross-Layering approach which allows different network layers to communicate and exchange information with each other, thus, it supports applying adaptive rate technique. The approach is applied between MAC and Application layers due to the fact that the voice packet size at the Application layer is adapted according to the transmission data rate at the MAC layer.

3- Measuring different VoIP quality parameters incorporating MOS, end-to-end delay, packet loss, jitter, and throughput when applying CLAA agent. This measurement is performed to evaluate the effect of CLAA agent on VoIP performance. Hence, evaluating the effect of adaptive voice packet size on VoIP quality. Furthermore, it also examines the effectiveness of CLAA agent by performing further evaluation of its performance in different network scenarios of different levels of congestion.

1.7 Organization of the Thesis

The rest of the thesis is organized as follows:

Chapter 2 provides an essential background study on VoIP system including its components, protocols, QoS issues, and evaluation assessment techniques. It also discusses the network type used for this study, which is WLAN network, and explains its standard, techniques, and issues. The challenges of applying VoIP system over WLAN are presented in this chapter as well. Furthermore, chapter 2 covers several research works that are related to the work of this research.

Chapter 3 describes the methodology that is followed in this work to achieve the stated objectives of the study. In this chapter, the CLAA agent model, its operation and its algorithm are also explained in details. Besides, it discusses the simulation tool and its methods, set-up, and necessary configuration that are implemented in order to obtain the required results.

Chapter 4 focuses on analyzing and discussing the obtained simulation results. These results include the preliminary analysis study results obtained from examining the effect of voice packet size on VoWLAN for the different transmission data rates and the results obtained from evaluating the performance of the proposed agent. In this evaluation, a performance comparison between a network using the proposed agent and the same network setup without using the agent is made. The chapter also presents more results obtained from performing further examination on the agent performance. These results evaluate the agent effectiveness under different levels of congestion caused by the multi-rate issue. Furthermore, all evaluation results obtained in this study are based on five parameters that the agent attempts to enhance, which are MOS, end-to-end delay, packet loss, jitter, and throughput.

Chapter 5 draws a conclusion of the work by summing up the achieved objectives, findings and contributions of this research. It also describes the work limitations and proposes some future works.

The thesis reading outline is illustrated in Figure 1.3.

Appendix A lists a set of publications that has been contributed throughout this research.

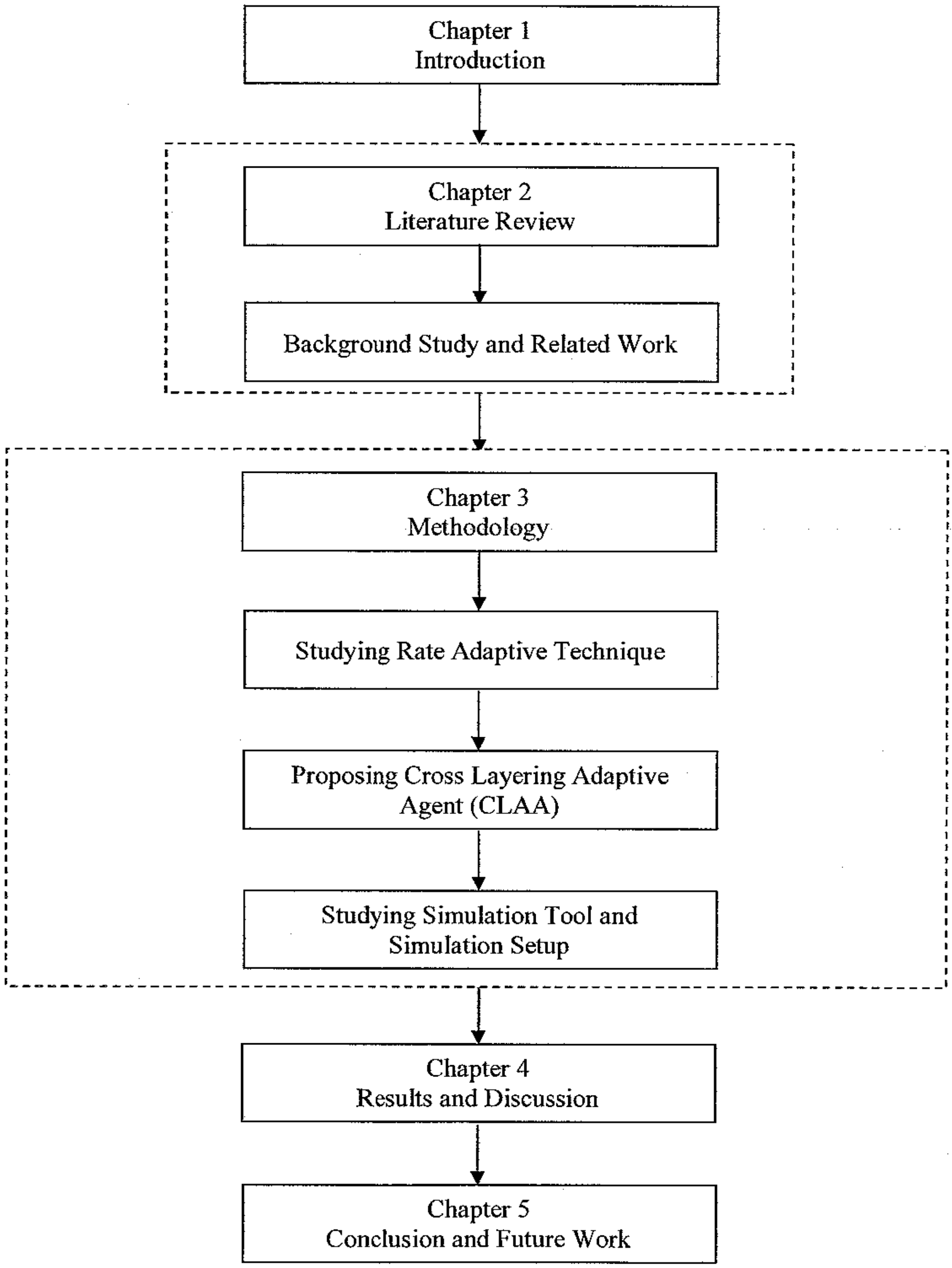


Figure 1.3: Thesis Outline

1.8 Summary

The chapter introduced the VoIP technology and its implementation over WLAN network, which is known as VoIP over WLAN (VoWLAN). It covered the issue introduced by the multi-rate transmission of WLAN network, which is addressed in this work. It also discussed the research objectives and research approach used to achieve them. Lastly, it presented the organization of the rest of thesis; Chapter 2: discusses literature review, Chapter 3: presents research methodology, Chapter 4: discusses and analyzes simulation results, and Chapter 5: concludes the thesis.

CHAPTER 2

LITERATURE REVIEW

This chapter covers the technical fundamentals of VoIP including its protocols and its main components. The chapter, thereafter, provides a discussion on WLAN network and its IEEE 802.11 standard. The challenges of implementing VoIP over WLAN network are discussed as well. Also, the chapter presents the VoIP QoS issues, the important parameters that measure this quality, and the assessment techniques used to evaluate it. The chapter, then, highlights a brief description on Cross-Layering approach, which is used in the study. Finally, the chapter discusses and criticizes the most related works to the proposed algorithm in this study.

2.1 Overview of Voice over IP (VoIP)

2.1.1 VoIP System

VoIP system basically converts voice analogue signals that are captured by a microphone device to digital signals that can be read and processed by the computer. These digital signals are then gathered and grouped in packets to be transported over the IP-based network. There are several components used for this process of voice signals before they are sent to the network. The most integral components are voice CODEC (Coder/Decoder), packetizer and play-out buffer [19].

Generally, VoIP system starts its process with collecting an adequate sample of analogue voice signals by a sound card and then converting them into digital signals. The digital samples are now grouped into frames to be encoded and compressed by the voice CODEC. The CODEC defines the number of samples per frame and the mathematical compression algorithm that is used to translate the raw samples into compressed data. There are many different types of voice CODECs that are developed and standardized by the International Telecommunication Union-Telecommunication (ITU-T) such as G.711, G.729, G.723.1a, etc. The encoded frames are now ready for the next process, packetization in which the frames are fragmented into equal size of voice packets. Conversely, throughout the system process different protocol headers from different layers of the network model will be attached to each voice packet as illustrated in Figure 2.1. The protocols' headers added to the voice packets are Real-time Transport Protocol (RTP), User Datagram Protocol (UDP), Internet Protocol (IP) and Data Link layer headers.

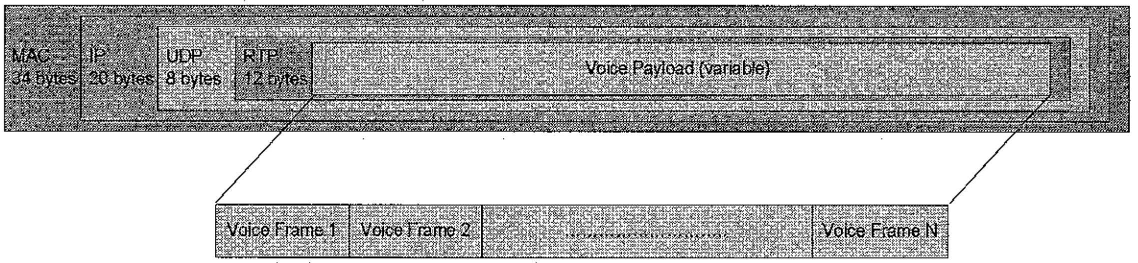


Figure 2.1: Voice Packet

The packets are then transported over the network to their destination. At the receiver side, the opposite process of depacketizing and decoding the received packets is performed. Furthermore, during the transmission process, packets may be lost or delayed causing time variation (jitter) in the packets' arrival. Hence, a play-out buffer is placed for the received packets to be queued for a play-out time before being played to the user. Therefore, the play-out buffer mitigates the incurred jitter for a smooth play-out. However, those packets arrive later than the predefined play-out time will be discarded. VoIP system and its components are illustrated in Figure 2.2.

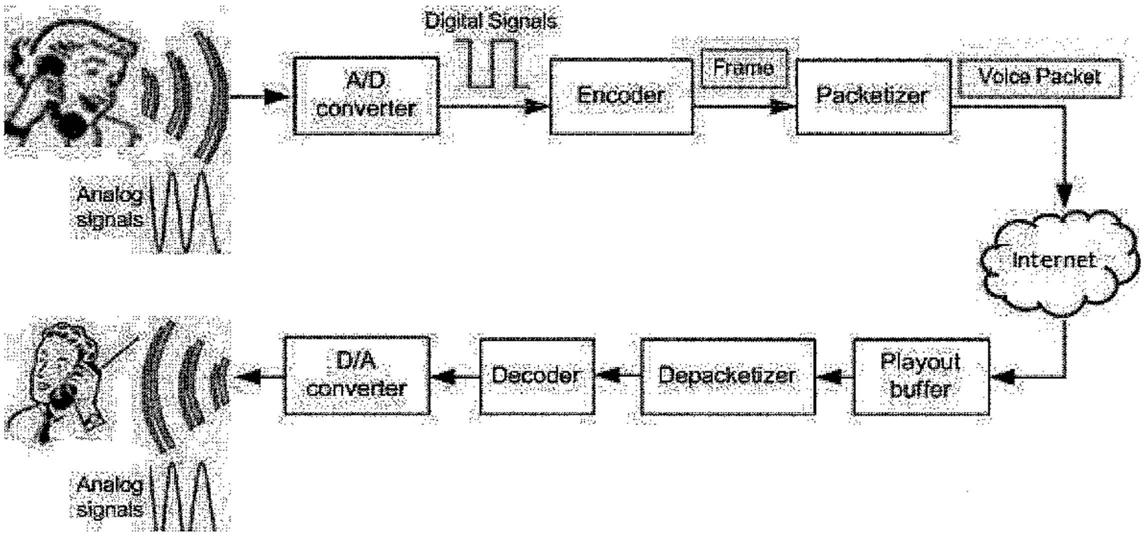


Figure 2.2: VoIP System

2.1.2 VoIP Protocols

For VoIP system to work, several components have been developed. A set of these components is the different protocols designed to enable the communication between VoIP users. Figure 2.3 illustrates VoIP protocols particularly over WLAN network, which is the focus of this study. The most commonly used protocols in VoIP systems are described in the following sub-sections.

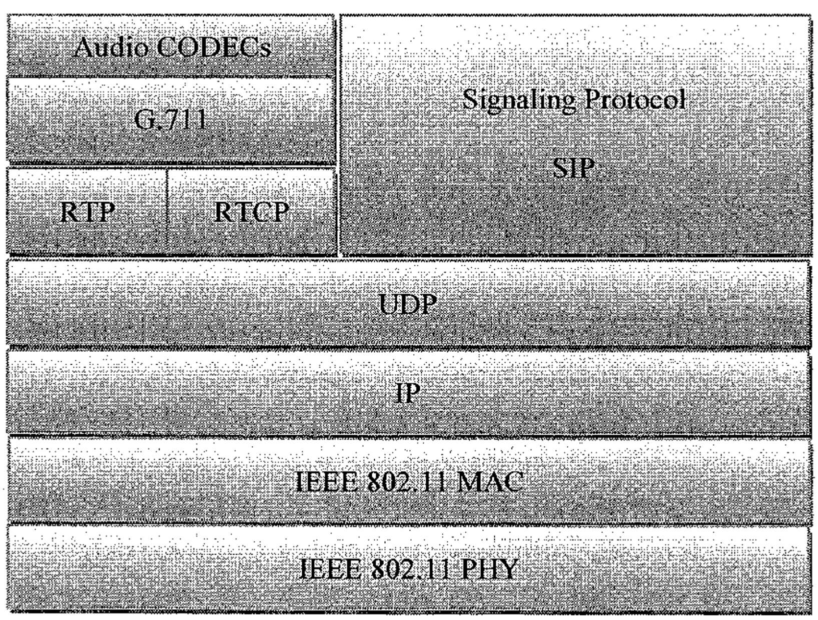


Figure 2.3: VoIP over WLAN IEEE 802.11 Protocol Stack

2.1.2.1 Session Initiation Protocol (SIP)

In order to carry VoIP calls over IP-based networks, two types of protocols are required: signaling protocols and media transport protocols. The signaling protocols are responsible of transferring signaling information and controlling VoIP calls, which mainly consists of setting up, managing, and terminating the calls [4]. They are also important to locate the callee, especially that the client's IP address can be changed from one session to another, to invite the callee to the call session, and to establish the call session based on the receiver's decision whether to accept, reject, or redirect the call. Moreover, upon the call establishment, the signaling protocol enables the endpoints to negotiate and agree upon a set of parameters for instance the CODEC type to be used in the media transmission. In case of a session that involves more than one participant, it manages the requests of adding or leaving participants. After finishing the call as the endpoints end it up, the signaling protocol tears down the call session.

Several signaling protocols have been standardized over the years. However, the most well-known two signaling standards are H.323 and SIP. H.323 is the first signaling protocol that was standardized to support the functions of real time applications over LAN networks. H.323 protocol was standardized by the ITU-T [20]. On the other hand, Session Initiation Protocol (SIP), which has been standardized as RFC3261 by the Internet Engineering Task Force (IETF) [21], became the de-facto signaling standard for VoIP transmission. Although these two signaling standards have the same objective, they differ in their approaches. While H.323 protocol is based on circuit-switched approach, SIP protocol is based on Internet protocols. Nonetheless, SIP is gaining more popularity for it is more lightweight, flexible, and scalable and it has less complexity [22].

SIP is the industry leading standard of VoIP signaling protocol as it is developed with the aim of supporting calls control for IP-based communications. It is a text-based application-layer signaling protocol that is mostly implemented on top of UDP or TCP transport-layer protocols [23]. The root of designing SIP is both Hypertext Transfer Protocol (HTTP) and Simple Mail Transfer Protocol (SMTP) and it operates based on request-response method where VoIP user sends a request message to SIP server and the server sends back a response to the user. Moreover, SIP Identifier called SIP Uniform Resource Identifier (SIP URI) is in the form of email address containing a user name and a host name.

The primary components of SIP protocol are:

- User Agent (UA): is an endpoint entity of VoIP session. If it plays the role of client, it creates and sends requests to initiate the call session, whereas it receives requests and responds back if it is the server.

- SIP Registrar: is where the user registers the SIP URI and location. It also keeps track of the user's location.

- SIP Proxy Server: is an intermediate element that is in charge of transmitting the signaling packets to the proper entities. It works as a router for voice signals gaining the capability to follow the call session status. It also has the ability to modify the requests before forwarding them.

- Redirect server: enables the call to be redirected to an alternate SIP URI.

The exchanged SIP packets between these components are called "messages". The request message contains "method" that defines the type of request while the response is carrying "response code". The basic methods of SIP request messages are as listed below in Table 2.1.

Table 2.1: Basic SIP Messages

SIP Request	Description
INVITE	Request to invite a client to a call. It is mainly used to initiate a call session.
UPDATE	Request to modify or update the description of a call session.
CANCEL	Request to remove any pending invitation as no response is received after sending the INVITE request.
ACK	Request to notify that the response has been received.
REGISTER	Request to register a new user in the server indicating some information such as user's IP address.
OPTIONS	Request to inquire about the capabilities of the caller prior to establishing the call session.
BYE	Request to terminate a call.

On the other hand, there are numerous of SIP responses, which can be divided into different categories based on the first digit of the response code as illustrated in Table 2.2. For example, the response code 180, which defines the ringing process, starts with the digit 1 indicating that the response is of "information" type. 200 OK is a common response, which verifies that the request is accepted. The response codes are explained in the standard in [21].

Table 2.2: Basic SIP Responses

SIP Response	Description
Provisional: 1xx	Information category that defines that the request is in process and there is no response is identified yet.
Successful: 2xx	The request is successfully received.
Redirection: 3xx	It provides services that help to redirect the call.
Request Failure: 4xx	Server failed to process the request for reasons such as syntax errors.
Server Error: 5xx	Server failed to process the response for reasons such as internal errors.
Global Failure: 6xx	It indicates a global failure that cannot be processed by any server.

Generally, to establish a call session, the endpoints have to agree upon a set of information that is common between them, for example IP address, port number, CODEC, etc. Therefore, a protocol is required to provide a way for the involved parties to negotiate. SIP uses Session Description Protocol (SDP) protocol for such negotiation. SDP message, which carries out a set of properties and parameters called "session profile", is attached to the body of SIP INVITE message.

In explaining how SIP works, let us assume a caller (Client A) is initiating a call session with a callee (Client B). The signaling process starts off with sending an INVITE request that carries the session characteristics for both parties to agree upon, to the proxy server (or directly to Client B in case the communication type is peer-to-peer). When the proxy server forwards the request to Client B, Client B receives a 'ring'. As Client B picks up the call, a response message with the code 200 OK will be sent back to the proxy server, which in turn forwards it to Client A. After that, a notification ACK method is sent back from Client A as a confirmation of call session establishment. After completing the call setup process, the media transmission starts between both clients so they start the conversation. When the clients finish talking and the call ends up, Client B sends a BYE message to Client A, which then sends back

OK message. In addition, SIP enables session modification while the call is ongoing by re-negotiating the call session parameters. For doing that, the caller should send a RE-INVITE message following the same above signaling setup procedure. Figure 2.4 demonstrates the basic setup signaling process.

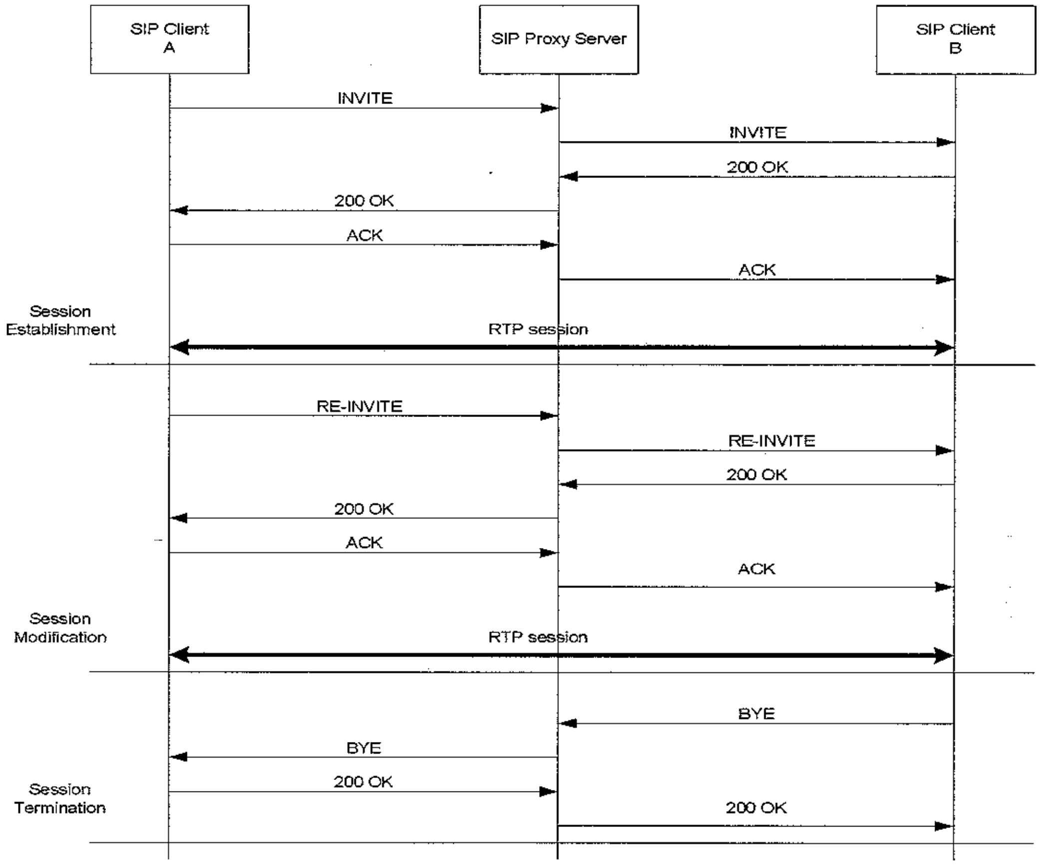


Figure 2.4: Basic SIP Call Operation

2.1.2.2 Real Time Protocol/ Real Time Control Protocol (RTP/RTCP)

As a call session is established between VoIP endpoints, the conversation stream starts to flow in the network from one end to the other. The protocol used to transport this stream across the network is called Real time Transport Protocol (RTP).

RTP protocol is defined in RFC 3550, which was last updated in 2003 [24]. RTP is designed to transport real time stream over IP networks, which by nature disorders the sequence of packets and introduces jitter. However, RTP protocol can compensate these problems with the end-to-end delivery services it provides such as timestamping, sequence numbering, etc. In fact, RTP does not guarantee quality of service for real time applications nor does it ensure packets arrive in sequence at the receiver end. However, RTP provides services that the Application layer can use to address such issues. For example the packet sequence numbering service helps the receiver to solve issues such as rearranging the packets to be in order and counting the number of lost packets. Also, the time-stamp included in each packet helps the receiver to synchronize the play-out and calculate delay and jitter.

RTP is very often accompanied by RTP Control Protocol (RTCP). RTCP protocol does not transport the stream by itself, but it supports its delivery. The main functions of this protocol are providing quality feedbacks such as number of lost packets, delay and jitter that can help in monitoring the network state, providing an identifier called canonical name or CNAME for RTP that helps identifying users or sessions, enabling VoIP user to identify the number of users in the session, and providing control information of a session.

Additionally, RTCP feedback reports or messages are of different types and for different purposes. For instance, the reports generated at the receiving side (Receiver Reports) are sent periodically providing reception quality statistics including jitter, packet loss and other information to the sender. Table 2.3 lists the most important types of RTCP messages [25]. On the other hand, RTCP does not control the QoS parameters, it only sends quality-related information that can help the receiver to monitor and improve the QoS when required. Furthermore, RTCP packets consume 5% of the call session bandwidth with a transmission interval rate of 5 seconds.

Table 2.3: RTCP Types of Messages

RTCP Message Type	Description
Sender Report (SR)	It is sent by the sender reporting the timestamp and amount of data that has been sent.
Receiver Report (RR)	The receiver reports quality statistics such as number of packets lost, delay and jitter.
Session Description (SDES)	It provides information related to the source such as email address, phone number, etc.
Bye	An announcement message for disconnection.
Application Specific (APP)	For designing Application specific RTCP extensions.

Although TCP transport protocol is commonly used to transport data traffic over the internet, RTP and RTCP protocols commonly run on top of the connectionless unreliable UDP transport protocol. That is because UDP protocol is preferred for the delay-sensitive real time applications [26]. The connection-oriented TCP protocol is not suitable for delay-sensitive packets for it applies the acknowledgement (ACK) scheme and retransmission technique of lost packets, which introduce delay. Besides, the ACK scheme that requires the receiver to send a notification message of each received packet to the sender and retransmit every lost packet consume more network bandwidth. Alternatively, UDP, which does not apply ACK scheme, has less network bandwidth consumption and lower overhead on the network, thus more suitable for VoIP applications. However, UDP lacks reliability and congestion control techniques, which RTP and RTCP reports can help to address these issues [27].

2.1.3 VoIP CODEC

As a call session is established and the caller starts speaking, his/her voice, which initially goes through the microphone, is basically analog waveforms. For the voice to be transported by the RTP protocol over the internet, it must be digitized first. The conversion process of analog signals to digital form is performed by the CODEC component, which is built in a microchip called Digital Signal Processor (DSP). The CODEC basically converts analog signals to digital signals at the sending side and carries out the reverse process at the receiving side as depicted in Figure 2.5. Moreover, CODEC is a key component of VoIP systems for the reason that it defines the amount of speech to be placed in a voice packet that traverses the internet, hence defining the VoIP bandwidth requirement. Besides, the accuracy of its analog signals reproduction process also plays role on voice quality.

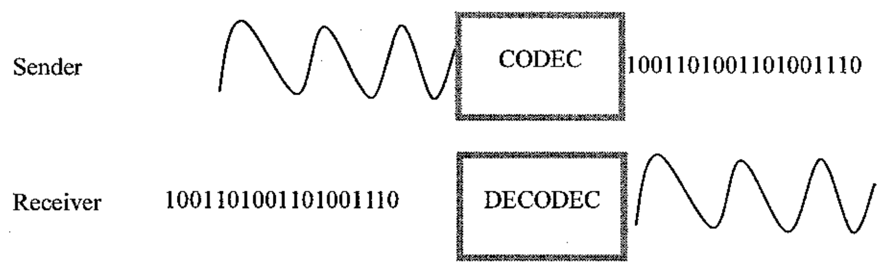


Figure 2.5: CODEC: Analog-to-Digital Conversion

In fact, the analog signals are continuous-time signals and corresponds to a set of continuous values while the digital signals are discrete-time signals that take discrete values at uniform intervals [28]. In order to transform the analog signals to digital signals that can be processed by the computer, the CODEC performs three main steps: sampling, quantizing and coding [5].

Sampling refers to the process of collecting samples of analog signals at discrete time intervals. The number of samples collected per second is called sampling rate or sampling frequency, which is defined in the unit of Hz. Deciding on the sampling rate is based on Nyquist theorem that defines a method of selecting the sampling rate. The theorem proves that for the reproduction of original analog signal from digital signal, sampling rate should be greater than or equal to twice the highest frequency of the analog signal [29]. Accordingly, sampling rate of voice signals is 8 kHz [30] since human speech frequency falls below 3000 Hz.

The continues values of the discrete-time sample are then mapped into a finite number of discrete values in the process of quantization [31] where sample values are represented by bits. Quantization also defines the number of bits (bit rate) to be produced as a result of its process. Additionally, throughout the process of analog to digital conversion, some data of the original analog signals are lost and cannot be recovered. Therefore, selecting the sample rate and quantization method is important to increase the level of accuracy. On the other hand, CODECs intend to minimize the produced bit rate with maintaining the signals quality using less complex algorithms [32]. For that reason, different coding techniques have been introduced aiming to meet the challenge created from the trade-off between these parameters.

The coding process is performed next where the discrete value is represented in binary format called Pulse Code Modulation (PCM). As shown in Figure 2.6, it gathers a number of samples for a period of time and encodes them into a set of bits forming a frame, which is then inserted in a voice packet that is transmitted over the network. The encoding process can also incorporate a compression technique, which compresses the frame to reduce its size thus reducing the bandwidth requirement. Also, the CODEC capability to determine the number of bits of a frame (frame size), makes it play role in determining the bandwidth requirements. As a matter of fact, increasing the frame size or the number of frames inserted in a voice packet reduces the required amount of bandwidth for VoIP transmission. That is because of the reduction happens in the number of headers attached to every transmitted packet as will be explained later in this thesis. Furthermore, the different compression techniques of CODECs influence the voice quality. Although the higher compression method reduces the congestion level in the network, it reduces the voice quality.

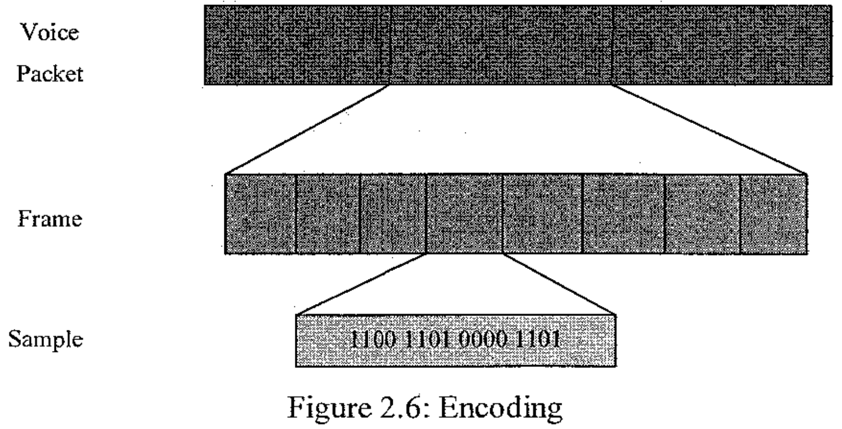


Figure 2.6: Encoding

Generally, CODEC produces three types of delay during its process - framing size, data processing, and look-ahead [33]. Framing size delay can be defined as the time interval required to process a frame, data processing delay refers to the time the CODEC algorithm takes to process a sample and look-ahead, which is applied in some CODECs, is the time it takes to observe the next sample. Different CODECs with different delays and characteristics have been introduced for example G.711 CODEC, which is considered in this study.

G.711 is a PCM-based algorithm that is standardized by ITU-T [34]. It encodes signals at bit rate of 64 kbps producing samples of size 8 bits with sampling rate of 8 kHz [1]. It is the most common standard in VoIP and it is also widely used in PSTN systems. It provides the best voice quality as the higher the bit rate, the better the quality but the less the transmitted data. Furthermore, the standard defines two forms of the algorithm, which are μ-Law used in North America and Japan and A-Law used in Europe and the rest of the world. Both μ-Law and A-Law encode 14 bit long linear PCM samples and 13 bit long linear PCM samples respectively into 8 bit samples. Some of G.711 parameters including delay parameters are listed in Table 2.4.

Table 2.4: G.711 parameters [35, 36]

Parameter	**G.711**
Bit-rate (kb/s)	64
Bits per frame	8
Compression type	PCM
Algorithmic delay (ms)	0.75
Look-ahead delay (ms)	0
Framing size (ms)	0.125
Complexity (MIPS)	<<1
MOS	>4.0

2.1.4 Effect of Packet Size on VoIP

After the encoding process, the generated frames are inserted into packets to be sent to the network. As illustrated in Figure 2.1, the voice packet may contain one or more frames based on packetization, which decides on the number of frames (amount of speech) to be included in the voice packet. The time it takes to produce two successive packets is called packetization interval. Basically, there is a relation among these parameters; the longer the packetization interval, the larger the packet size and the more number of frames inserted in the voice packet, and vice versa. Accordingly, packetization is an important parameter to determine the required network bandwidth for transmitting voice packets as has been simply shown in the following equation [37]:

$$NB = H + N^* FS / N^* T \tag{2.1}$$

where NB = Network Bandwidth, H = Packet Header Size (bits), N = Number of Frames, FS = Frame Size (bits), and T = Frame Delay (msec).

As can be seen from the equation, the header parameter has an impact on the network bandwidth requirement. As mentioned previously in Section 2.1.1, headers overhead of RTP, UDP, IP, MAC and PHY layers are attached to each voice packet before it is transmitted. If the packet size is small, the overheard ratio will be high and will consume more bandwidth since more than 74 bytes will be attached to each packet. Therefore, the call traffic rate will increase causing the voice quality to decrease. However, if the packet size is large, the transmitted overheads will consume less bandwidth. Therefore, the method of applying a fixed packetization interval is not effective in the variable WLAN environment, since a fixed voice transmission rate does not match the variability of the available network bandwidth. Hence, it is important to adapt the packet size parameter according to the network condition to obtain better voice quality.

On the other hand, producing large packets would add to the end-to-end delay due to the large packetization interval. If a voice packet that is large in size is transmitted in a network with a large delay, it will increase the delay of the network causing degradation in the voice quality. Conversely, the packet size has an effect on packet loss parameter as well. The effect of losing a large packet is more significant than the effect of losing a number of packets of small size [38]. However, the required retransmission of the lost small packets will require more bandwidth and time than retransmitting one large packet. Furthermore, the transmission of large packets decreases the delay jitter while small packets causes the delay jitter to increase but throughput to decrease as investigated in [39].

Several studies have investigated the effect of packet size on voice quality. Author of [40] has performed a comparison experiment of delay when transmitting different packet sizes. It was concluded that the packets of small sizes are not efficient for VoIP transmissions while large packets increase the delay. Another study in [41] has developed an adaptive quality management technique, which is based on a comparison of encoding parameters made in [38]. The comparison was conducted by analyzing the QoS parameters when applying different encoding parameters in the presence of background traffic. It has also evaluated the effect of applying different voice-to-data ratios on voice quality. The study has concluded that when voice traffic ratio is small in the network, adapting the packet size or the CODEC compression ratio does not enhance the quality. However, when voice traffic ratio is not small, varying packet size in a congested network improves the quality while increasing CODEC compression ratio works better when the number of calls increases.

The work in [27] has also analyzed the network behavior when it gets congested due to either packetization or increasing number of calls in the network with the existing of data traffic. The study has conducted simulation of two scenarios with different AP capacities 1Mbps and 2Mbps and with varying the data traffic. The obtained results have proven that increasing the packet size causes the packetization delay to increase but queuing delay to decrease, which then would not affect the end-to-end delay. However, after an optimum point at which the queuing delay reaches a steady state, the increase of packetization delay causes the total delay to increase.

Moreover, from the bandwidth requirement point of view, an investigation study has been carried out in [37] on the relation between packetization, network bandwidth requirement, and end-to-end delay. The study has concluded that as the network load increases, which decreases the available bandwidth, packetization should be increased to decrease the required network bandwidth for voice transmission. As for the less loaded network, small packetization interval reduces the packetization delay in the expense of increasing the network bandwidth requirement.

2.1.5 Quality of Service Issues in VoIP

Despite the advantages and features VoIP systems offer, voice quality is the key parameter for the success of VoIP. Therefore, developing VoIP systems with high quality is a focal area in recent research activities. The quality of service can be defined as a measurement technique for user satisfactions; the higher the QoS VoIP offers, the more satisfied the user becomes. For VoIP to provide high QoS, it is required to solve the challenges introduced from the transmission of voice over IP networks. It is even worse when transmitting voice packets over wireless channels for its unreliability. QoS was not a big issue when wireless networks were used to transmit data packets. However, with the growing demand of applying VoIP applications over wireless networks, particularly WLAN, QoS has become a critical issue [42]. That is because VoIP applications, unlike data applications, are real time applications that are very sensitive to delay. Besides VoIP is expected to replace PSTN systems in the future, which requires VoIP to deliver the same quality as PSTN quality of service or better. Hence, providing high QoS is an important concern that ensures voice packets are not delayed, lost or dropped during the transmission over IP networks, which still cannot meet the stringent QoS that VoIP requires. The major parameters that affect QoS are delay, packet loss and jitter and improving QoS can be achieved by controlling the values of these parameters to be within the acceptable ranges that are defined by their standards.

2.1.5.1 Delay

End-to-end (E2E) or mouth-to-ear delay can be defined as the total time it takes for voice communication from the moment a person speaks words at one side till they are heard at the other end. The E2E delay is one of the most significant factors that has an impact on user's perception of quality [43] because it affects the communication interactivity. Consequently, the total delay of voice transmission should not exceed the defined standard limits. As recommended by ITU G.114 [44] for good voice communication, delay should be less than 150ms [45]. An E2E delay exceeding this value up to 400ms is still acceptable for long distance calls, but above 400ms is not acceptable. Furthermore, there are several parameters that can add to the total delay, which can be categorized into: delay at the source, delay at the receiver and network delay. While the delay values of some of these parameters are consistent, they vary for some others.

a) Delay at the source: the sources of delay at the sender side are encoding delay, packetization delay, and transmission delay. The voice CODEC introduces some delays when processing the conversion of analogue to digital signals and the compression techniques that are used for efficient utilization of bandwidth. There is a trade-off between compression delay and bandwidth efficiency as the more bits compressed, the less bandwidth needed, but the longer delay added. For instance, Table 2.5 shows delay values for some CODECs of different compression techniques. For the packetization delay, it takes place when inserting the chunks of frames in the packet to be transmitted across the network. Delay is also added when the computer transmits the packets into the network to travel to their destination.

Table 2.5: VoIP CODECs one-way delay and loss tolerance

Codec Standard	G.711	G.729
Encoding Delay (ms)	0.13 (negligible)	about 25
Loss Tolerance (%)	7-10	<2

b) Delay at the receiver: the processing time of receiving, decoding, and decompressing voice packets is the delay component at the end side. Additionally, playback delay is incurred by the play-out buffer that gathers and stores packets before playing them out.

c) Network delay: network delay is induced by queuing time in some intermediate nodes in the network, transmission time, and propagation time. The propagation delay is the time packets take to travel through the network media while transmission delay is from network elements such as router's delay and MAC retransmission delay [46].

2.1.5.2 Packet Loss

During the transmission of packets over the network, packets could be lost or corrupted, or arrive late [6]. The packets could be also discarded by the network router buffers due to overflow or by the play-out buffers for late arrival. In wireless networks, unlike wired networks, packets suffer high bit error rate (BER) due to channel fading and interference resulting an increase in packet loss rate. For voice communications, VoIP is able to tolerate packet loss to some extend: for roll quality, 1% or less is acceptable, while 3% or less is acceptable for business quality. Hence, packet loss of greater than 3% causes speech quality to degrade. Packet loss also has an effect on the operation of decoding, which depends on prior or next packet when processing a packet. Consequently, several error concealment and loss recovery mechanisms have been developed to resolve packet loss issue. For instance, the acknowledgement scheme at MAC layer is used to retransmit lost packets, whereas redundancy technique sends packets several times consuming more bandwidth to ensure packet's arrival. The mathematical forward-error correction (FEC) technique facilitates in reconstructing lost packets from the previously received packets. In addition, CODECs also employ error concealment techniques called packet-loss concealment (PLC) to recover the lost or corrupted packets. In packet loss techniques, however, there is a trade-off between complexity and quality, where the more complex techniques produce less packet loss rate.

Moreover, fragmentation technique that MAC layer applies can also affect packet loss rate because of the relationship between packet size and packet loss rate. If the received packets have errors, then the probability of being discarded is higher for packets of big size than those of small size [47].

2.1.5.3 Jitter

Basically, IP networks do not guarantee consistency in packets delivery time. Despite the fact that packets are sent with regular intervals, these intervals become uneven at the receiving side due to different factors such as network congestion. This variation in delay of received packets is known as jitter, which should not go above 30ms for good network quality [48]. Therefore, voice packets of the same flow do not arrive at the same time and playing them once they are arrived results in hearing broken voice, which is not an interactive conversation nature. For this reason, jitter buffer or play-out buffer is introduced in order to reduce the jitter effect and make the conversation becomes smooth. As illustrated in Figure 2.7, the voice packets are collected and queued in the play-out buffer before the play-out. Furthermore, the length of play-out buffer can be fixed or adaptive and the later helps to mitigate the risk of buffer overflow or underflow from happening. The buffer overflow describes the state when the number of arrived packets has become more than the buffer can hold causing the packets to drop, while the underflow issue occurs when the buffer is empty and packets are needed for play-out. Although, the play-out buffer attempts to compensate jitter but at the expense of adding delay to packets that arrive on time or losing packets that arrive late [49].

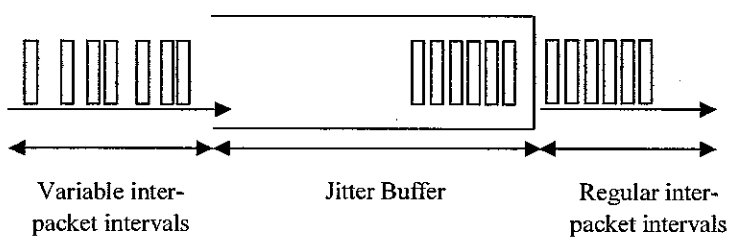


Figure 2.7: Jitter Buffer to Smooth out the Conversation

2.1.5.4 Throughput

The throughput parameter defines the maximum number of received bits compared to the total number of transmitted bits. A high throughput achieved is a reflect of a good network quality [50]. Although the IEEE 802.11 network standards specify the transmission bit rate such as IEEE 802.11b transmits at 11Mbps, 5.5Mbps, 2Mbps and 1Mbps, the achieved throughput is still less than that specified in the standard due to several factors such as the overhead introduced by the WLAN network protocols , the packet size, and the channel condition [51]. Therefore, several methods have been introduced aiming to improve the throughput of WLAN network such as aggregation technique [52] and adaptive techniques [53].

2.1.6 Quality Evaluation Technique for VoIP: E-Model

In order to ensure that the transmitted voice signals meet the required quality, they go through assessment when they reach their destination. The performance assessment of VoIP can be evaluated by different methods, which are classified into two categories: subjective assessment methods and objective assessment methods [54]. As a subjective measurement, Mean Opinion Score (MOS) is defined by the ITU-T P800 [55] as a voice quality rating technique that measures the users' perception. As listed in Table 2.6, the MOS technique defines a set of numerical values that range from 1 to 5, where 1 indicates the worse quality and 5 is the best quality. The technique is performed by a panel of listeners who listen to voice samples and rate the voice quality by voting on the predefined MOS scale. After that, the average of collected votes from listeners is calculated to get the MOS value (MOS score). Nevertheless, this method is expensive, hard to implement, unrepeatable, and time consuming [56]. Comparatively, E-Model is an objective measurement technique that is easier to implement, not time consuming, inexpensive and repeatable [57]. Hence, E-Model is the most commonly used for evaluating the service quality of VoIP.

Table 2.6: MOS Score

Quality of Speech	Bad	Poor	Fair	Good	Excellent
MOS Score	1	2	3	4	5

2.1.6.1 E-Model

E-Model technique is developed by the European Telecommunications Standards Institute (ETSI) [58] and standardized by the ITU-T in 1998 [59] and the International Telecommunications Industries Association (TIA). The E-Model is a computational technique that calculates a quality rating factor (R-factor/R-value) that goes from 0 to 100, where R = 0 refers to extremely bad voice quality and R = 100 means the voice quality is very high. When R value is greater than 70, it is considered as an acceptable quality [60, 61]. Furthermore, with the aim of predicting the subjective voice quality the value of R is calculated based on different parameters of speech signals, network transmission, and terminals that affect the voice quality. Therefore, E-Model is categorized as non-intrusive speech quality because it does not require the original signals in order to calculate voice quality. The following equation is applied in order to calculate the value of R-factor:

$$R = R_0 - I_s - I_d - I_e + A \tag{2.2}$$

where:

- R_0 represents signal-to-noise ratio impairments including room noise and circuit noise.
- I_s refers to impairments factors, which occur simultaneously with voice signal such as loud speech, side tone and quantization noise.
- I_d represents mouth-to-ear delay and echo effects.
- I_e is equipment impairments due to low bit rate voice encoding and packet losses.
- A is advantage of access that compensates impairments factors when other advantages of access are offered for users e.g. users would not expect the same voice quality as traditional phones when using mobility service. The A value lies between 0 and 20 [56] ; where A = 0 in wirebound networks, A = 5 in cellular networks, A= 10 in geographical area or a moving vehicle , and A= 20 in hard-to-reach locations such as satellite connections.

Based on Cole and Rosenbluth's model [62], the E-Model equation is simplified taking into account the default values of all parameters but two: delay and packet loss which transformed into delay and equipment impairments. The simplified expression of E-Model is:

$$R = 94.2 - I_d - I_e \tag{2.3}$$

where: I_d represents quality degradation due to one-way delay and can be calculated as:

$$I_d = 0.024delay + 0.11(delay\text{-}177.3)H(delay\text{-}177.3) \tag{2.4}$$

The *delay* parameter represents the one-way or "mouth-to-ear" delay and H (x) is the Heavyside step function:

$$\text{where} \begin{cases} H(x) = 0 & \text{if } x < 0 \\ H(x) = 1 & \text{if } x \geq 0 \end{cases} \tag{2.5}$$

And, the I_e refers to impairment factor due to low rate CODEC and packet losses. Appendix I in [63] provides values of I_e for several CODECs as a function of packet loss.

Moreover, the output of E-Model, R-factor, is then mapped to an equivalent value of MOS to obtain the subjective quality value by using the following equation [64]:

$$MOS = \begin{cases} 1, & R < 0 \\ 4.5, & R > 100 \\ 1 + 0.035\,R + 7.10^{-6}\,R(R\text{-}60)\times(100\text{-}R), & 0 \leq R \leq 100 \end{cases} \tag{2.6}$$

E-model values are then compared to their corresponding values of MOS and user satisfaction category, which shown in Table 2.7 [65].

Table 2.7: R-Factor vs. MOS score

User Satisfaction	**R-Factor**	**MOS**
Very satisfied	90	4.3
Satisfied	80	4.0
Some Users Dissatisfied	70	3.6
Many Users Dissatisfied	60	3.1
Nearly All Users Dissatisfied	50	2.6
Not recommended	0	1.0

Due to the advantages of this technique and to the fact that it can be computed in real time, E-model technique is applied in the simulation of this work in order to obtain MOS values that evaluate the QoS of VoIP.

2.2 Overview on VoIP over WLAN (VoWLAN)

WLAN network, also known as Wi-Fi, has become an important technology in the wireless networks. For its economical advantage and easy implementation [60], it is being spread worldwide allowing tetherless access not only in homes but also in airports, offices, hospitals, cafes, roads, hotels, shopping malls and more of previously unconnected places. This spread has led users to increasingly demand applying real time applications such as VoIP on WLAN networks. WLAN allows wireless devices, also called nodes, within the local radio coverage area (aka Basic Service Set (BSS)) to communicate with each other wirelessly. Multiple BSSs can be also linked together by the WLAN component Extended Service Set (ESS). Generally, the architecture of WLAN can be classified in two modes: infrastructure less mode and infrastructure mode. The first mode is called Ad hoc or peer-to-peer (P2P) network, which enables connected nodes to directly communicate with each other without an administration from a third party. Alternatively, in the second method, as illustrated in Figure 2.8 the transmission between two or more nodes goes through an intermediate node called Access Point (AP). The AP node is placed in a central location as it receives voice packets from the sender nodes and forwards them to their destinations. The AP node has two interfaces, wireless and wired. Therefore it can be used to extend the wired LAN network with the wireless network and act as an interface between the two types of network. The infrastructure type of WLAN network that is commonly deployed [66] is the network architecture studied in this research work.

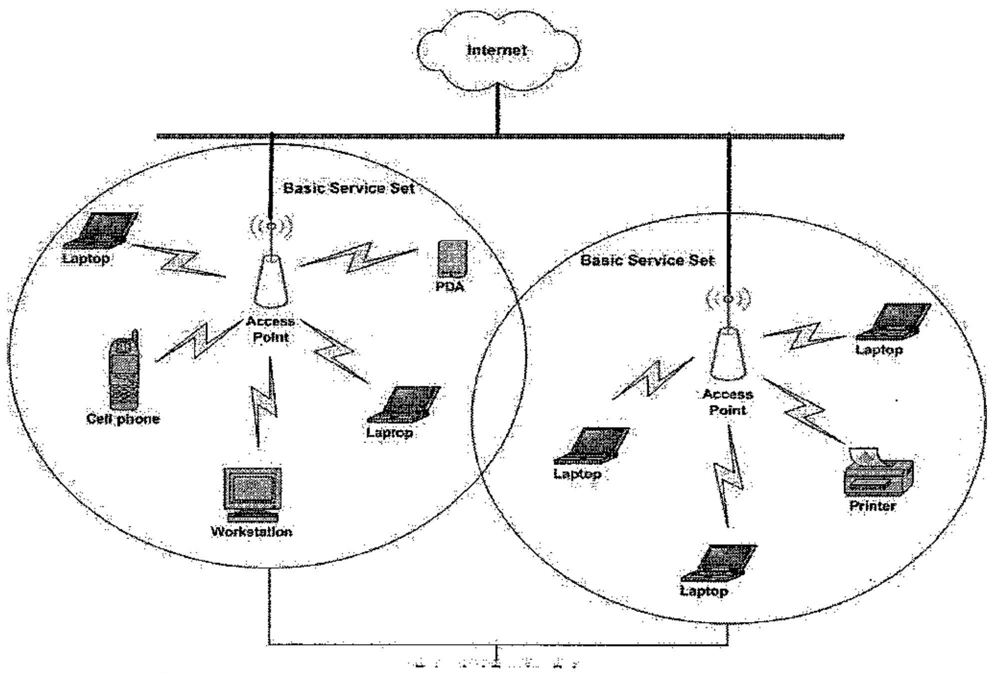


Figure 2.8: WLAN Architecture: Infrastructure Mode

2.2.1 IEEE 802.11 Overview

Fundamentally, wireless is a shared medium in which multiple stations can access the medium at the same time. However, if several stations try to transmit simultaneously, collision will take place and signals will become garbled. For that reason, Medium Access Control (MAC) protocol of the wireless network controls the nodes' access to the medium enabling the wireless devices to communicate with each other efficiently [67]. In WLAN networks, IEEE 802.11 standard defines the MAC layer and the Physical (PHY) layer specifications. The widely deployed IEEE 802.11 [68] wireless technology is a set of IEEE 802.11 series namely 802.11, 802.11a, 802.11b, 802.11g, 802.11e, and 802.11n. The standard was ratified in 1997 [25] and its name came after a committee name "802.11", which is responsible of WLAN development in IEEE organization and the letter appended to the name identifies the group that was assigned a specific task in the development [69].

2.2.1.1 IEEE 802.11 PHY Layer

IEEE 802.11 PHY layer consists of two sub-layers: Physical Layer Convergence Protocol (PLCP) and Physical Medium Dependent (PMD). The PLCP sub-layer is responsible of two tasks: attaching the Physical layer header to the packet that will be transmitted and interacting with MAC layer as required. Conversely, PMD that facilitates the communication with the medium is responsible of implementing modulation and encoding techniques. The modulation schemes convert the frame from digital to analog signals to be transmitted over the medium; examples of this scheme are Phase Shift Keying (PSK) and Complementary Code Keying (CCK) techniques.

Basically, IEEE 802.11 PHY layers support different transmission rates that operate in different modulation techniques; in other words each PHY layer standard supports a different set of transmission rates. For example, the common standard 802.11b operates at 1, 2, 5.5 and 11 Mbps with the modulation techniques CCK, CCK, DQPSK (Differentially-encoded Quadrature PSK) and DBPSK (Differential Binary PSK) respectively. According to the standard specifications, a mobile node is allowed to switch between its transmission data rates while communicating with other node(s) by using Link Adaptive (LA) technique in order to cope with the signal-to-noise ratio (SNR) variation in the wireless channel. Although, this Physical layer rate adaptation in IEEE 802.11 standard aims to improve MAC throughput by adapting its transmission data rate according to the channel condition, it leads to a multi-rate issue which will be further discussed in Section 2.2.2.1. Nevertheless, there is no particular way in the standard that specifies when to switch between the transmission data rates. As a result, a number of LA techniques has been proposed by manufacturers and research communities: Auto Rate Fallback (ARF) is the first rate adaptation algorithm published [16].

In addition, 802.11 standards operate in different frequency bands. The frequency band is divided into channels where a number of them are non-overlapping channels. These non-overlapping channels are based on the frequency reusability in areas that are far from each other with a minimum distance that is determined by the number of non-overlapping channels.

2.2.1.2 IEEE 802.11 MAC Layer

At MAC layer, IEEE 802.11 defines two medium access functions, Distributed Coordination Function (DCF) and Point Coordination Function (PCF). The DCF function provides a contention-based transmission and it is implemented in all the WLAN stations allowing them to control the medium access with no central coordination. DCF operates based on Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) protocol. In contrast, the optional PCF function is a centralized polling-based access protocol that supports contention free transmission. PCF is implemented in the AP, which manages the access to the medium by sending a poll message to each station giving it a transmission opportunity. Although PCF was designed to support real time traffic, it is not commercially implemented in AP [25] for its complex and inefficient polling schemes and QoS provisioning limitation [70, 71]. Hence, DCF is considered in this study as it is widely implemented for the easy implementation, less complexity and small overhead features it has [72].

Furthermore, in CSMA/CA protocol, when a station wants to transmit a packet, it first senses the medium to determine if it is idle or busy. When the medium is sensed idle for a time interval equals to DCF Inter-Frame Spacing (DIFS), the station starts the transmission immediately. However, if the medium is sensed busy, it stops the transmission and waits until the medium status becomes idle. As the medium is sensed idle, it waits for an additional period of DIFS. After the DIFS time is expired, the station also waits for a random time interval called back-off period. The back-off duration is a number of time slots; time slot is a fundamental unit of time in 802.11 PHY standards. The slot time is the minimum time wherein a station can detect transmission from any other station in the network and the number of these slots is determined by a back-off counter (BC). BC is uniformly distributed between a range of [0, CW], where CW refers to Contention Window and it is initially set to CWmin. The BC decreases the number of slots by one only when the medium is detected idle, otherwise it stops the counting down and continues to decrease the slots from the previous value when the medium is sensed idle again. As the back-off time reaches zero, the transmission begins. In fact, when the back-off period reduction pauses till next time the medium is sensed idle, it guarantees priority for the stations that were competing for access over the ones have just started to compete. Furthermore, during this process of medium access, the back-off timers and DIFS achieve the collision avoidance mechanism.

According to the acknowledgment scheme that 802.11 standard employs, an ACK packet is sent by the receiver station as a notification of a successful packet reception. Therefore, upon receiving the data, the receiver waits for a period of Short IFS (SIFS) time interval before transmitting the ACK packet. The SIFS interval is smaller than DIFS interval as SIFS transmission has a higher priority. However, if the transmitter does not receive the ACK packet, it will keep resending the packet until it gets an ACK packet or drops the packet when the retransmission threshold is met. Moreover, after every unsuccessful transmission, the CW value is doubled till it reaches a value of CWmax and when the transmission is succeeded it will be reset to CWmin. The values of CWmin and CWmax are identified by the PHY layer standard. Figure 2.9 illustrates a basic operation of the DCF function.

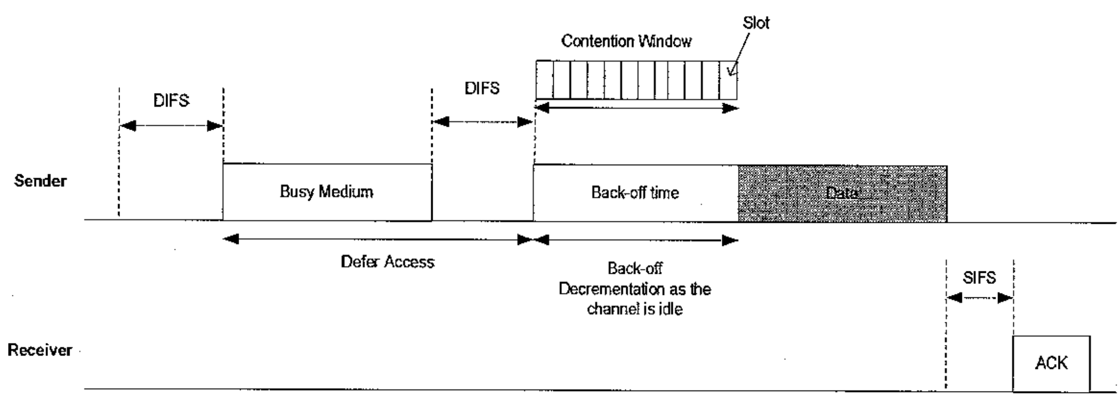


Figure 2.9: IEEE 802.11 DCF Function of MAC Protocol

During the stations' communication, not all the station nodes in the ESS wireless network can hear each other causing the challenges of hidden-node and exposed-node. The hidden-node can occur when different nodes from two different BSSs want to communicate simultaneously with a node that exists in between the transmission ranges of both BSSs. Referring to Figure 2.10, there are three different BSSs, each with a transmission range shown in a circular fashion. Assuming that node A senses the medium idle and starts the communication with node B and at the same time as node C wants to transmit data to node B, node C will sense the medium idle as well if no transmission happening in its BSS. Also, node C cannot hear the transmission going between node A and node B since node A is hidden to node C. Consequently node C starts the transmission to node B causing collision. The other issue, exposed-node problem, arises when node C wants to communicate with node D while node B is transmitting data to node A. when node C senses the medium, medium will appear busy to node C since node B is communicating with node A. Therefore, the transmission of node C is deferred unreasonably.

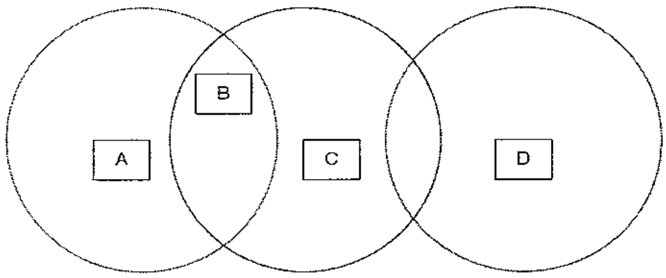


Figure 2.10: Hidden-node and Exposed-node Issues

In order to solve these two issues, 802.11 standard defines the collision avoidance or RTS-CTS (request to send - clear to send) mechanism. In this mechanism, the channel is reserved by exchanging control packets between the sender and the receiver for a period that it requires to transmit the data. As the sender node gets access to the medium, it sends RTS packet to the receiver containing the time it will need to use the channel. In response, the receiver will send CTS packet that contains the time it will reserve the channel too. Accordingly all other nodes in both the sender BSS and the receiver BSS will set an indicator called Network Allocation Vector (NAV) to the time that the channel will be busy. Although this proposed scheme helps to reduce the probability of collision occurrence, it still has its limitations which are out of this study scope.

2.2.2 VoWLAN Challenges: QoS and Capacity

The transmission of voice packets over WLAN poses several challenges essentially QoS and capacity [73]. The QoS and capacity issues are correlated; when there is enough capacity in the network to support more calls, the quality will improve. The capacity can be defined as the maximum number of simultaneous calls that the network can support. According to [74], WLAN network can actually support more than 100 VoIP calls at the same time sharing the bandwidth with other types of data. However, in reality WLAN network supports only about 12 calls. The reasons behind that are various factors that affect the performance of VoIP:

- The added headers overhead to VoIP packets in various network layers: the total size of headers added to each voice packet is 74 bytes exclusive the PHY layer headers. Generally, the size of voice packet is smaller than the size of the attached headers overhead, which affects the efficiency of the transmission. As calculated in [74], for a packet of size 1000 bytes with a transmission rate of 11Mbps, the transmission efficiency dropped to 46.5% due to headers overhead. Thus, the additional headers overhead added to voice packets has an impact on the transmission of VoIP.

- The inefficiency of WLAN MAC layer [75] and DCF mechanism overheads: DCF overhead plays role in reducing the available bandwidth, which leads to decreasing the number of calls [76]. Also, its implementation of ACK scheme where an ACK packet is sent after every successfully received packet affects the system capacity and performance [77].

- The fairness feature of MAC: another issue that affects the system capacity is the fairness among all WLAN nodes. WLAN station nodes and AP implement the same MAC layer functions including the medium access mechanism, thus they all have the same opportunity to access the medium. In spite of the fact that AP node holds more traffic than other nodes in the system, there is no priority given to AP node, which could cause network congestion [70]. This MAC fairness feature works for data applications but not for real time applications whose uplink and downlink traffic require almost the same bandwidth [78].

- The packet size: this parameter has an impact on the quality as well as the system effective bandwidth as analyzed earlier in Section 2.1.4.

- The transmission data rate: different transmission data rates can support different number of calls and obviously the higher transmission data rate, the more number of calls.

IEEE 802.11 lacks of supporting QoS for real time applications. Accordingly, it is enhanced to IEEE 802.11e standard [79] with the aim of supporting QoS. This support is achieved by classifying the traffic of applications into categories with different access priorities according to their QoS requirements. Its MAC operates based on a new introduced function called Hybrid Coordination Function (HCF), which supports upward compatibility with the DCF and PCF functions [7]. The HCF function enhanced the DCF function into Enhanced DCF (EDCF) that classifies the different internet applications into four access categories (AC) namely voice, video, best-effort, and background, where each class uses its different queue. Although this technique intends to support the QoS of real time applications to a certain level by applying the traffic prioritization scheme, it is still not adequate enough to support the strict QoS requirements of VoIP applications particularly in congested networks, nor it solves the issue of packet transmission overhead [76, 80].

Since 802.11e mechanism cannot avoid network congestion, a new scheme called Call Admission Control (CAC) is proposed to enhance the performance of VoIP. This scheme attempts to prevent the network congestion from happening as it rejects new calls to be connected if the QoS is not guaranteed [81]. Moreover, it ensures that a new call is allowed only if its QoS is guaranteed and the QoS of existing calls is not affected [82]. The CAC schemes can be categorized into different classes that differ by the parameters they based on the CAC decision such as the measurements of traffic and network status, MAC parameters, etc. Details of these classes can be found in [82]. Each method has it is own advantages and drawbacks that some other methods attempt to address. Yet, CAC scheme still cannot meet the strict requirements of VoIP applications especially for the wireless network, which differs from the wired LAN network in its unique features [83]. Besides, the CAC mechanism intends to admit or reject new calls based on the rules it defines but it does not maintain or improve the QoS of those already connected calls if any changes occurred in the dynamic WLAN environment.

2.2.2.1 *VoWLAN Multi-Rate*

In multi-rate WLAN networks, the transmission data rate of a mobile node is dynamically changing by LA technique according to the channel state in order to lower the bit error rate and efficiently utilize the radio resources [15]. LA techniques perform this adaptation based on different measurements such as packet error rate or signal strength that can be affected by different factors such as the user moving away from the AP. As a node moves further from the AP and enters a lower coverage area, the traveling signal strength decreases causing the PHY layer to select a lower transmission data rate [84] in order to reduce the error rate and protect the transmitted packets. Therefore, the PHY layer of the node selects a lower data rate with which it transmits longer packets to carry the same amount of data that would be sent from near the AP, and also with the same amount of headers overhead. Consequently, the node transmits from low coverage area will increase the load in the network reducing the bandwidth availability. As a result, the performance of other nodes with high transmission data rate is also affected. Therefore, transmission data rate reduction of one node causes all other active calls to face quality degradation affecting the whole system performance [85].

Normally this issue is solved by blocking the call of the node suffers rate variation (using CAC mechanism) resulting less number of calls in the network. Nevertheless, this is not an optimal solution and, hence, an optimal solution is required to reduce the effect of multi-rate feature of WLAN on VoIP systems. The solution should aim at improving the QoS of VoIP as well as maintaining if not increasing the network capacity for more number of calls. Consequently, several studies focus on proposing solutions, for instance voice rate adaptation, to alleviate the multi-rate effect on VoIP performance and prevent blocking the calls.

Voice data rate adaptation is a technique where the CODEC/ CODEC parameters are adjusted according to the network condition. Basically, adapting voice data rate at the Application layer according to the transmission data rate is claimed to be effective since no specific CODEC/CODEC parameter is suitable for all WLAN network conditions and this technique allows the transmission of dynamic voice data rate with different bandwidth consumptions. In view of that, sharing information between MAC and Application layers is required for the implementation of data rate adaptive technique. The Application layer should be able to interact with MAC layer to adapt its parameters according to the network conditions detected at MAC layer. This could happen by applying Cross-Layering approach. The Cross-Layering approach has been widely applied to enable the exchange of information between the different network layers.

2.3 Cross-Layering Approach

Although the layered architecture of the network model is designed for wired networks and it serves that purpose well, it is still not efficient enough for wireless networks [86]. In this layered architecture, each layer handles specific functions independently in the network. However, due to the dynamic environment of wireless networks, wireless networks face different issues than LAN networks. In order to address the issues of wireless networks, involvement of different layers in the network is sometimes required. Therefore, the complex layered structure of network has an effect on the performance of wireless networks. Hence, it is not sufficient enough to support the strict QoS requirements of real time applications in particular VoIP transmission over WLAN.

Consequently, the Cross-Layering approach has been introduced to improve the transmission of real time applications over wireless networks. The concept of this approach is that it allows neighboring or non-neighboring network layers to communicate and exchange information with each other to achieve better network quality. Non-adjacent layers interaction generally can be categorized as [87, 88]:

- Bottom-to-top: lower layers provide information to higher layers that would help the latter to adjust its parameters.
- Top-to-bottom: higher layer's information is available to lower layers to adjust its functionality accordingly.

Studies and surveys show that the Cross-Layering approach has a great impact on improving the performance of real time transmission. Accordingly, there are several research works that have been conducted applying this approach to improve the quality of real time applications over WLAN networks, for instance the proposed cross layer adaptation technique for video applications in [89], where the cross layering happened between the Network and the Application layers. The work of [90] presented adaptive source-network rate control scheme (ASNR) for VoIP applications in Ad Hoc environment. This scheme adapted the coding rate based on the network bandwidth. Another Cross-Layer framework was proposed in [16]. It was based on top-bottom category in which controlling the transmission data rate at MAC layer was adapted according to information of the traffic type extracted from the Transport layer. Furthermore, the Cross-Layering approach has been also applied in addressing the multi-rate issue as will be seen in the next section of related work.

2.4 Related Works

Several research studies have been carried out to address the multi-rate issue and its effect on the QoS of VoIP. Most of these studies have applied CODEC adaptation scheme, where the voice rate is adapted by changing the type of VoIP CODEC. Authors in [15] proposed Link Adaptation Codec Adaptation Mechanism (LACAM) that aims to mitigate the congestion problem that LA mechanism can cause by using the CODEC adaptation scheme. Once a change in the transmission rate is occurred, the LACAM algorithm starts to operate with checking if the LA has caused WLAN network to become congested. Due to the difference in delay values before and after congestion, the delay parameter (measured by jitter value) is used to detect network congestion. If network congestion is detected, the algorithm then decides to adapt a different CODEC. The performed CODEC selection technique is based on CODECs' channel occupancy times. When there is a CODEC adaptation required, the algorithm will change the current CODEC to another, which has the closest value of channel occupancy time. Different CODECs with various packetization intervals and their channel occupancy times are listed in a lookup table for the algorithm's reference when selecting a new CODEC. The algorithm simply works only with delay. It also focuses its studies on the downlink statistic since download streams suffers more delay than uplink streams when the network exceeds its capacity limit. Furthermore, it does not proactively response to the change of transmission data rate since it starts the process with checking the effect of LA technique on the network before performing any adaptation.

The work in [18] developed an adaptive Multi-Rate VoIP (MRV) scheme that optimizes the Cross-Layering approach as well. In this scheme, uplink and downlink streams are studied, where wireless stations are communicating with remote stations. The MRV mechanism applies a function in the wireless station that obtains the transmission data rate information from PHY layer and accordingly changes the CODEC rate and packetization interval at the Application layer. The mechanism also incorporates a function called media gateway (MGW) with the AP to translate any changes to the CODEC rate to the initially specified CODEC rate between the two connected stations. However, MGW function adds more delay and overhead for the rate translation process at the AP and no synchronization applied after the translation. Moreover, MRV scheme still suffers packet loss and delay under small fluctuated SNR intervals when the traffic load becomes high (maximum calls 10). Furthermore, the MRV algorithm was extended to work with the existing of best-effort traffic using IEEE 802.11e standard [91]. The simulation of this study has shown that applying both IEEE 802.11e standard and MRV mechanism results better quality of VoIP and higher data traffic throughput under fluctuated wireless channel.

With the aim of minimizing the effect of multi-rate issue on the performance of VoIP and avoiding dropped calls that suffer from transmission rate variation, Sfairopoulou et al. [92, 93] proposed a new algorithm called Codec Adaptation Algorithm (CAA) that changes the CODEC type of those nodes caused the multi-rate issue and of some other nodes if required. The CAA algorithm utilizes information gathered from MAC layer about the transmission data rate and from RTCP reports about VoIP quality parameters such as delay, packet loss and jitter. The algorithm operates based on three phases that starts with monitoring phase, where the predefined information is collected and observed. If a change in the transmission data rate or degradation in one of the QoS parameters is detected, the algorithm goes to the next phase, which is adaptation phase. In adaptation phase, the CODEC type is changed to prevent the happening of network congestion. The last phase of CAA, recovery phase, observes the network state after performing the CODEC adaptation to ensure that applying a new CODEC improves the system performance. The authors examined the algorithm in both distributed and centralized modes. Although the distributed mode is easier to be implemented since the algorithm is implemented in every node and every node independently adapts its CODEC according to the algorithm, there will be more number of nodes than required that go through the CODEC adaptation. However, the load of the algorithm process is distributed among the nodes. Alternatively, if the algorithm is installed in the AP, it is in a centralized mode in which the AP monitors all nodes' transmission rates and CODEC types and makes decision on how many and which nodes to adapt their CODECs. Although the centralized mode provides better network monitoring, it could be more difficult to implement and it adds extra processing work and overload to the AP node.

Another adaptation mechanism called Scalable Speech/Audio Coder Control (SSACC) was proposed in [94]. The SSACC controller simply utilizes and controls the G.729.1 CODEC, which operates in 12 different bit rates, by switching its bit rate from one to another according to the network condition. The controller consists of two modules inserted to the AP functionality and another two were placed in the wireless nodes. The SSACC scheme operates by transmitting periodically beacon frames that carry information of media access delay time and number of beacon frames (NoB) from the AP to the stations. Accordingly, the network stations would detect if the AP gets congested and therefore adapt the CODEC rate based on the NoB. This scheme is found to have no major functionality of the component "Cross Layered Monitoring" that is added in the wireless node as it only receives network condition parameters from MAC layer and delivers them to another component at the Application layer to adjust the CODEC bit rate if required. Although it applies the CODEC rate adaptation scheme, it does not address the multi-rate issue and the RTCP protocol is not utilized. The scheme also requires hardware modification at MAC layer. Furthermore, it measures only the media access delay and VoIP E2E delay parameters.

Applying the adaptive CODEC rate technique using Cross-Layering approach was also conducted by the author of [95] who designed a cross layer mechanism that links between IEEE 802.11e MAC and Application layers. At the Application layer, the proposed algorithm dynamically switches between CODECs according to information gathered at the MAC layer. Basically, the adaptation is based on both transmission data rate and network load, where it calculates if the network channel can support the current CODEC else it changes it to another CODEC. The simulated network consists of only two VoIP nodes and eight other nodes that generate background traffic and the system evaluation is performed using only two CODEC types. The algorithm, however, does not evaluate the system considering more number of VoIP calls and it also adds a calculation overhead at the MAC layer. Moreover, it does not utilize the RTCP protocol.

In order to maximize the throughput of multi-rate WLAN network, a centralized mode mechanism was proposed in [51]. Its analytical algorithm operates according to the state of the wireless channel whether it is an idle channel or an error prone channel. The proposed algorithm applies CODEC and frame size adaptation technique based on information obtained from both MAC layer and RTCP feedback. The algorithm decision of the number of CODECs and/or frame sizes to be changed is made according to the network capacity if it has reached its limit or not. Additionally, the algorithm adds an additional phase to change the frame sizes of different nodes if the channel is an error prone channel. Although, the study focuses on the throughput results of the algorithm, there are no results obtained for the QoS parameters that evaluate the quality of VoIP.

Although these studies attempted to address the multi-rate issue, they still have their drawbacks as illustrated in Table 2.8 that summarizes the related studies to this research work. Besides, they focus on utilizing the CODEC adaptation scheme. This scheme requires VoIP system to have more than one type of CODECs with different compression ratios. However, CODECs with high compression ratios are not free as they are licensed. Therefore, the VoIP user has to purchase more than one CODEC for such approach to be implemented. More to the point, as the compression ratio of a CODEC increases, the VoIP quality decreases. Hence, this study attempts to look at a different parameter that would change the voice rate according to the network condition without charging the user for more CODECs. As found from the literature review of this work, the voice packet size can change the voice rate at the Application layer and it has an impact on the network bandwidth requirement. Therefore, this parameter is considered in this study. Furthermore, applying the voice packet size adaptation scheme has the advantage of utilizing the available resources in the network.

Table 2.8: Summary of Related Works

Reference	Description	Drawbacks
[15] (2006)	» It intends to alleviate congestion problem caused by LA mechanism. » When MAC data rate is changed, it checks if congestion has been caused as indicated by increasing the jitter delays. If the congestion is detected, it will adapt the CODEC. » It performs the CODEC selection based on CODEC's channel occupancy time.	» It works with only one parameter: delay. » It does not proactively response to the change of MAC transmission rate since it starts its process with checking the effect of LA on the network before performing any adaptation. » It studies downlink statistic only.
[18,91] (2006 - 2007)	» It adapts the CODEC according to the change of MAC data rate. » A built in MGW function at the AP is proposed that translates any changes in the CODEC rate to the initially specified CODEC rate between the two connected stations.	» It adds more delay for the process of rate translation performed at the AP node and no synchronization is applied. » It still suffers packet loss and delay when traffic load becomes high.
[92,93] (2006,2008)	» Its CODEC adaptation is based on: ▪ MAC data rate changes & RTCP report. » It consists of three phases: monitoring, adaptation and recovery. » It maintains the overall network quality.	» It illustrates a good work as the adaptation is performed based on different parameters. Yet, its process goes through three phases.
[94] (2008)	» It utilizes the G.729.1 encoder, which operates in 12 different bit rates. » It operates by transmitting periodically beacon frames that carry information of media access delay and number of beacon frames (NoB) from the AP to the stations. Accordingly, the stations would detect if the AP gets congested and therefore adapt the CODEC rate.	» It requires hardware modification at MAC layer. » It measures delay parameters only. » It adds overhead by transmitting the beacon frames.
[95] (2009)	» It dynamically switches between two CODECs according to calculations that made based on information gathered at MAC layer. » The adaptation is based on MAC data rate and network load.	» The network topology is very simple as it consists of only two VoIP nodes. » It adds a calculation overhead at MAC layer.
[51] (2010)	» It operates according to the state of the wireless channel. If it is idle, CODEC adaptation scheme is applied based on information obtained from both MAC and RTCP report. » The decision of number of CODECs and/or frames to be changed is made according to the network capacity if it has reached its limit or not.	» It does not show measurements for QoS parameters as it focuses on throughput parameter only.

2.5 Summary

In this chapter, VoIP system and its main components and protocols were presented showing the process of transmitting voice packets over IP networks. VoIP CODEC and its packet size parameter were also elaborated and analyzed. On the other hand, QoS is an important factor for the success of VoIP. Therefore, the main parameters of QoS including delay, jitter, packet loss, and throughput were discussed. The chapter also presented E-Model technique that is used to evaluate VoIP quality. Moreover, the chapter covered IEEE 802.11 standard of WLAN, its architecture and its main specifications of PHY and MAC layers. It also explained the challenges that VoIP over WLAN faces and highlighted the Cross-Layering mechanism that is being recently used to address WLAN issues. Eventually, the chapter presented the related works to this research study, which attempt to address the multi-rate issue.

CHAPTER 3

RESEARCH METHODOLOGY

This chapter discusses the methodology applied in order to achieve the objectives of this study. It is mainly divided into two parts; the first part presents the proposed CLAA agent, its model, and its functions. The second part covers the simulation study including the simulation tool and the system configuration steps.

3.1 Introduction

As illustrated in Figure 3.1, the research starts with studying the rate adaptive technique, the Cross-Layering approach, which is used in developing the rate adaptive technique, and the effect of packet size on VoIP as it is the key parameter of this study. Accordingly and based on the conducted literature review, CLAA agent has been developed intending to improve the QoS of VoIP over a multi-rate WLAN network. For the purpose of validating the proposed CLAA agent, this study intends to perform simulation work. Therefore, the comprehensively studied simulation tool is used to setup the WLAN network topology of this research and the VoIP system operating on this network. It is also used to perform a preliminary analysis study that helps to define a range of packet sizes for CLAA algorithm to apply when making its adaptive decision. Finally, the results of simulation runs are collected, analyzed and discussed in the subsequent chapter.

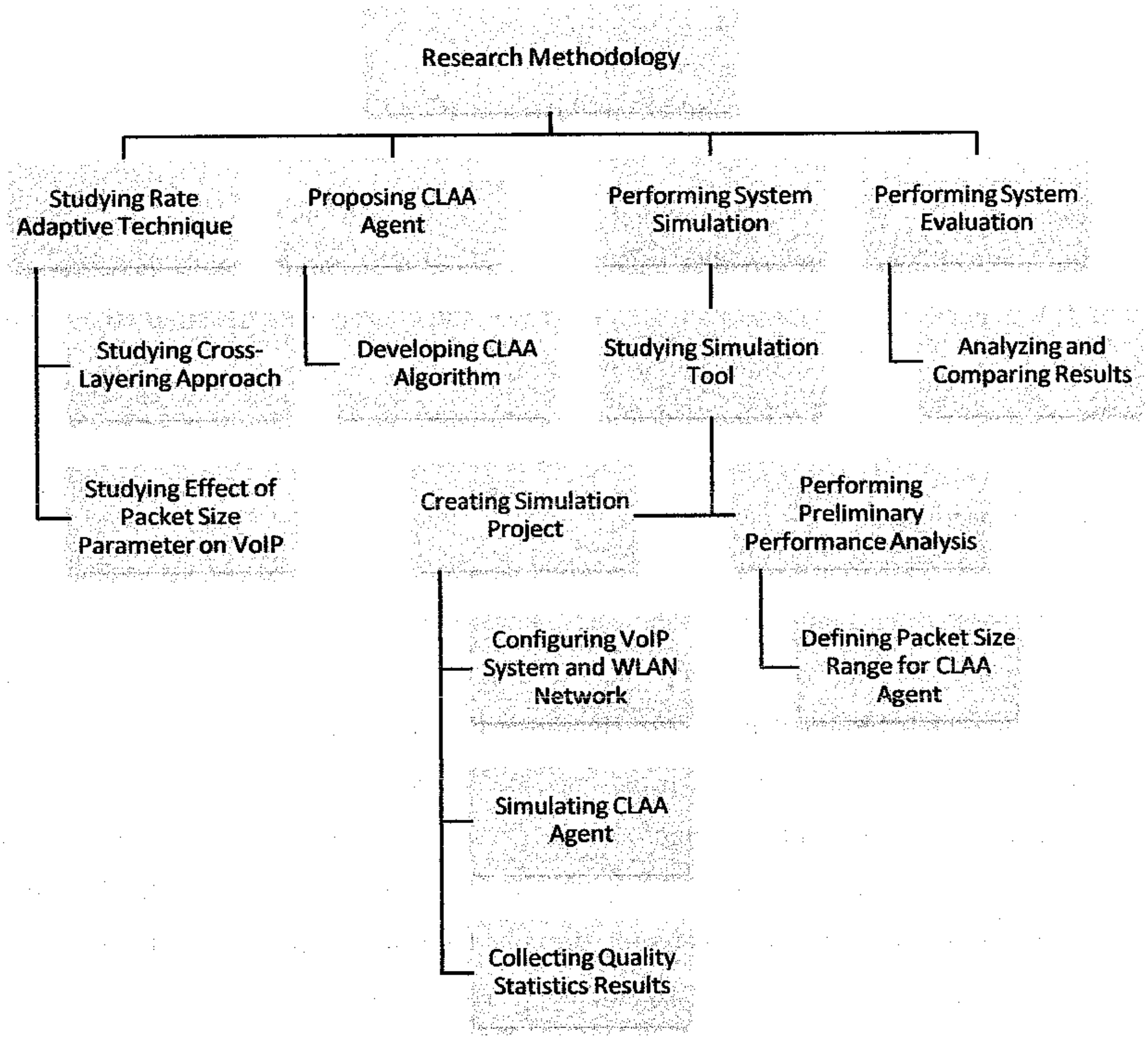


Figure 3.1: Research Methodology Model

3.2 Rate Adaptive Technique

Since the transmission of packets at constant rate does not work well in all network conditions [35], developing rate adaptive techniques e.g. CODEC adaptive technique has been introduced for real time applications. To define the rate adaptive technique, the CODEC or CODEC parameters are adjusted according to the network condition so that the voice rate is lowered when the network is getting congested. Therefore, it allows the transmission of voice rate with different bandwidth consumptions. Additionally, research studies have shown that applying rate adaptive schemes perform better than the transmission at a constant rate [96] [97].

Generally, the CODEC adaptation technique has been effectively implemented in the wired networks particularly to deal with network load parameter [10]. However, it is still a challenge in the wireless network that suffers from high overhead associated with packets, which reduces the network bandwidth availability [10]. Although the development of this technique for wireless networks is in its early stage, there are several mechanisms that have been proposed mostly for real time applications. Furthermore, it was suggested that adapting the parameter voice packet size in rate adaptive technique is satisfactory with keeping the same CODEC type in use for better VoIP quality in congested links of WLAN networks [98].

Basically, the rate adaptive technique is implemented at the Application layer and it operates according to the network condition. It utilizes the fact that the lower layers of the network are able to identify the network state. Therefore, the Application layer should be able to interact with the lower layers and adapt its parameters according to the network conditions as detected by them. For such interaction to happen to achieve the development of rate adaptive technique, Cross-Layering approach has been widely applied because it allows the sharing of information among the different network layers. Hence, this study applies the rate adaptive technique with utilizing the information of MAC data rate and RTCP QoS parameters using the Cross-Layering approach. It focuses on adapting the CODEC parameter 'packet size' to the changes occur in the multi-rate WLAN network in order to improve the performance of VoIP.

That is for the reason that adapting the voice packet size to the network condition is an important parameter and would produce results of better quality [41].

3.3 Cross-Layer Adaptive Agent (CLAA)

Generally in VoIP systems, it is configured to apply one of the available CODECs with a fixed packet size for the encoding and packetization process. This parameter with the same size will be used till the end of the call without being changed even though the network condition is changing causing variation in the network bandwidth. This variation of network affects the quality of VoIP calls since the packets consume the same network bandwidth while the available network bandwidth keeps changing. Therefore, it is logic to vary the bandwidth requirements of voice packets when the network bandwidth availability is changed, and this reason was the driven force behind the development of rate adaptive techniques.

Due to the fact that packet size is a key parameter in determining the required amount of network bandwidth for the transmission of voice packets [37], as explained in the preceding Chapter 2 Section 2.1.4, adapting the packet size parameter using rate adaptive technique can play role in solving the variation issue of network bandwidth. Hence, in order to achieve the research objective of improving the QoS of VoIP calls by mitigating the issue of multi-rate in WLAN network that causes the variation of bandwidth, a Cross-Layer Adaptive Agent (CLAA) is proposed that applies packet size adaptive technique. The CLAA agent is an entity that runs based on the Cross-Layering approach allowing the communication between MAC and Application layers. It basically highlights the positive impact of voice packet size on the performance of VoIP calls and the importance of changing this parameter at the Application layer in a dynamic manner. The adaptation of voice packet size happens according to the variable network condition in order to minimize the channel congestion in WLAN network and hence reducing the effect of channel congestion on the quality of VoIP calls. As the model of CLAA agent demonstrated in Figure 3.2, the agent is located between the two targeted network layers for communication, which are MAC and Application layers so that the required information from MAC can be shared with the Application layer to adapt the voice data rate accordingly.

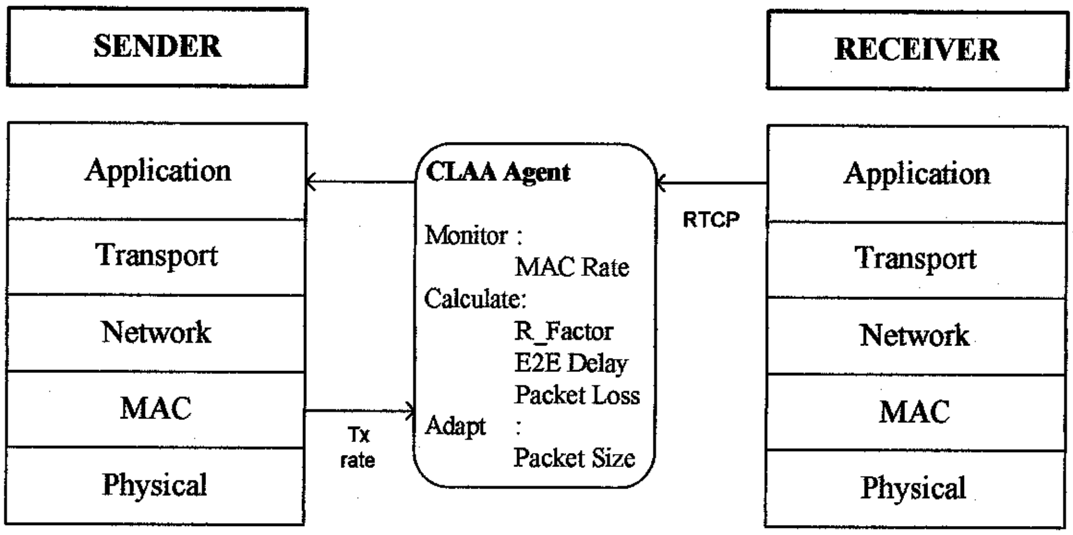


Figure 3.2: Cross-Layer Adaptive Agent (CLAA) Model

Generally, the main function of CLAA agent is to continuously monitor and observe MAC layer to detect if there is a change in the transmission data rate caused by LA technique. If so, it will communicate with the Application layer using Cross-Layering approach to inform it of the transmission data rate change at MAC and whether this rate change is to a lower or higher rate. In response to the received information, the Application layer will perform rate adaptive function accordingly and increase or decrease the voice packet size to reduce the effect of multi-rate situation occurred in the network. Furthermore, the agent function is also to monitor the state of the call quality from the generated RTCP reports at the callee side, which help in calculating some of the quality statistics of VoIP. From these calculated statistics, the agent can observe degradation in the call quality. If the condition of call quality is noticed to be getting worse, then the agent requests the Application layer for more adaptation in an attempt to improve the quality condition. Further details on the functions of CLAA agent will be explained in the next Section 3.3.1.

3.3.1 CLAA Agent Operation

As illustrated in Figure 3.3, CLAA agent starts operating by monitoring MAC layer to detect any transmission data rate changes that could happen. The agent reads the current transmission rate (Curr_Tx_R) of MAC and compares it with the previous transmission rate (Prev_Tx_R). If it finds their values are not exactly the same, a transmission data rate change is detected. Subsequently it will check if the current transmission data rate is lower or higher than the previous transmission data rate. In the former case where the transmission data rate is lowered, the network bandwidth availability is reduced due to the fact that the transmission with low data rate consumes more bandwidth. Therefore, the agent will inform the Application layer to increase the voice packet size so that the bandwidth requirement is reduced. If this reduction was not enough to improve the quality of VoIP, it will be detected and improved by the RTCP observation process, where the agent keeps monitoring the call quality through reading the RTCP reports and adapts the packet size of the calls with low quality. Moreover, to recover the network performance, it will be also required to adapt some of the other effected nodes in the network besides adapting those nodes caused the multi-rate issue.

However, if the transmission rate went up and the packet size is not lower than the lowest packet size, it will reduce the length of packet size to utilize the increased amount of the available network bandwidth. Following this adaptation, the agent calls a retrieve state function. In this function of the algorithm, it will search the network for a node that has the lowest packet size that is lower than 5 to increase its packet size and a node with the highest packet size, which is higher than 5 to decrease its value. In this function, the agent attempts to return the nodes' packet sizes to their initially configured packet sizes since the network state is improving. Furthermore, this quick adaptation response to the transmission data rate changes shows the agent attempt to prevent the effect of multi-rate network on voice performance. Besides, the agent operates fairly since it first starts the adaptation process of voice packet size for the nodes that cause the multi-rate issue. If the network condition still requires further adaptation, only then it will adapt the voice packet size of other nodes that are also affected in the network.

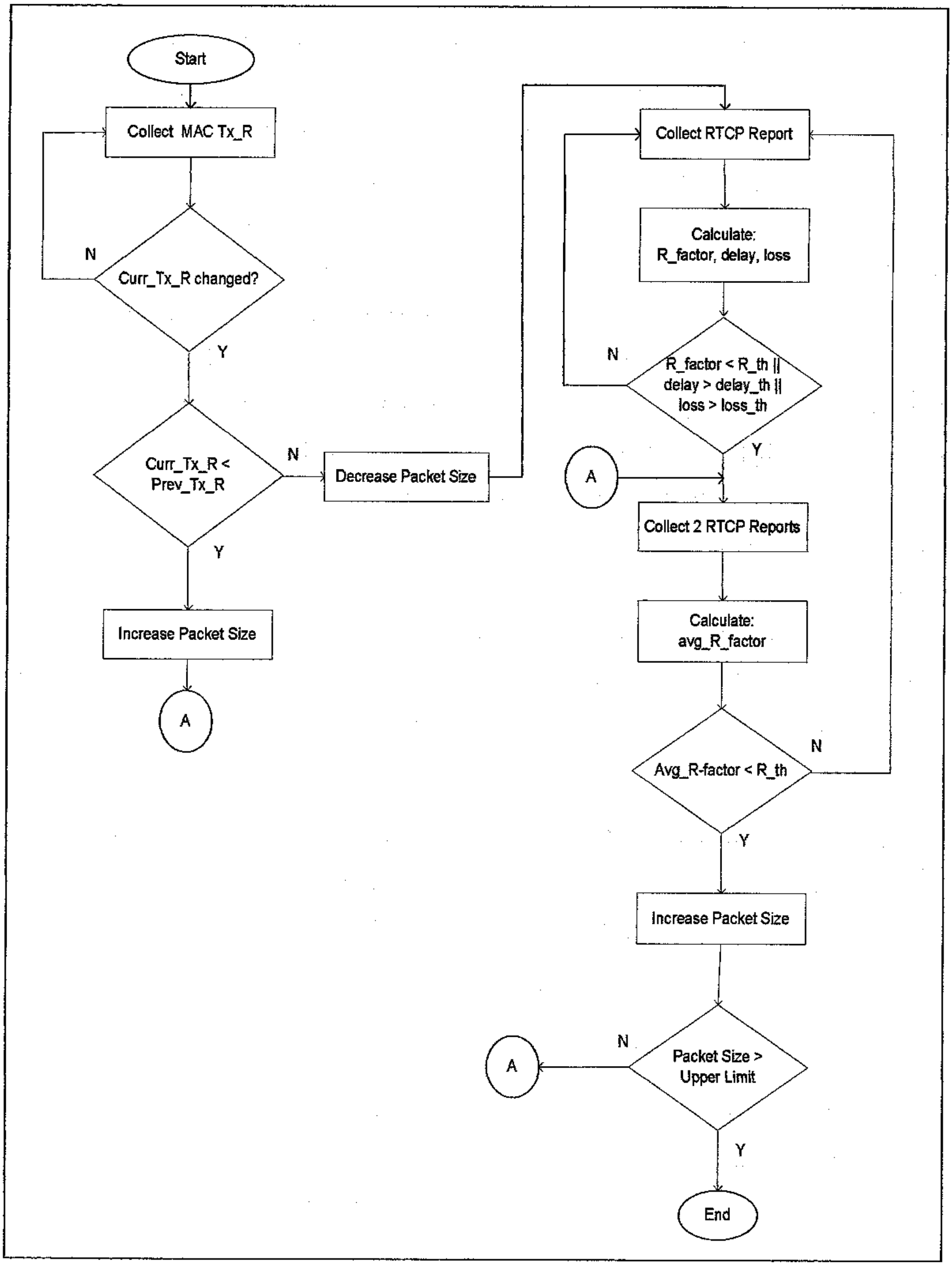


Figure 3.3: CLAA Flowchart

As explained earlier, increasing the packet size would consume less bandwidth since it reduces the bandwidth requirements of the headers overhead, thus reducing the total traffic load in the network. Therefore, it is better to increase the voice packet size when the available bandwidth is decreased. In contrast, a small packet is selected when the bandwidth availability is increased for the reason that the transmission of small packets requires more bandwidth for the higher accumulated overhead ratio. On the other hand, a range of voice packet sizes should be carefully defined for the agent to select from during the adaptation process. That is due to the fact that enlarging the packet size will increase the packetization interval, which adds delay to the total network delay while producing packets of small size will decrease the packetization interval, hence reducing the delay though it consumes more bandwidth. Therefore, an analysis study on the performance of VoIP network with different voice packet sizes has been carried out in order to determine the optimum range of voice packet sizes to be used by the agent when required. In this analysis study, a set of preliminary simulation runs were performed examining the different voice packet sizes. It aimed to find the lowest and highest packet sizes, which can help defining the optimum range of packet sizes that does not adversely impact the network performance. Further details on this analysis study of VoIP performance with different packet sizes will be explained in Section 4.2.

Besides the function of CLAA agent that monitors MAC layer, it simultaneously reads the periodically generated RTCP reports. From these reports, it calculates VoIP quality parameters including delay, packet loss and R-factor. The R-factor parameter is calculated using E-model technique as explained in Chapter 2 Section 2.1.6. Unlike subjective measurement techniques, the objective computational E-model technique has been utilized in this agent because it can be computed quickly and in real time. During the observation of RTCP reports, if the agent detects network quality degradation by finding at least one of the monitored voice quality parameters fell below its threshold (R-factor < 70, delay > 150, and packet-loss > 3%), it calls for adaptation. In this adaptation process, the agent collects more than 1 RTCP report for a better evaluation of the network condition, 2 RTCP reports, with short interval (1 second). It then calculates the average of R-factor and compares it with its threshold value (R_th), which is set to 70 as recommended by the ITU-T. If the average value is found to be equal or larger than the threshold value, the network quality is observed to be in a good condition. Therefore, the agent does nothing but loops back to the start to monitor the next generated RTCP reports. Conversely, if the calculated average R-factor value is smaller than its threshold, this means that the network is congested and the quality of VoIP calls is observed to be degraded. As a result, the agent requests the Application layer to increase the voice packet size so that the network load does not increase when it is congested.

Based on RTP standard, the RTCP reports are generated within an interval period of 5 seconds to ensure that the RTCP reports do not consume more than 5% of the network bandwidth as mentioned earlier in Chapter 2 Section 2.1.2.2. However, when the CLAA agent detects quality degradation and applies packet size adaptation, the extended RTCP standard [99] is applied allowing the RTCP reports to be collected within a shorter period of 1 second. This is to ensure a fast response to the quality degradation and to minimize the delay of the adaptation process to 2 seconds for collecting the 2 RTCP reports. As a matter of fact, the extended RTCP standard permits modifying the RTCP transmission interval in order to support adaptation mechanisms that require receiving an immediate feedback.

Furthermore, as the agent detects quality degradation, it will check the average of two collected report values before it triggers the sender for adaptation. Basically, the agent does not immediately request for adaptation when it observes for the first time such quality degradation considering that the sender might have adapted its packet size if its transmission data rate is changed. It is also to ensure that the quality degradation is not an error that can discontinuously occur. Therefore, the agent keeps checking the quality of VoIP call for few seconds to give it some time if it can recover its performance. Moreover, this adaptation process happens in a short time so that the quality degradation can be improved within unnoticeable time to the users.

A descriptive flow chart of CLAA functions is shown in Figure 3.3. Also, the CLAA algorithm is demonstrated in Figure 3.4.

Global Initialization

set r_min = 70

set delay_max = 150

set loss_max = 3

set nodes = number of nodes in the network

CLAA_Agent main algorithm

```
while nodes ≠ 0
        Read mac_rate_now
        if mac_rate_now ≠ mac_rate_previous
          if mac_rate_now > mac_rate_previous
             set new_packet_size = packet_size – b
             call retrievestate_function
          else if mac_rate_now < mac_rate_previous
             set new_packet_size = packet_size + b
             call adapt_function (node_id)
          endif
        else
          do nothing
        endif
        collect rtcp_report
        calculate r_factor parameter
        calculate delay parameter
        calculate loss parameter
        compare parameters with their thresholds
        if r_factor < r_min OR delay > delay_max OR loss > loss_max
           call adapt_function (node_id)
        endif
set packet_size = new_packet_size
nodes = nodes - 1
endwhile
```

adapt_function (*node_id*)

collect two rtcp reports

calculate the average of *r_factor*

if *avg_r_factor* < *r_min*

  set *new_packet_size* = *packet_size* + a

endif

set *packet_size* = *new_packet_size*

---

retrievestate_function

while *nodes* ≠ 0

  Find *lowest_pkt_size*

  Find *highest_pkt_size*

  nodes = *nodes* - 1

endwhile

if *lowest_pkt_size* < *initial_pkt_size*

  set *new_packet_size* = *packet_size* + a;

endif

if *highest_pkt_size* > *initial_pkt_size*

  set *new_packet_size* = *packet_size* - b;

endif

set *packet_size* to *new_packet_size*

Figure 3.4: CLAA Algorithm

As mentioned earlier, the algorithm starts with monitoring the MAC transmission data rate to perform adaptation of voice rate if the transmission data rate changes. Accordingly, the packet size will be decreased or increased by "b" value, which is equal to 2. This is to ensure a fast adaptation response to the new changes happening in the network. Following this adaptation, RTCP report will be collected to observe the adaptation effect on the network condition. From this report, the R-factor, delay and packet loss parameters will be calculated and compared with their threshold values. According to the standard, the value of R-factor threshold is set to 70, the delay threshold value is set to 150 and the loss threshold value is set to 3. If any of these three parameters is found beyond its threshold value, 'adapt_function' will be performed. This function collects another two RTCP reports and observes the R-factor value. If the average value of two R-factor values calculated from these two reports is less than "r_min", which is set to R-factor threshold 70, the packet size will be increased by "a". Since the packet size has been increased by 2 before this process, the variable "a" is set to 1. This is to further adapt the packet size in order to improve the network condition.

Furthermore, as the MAC transmission data rate increases, the algorithm calls the "retrievestate" function. The aim of this function is to return the packet size to their initially set values when the transmission data rates changes to high values. The function looks for the node with the lowest packet size that is lower than '*initial_pkt_size*' value and increases it by "a" value. It also looks for the node with the highest packet size that is higher than '*initial_pkt_size*' value and decreases it by "b" value. The '*initial_pkt_size*' variable is set to the value 5, which is the initial value set to packet sizes of network nodes. During the adaptation process, the packet size decreases only when the MAC transmission data rate increases. However, the packet size may increase two times when the transmission data rate decreases since the "adapt_function" decreases the packet size after it is decreased by the MAC monitoring process. Therefore, the "retrievestate" function does the increment by one while the decrement by two to ensure that the packet size values change to the initial values, which is 5.

The advantage of this algorithm is that it is simple to be implemented as it does not require any hardware or protocol amendments and it makes use of the available information in the network, which are MAC transmission data rate and RTCP reports. Moreover, unlike COEDEC adaptation, applying the packet size adaptation does not require the active nodes in the network to have multiple CODECs. Instead, it requires having only one CODEC in the system.

3.4 Simulation Tool

A network topology could be a large network that consists of many components. The implementation of such real network set-up for network evaluation when conducting a research study could be time consuming, expensive, and complex. Hence, simulation is being widely used by either researchers or network engineers prior to deploying the real networks. There are several network simulation tools such as NS2, OPNET, etc. that have been developed to assist in simulating and modeling the required network of study. In this study, the evaluation of the proposed agent is based on simulation using OPNET Modeler. Although OPNET Modeler is more complicated than open source tools, it provides accurate results [100].

OPNET (Optimized Network Engineering Tool) Modeler is a powerful network modeling and simulation tool that has a wide variety of features and a rich library of models, protocols, graphical results, etc. OPNET tool is based on discrete event and supports modeling different types of networks including WLAN networks. In discrete event mechanism, the system operation is simulated by modeling its events according to the specification of the system scenario. The events are represented as activities or tasks that are performed at specific times. These events are globally scheduled in OPNET in a sequenced list to be invoked and executed one by one according to the sequence.

For developing simulation models, OPNET consists of three hierarchical domains namely Network, Node and Process domains, which can be accessed via four editors; Network editor, Node editor, Process editor, and Parameter editor [101] as illustrated in Figure 3.5. The models that are developed at any level of the hierarchy can be used by other models at a higher level. Besides OPNET advanced simulation capabilities, it also supports data analysis, statistics collection, and system debugging. Furthermore, a graphical interface is provided by OPNET for its users to easily design the simulated models.

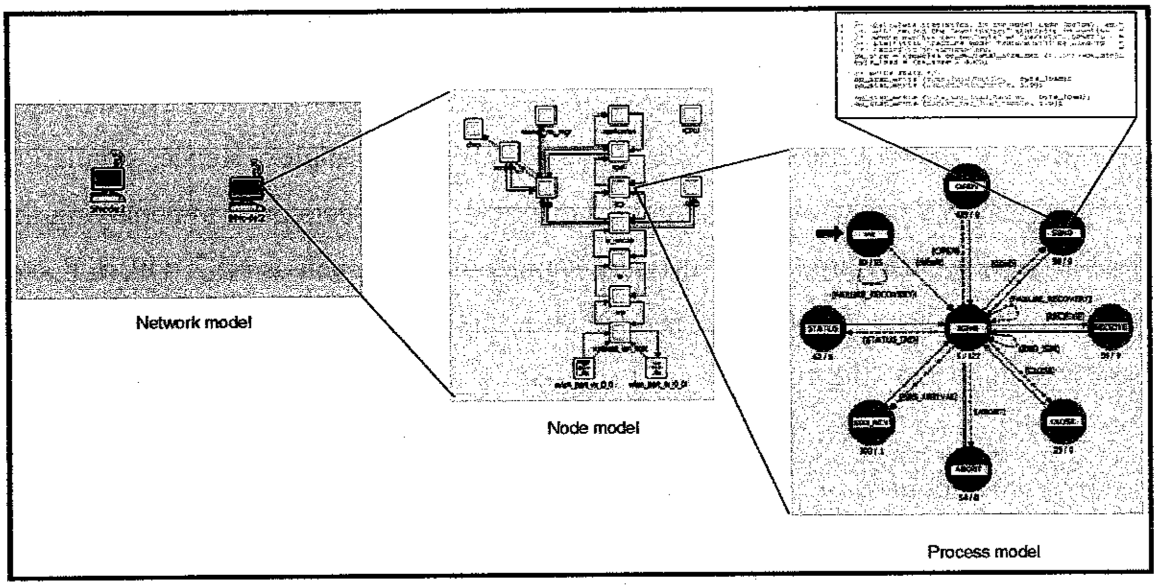


Figure 3.5: OPNET Modeling Domains

3.4.1 Modeling domains

As demonstrated in Figure 3.5, OPNET consists of three main domains:

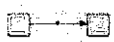
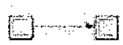
- Network Model

Basically network modeling in OPNET is created using project and scenario approach. Therefore, more than one scenario can be created in a project, which gives an opportunity of exploring a network design with different characteristics as will be utilized in the preliminary performance analysis study of this work. The network scenario is the top-level of the hierarchy domains of OPNET and it is called network model. The network model is the key staging area to develop a network topology that consists of different components for instance links, and node models such as workstations and routers. In addition, the network model defines the nodes' attributes to be configured according to the requirements of the simulated network, the results format, the simulation attributes, and so on.

- Node Model

Node models specify objects or nodes that are designed in node editor and deployed in the network model. A node model is made up of a set of modules, which defines the behavior and functionality of the node. For example, a module may represent a network protocol such as MAC protocol. While some modules are configurable via built-in parameters for instance transceivers, some others are programmable such as processors and queues. Table 3.1 illustrates the main components of node model.

Table 3.1: Node Model Components

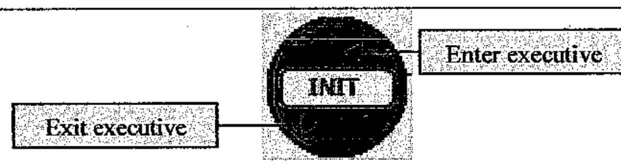
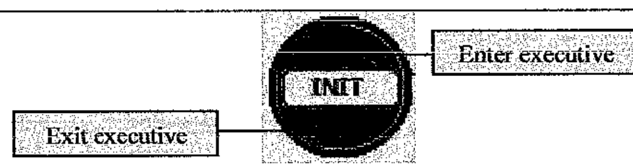
Processors		Programmable to execute processor activities
Queues		Work as buffers
Transceivers		Interfaces between nodes
		
Packet streams		Carries packet between connected modules
Statistic wires		Transfers a numerical signal between connected modules

- Process Model

Process model is the lowest level of the models hierarchy that can be defined as a platform for implementing the programmable modules of node model using processes. It is supported by Proto-C language, which offers several features such as Finite State Machines (FSMs) also known as State Transition Diagrams (STDs). The STDs are represented by graphical and textual formats. In the graphical layout, STDs consists of icons of states and lines of transitions between the states. On the other hand, the textual representation describes the state's operations or actions and it is based on C/C++ language and a powerful library of commands called Kernel Procedures.

Furthermore, the states of STD are also called state executives and each STD state is divided into two blocks called Enter and Exit executives. Generally, there are two types of states: forced and unforced states. The forced state is symbolized by green icon, while the unforced state icon is in red color as shown in Table 3.2. The difference between these two states is in the execution timing. For the forced state, when it is invoked, the process invokes and executes the Enter executive then moves directly to the Exit executive to execute its code. On the other hand, a process that invokes unforced state leaves a mark at the middle of the state after executing its Enter executive code and hands the control back to the simulation kernel, which manages the events executions. As the next event is to be executed, the process starts at the mark of the unforced state to invoke its Exit executive.

Table 3.2: State Transition Diagrams

Forced state	
Unforced state	

3.4.2 Simulation Setup in OPNET

3.4.2.1 Developing Network topology

The WLAN networks are implemented in many organizations, universities, etc. As a realistic scenario for this research study, it is assumed that VoIP services are provided over a WLAN network of an organization or university department for its employees to communicate with each other. To simulate this network in OPNET, the standard 802.11 WLAN is one of its built-in network models that contain several node models including WLAN Workstation Nodes (WNs), WLAN Station Nodes (SNs), routers, etc. Each node model is developed of several process models that represent the different protocols of WLAN network. The key protocol MAC of the standard is represented by the process model wireless_lan_mac that defines the functions and features of the standard in the STD and Proto-C representations. Figure 3.6 illustrates the process model STD representation of MAC based on its IEEE 802.11 standard specifications as explained earlier in Section 2.2.1. In addition, for configuring the WLAN network, while some of the WLAN parameters are assigned their values at the lower-layer of the process model Proto-C text as shown in Figure 3.7 (a), some others are given their values from an attribute window of the node at the network model as shown in Figure 3.7 (b).

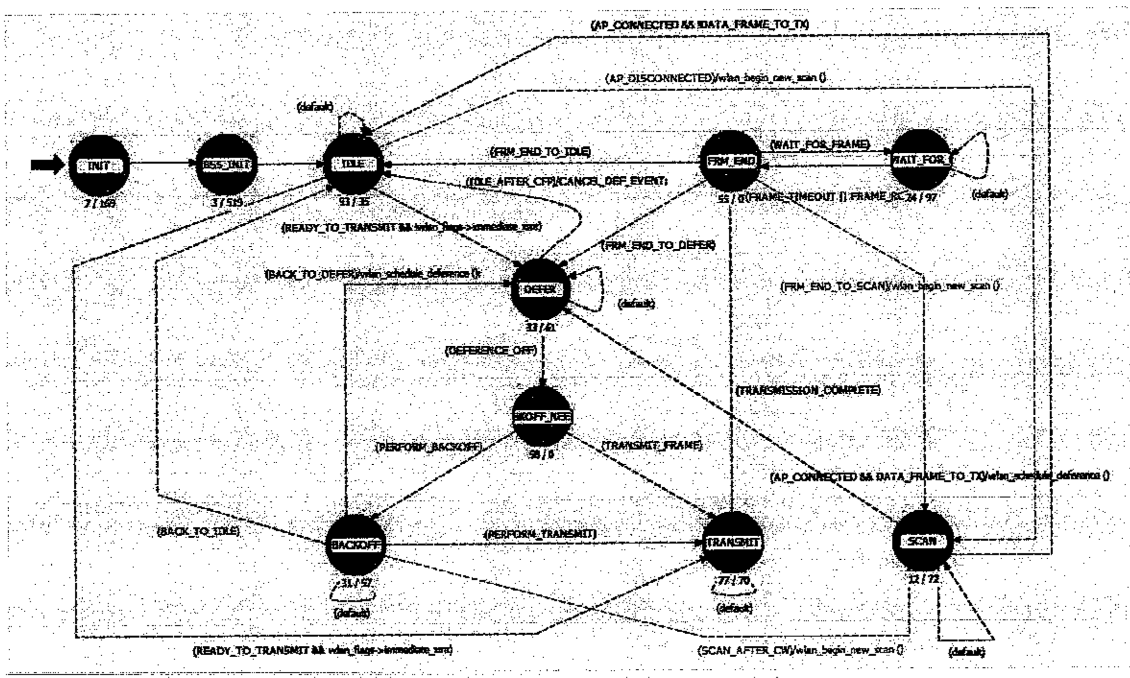
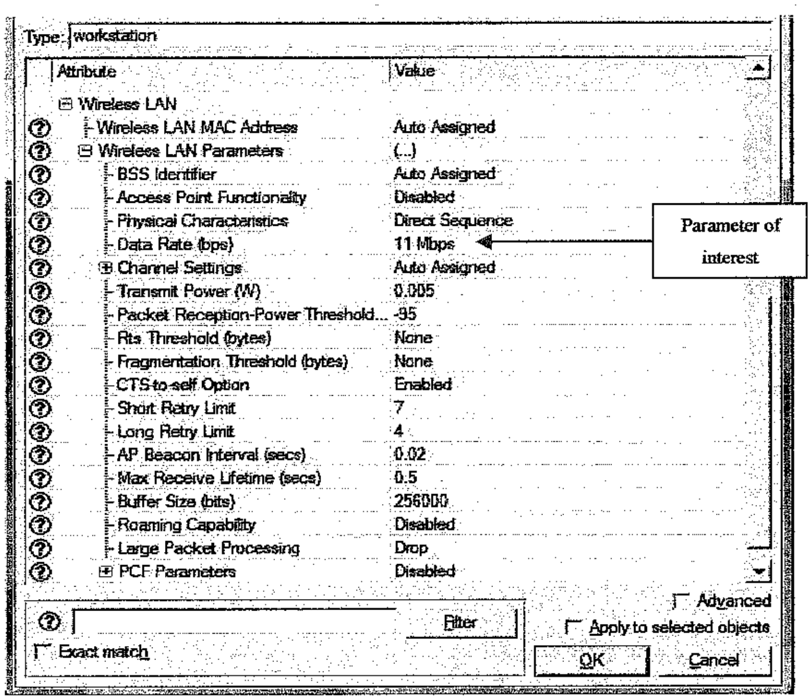


Figure 3.6: OPNET STD Representation of IEEE802.11 Standard

```
case wlanC_Direct_Sequence:
    {
    /* Slot duration in terms of seconds.                                    */
    slot_time = 20E-06;

    /* Short interframe gap in terms of seconds.                             */
    sifs_time = 10E-06;

    /* PLCP overheads, which include the preamble and header, in            */
    /* terms of seconds.                                                     */
    plcp_overhead_control = WLANC_PLCP_OVERHEAD_DSSS_LONG;
    plcp_overhead_data    = WLANC_PLCP_OVERHEAD_DSSS_LONG;

    /* Minimum contention window size for selecting backoff slots.          */
    cw_min = 31;

    /* Maximum contention window size for selecting backoff slots.          */
    cw_max = 1023;

    /* Set the PHY standard as 11b for the technologies specified           */
    /* in 802.11 and 802.11b.                                                */
    phy_type = WlanC_11b_PHY;
    break;
    }
```

(a) At Process Model Level



(b) At Network Model Level

Figure 3.7: WLAN Parameters in OPNET

As illustrated in Figure 3.8, the network model scenario developed for this study is a WLAN network that consists of twenty two (22) WNs, a wlan_ethernet_router node operates as an AP, a Proxy_Server node, an application node, a profile node and a customized node that holds the functions of CLAA agent. The WNs are classified into two categories: callers and callees that represent the two ends of VoIP calls. The caller or sender nodes initiate the call sessions, while the destinations of the calls or callees receive the calls. To differentiate between the nodes in the network scenario, the nodes of caller category are labeled with the prefix (Sender). Conversely, the nodes of callee category are labeled with the prefix (Receiver). According to WLAN network standard, traffic of all calls established among the WNs go through the AP node. Moreover, it is assumed in this study that there is only voice traffic in the network. Therefore, there is no other traffic coexist in the network with the voice traffic. Besides, all generated calls are of peer-to-peer calls and no voice conferencing are made. This allows the focus of the study to be on the performance of voice traffic and the impact of changing packet sizes on voice quality when running over WLAN network.

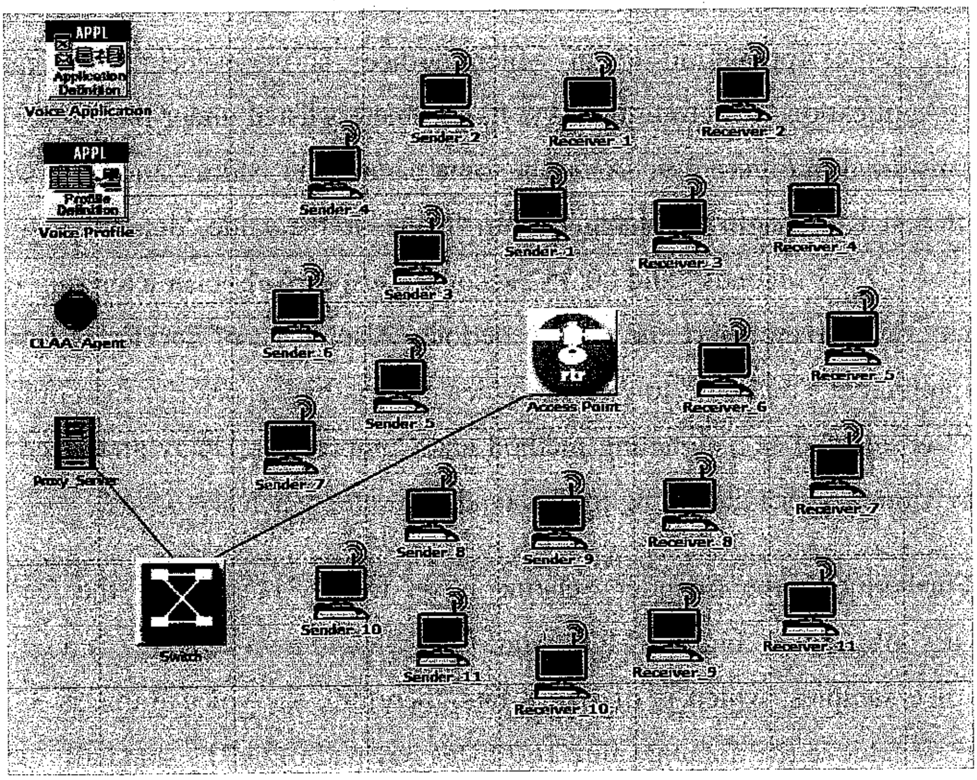


Figure 3.8: VoIP over WLAN Network Simulation Model

3.4.2.2 VoIP Traffic

For VoIP, the two types of generated traffic are: signaling and voice traffic. The method of generating signaling traffic in OPNET will be explained in the following Section 3.4.2.3. Conversely, the voice traffic is modeled as a voice application using the object "Application Config" node. The Application Config or application node consists of several predefined applications called Standard Applications that are most commonly used applications. Examples of these built-in applications are Database, FTP, Voice, etc. which listed under the attribute "Application Definitions". Basically, the standard application is modeled with the same traffic characteristic of the corresponding real network application including packet size, transport protocol, etc. Consequently, the standard application contains different parameters that are assigned to default values by OPNET. These values, however, can be modified by the user.

In our simulation, the voice traffic is modeled by defining a standard application named "VoiceApplication_1" in which 'Voice' is the selected application. The VoiceApplication_1 is configured via its parameters listed under the (Voice) Table as demonstrated in Figure 3.9. As seen, some parameters' values are modified; Silence length and Talk length are changed from "None" to the value "default", which are the values of "0.65 seconds" and "0.352 seconds" respectively characterizing the speech activity of alternate spurt and silence periods. Also, Type of Service parameter is set to "Interactive Voice (6)" to give the voice packets higher priority than other packets. "SIP" is selected as a signaling method and Compression Delay and Decompression Delay values are set according to the standard value, which is 0.75ms.

Furthermore, the Encoder Scheme is set to "G.711" due to the fact that this CODEC obtains the highest MOS value. In OPNET, a voice frame is a collection of 32 voice samples, which is of size 8 bits, therefore, the voice frame is 32 bytes. In order to achieve the default value 160 bytes payload of G.711 standard, the value of Voice Frames per Packet parameter is set to "5" [102] [103]. A summary of these parameters' values is listed in Table 3.3.

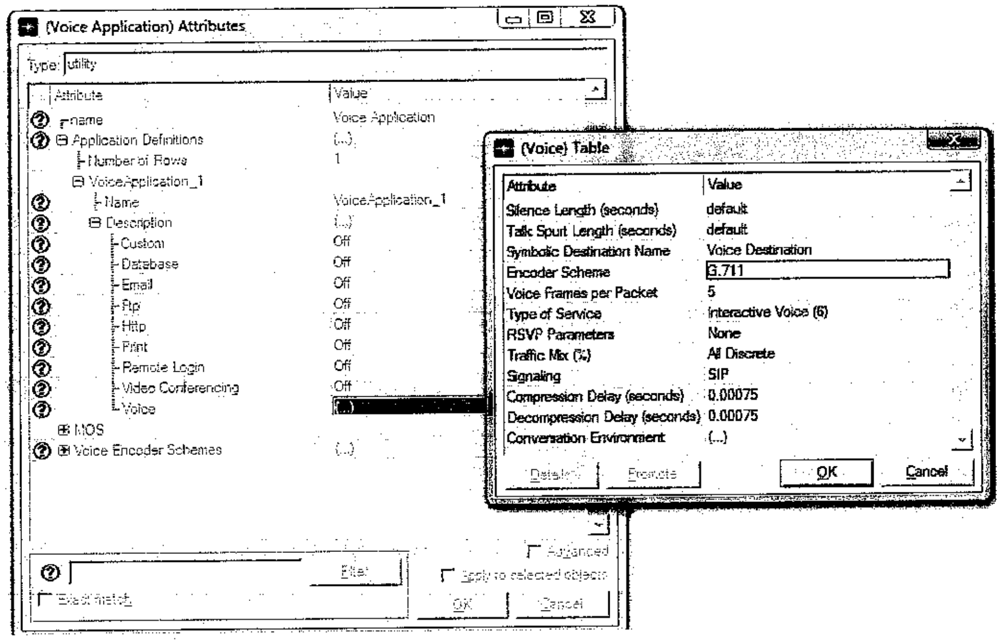


Figure 3.9: VoIP Application Simulation Model

Table 3.3: Application Node Attributes

Application Name / **Attributes**	**VoiceApplication_1**
Encoder Scheme	G.711
Voice Frames per Packet	5
Type of Service	Interactive Voice (6)
Compression Delay (Seconds)	0.00075
Decompression Delay (Seconds)	0.00075
Playout – Nominal value	80
Silence Length (seconds)	Default
Talk Spurt Length (seconds)	Default

In order to define the behavior of voice traffic or call sessions among the WNs, OPNET provides "Profile Config" node object that is designed to define factors that affect the way the traffic is modeled. The Profile Config or profile node allows the user to create one or more profiles with different specifications using its attribute "Profile Configuration". A profile is configured by different parameters such as supporting one or more applications, the start time and end time of the application, for how long it should last, etc. To specify which WN in the network runs which profile, each WN particularly the calling WN is configured to support one of the defined profiles. This configuration is made by adding one of the profiles to the workstation's attribute "Application: Supported Profiles". As shown in Table 3.4, for this study, one profile is created named "VoiceProfile_1", which starts to run the voice application in 20 seconds from the simulation start time with no repetition of the application.

Table 3.4: Profile Node Attributes

Attributes	**Profile Name**	**VoiceProfile_1**
Profile attributes	Application Name	VoiceApplication_1
	Operation Mode	Simultaneous
	Profile Start Time	constant (10)
	Duration	End of Simulation
	Profile Repeatability	Once at Start Time
Application attributes	start time	constant (10)
	Inter-repetition Time	exponential (300)
	Number of Repetitions	constant (0)
	Repetition Pattern	Serial

3.4.2.3 Configuring SIP protocol

SIP is a signaling method used to manage the signaling traffic that establishes VoIP call sessions as elaborated in Section 2.1.2.1. In order to configure the SIP signaling protocol in OPNET, the Proxy_Server node is added to the network.

Besides, SIP related attributes of the workstation, Proxy_Server and application nodes are set to enable SIP protocol services. The attributes of each type of the nodes and their assigned values are listed in Table 3.5.

Table 3.5: SIP Configuration in OPNET

Node	**Attribute**	**Value**
Application Config	Application Definitions→ Descriptions→ Voice → Signaling	SIP
wlan_wkstn_adv [Senders]	SIP → SIP UAC Parameters → UAC Service	Enabled
	SIP → SIP UAC Parameters → Proxy Server Specification → Proxy_Server → Proxy Server Address	Proxy_Server
wlan_wkstn_adv [Receivers]	SIP → SIP UAC Parameters → UAC Service	Enabled
Proxy Server	SIP → SIP Proxy Server Parameters → Proxy Service	Enabled
	Server Address	Proxy_Server

3.4.2.4 Configuring E-Model in OPNET

For the evaluation assessment of VoIP quality, E-Model technique is employed and MOS value is calculated based on its R-factor value. The E-Model technique as described earlier in section 2.1.6 is also implemented in OPNET. As demonstrated in Figure 3.10, the values of its key parameters used to calculate the R-factor are assigned in "Voice Conversation Environment" attribute of the application node. Furthermore, the value of Nominal Value parameter of the attribute Playout Delay is changed from "200" that is set by OPNET to the value "80". The Nominal Value defines the amount of delay that the jitter buffer applies at the beginning of a call. If the jitter buffer is a fixed mode, it is the maximum length of the jitter buffer. This value has been changed because it is suggested that the play-out delay value for VoIP systems should not be more than 80 Seconds [104].

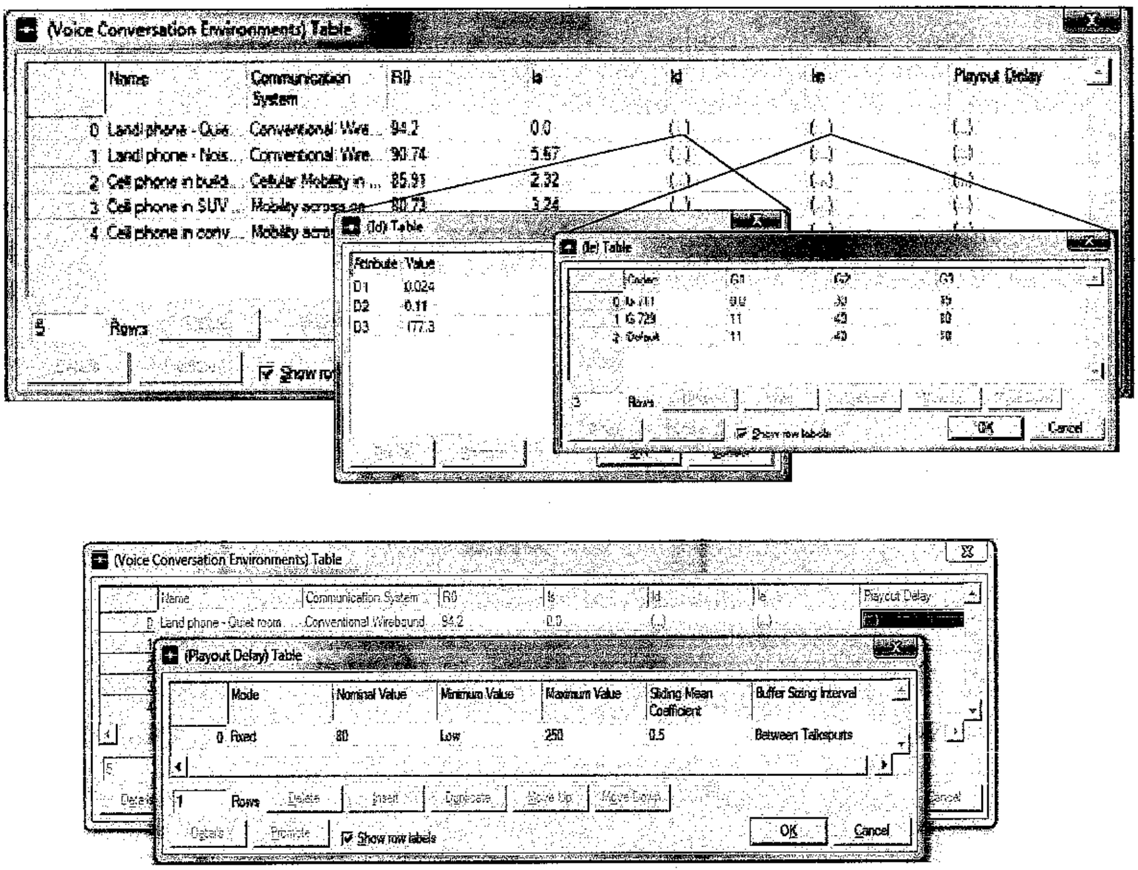


Figure 3.10: E-Model Parameters in OPNET

3.4.2.5 Collecting Statistic Results and Reports

In order to obtain the required results, statistics are configured before running the simulation scenario so that they are collected throughout the simulation run. There are many available statistics that are built-in OPNET and they are selected via its GUI. Figure 3.11 illustrates some statistics that can be configured in OPNET.

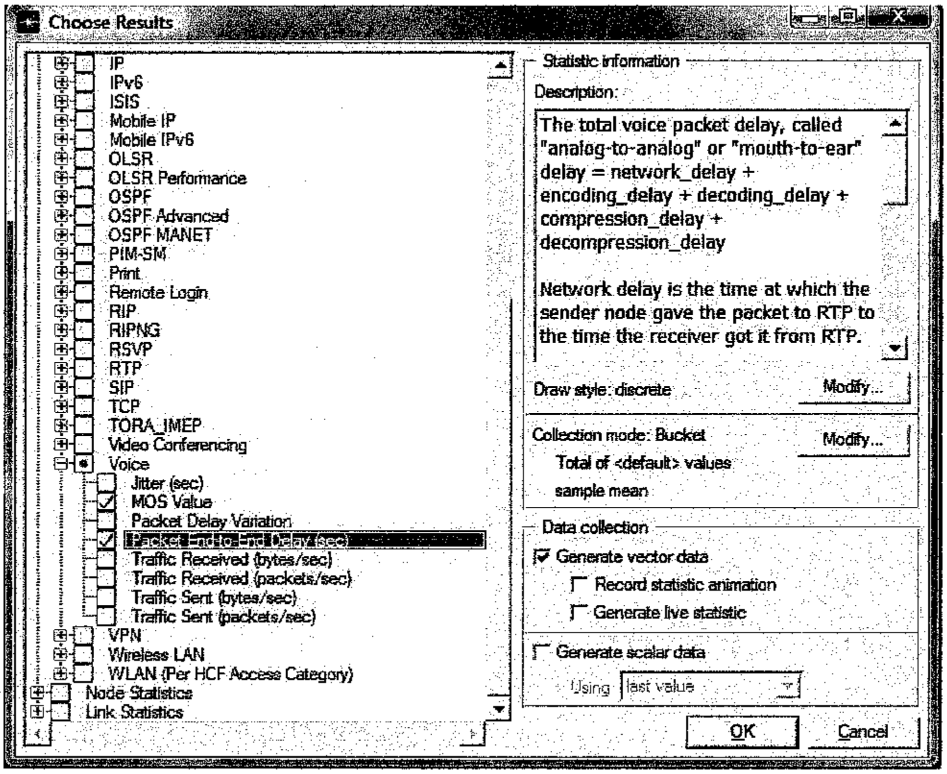


Figure 3.11: Statistics Configuration in OPNET

3.4.2.6 Preliminary Performance Analysis

For the purpose of performing the preliminary analysis study, another project is created in OPNET. The project consists of several scenarios that have the same settings with changing only the packet size and/or the MAC transmission data rate to study the impact of different packet sizes with each transmission data rate on VoIP quality. In this project, a WLAN network topology is created with 4 WNs, an AP, an application node and a profile node. The WNs make the calls where two of them, which named "Sender_1" and "Sender_2" act as the sender nodes and the other two named "Receiver_1" and "Receiver_2" act as the receiver nodes. The behavior of the voice traffic is described by the application and profile nodes by setting the values shown in Table 3.6 and Table 3.7 to their attributes.

Table 3.6: Application Node Attributes for Performance Analysis Study

Attributes \ Application Name	VoiceApplication_1
Encoder Scheme	G.711
Voice Frames per Packet	Variable
Type of Service	Interactive Voice (6)
Compression Delay (Seconds)	0.00075
Decompression Delay (Seconds)	0.00075
Playout – Nominal value	80
Silence Length (seconds)	Default
Talk Spurt Length (seconds)	Default

Table 3.7: Profile Node Attributes for Performance Analysis Study

Attributes \ Profile Name		VoiceProf_1	VoiceProf_2
Profile attributes	Application Name	VoiceApplication_1	VoiceApplication_1
	Operation Mode	Simultaneous	Simultaneous
	Profile Start Time	Constant (10)	Constant (20)
	Duration	End of Simulation	End of Simulation
	Prof Repeatability	Once at Start Time	Once at Start Time
Application attributes	start time	Constant (10)	Constant (10)
	Inter-repetition Time	Constant (20)	Constant (20)
	Number of Repetitions	Unlimited	Unlimited
	Repetition Pattern	Concurrent	Concurrent

The VoiceProfile_1 profile is defined for the first calling node (Sender_1), which starts generating the first call at the 20th second of simulation time. After every 20 seconds from the time of previous call, it increases its number of calls concurrently by one call. The call increment happens by configuring the parameters of the attribute Repeatability, which falls under the Applications attribute of the defined profile. On the other hand, the VoiceProfile_2 is defined for the second calling node (Sender_2). It establishes its first call at second 30 of simulation time and adds concurrently one call after every 20 seconds. Therefore, from the two profiles VoiceProfile_1 and VoiceProfile_2, the first generated call in the network is at second 20 from the simulation start time and the number of calls will increase by one every 10 seconds so that we can measure at which number of call the network will get congested and QoS of VoIP will be degraded. Hence, from the collected results of this analysis study, the capacity of VoIP can be defined for each transmission rate and an optimum range of packet sizes that does not degrade the QoS of VoIP when its capacity is met can be defined to be used by the proposed agent.

3.5 Summary

This chapter demonstrated the followed research steps in order to address the multi-rate issue in WLAN network. A CLAA agent was proposed with the aim of improving the QoS of VoIP when this issue causes network congestion. The model of CLAA agent and how it operates were elaborated. It also showed how the algorithm of CLAA agent monitors MAC layer to detect transmission data rate changes and accordingly informs Application layer to adjust its voice packet size. It also utilizes the RTCP reports to observe the network quality. The chapter also covered the simulation implementation using OPNET Modeler tool. The network topology setup of this research work was presented with demonstrating in details all the required configurations of WLAN, VoIP, its protocols, E-Model technique and statistic results to be collected.

CHAPTER 4

RESULTS AND DISCUSSION

In this chapter, simulation results are presented and analyzed. The results are grouped into three categories; the first category is the results of a preliminary analysis study, the second category is for evaluating the performance of the proposed CLAA agent, and the third category presents the results of investigating CLAA agent effectiveness under different levels of network congestion.

4.1 Introduction

This chapter starts with discussing the results of the preliminary performance analysis study. In this study, the impact of packet size with different transmission data rates on the quality of VoWLAN is demonstrated. Basically, the results obtained in this section are utilized to determine an optimum range of packet size values. The CLAA agent makes use of this range when required to make its adaptive decision in changing the currently used packet size to another value to cope with the current network condition. After that, the performance of CLAA agent is evaluated by conducting an extensive simulation study. It is happened by illustrating a performance comparison of VoWLAN network using CLAA agent to that of the standard network without agent in order to assess the proposed algorithm enhancement on VoIP. Lastly, the results of the third category are discussed and analyzed. These results examine the effectiveness of CLAA agent under different levels of congestions. This change of congestion levels happens by varying the number of slow nodes, which change their transmission rate in the network. Additionally, the network performance in this study is measured and analyzed based on different parameters, which are MOS value, E2E delay, jitter, and packet loss due to buffer overflow as well as throughput parameter.

4.2 Preliminary Performance Analysis

A series of simulation scenarios has been performed to study the effect of packet size or packetization on VoWLAN network with different transmission data rates. The aim of this analysis study is to find the lowest and highest packet sizes that help defining an optimum range of packet sizes, which would be applied in the adaptive CLAA agent. In order to achieve this analysis study, the values of voice quality produced when applying different packet sizes with each MAC data rate are compared. In other words, each MAC data rate is set with different packet sizes that starts with 1 frame per packet and then increases by another 1 frame in each other scenario when using the same CODEC type G.711 in all scenarios. Initially, each scenario starts with 1 call and then the number of calls regularly increases by 1 until congestion is caused in the network as it reaches its maximum capacity. It, therefore, helps to define the maximum calls capacity of each transmission data rate with each different packet size. This would assist defining the lower limit of the optimum range as the low sizes that support very few calls would be excluded. On the other hand, as the number of frames inserted in each packet increases, it reaches to a number that would negatively affect the network quality. This large size of packet that degrades the network would define the upper limit of the optimum range of packet sizes. The following sections explain the results of each scenario and show that different packet sizes produce different values of voice quality and the network congestion occurs after a different number of calls that differs depending on the two factors; transmission data rate and packet size.

Figure 4.1 illustrates the results obtained from the scenarios where the data rate is set to the highest value of 802.11b standard, 11 Mbps. This data rate can support more calls compared with the other data rates as congestion occurs after more number of calls. In comparing the voice quality produced by the different packet sizes with 11 Mbps data rate, it is noticed from Figure 4.1 that, the lowest packet size value, which is 1 frame per packet produces the highest MOS value, which is almost 4.4. However, congestion occurs when the number of calls reaches only 3 calls. That means that the packet size of 1 frame can support only 2 calls in the network with an acceptable voice quality. The second lowest value, which is 2 frames per packet, increases the number of supported calls in the network to 5 calls. Similarly, every next higher packet size increases the number of calls by 1 or 2 calls. Table 4.1 summarizes the number of calls each packet size can support when data rate is set to 11 Mbps. Hence, as the number of frames inserted in voice packet increases, it allows more number of calls with an acceptable voice quality. However, this increase of number of calls will degrade at a certain point where the effect of the increased packetization delay is noticed in the network and hence degrades the network quality.

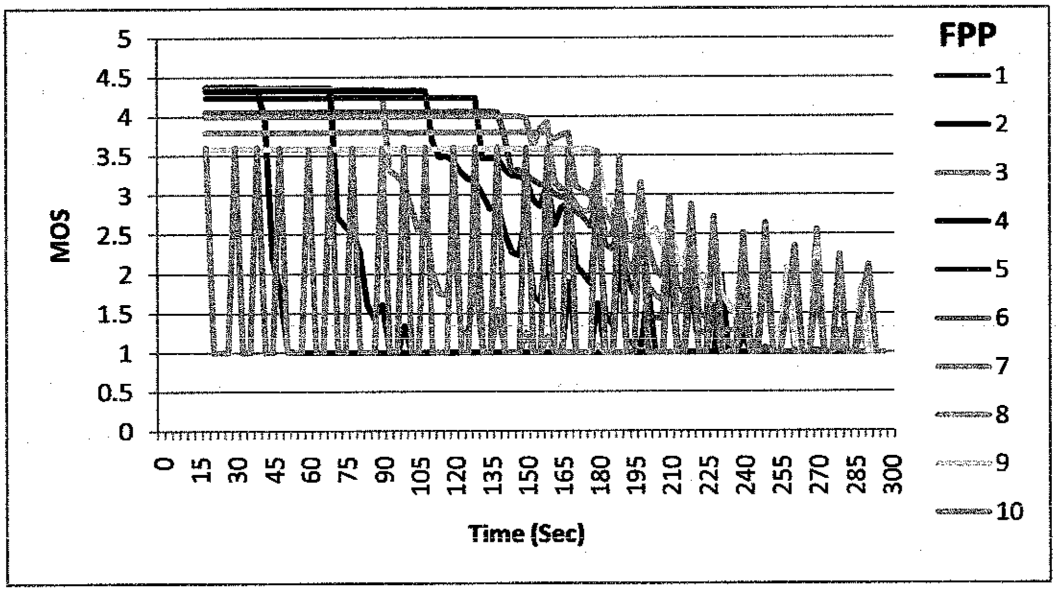


Figure 4.1: Effect of Packet Size on VoIP with Data rate 11 Mbps

Table 4.1: Number of Calls of different Packet Sizes with Data Rate 11 Mbps

Number of frames per packet	Number of calls with accepts quality
1	2 calls
2	5 calls
3	7 calls
4	9 calls
5	11 calls
6	12 calls
7	14 calls
8	15 calls
9	16 calls
10	Quality is not stable at all time

Although producing small sized packets reduces the packetization delay, it allows the transmission of more overhead, which increases the network traffic load. Because of this reduction in the bandwidth availability, less number of calls can be carried in the network. As the packet size is increased, the traffic load is reduced in the network due to reducing the total overhead of all transmitted packets. Therefore, the available bandwidth increases allowing the network to carry more calls. However, increasing the packet size, which means more frames are inserted in the voice packet, adds more packetization delay to the total network delay. This added delay does not have any negative impact on the network performance until the packet size reaches 10 frames where the whole network performance is significantly degraded. As seen in the figure, the network quality variation is a result of the packetization delay effect that drops the network quality every time a new call starts.

The same conclusion from the previous section when data rate was set to 11 Mbps can be drawn in this section, which presents the results of the scenarios applying the data rate 5.5 Mbps. As shown in Figure 4.2, the lowest value of packet size allows 2 calls in the network and when the packet size increases, more number of calls are accepted in the network, which are a little bit less than the total number of calls allowed when the data rate was higher, 11 Mbps. The number of calls that can be supported by the data rate 5.5 Mbps with the different packet sizes is summarized in Table 4.2. Moreover, it is also observed that the MOS value remains acceptable when increasing the number of frames in the voice packets. However, once the size of the packets reaches 10 frames per packet, the voice quality of the whole network degrades due to the effect of high packetization delay and therefore the calls can't be carried in the network.

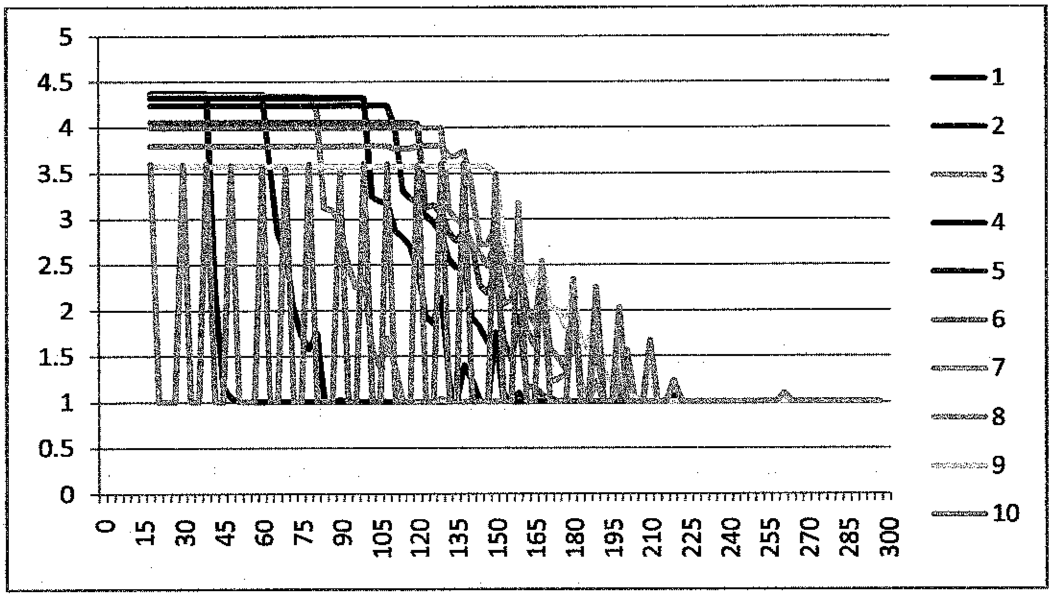


Figure 4.2: Effect of Packet Size on VoIP with Data rate 5.5 Mbps

Table 4.2: Number of Calls of different Packet Sizes with Data Rate 5.5 Mbps

Number of frames per packet	Number of calls with accepts quality
1	2 calls
2	4 calls
3	6 calls
4	8 calls
5	9 calls
6	10 calls
7	11 calls
8	12 calls
9	13 calls
10	Quality is not stable at all time

Since the lower transmission data rate, the less bandwidth availability, the network congestion occurs after less number of calls for the simulation scenarios of 2 Mbps and 1 Mbps data rates as shown in Figure 4.3 and Figure 4.4 respectively. The maximum numbers of calls that 2 Mbps and 1 Mbps data rates can reach are 8 calls and 4 calls respectively, which are much less than what 11 Mbps and 5.5 Mbps data rates can achieve. Furthermore, the packet size of 10 frames causes the whole network quality to degrade in these scenarios as well. The number of supported calls in the networks with 2 Mbps and 1 Mbps data rates are summarized in Table 4.3 and Table 4.4 respectively.

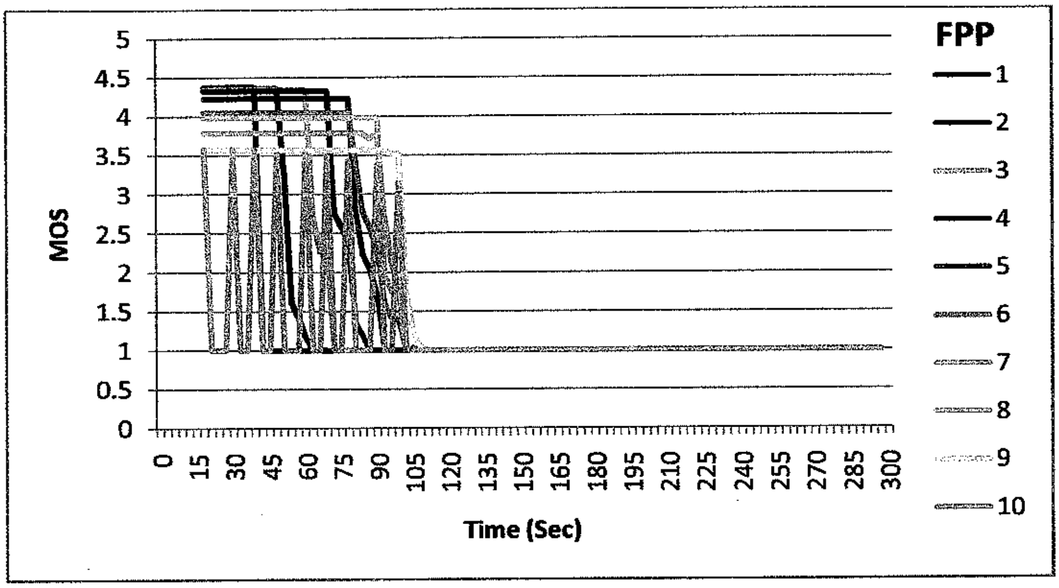


Figure 4.3: Effect of Packet Size on VoIP with Data rate 2 Mbps

Table 4.3: Number of Calls of different Packet Sizes with Data Rate 2 Mbps

Number of frames per packet	Number of calls with accepts quality
1	2 calls
2	3 calls
3	4 calls
4	5 calls
5	6 calls
6	6 calls
7	7 calls
8	7 calls
9	8 calls
10	Quality is not stable at all time

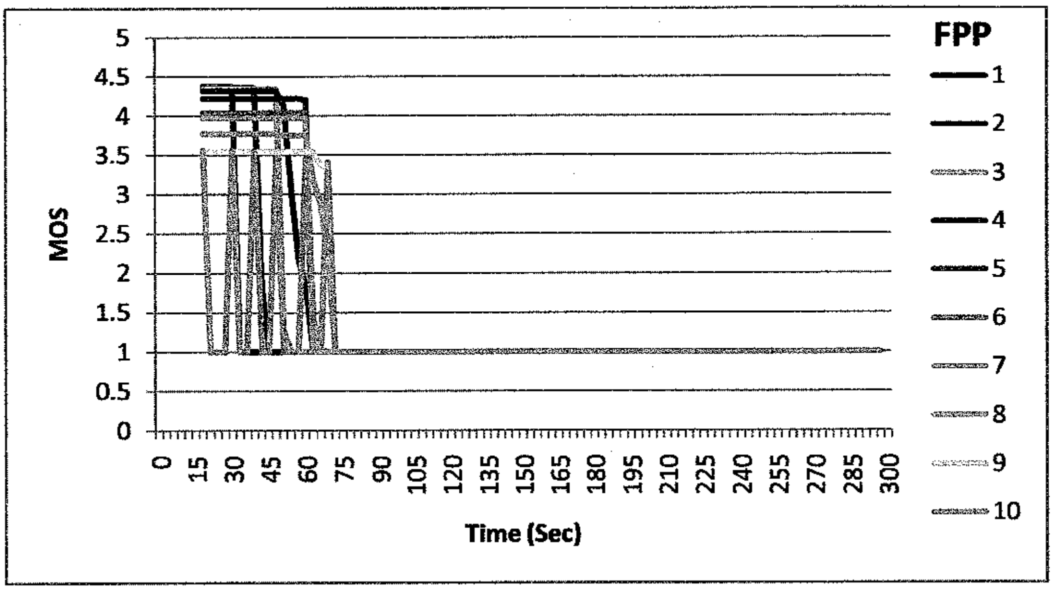


Figure 4.4: Effect of Packet Size on VoIP with Data rate 1 Mbps

Table 4.4: Number of Calls of different Packet Sizes with Data Rate 1 Mbps

Number of frames per packet	Number of calls with accepts quality
1	1 calls
2	2 calls
3	3 calls
4	3 calls
5	4 calls
6	4 calls
7	4 calls
8	4 calls
9	4 calls
10	Quality is not stable at all time

From this preliminary study of VoWLAN network performance analysis, it was observed that different packet sizes obtained different network qualities and supported different number of calls. It was also shown that MAC transmission data rate played role in increasing or decreasing the number of calls supported in the network; the higher data rate supports more number of calls. On the other hand, from the obtained results of this category simulation, it was also found that the network quality was variable and not acceptable when the number of frames inserted into the voice packet reached to 10 frames. Also, the packet size of 1 frame supported only 2 users and only 1 user in case of 1 Mbps data rate. Therefore, these two packet size values, 1 frame and 10 frames, are excluded in this study and the values between them are found to be suitable to use in the proposed algorithm. Hence, the range from 2 to 9 (inclusive) frames per packet is defined for the CLAA agent to select from when required to make decision in adapting the packet size to the variable network condition.

4.3 CLAA Agent Function

This section presents a set of results that highlights the main functionalities of the proposed CLAA agent. It shows that CLAA agent offers better VoIP performance by improving its QoS in the multi-rate WLAN environment. In order to demonstrate the significant performance enhancement that CLAA agent can achieve, a set of simulation scenarios is conducted to evaluate the algorithm when a number of slow nodes that change their data rates to lower values cause congestion due to creating a multi-rate issue.

4.3.1 Simulation Scenarios Description

As described in Section 3.4.2.1, this study considers a WLAN network that can be set up at a university department. The WLAN network is based on IEEE 802.11b standard with its key parameters that are configured as stated in Table 4.5. The network accommodates a total number of 11 call sessions, which are established among 22 nodes. The total number of nodes are equally divided into two types, 11 nodes act as VoIP senders and the other half act as VoIP receivers as depicted in Figure 3.8. As for the simulation duration, it is set to five minutes where all call sessions are being active simultaneously throughout the simulation time with initially transmission data rate of 11 Mbps. However, with the purpose of simulating the problem being addressed in this research study, the data rate for some of the nodes will be dropped to 1 Mbps for a specific period, from the time 70 seconds to 120 seconds, creating a multi-rate environment. This drop of data rate will cause congestion in the network, which degrades the quality of VoIP network. In fact, according to [15] network congestion in 802.11 does not occur gradually and a little of traffic is enough to cause a sudden change from a non-congested network to a congested network. Furthermore, it is assumed that only voice packets are transmitted in the network in order to avoid the issues that might occur due to the integration of voice and data packets in the same network.

Table 4.5: IEEE802.11b Parameter Values

Parameter	Value
Operating Frequency (GHz)	2.4
Data Rate (Mbps)	1,2,5.5,11
Spread Spectrum	PSK / CCK
Nonoverlapping Channels	3
Basic Rate (Mbps)	DSSS
DIFS (μsec)	50
SIFS (μsec)	10
Slot Time (μsec)	20
CWmin	31
CWmax	1023
PHY Header (bytes)	192
MAC Header (bytes)	34

Additionally, as for configuring the voice application parameters, all nodes are set to use G.711 CODEC and 5 frames per packet. This number of frames that represents the voice packet size is going to be changed by the proposed CLAA agent when required. As soon as the CLAA agent is alerted to a change in the transmission data rate or quality degradation in the network, it reacts and adapts the voice packet size accordingly to return the network quality to a stable state again. In order to evaluate the quality performance of CLAA algorithm, a comparison of the network behavior is made between a multi-rate network applying CLAA algorithm and a multi-rate network without the algorithm. This comparison is based on the measurement parameters MOS value, E2E delay, jitter, packet loss and throughput. The comparison results of these parameters are illustrated and analyzed in the next section.

4.3.2 Results Analysis

4.3.2.1 MOS Value

In order to investigate the CLAA ability to maintain the voice quality in accordance with VoIP user satisfaction, E-Model evaluation technique is applied. The E-Model is more friendly and suitable to monitor VoIP network performance. This technique qualifies the network performance by computing the R-factor parameter as a function of different network impairments that affect the voice quality. The computation of R-factor has been explained earlier in Chapter 2 Section 2.1.6.1. After that, the R-factor value is mapped to the traditional speech quality measurement MOS value to show the degree of perceived quality with user satisfaction. For a good VoWLAN network quality, MOS value should be kept above 3.5 as for a value less than 3.5 is termed unacceptable for many users.

As a first case of this simulation study, in a saturated network with capacity fully occupied by the 11 call sessions, two nodes dropped their data rates from 11 Mbps to 1 Mbps causing network congestion. The congestion happened because of increasing the load in the network that reduced the available network bandwidth. This caused lack of the amount of available bandwidth affected the performance of all nodes in the network and consequently led to performance degradation of the whole network. This performance degradation of the multi-rate network can be illustrated by the MOS value. As shown in Figure 4.5, the MOS value of the network without applying the proposed algorithm has been degraded from above 4 to less than 3.5 during the period of multi-rate occurrence, which is from 70 to 120 seconds. It is observed to be degradation in the network from an acceptable quality to unacceptable quality when the transmission data rates were dropped to lower rates. However, when applying the proposed algorithm CLAA in the network, it is found that the network recovered its health shortly after the drop of transmission data rates seeing that the MOS value increased to above 4 regaining the network stable state. Figure 4.5 shows the MOS value comparison of the network with and without applying the CLAA algorithm when congestion occurs.

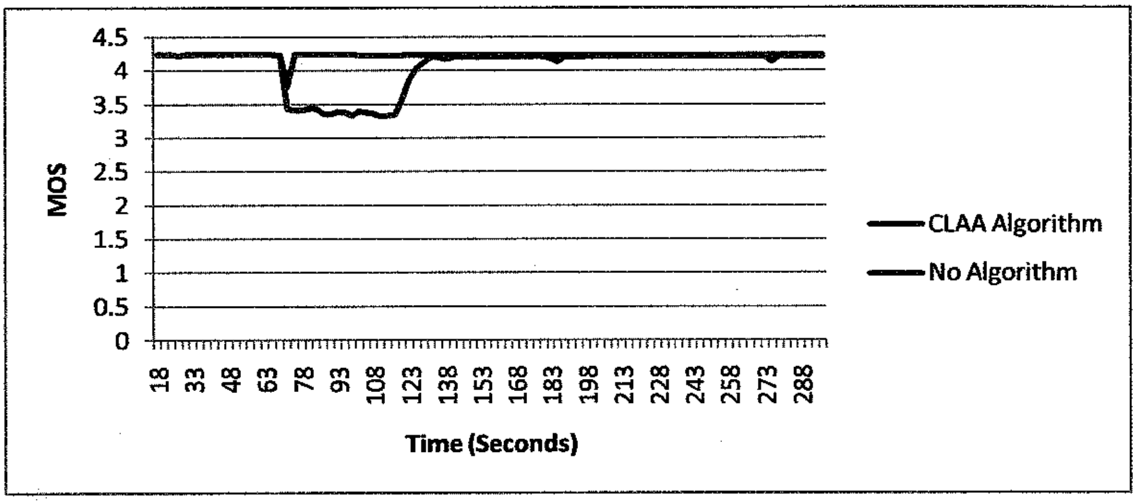


Figure 4.5: Average MOS Value of VoIP over WLAN

It is likely that, when transmitting a fixed number of frames per packet over the network throughout the call session, it does not cope with the variation of the wireless channel the same as when transmitting a variable number of frames per packet. The CLAA agent was able to vary the voice rate by adapting the packet size according to the sudden changes in the network to recover its quality. It was also able to keep the network stable even after the transmission data rates went high again after the simulation time 120 seconds when the multi-rate issue was ended in the network. Moreover, the algorithm reaction to the changes of the data rate from low to high rate happens in order to utilize the increased available bandwidth in the network.

4.3.2.2 End-to-End Delay

The E2E delay factor plays role in assessing the perceived quality and interactivity of VoIP conversation as explained in Chapter 2 Section 2.1.5.1. As recommended by ITU [44], a threshold of 150ms delay is set to define a good network quality. That means that the E2E delay should not reach a value higher than 150ms for a good network quality. Basically, the three key components of concern in this study that add to the total E2E delay are compression/decompression delay, packetization delay, and network delay, which is mainly affected by the queuing delay. On the other hand, the delay incurred by the algorithm for the time it spends during its process to sense the network deterioration, react to it, monitor the network quality by collecting RTCP reports, and computing the required quality measurements to recover the system should not be high so that it does not become noticeable to the user. In other words, the algorithm should not take more than few seconds to recover the network quality when it gets degraded.

In this study, the compression/decompression delay is considered negligible due to the fact that G.711 CODEC does not use a compression technique [105]. Therefore, the compression/decompression delay does not have a significant influence on the E2E delay. Conversely, packetization delay and network delay can be regarded as the effecting delay components on the E2E delay parameter in this study. That is because the algorithm adopts the dynamic packet size mechanism in which enlarging the packet size increases the packetization delay by reason of taking longer time to insert more frames into the packet. Whereas transmitting small-sized packets increases the queuing delay at MAC layer for the overhead it adds to the network load and that leads to increasing the network delay. Hence, varying the packet size in the network should be implemented carefully so that it does not reach to a point where the effect of either of these two components on E2E delay is significant.

Figure 4.6 illustrates the E2E delay of the WLAN network without the CLAA algorithm compared with the E2E delay of CLAA agent adopted in the network. The difference of the values gathered from both compared networks is easily observed. The occurred congestion due to the change of data rates in the first network caused the E2E delay to increase drastically to reach a value above 280 ms (and close to 300 ms), which is way far from the acceptable delay threshold value. That means that the total delay became unacceptable to many users. Nevertheless, in the second network where CLAA was adopted, the E2E delay dipped from above 280 ms to less than 150ms. This is an evidence of CLAA algorithm capability to improve the network performance by minimizing the total delay when adapting the packet size to the network condition. In addition, this packet size adaptation is performed without letting the packetization and network delays affect the total network delay negatively. As elaborated in Chapter 2 Section 2.1.4, despite the fact that transmitting large sized packets increases the packetization delay, it helps to reduce the overhead load in the network, therefore, decreases the queuing delay. As a result, the agent was able to enlarge the packets size of the slow nodes that changed their data rates and some of the other network nodes with the limit that the increase in packetization delay does not have a negative effect on the total E2E delay. Furthermore, the agent selects the packet sizes from the range that has been predefined to be an optimum range for such WLAN network. Hence, as long as enlarging the packet size does not reach a value out of this range; it will not cause the packetization delay to negatively affect the network delay. On the other hand, after the two slow nodes increased their data rates to 11 Mbps at 120 seconds, CLAA agent decreased the number of frames in the voice packet to utilize the increased available bandwidth in the network.

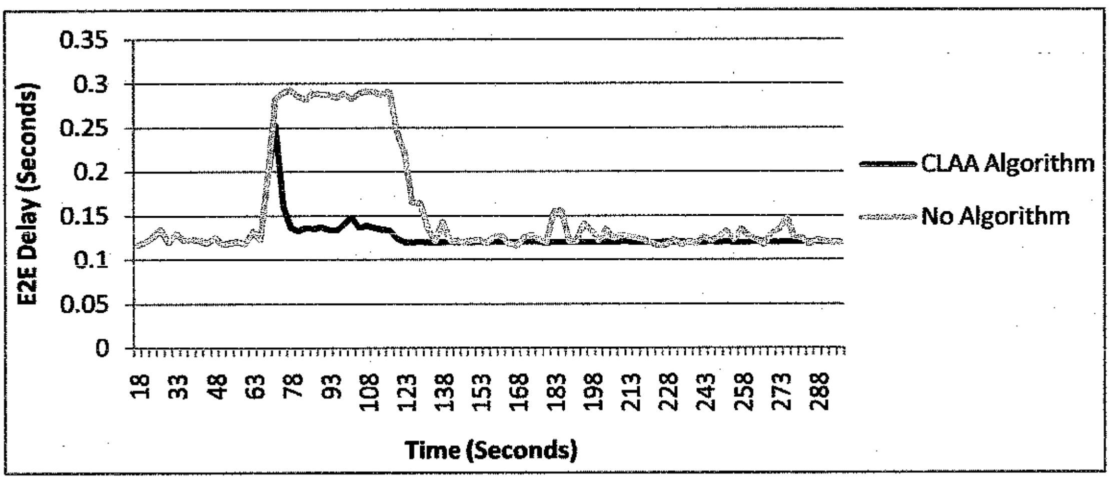


Figure 4.6: Average End-to-end Delay of VoIP over WLAN

Furthermore, the graph shows that the process time that the CLAA algorithm spends till it regains the good network quality after the degradation is not more than 3 seconds. This few seconds taken to recover the network degradation is small to be noticeable by the user and therefore it would not lead to a call drop.

4.3.2.3 Packet Loss

Packet loss is another important factor of measuring the quality of VoIP network and varying the packet size can affect this parameter. As a matter of fact, the transmission of small-sized packets means that the packetization interval is small and more packets will be generated at the buffers within a short time besides increasing the overhead load. Therefore, this increased network load will induce a buffer overflow to occur causing the packets to be dropped especially when the network capacity is not adequate to accommodate such high network load. As a result, the packet loss rate increases affecting the network performance adversely. Therefore, it is better to transfer large voice packets. Nonetheless, packet size should not be too large as well because large-sized packets increases the packetization delay and also they are more likely to be discarded at the receiver end if they encountered bit errors or arrived late as mentioned in Chapter 2 Section 2.1.5.2.

Essentially, dropping voice packets owing to buffer overflow is an important element causing packet loss rate to increase. Hence, statistics of data dropped in the network are collected to study the effect of CLAA agent behavior on packet loss. The results of data dropped due to buffer overflow are plotted in a graph as illustrated in Figure 4.7. The figure compares between a curve representing data dropped in the network that does not apply CLAA algorithm with another curve plotting the data dropped in the network with the algorithm. Like other measured parameters, the rate of data dropped is significantly affected by the happening of multi-rate issue. That can be observed from the graph where the curve of data dropped (represented by light green color) went high at the time the multi-rate situation is occurred because of the buffers became full. On the contrary, as shown in the second curve plotting the results of the network adopting CLAA algorithm (represented by dark red color), the curve sudden rising at the time of multi-rate was fallen shortly validating the reduction of data dropped rate. Thanks to CLAA agent potential to adapt the packet size to the network condition. It helped to reduce the traffic load by reducing the overhead load when the available network bandwidth is decreased, which led to reducing the data dropped rate attributable to buffer overflows.

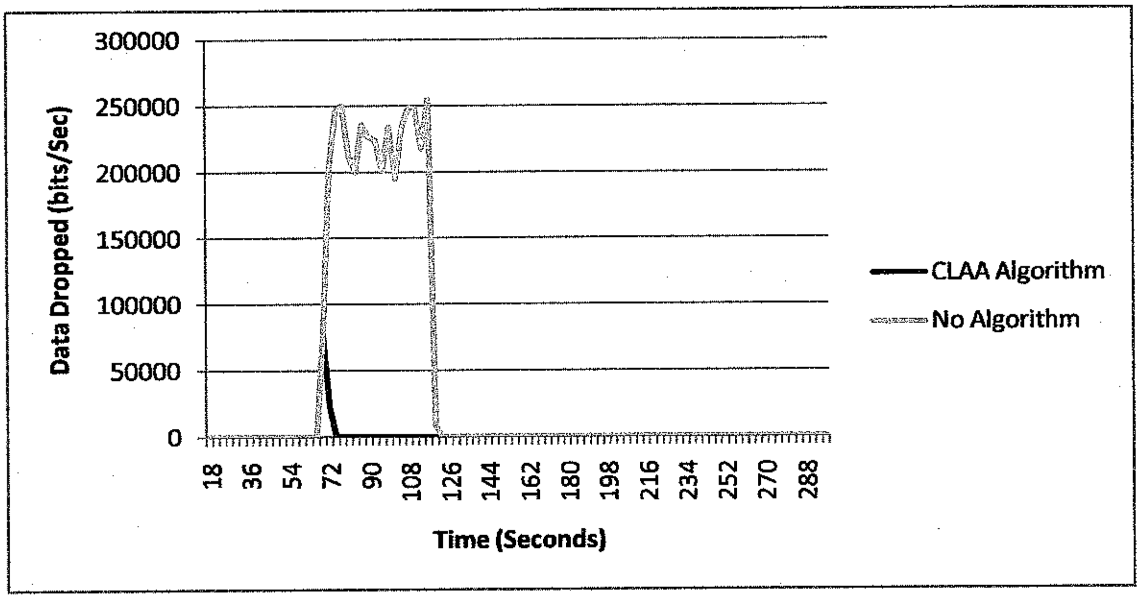


Figure 4.7: Average Data Dropped (Buffer Overflow) of VoIP over WLAN

4.3.2.4 Jitter

As jitter has been elaborated in Chapter 2 Section 2.1.5.3, although voice packets are being transmitted over the network at consistent intervals, a jitter or delay variation happens where the inter-arrival times between consecutive packets are variable. The jitter factor is also not desirable in real time applications due to the fact that jitter makes the speech to be heard as broken voice. Therefore, a play-out buffer has been introduced at the receiver side to compensate for jitter. However, packets have to arrive at the play-out buffer within a predefine delay otherwise they will be discarded by the play-out buffer, which causes the packet loss rate to increase. Furthermore, for good network quality, jitter value should not go above 30ms.

Several factors such as network congestion can cause jitter. Besides, the variation of voice packet size in the network can also introduce jitter. The jitter is calculated using the packets time stamps at the sender and receiver sides based on the following equation:

$$Jitter = (R_c - R_p) - (S_c - S_p) \quad (4.1)$$

where R_c is the arrival time of current packet, R_p is the arrival time of previous packet, S_c is the timestamp of current packet, and S_p is the timestamp of previous packet. The jitter or delay difference of two consecutive packets at the receiver side can be less or more than the delay difference of these packets at the source when generating them. For this reason, the equation will return either a positive value or a negative value. In fact, high values of both positive and negative types are not desirable and 0 is the ideal value for jitter.

As illustrated in Figure 4.8 where jitter results are plotted, the curves increase and decrease from the value 0. The curve representing the jitter results of the network without CLAA algorithm shows that jitter values are not very close to 0 particularly from the time the congestion happened due to the multi-rate. At that particular time, the jitter increases to a value higher than 2.5ms. This time when the congestion started is the changing point of the network condition from good to bad. At this point, the delay increases before the next packet is sent, which causes the inter-arrival time between its received time and the previous received packet to increase to a high value. After that, the network condition does not help the jitter to maintain its values. As the congestion period is over and the network is changed to a better condition with more bandwidth, the reverse effect happened where the delay gap between the currently received packet and the previous packet decreased. For that reason, the curve fall down to a negative value far from 0 as the network bandwidth improved and packets queuing delay decreased at time 120 seconds when the two slow nodes increased their data rates to 11 Mbps. Contrariwise, the curve representing the network with CLAA agent shows that after the jitter is increased to a high value at the starting point of the congestion, the reversed effect happened at the time the network condition is improved where jitter is changed to a negative value. The agent improvement in the network bandwidth availability decreased the delay gap between the packets received right before this improvement and the consecutive packets received immediately after the improvement. Furthermore, the two compared curves in the figure show that after eliminating the network congestion effects by the agent, the jitter results are improved throughout the transmission time even after the multi-rate period as the obtained values are closer to 0 than those of the network without algorithm. It is an evident that

CLAA agent improved the jitter parameter as it maintained its value to be as close to the ideal value as possible.

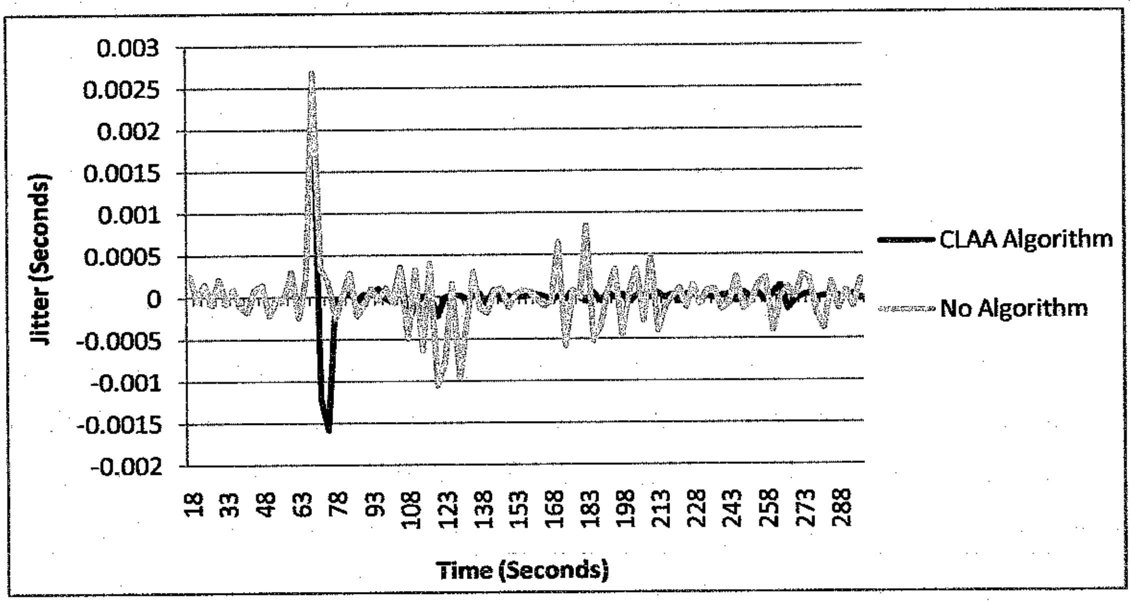


Figure 4.8: Average Jitter of VoIP over WLAN

4.3.2.5 Throughput

Since the quality parameters have an impact on network throughput, minimizing the total delay, jitter, and packet loss would maximize the throughput and vice versa. In view of the fact that CLAA agent has improved the voice quality and its parameters, so it should increase the network throughput. Basically, when the congestion happens in a multi-rate network with full capacity, the network bandwidth exceeds its saturation state where there is no enough bandwidth to carry all the traffic in the network causing delay and data drops, therefore, the total network throughput is reduced. However, applying the scheme of adaptive voice packet size on the variable network bandwidth assists to change the bandwidth requirement of voice traffic according to this variable network condition. Consequently, with CLAA agent, the available network bandwidth is increased during the network congestion to be able to carry more traffic in the network, hence, increasing the network throughput. This achieved throughput improvement when applying CLAA agent in the network can been seen in Figure 4.9, which compares the total network throughput of a multi-rate network without applying the agent to that of a multi-rate network with applying the agent.

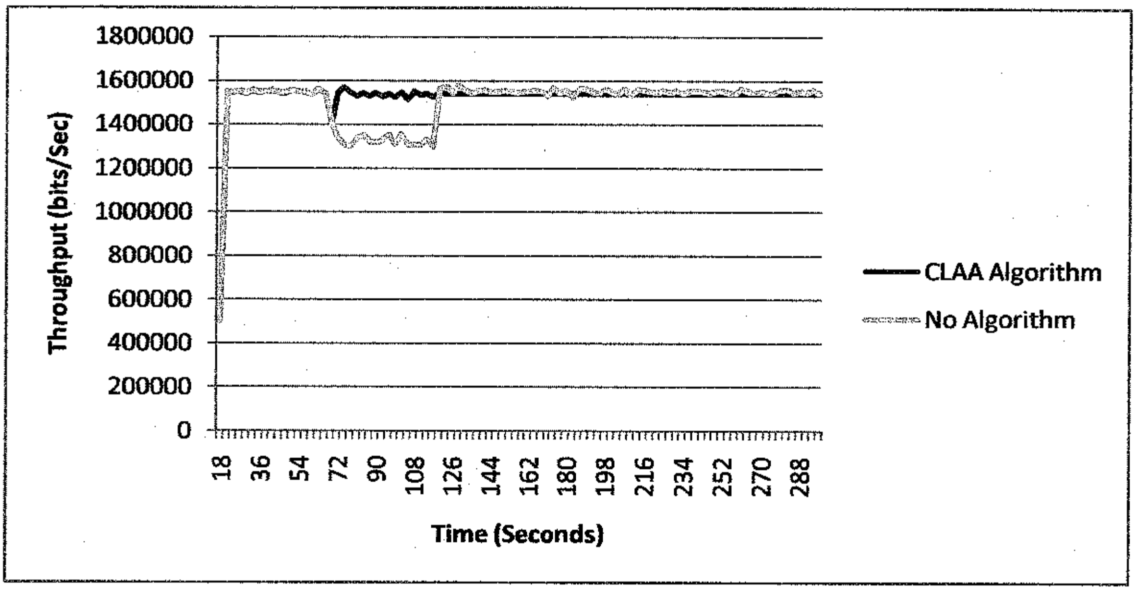


Figure 4.9: Average Throughput of VoIP over WLAN

As seen in the graph, the throughput of the network without applying CLAA agent, plotted with a light green line, is decreased when the two slow nodes lowered their data rates as an evident of the effect of multi-rate on the network throughput parameter. Since the network capacity is fully occupied, adding even a little of traffic load into the network can cause congestion and packet loss, hence, reducing the throughput or the number of bits received by the receiver. Whereas the dark red line on the graph plotting the throughput of the network with the agent shows that the decreased throughput when the congestion happened is then increased shortly after the congestion to be at the same level of the throughput values obtained before the network degradation. The throughput improvement is achieved owing to decreasing the overhead traffic load in the network by increasing the packet size. In other words, the agent managed to match the transmission bandwidth requirement to the available variable network bandwidth. It also makes a provision of transmitting more data bits in the network with sending less overhead. Furthermore, like previous parameters of analysis, the network degradation at the time the nodes change their transmission data rates lasts for a few seconds, which is short to be not noticed by the user.

4.4 Effect of Increased Number of Slow Nodes on CLAA Performance

Although the demonstrated results thus far proved the efficiency of the algorithm in improving the VoIP network quality during the presence of multi-rate issue, further investigation on the effectiveness of the algorithm has been carried out. In fact, the previous results are obtained when only two of the network nodes cause the addressed multi-rare issue. However, to know how the algorithm would respond if the number of slow nodes causing the network congestion is increased, more results are obtained from different scenarios. The network setup of these scenarios is similar to the previous scenarios as described in Section 4.3.1 inclusive parameters values stated in Table 4.5 with changing only the number of slow nodes that cause the transmission rate variation issue. A set of 5 different simulation scenarios has been executed with an increasing interval of two slow nodes different from a scenario to another; two slow nodes, four slow nodes, six slow nodes, eight slow nodes, and ten slow nodes respectively. This is, unlike other research works, to examine the effectiveness of the algorithm with higher levels of congestion caused by increasing the number of slow nodes in the WLAN network.

In comparing the MOS quality results of these different scenarios for both cases when applying CLAA agent in the network and when the agent is not applied, the MOS values during the time of multi-rate in the network of each scenario is calculated and then plotted together for each case as seen in Figure 4.10. For the case of the network applying CLAA agent, as noticed from the figure, when the number of slow nodes increases, which means the congestion level increases as well, the performance of the algorithm decreases. In this case, when the number of slow nodes reaches eight nodes, the quality drops very low to a value close to 1. The MOS value of 1 is not acceptable in the standard of VoIP quality. That is because the load in the network is getting higher than what CLAA agent can handle seeing that its attempt to further adapt packet size cannot free enough bandwidth for the increased network traffic. Hence, the CLAA algorithm can improve the network quality degradation caused by a small number of slow nodes. And, when the level of congestion caused by multi-rate is very high that cannot be handled by the algorithm, it will be required to drop a number of calls to regain a stable network performance.

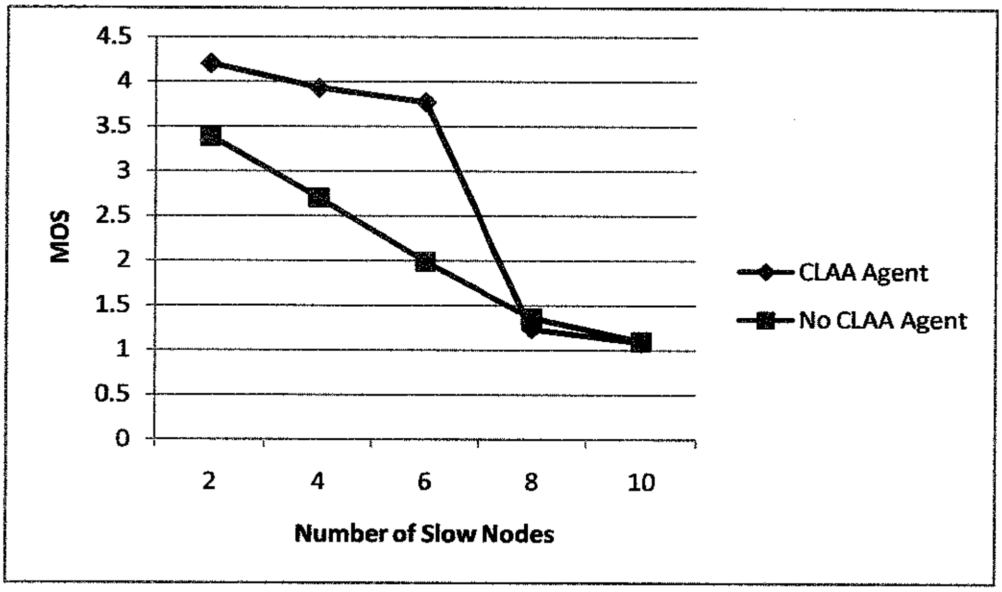


Figure 4.10: the average of the MOS results of network with and without CLAA agent during the happening of multi-rate issue by different number of nodes

Although the agent performance degrades as the number of slow nodes increases, it still improves the MOS value to be above the acceptable value rather than when the CLAA agent is not applied in the network. As shown in the figure, MOS values of the first case are higher than those of the second case for the first three scenarios. The MOS values are all below 3.5 for the second case and it is very low when the number of slow nodes is 6 nodes and more.

Similarly, in order to examine the E2E delay parameter for each case, the values of E2E delay obtained during the time the multi-rate occurred is calculated for each scenario and compared together in a graph as shown in Figure 4.11. The plotted graphs show that the delay values gradually increase as the number of slow nodes increases. In the first case, the delay values start to slightly exceed the value 150ms when there are 4 slow nodes in the network to reach values close to 200ms when the number of slow nodes is 6 nodes and it exceeds the value 200ms when the number of slow nodes is 8 nodes or more. Although E2E delay should be less than 150 for good quality, it is still acceptable if it reached 200ms as recommended by the ITU-T and also by Cisco particularly for private networks. However, it is considered annoying if it exceeded this value (250 is the limit for Cisco). Therefore, the achieved delay values in the case of having less than 8 slow nodes are still acceptable.

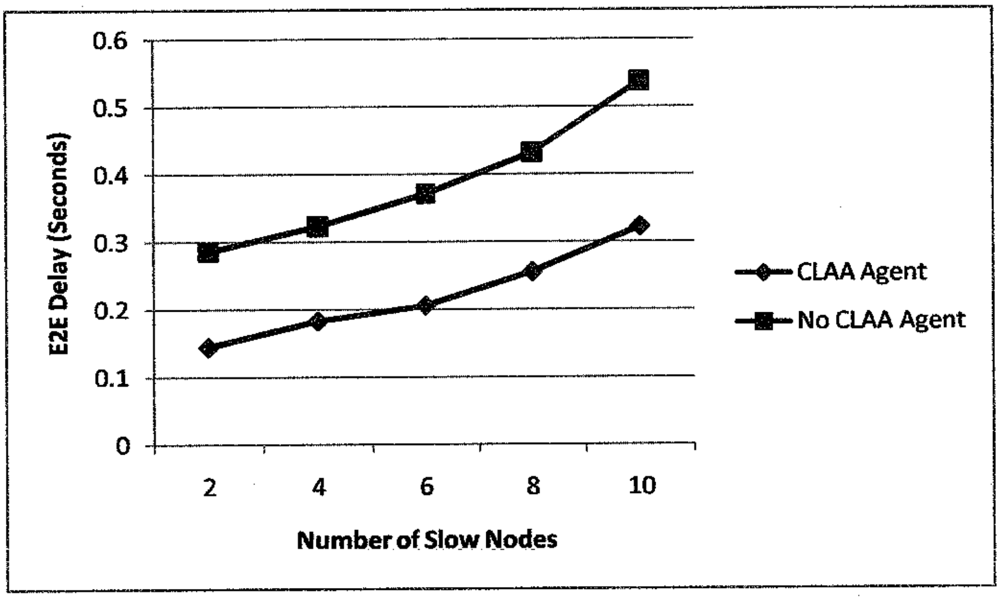


Figure 4.11: the average of the E2E Delay results of network with and without CLAA agent during the happening of multi-rate issue by different number of nodes

However, the E2E delay values obtained in the second case are very high and not acceptable in all scenarios as all of them are even above 250ms to reach a very poor performance at the last two scenarios of having 8 and 10 slow nodes in the network respectively. The percentage difference between the two cases is measured to be around %43, which shows the good enhancement of CLAA agent on the network delay.

Obviously, the happened increase of delay was due to increasing the traffic load in the network, which yielded to fill the buffers quickly. Since the number of slow nodes is increasing and capacity of the network is full, competition among nodes to access the network media to transfer its data becomes high and data has to queue for long in the MAC buffer till it gets the chance to access the network media. This is, in turn, causing E2E delay to increase. Furthermore, as the buffers become full with packets queuing to be transmitted in the network and have no free room to accommodate more packets, they will not accept the new received packets and will drop them.

Moreover, the number of packets being dropped will increase with the increase of traffic load or congestion in the network. This can be also noticed from the results obtained of data dropped for both network cases. Following the same previous steps of comparison in this section, data dropped, which represents packet loss in bits per second due to buffer overflow is plotted in Figure 4.12. The graphs show that increasing the number of slow nodes increases the amount of lost bits due to increasing the traffic load in the network as well as the delay. The increase of packet loss in the second case is very high and it is reduced in the first case. Therefore, CLAA agent has an impact on minimizing the rate of voice packet loss in the network.

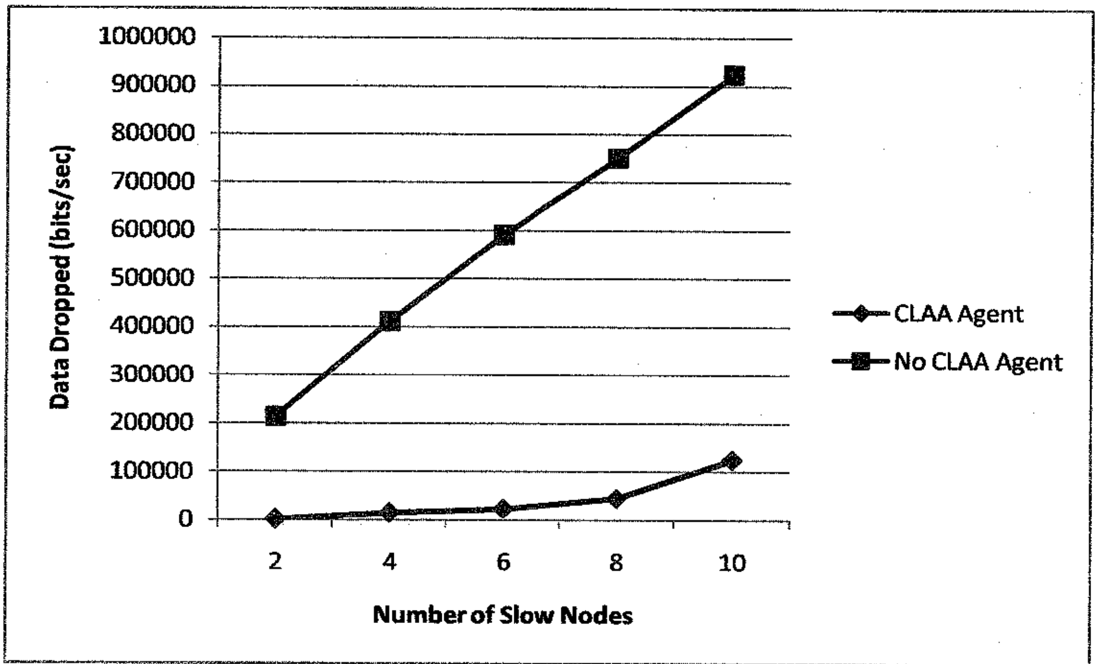


Figure 4.12: the average of the Data Dropped results of network with and without CLAA agent during the happening of multi-rate issue by different number of nodes

Principally, increasing the number of dropped packets affects negatively the throughput of the network. The throughput is a key metric in achieving good network performance and the higher gained throughput, the better. Like the previous compared parameters, throughput value is also affected negatively as it is decreased when the number of slow nodes is increased. Generally, increasing the load in the network causes the throughput to increase until it reaches its peak when the network gets saturated then it is started to decrease due to different reasons such as network collisions that happen due to increasing the network traffic load. When network collisions happen, the dropped MAC frames are retransmitted, which in turn increases further the network load. Also, the buffers become full and cannot store more packets so they are dropped causing decrease in the throughput.

Figure 4.13 shows the scenarios comparison of the throughput parameter for both network cases. As illustrated in the first case graph, the throughput decreases gradually that the percentage difference between the throughput of the networks with two slow nodes and with four slow nodes is around %1.67, between the four slow nodes network and six slow nodes network is around %1.266, between the networks with six slow nodes and with eight slow nodes is around %2.15, and between the networks with eight slow nodes and with ten slow nodes is around % 5.94. As noticed, the gab difference in the decreased throughput is increasing particularly when the network includes more than eight slow nodes. Furthermore, the percentage difference between the network with two slow nodes and the network with less than eight slow nodes is less than %3. However, the difference between two slow nodes network and eight or more slow nodes network is larger than %5 as it reaches around %10.56 when compared with ten slow nodes in the network. On the other hand, the throughput was highly decreasing when increasing the number of slow nodes in the second case with a percentage difference around %34 between both compared cases. This is just another parameter that CLAA agent improved when multi-rate issue is occurred.

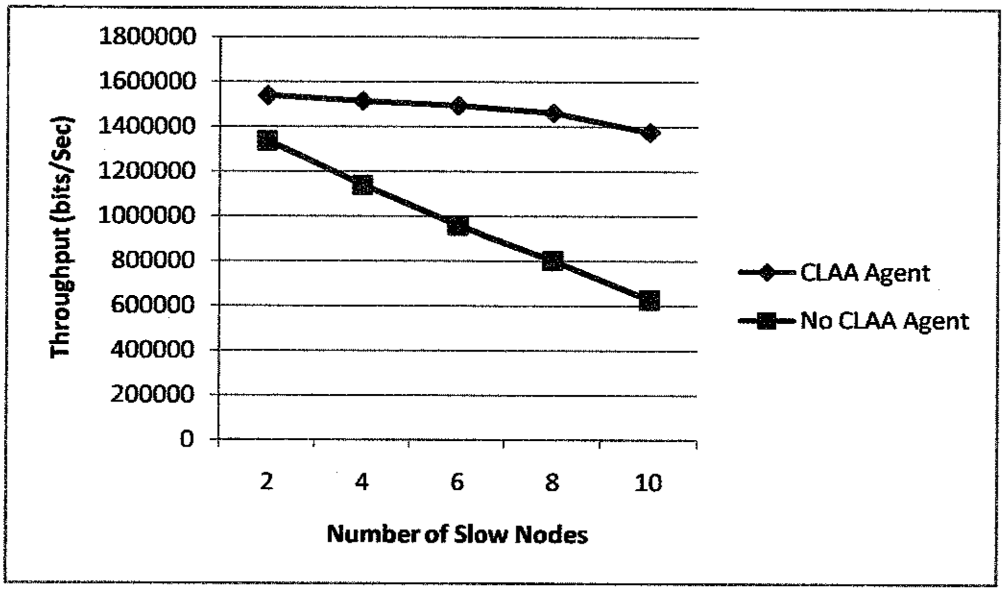


Figure 4.13: the average of the Throughput results of network with and without CLAA agent during the happening of multi-rate issue by different number of nodes

Lastly, the jitter parameter is evaluated and compared as well. As mentioned earlier, for better performance, jitter values should be close to the value 0. The values of jitter are calculated for the different scenarios for both network cases and plotted in graphs as shown in Figure 4.14. As illustrated in the figure, although the jitter values of the second case are closer to the value 0 than the values of the first case, all values are within the threshold limit. Increasing the number of slow nodes in the network increases the network congestion and therefore the network variation. Besides, as the CLAA agent attempts to adapt the packet size to cope with the network congestion, it finds difficulty in balancing the network load with the available network bandwidth when the number of slow nodes increases in the network particularly more than seven nodes. The frequent changes of packet size that the agent performs in attempt to improve the network performance lead to increase the delay variation in the network, hence, increased jitter.

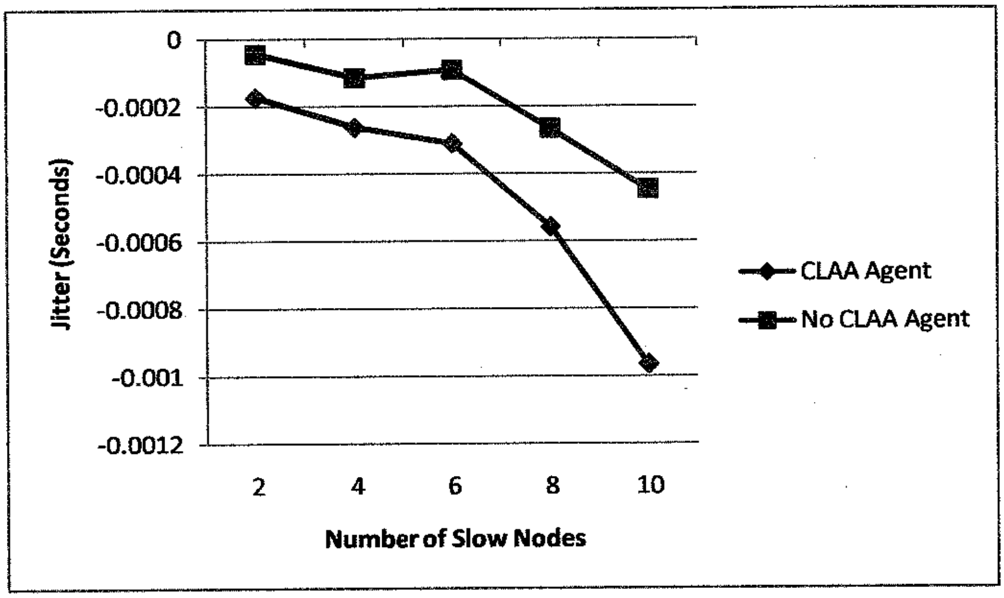


Figure 4.14: the average of the Jitter results of network with and without CLAA agent during the happening of multi-rate issue by different number of nodes

4.5 Summary

This chapter started with a preliminary analysis study of VoWLAN performance that defined the optimum range of packet sizes to be used by the proposed CLAA agent. It is then followed by a discussion on the evaluation results of CLAA algorithm, which was measured based on the QoS parameters; MOS, delay, packet loss, jitter, as well as throughput. The results presented and analyzed in the chapter have proven that CLAA agent enhanced the VoIP quality when it is degraded because of the multi-rate issue. Also, a simulation study was conducted to investigate the effectiveness of CLAA agent. Unlike other works, it was studied with the existing of different number of slow nodes in the network and found to be a good solution to address the research issue when the number of slow nodes is not larger than seven nodes. Yet, the quality produced from the network with CLAA is better than the quality produced from the network with no CLAA even if the number of slow nodes is more than seven nodes.

CHAPTER 5

CONCLUSIONS

This chapter concludes the research work by summarizing what has been done in this study. It also presents the achieved research contributions and what conclusions can be derived from the research work. Finally, the chapter also discusses some limitations of the work and suggestions of future work.

5.1 Summary

Due to the characteristics and features that WLAN networks can provide for VoIP applications such as mobility and simplicity, the integration of VoIP and WLAN technologies is growing rapidly. Yet, unlike wired networks, WLAN networks face different challenges that can affect the quality of VoIP such as bandwidth variation. The variation of network bandwidth can be caused by the multi-rate feature of WLAN network. This multi-rate feature occurs due to applying LA technique, which enables the mobile node far from the AP to change its transmission rate to a lower rate. Therefore, it leads to having a multi-rate network where different nodes in the same WLAN network apply different transmission rates. This leads to bandwidth variation in the WLAN network, which impacts the performance of VoIP adversely.

Since the multi-rate issue occurs as the LA technique changes the transmission data rate at the MAC layer, it is found effective to address this issue using rate adaptive technique. The rate adaptive technique performs voice rate adaptation at the Application layer according to the transmission rate changes at the MAC layer. As illustrated in Table 2.8, most of the proposed studies so far that aimed to address the multi-rate issue utilized CODEC rate adaptation at the Application layer as a key parameter of rate adaptive technique. The CODEC rate adaptive scheme applies a different CODEC with different bit rates for a different transmission data rate. Therefore, this scheme requires the system to have multiple CODECs. Furthermore, the CODECs with high compression ratios that produce low bit rates are licensed and must be purchased. Also, as the CODEC compression ratio increases, the VoIP quality decreases.

The voice rate can also be adapted using the voice packet size. As elaborated in Section 2.1.4, the packet size parameter plays key role in determining the required amount of network bandwidth for voice transmission. Large voice packets consume less network bandwidth than small voice packets due to the increase of overheard ratio when transmitting small voice packets. In addition, several studies have shown the effect of this parameter on voice quality parameters and how important it is to adapt this parameter for better VoIP quality.

Therefore, this work aimed to develop a new algorithm that addresses the issue of multi-rate by applying voice packet size adaptation scheme. An agent named Cross-Layering Adaptive Agent (CLAA) was developed based on rate adaptive technique with the purpose of adapting the voice packet size at the Application layer according the network bandwidth variation that is caused by the multi-rate of WLAN. The packet size adaptation scheme was able to adapt the network bandwidth requirement by increasing or decreasing the packet size according to the network variation, hence improving the VoIP quality. The agent also applied the Cross-Layering approach to monitor the MAC layer data rate and accordingly inform the Application layer to adapt the packet size when detecting transmission data rate changes. Furthermore, the agent utilized the statistics calculated from the RTCP reports. From these statistics, the agent monitored the network quality and if it observed any quality degradation happened, it would apply the rate adaptation as required to maintain the VoIP quality in a good condition.

However, although adapting the packet size can improve the network condition, a very large or a very small voice packet can negatively affect the VoIP quality. That is because a large packet size can increase the packetization delay, which therefore increases the network total delay. On the other hand, too small packet size can increase the network load and the overhead traffic, which as a result increases the queuing delay that can affect the total delay as well. Therefore, an analysis study on the impact of voice packet size on MAC data rates was conducted. The study was performed through running a set of preliminary simulations to evaluate the impact of each packet size with each transmission data rate on VoIP performance. The results of this investigation showed that the range of inclusively 2 to 9 packet sizes produces good network quality with all transmission data rates. For VoIP, hence, this range of packet size values was defined for the algorithm of CLAA to use when making its adaptive decision.

Furthermore, the proposed algorithm in this work was extensively evaluated through simulation study measuring the quality parameters of MOS, delay, packet loss, jitter, and throughput. The results of these measured parameters showed that the algorithm improved the network performance from a congested state to an uncongested state as the CLAA agent returned the network quality to be as good as it was before the network congestion happened. This is evidence that the packet size parameter plays role in improving the quality of VoIP since it can adapt the voice network bandwidth requirements to the available network bandwidth. Hence, the main research objective of improving the QoS of VoIP over a multi-rate WLAN network is achieved.

5.2 Research Contributions

The contributions of this research can be highlighted as follow:

1- A preliminary simulation study of VoIP over WLAN network performance with different MAC transmission data rates and voice packet sizes was conducted. This analysis study of these parameters and their impact on VoIP quality defined an optimum range of voice packet sizes. The optimum range of packet sizes that produces acceptable voice quality in all cases of applying different transmission data rates was set to inclusively 2 to 9 frames per packet. The outcome of this study was utilized by the proposed algorithm of CLAA agent for its adaptive decision. Hence, the first research objective was met.

2- A Cross-Layering Adaptive agent (CLAA) was proposed. It addressed the effect of multi-rate issue of WLAN network on VoIP quality by enhancing the quality parameters. The agent operated mainly by monitoring the MAC data rate and some statistics collected from the RTCP reports by which the network quality can be detected. If it observed any changes or quality degradation happened, it would adapt the voice packet size. For the agent to read the MAC layer information and accordingly adapt the Application layer parameter, it applied the Cross-Layering approach, which allowed the interaction between both MAC and Application layers. The proposed agent is transparent and does not require any kind of changes on any protocol, standard or hardware. Thus, the second objective of this study is met.

3- The CLAA agent was evaluated by measuring its effect on different VoIP quality parameters, which are MOS value, E2E delay, packet loss, jitter, and throughput, through extensive simulations. Also, the effectiveness of CLAA agent was evaluated by comparing its performance in different simulation scenarios that differ by the level of network congestion. The obtained evaluation results proved that the CLAA agent has contributed in enhancing VoIP quality. By this, the third research objective is met.

5.3 Conclusions

A new algorithm was proposed addressing the issue of multi-rate WLAN network. The evaluation performed through simulation proved that the main research objective is achieved as the measured quality parameters showed that the agent improved the network performance from a congested state to an uncongested state. It also proved the important role of voice packet size parameter in adapting the network bandwidth requirements to the available network bandwidth, thus improving VoIP quality in congested network.

5.4 Limitations and Future work

In this research work, the algorithm was evaluated in a network of only voice traffic. Therefore, it can be examined when TCP traffic coexists with the VoIP traffic in the same network. TCP traffic has an impact on VoIP traffic and a different ratio of each type of traffic in the network may lead to different results. The algorithm may not be able to significantly improve VoIP quality when a high ratio of TCP traffic coexists. Hence, it is recommended to enhance the algorithm to cope with the coexisting of TCP traffic. Similarly, the algorithm can be evaluated with the coexisting of video traffic as well.

The algorithm was able to enhance the performance of VoIP even when the number of slow nodes that apply low transmission rate increased in the network. Yet, it did not significantly improve the quality when the number of slow nodes was large. Therefore, further work to extend the algorithm to be able to cope with the existing of a large number of slow nodes in the network is needed. For that, it is suggested to integrate the algorithm with other adaptive algorithms such as CODEC adaptation.

In addition, this research work was simulated with making 11 calls in the network, which is the capacity limit of the studied network. The algorithm can also be enhanced to support more number of calls. The suggested integration of this algorithm with other adaptive algorithms may also support the increase of network capacity for more number of calls. Furthermore, due to the fact that the length of jitter play-out buffer can be fixed or adaptive, this parameter can also be examined and utilized to work with the proposed algorithm.

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APPENDIX A

LIST OF PUBLICATIONS

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