PPU College Of Engineering and Technology
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Electrical and Computer Engineering Department

Communication Engineering program

Bachelor Thesis

Graduation Project

The effect of mobility on VOIP over PPU WLAN

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كلية الهندسة والتكنولوجيا
دائرة الهندسة الكهربائية والحاسوب

اسم المشروع

The Effect Of Mobility On VOIP Over PPU WLAN

أسماء الطلبة

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....

وفاء محمد نصار

....

بناء على نظام كلية الهندسة والتكنولوجيا وإشراف ومتابعة المشرف المباشر على المشروع وموافقة أعضاء اللجنة الممتحنة تم تقديم هذا المشروع إلى دائرة الهندسة الكهربائية والحاسوب , وذلك للوفاء بمتطلبات درجة البكالوريوس في الهندسة تخصص هندسة الاتصالات والالكترونيات .

توقيع المشرف

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توقيع اللجنة الممتحنة

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توقيع رئيس الدائرة

....

الإهـــداء

إلى من جرع الكأس فارغاً ليسقيني قطرة حب

إلى من كلّت أنامله ليقدم لنا لحظة سعادة

إلى من حصد الأشواك عن دربي ليمهد لي طريق العلم

إلى القلب الكبير (والدي العزيز)

إلى ملاكي في الحياة ... وإلى معنى الحنان والتفاني

إلى بسمة الحياة وسر الوجود

إلى من كان دعائها سر نجاحي وحنانها بلسم جراحي

إلى أغلى الحبايب(والدتي الحبيبة)

إلى القلوب الطاهرة الرقيقة والنفوس البريئة إلى رياحين حياتي

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إلى الأخوات اللواتي لم تلدهن أمي

إلى من تحلو بالإخاء وتميزوا بالوفاء والعطاء

إلى من معهم سعدت ، وبرفقتهم في دروب الحياة الحلوة والحزينة سرت

إلى من كانوا معي على طريق النجاح والخير

(صديقاتي)

بطاقة الشكر

في مثل هذه اللحظات يتوقف اليراع ليفكر قبل أن يخط الحروف ليجمعها في كلمات . . . تتبعثر

الأحرف وعبثاً أن يحاول تجميعها في سطور

سطوراً كثيرة تمر في الخيال ولا يبقى لنا ونحن نخطو خطواتنا الأخيرة في الحياه الجامعية إلا وقفة

نعود بها إلى أعوام قضيناها في رحاب الجامعة مع أساتذتنا الكرام اللذين قدموا لنا الكثير

باذلين جهودا كبيرة في بناء جيل الغد لتبعث الأمة من جديد وقبل أن نمضي نقدم

اسمى آيات الشكر والامتنان والتقدير والمحبة إلى اللذين حملوا أقدس رسالة في الحياه

إلى اللذين مهدوا لنا الكثير لنخطو خطواتنا الأولى في غمار الحياه

إلى ....أساتذتنا الأفاضل

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Acronyms

VOIP	Voice Over Internet Protocol
PSTN	Public Switched Telephone Network
WLAN	Wireless Local Area Network
QOS	Quality Of Service
IP	Internet Protocol
AP	Access Point
Wi-Fi	Wireless Fidelity
IEEE	Electrical and Electronic Engineers
IBSS	Independent Basic Service Set
BSS	Basic Service Set
OFDM	Orthogonal Frequency Division Multiplexing
CCK	Complementary Code Keying
LLC	Logical Link Layer
MAC	Medium Access Control
VoWLAN	Voice over WLAN
codec	code decode compression
UDP	User Datagram Protocol
SIP	Session Initiation Protocol
RTP	Real-Time Transport Protocol

TCP	Transmission Control Protocol
ACD	Automatic Call Distribution
MTU	maximum-transmission unit
ISP	Internet Service Provider
BS	Base Station
GPRS	General Packet Radio Service
3G	third-generation

Chapter One

1

Overview

1.1 Introduction

1.2 General Idea About The Project

1.3 Objectives

1.4 Problem Statement

1.5 Proposed Solutions

1.6 Related Work

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1.8 Time Plan

1.1 Introduction

This chapter discuss the basic idea of the project which is studying the effect of mobility on VOIP over WLAN at PPU building B , then discuss about the objectives of the project , problem statement , proposed solutions , related works ,chapters content and time plane.

1.2 General Idea about the Project

The idea of this project is to improve the QOS on VOIP over WLAN at PPU building B , to discuss the challenges that faced VOIP system .

The main principle of the project study the effect of mobility in the call quality by study the QOS parameters like loss packet , jitter , end to end delay using simulation by OPNET simulator , and to study the current network in our university and its problems to put suggestions for improvement to be suitable for VOIP service .

1.3 objectives :

1) To better understand the problems of VOIP system to improve it.

2) To measure QOS parameter using simulation – OPNET modeler – for several different cases on VOIP calls.

3) Study the possibility of providing VOIP service in PPU which provide student free call service using Skype or other program in Wi-Fi mobile .

1.4 Problem Statement:

VOIP is a family of technologies, methodologies, communication protocol , and transmission techniques for the delivery of voice communication and multimedia sessions over Internet Protocol (IP) networks, such as the Internet . In our project, we will discuss the quality of service in VOIP network under three main parameters the first one is end to end delay, jitter and the throughput at different situations and different traffic , we will discuss the main things that affect voice quality in VoIP and what can be done to maximize quality ,and we will focus on the effect of mobility on those QOS.

1.4.1 The quality of service parameters are :

1.4.1.1 Throughput

In data transmission, throughput is the amount of data moved successfully from one place to another in a given time period which is often measured in packet per second.

1.4.1.2 Delay and jitter in VOIP

Delay is caused when packets of data (voice) take more time than expected to reach their destination. This causes some disruption in the voice quality. But the delay is not always constant, and varies depending on some technical factors. This variation in delay is called jitter.

1.4.2 The main parameters that affect voice quality in VoIP are :

1.4.2.1 Bandwidth

Bandwidth is a range of frequencies through which data is transmitted. A large bandwidth 'range' means that more data are transmitted at one point in time, and thus at greater speed. For voice communication, the bandwidth requirements are more important, since voice is a type of data which is bulkier than conventional text. This implies that the greater the connection speed, the better the voice quality you can get.

1.4.2.2 Equipment: (Hardware)

The VoIP hardware equipment that uses can greatly affect on quality of voice over VOIP calls .

1.4.2.3 Compression: (codec)

VoIP transmits voice data packets in a compressed form, so that the load to be transmitted is lighter. The compression software used for this are called codec's. Some codes are good while others are less good. Put simply, each codec is designed for a specific use.

1.4.2.4 Weather Conditions :

The weather condition affect on the quality of the VOIP calls.

1.4.2.5 Location of hardware :

When the hardware (mobiles ,laptops) closely to the AP in the floor the power taken from it will be large than the hardware that is in far location from the AP .

1.5 Proposed solutions

this project study the effect of mobility on VOIP over WLAN in PPU university building and provide solutions to the problems faced by VoIP service . when we moving on the same floor during a VOIP call there will have good voice quality in location near from access point compare with far location , and the call quality will depends on several factors including the number of users of the network in the call time , traffic size , and the number of access points and their location ,so while moving from floor to floor at the same building we will observe that the quality of the call will decrease so we will notice delay or jitter in the voice and sometimes the call will disconnect due to the poor coverage. We proposed that the possible solution to these problem by increasing the number of access points in some location or change their places to improve the quality of VOIP call .

1.6 Related work :

There are many research papers about this subject including the following five papers :

[1] A. Abdurrahman Lakas and B. Mohammed Boulmalf , Experimental Analysis of VoIP over Wireless Local Area Networks , JUNE 2007 .

This paper explain and measure the effect of the handover for both intra and inter mobility for VoIP traffic. The study was oriented towards the assessment of the variation of the throughput and the packet delay jitter during the handover operation.

They did some experiments in different cases, In these experiments they focused on the observation of the SNR, the traffic throughput and the delay jitter also analyzed the correlation between these parameters. The results presented in this show the effect of the handover on the voice transmission over an 802.11 based LANs, although the fact that the handover configuration in experiments does not include extra operations related to the authentication, encryption information exchange and QOS parameters transfer, the results indicate that:

- intra-domain handover can still impact the quality of voice through the jitter increase and the drop in the throughput.
- The latency incurred in re-establishing the forwarding path between the mobile device and the new access point decreases the VoIP quality. Therefore, new methods for intra and inter-domain handover are required. These methods should keep the latency to an acceptable minimum before VoIP can be successfully deployed at a large scale. [1]

[2] Jasmeet Singh , Quality of Service in Wireless LAN Using OPNET MODELER , published in JUNE 2009 .

This paper explain the basic concepts and issues of Wireless/Cellular network that can improve the QOS of a cellular WLAN.

Mainly focusing on Medium Access Control layer of Open Systems Interconnection (OSI) model .and study, the presently implemented schemes (the Point Coordination Function (PCF) of

IEEE 802.11, the Enhanced Distributed Coordination Function (EDCF) of the proposed IEEE 802.11e extension to IEEE 802.11),solves these issues and what can be done to improve them further , metrics used were Throughput, Data Drop, Retransmission and Medium Access Delay to analyze the performance of various MAC protocols in providing QOS to users of WLAN .

Two scenarios were created in the network simulation tool (OPNET MODELER) to obtain the results , the two scenarios with same Physical and MAC parameters, one implementing the DCF and other EDCF. The results showed that the performance of EDCF was better in providing QOS for real-time interactive services as compared to DCF, because of its ability to differentiate and prioritize various services. [2]

[3] A.Mona Habib and B.Nirmala Bulus, Improving QOS of VoIP over WLAN (IQ-VW), December 2002.

In this research project, they studied the inherent limitations of wireless networks, especially in the areas of QOS and security, as compared to wired standards they used VoIP as the multimedia benchmarking environment to explore the differences in the quality of service of a wireless vs. a wired network and attempt to identify the main challenge areas for enhancing the QOS of VoIP in a WLAN .

There test plans focused on measuring the network QOS factors(loss packet , delay, and jitter) on wireless networks as compared to Ethernet networks a comparison of the inter-packet delay times across different scenarios showed that the most common packet delay was approx. 20msec, which is consistent with the expected delay for the codec in use (G.729). Jitter time values were the least in the case of Ethernet-to-Ethernet communication, which were in the range of ±0.02msec. This range doubled to ±0.04msec in the cases of Ethernet-to-wireless and wireless-to-wireless.[3]

[4] A. Lin Cai , yang xiao , B. Xuemin Jon , VoIP over WLAN Voice capacity, admission control, QOS, and MAC , in 2006.

In this paper, they have presented an extensive survey on the voice capacity of an IEEE 802.11- based WLAN and the QOS enhancement mechanisms in the MAC layer. Only a limited number of voice connections can be supported in an 802.11 WLAN because of the overhead and the inherent inefficiency of the MAC protocol. Accurate voice capacity estimation is critical for effective and efficient admission control for VOWLAN.

When voice and data traffic share the wireless medium, it is important to design a MAC protocol with QOS support, implement appropriate queue management schemes, choose proper voice codec, and develop efficient play out buffer algorithms to satisfy the stringent QOS requirements of voice traffic. And they study the handoff and admission control issues for voice over WLAN/cellular systems .[4]

[5] A. Wang , VoIP Service Over Wi-Fi Networks , Sept 2009 .

This paper introduces basic elements of VoIP technology and Wi-Fi and explains how these elements influence voice capacity and quality over wireless networks.

The paper also provides guidelines for building a Wi-Fi network for maximizing VoIP performances and data capacity , paper is talking about VoIP characteristics and VoIP technology uses coder-decoder (CODEC) for compressing/decompressing the sampled voice signal.It also talks about VoIP Quality which is measured in MOS (Mean Opinion Score) values. The MOS takes into account the three most critical parameters for VoIP quality(packet loss, packet delay, and jitter) which is defined as the variance of packet delay.

The paper discussed the challenge of VoIP deployment over Wi-Fi networks which is threat the Wi-Fi protocol was optimized for best-effort data transmissions and not necessarily for real-time transmissions like VoIP traffic, which is delay/jitter and packet loss sensitive. And to overcome part of the problem, an extension to the Wi-Fi protocol standard was defined to enable over-the-air prioritization of real time traffic over data traffic. This is defined as IEEE802.11e standard. And its explain another challenge which is Collisions and retransmissions where the Wi-Fi protocol copes with collisions by retransmissions and those collisions and retransmissions affect the network performance in two ways: They reduce the total network capacity and They increase the delay and jitter of packet transmission and it discussed how to overcome those challenges.[5]

1.7 Chapters content:

This project it will present details of our work up to now in following chapters:

Chapter 2 will provide three main parts, the first part about wireless communication LAn, the second part about (VOIP) ,and the third talking about the OPNET modeler as Wireless Network Simulation.

Chapter 3 contain two part ,the first one discuses the current reality of the network on PPU and conduct an experiments related to calculate delay , jitter and power using an IP software system ,the other presents the models built for the several simulations and case studies used ,the work plan , and conceptual design by representing it in block diagram ,flow chart, and algorithms.

Chapter 4 contain the simulation networks on OPNET , the result and analysis of the network , it have many scenario on different network and traffic type , to get QOS for VPOIP application.

Chapter 5 it will have the conclusion get from the previous chapter , the future work and some recommendation to improve voice quality at the network .

1.8 Time Plan:

The following Scheme is describe the time planning over sixteen weeks:

Week / Task	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
Gives idea																
Collection Data																
Install OPNET program in laptops																
Doing experiment																
Writing& Printing																
Ppt&Hard copy																
Presentation																

Chapter Two

2

Theoretical background

2.1: Wi-Fi

2.1.1 Introduction to Wi-Fi

2.1.2 Wireless LAN Technology

2.1.3 Wireless channels

2.1.4 IEEE 802.11 Standard

2.1.5 Logical Architecture of WLAN

2.1.6 physical Architecture of WLAN

2.1.7 Mobility

2.1.8 The Nature Of Roaming in 802.11

2.2 VOIP

2.2.1 Introduction to VOIP

2.2.2 History Technology VoIP Charter Internet

2.2.3 Circuit Switching vs. Packet Switching

2.2.4 The principle of technical work in VOIP

2.1.1 Wi-Fi Introduction :

Wi-Fi (Wireless Fidelity) a wireless network uses radio waves , a mechanism for wirelessly connecting electronic devices. A device enabled with Wi-Fi, such as a personal computer, video game , or smart phone can connect to the internet via a wireless network access point.

An AP (or hotspot) has a range of about 20 meters (65 ft) indoors and a greater range outdoors. Multiple overlapping access points can cover large areas.[6]

2.1.2 Wireless LAN Technology :

Wireless local area networks (wireless LANs, or WLANs) Wireless refers to the transmission of voice and data over radio waves. It allows it users to communicate with each other without requiring a physical connection to the network And allows the network to go where wire cannot go.

Configurations are easily changed and range from peer-to peer networks suitable for a small number of users to full infrastructure networks of thousands of users that enable roaming over abroad area.[7]

AP which is the last wired stop on your network. Connected to the rest of the network via Ethernet cable, the AP translates the wired network traffic into radio signals and transmits it out via either the 2.4-GHz band (for 802.11b products) or the 5-GHz band (802.11a products).[7]

2.1.3 Wireless channels:

IEEE 802.11g/b wireless nodes communicate with each other using radio frequency signals in the ISM (Industrial Scientific and Medical) band between 2.4 GHz and 2.5 GHz. Neighboring channels are 5 MHz apart. [8]

However, due to the spread spectrum effect of the signals, a node sending signals using a particular channel will utilize frequency spectrum 12.5 MHz above and below the center channel frequency.

The preferred channel separation between the channels in neighboring wireless networks is 25 MHz (five channels). In the United States, only 11 usable wireless channels are available, so we recommended that we start using channel 1, grow to use channel 6, and add channel 11 when necessary, because these three channels do not overlap.[8]

radio frequency channels used are listed in Table 2.1 .

Channel	Center Frequency	Frequency Spread
1	2412 MHz	2399.5 MHz – 2424.5 MHz
2	2417 MHz	2404.5 MHz – 2429.5 MHz
3	2422 MHz	2409.5 MHz – 2434.5 MHz

Table 2.1:Radio frequency channel

2.1.4 IEEE 802.11 Standard :

The Institute for Electrical and Electronic Engineers (IEEE) developed the first internationally recognized wireless LAN standard: IEEE 802.11 wireless networks operate in two modes: ad-hoc or infrastructure mode. The IEEE standard defines the ad-hoc mode as Independent Basic Service Set (IBSS), and the infrastructure mode as Basic Service Set (BSS). The most widely used IEEE standards in the industry are the 802.11a and the 802.11b.The third standard 802.11g holds promise but has not yet been ratified by the IEEE.[7]

2.1.4.1 802.11a :

Operates in the 5 - 6 GHz range with data rates commonly in the 6 Mbps, 12 Mbps, or 24Mbps range. Because uses the orthogonal frequency division multiplexing(OFDM) standard data transfer rates can be as high as 54 Mbps.[9]

2.4.1.2 802.11b:

The 802.11b standard (also known as Wi-Fi)operates in the 2.4 GHz range with up to11 Mbps data rates and is backward compatible with the 802.11 standard. 802.11b uses a technology known as complementary code keying (CCK) modulation, which allows for higher data rates with less chance of multi-path propagation interference.[9]

2.4.2.3 802.11g :

Standard that operates in the 2.4 GHz range with data rates as high as 54 Mbps over a limited distance.

2.4.2.4 802.11e :

The IEEE 802.11e is providing enhancements to the 802.11 standard while retaining compatibility with 802.11b and 802.11a.The enhancements include multimedia capability made possible with the adoption of quality of service functionality as well as security improvements .It means the ability to offer video on demand ,audio on demand, high speed Internet access and (VoIP) services. [9]

2.1.5 Logical Architecture of WLAN:

WLAN works in the lower two layers of OSI model. First one is the physical layer which takes care of transmission of bits through a communication channel. Second one is the data link layer which is sub-divided into two layers: logical link layer (LLC) and Medium Access Control layer (MAC) . Only MAC layer is considered as the part of wireless LAN functions the primary function of a MAC protocol is to define a set of rules and give the stations a fair access to the channel for successful communication.[10]

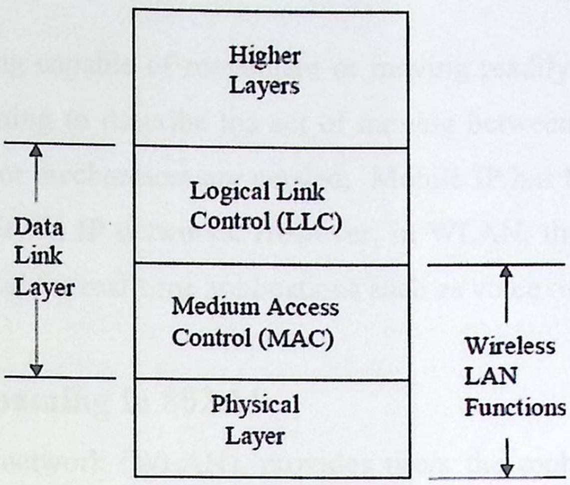

Figure 2.1 : Logical Architecture of WLAN

2.1.6 physical Architecture of WLAN:

There are two kinds of WLAN architectures:

- **Ad-hoc (Infrastructure less) architecture:**

An 802.11 networking framework in which devices or stations communicate directly with each other, without the use of an access point (AP). Ad-hoc mode is also referred to as peer-to-peer mode or an Independent Basic Service Set (IBSS). Ad-hoc mode is useful for establishing a network where wireless infrastructure does not exist or where services are not required.

- **Infrastructure architecture:**

With a wireless access point, This mode lets you connect wirelessly to wireless network devices within a fixed range or area of coverage. The access point has one or more antennas that allow you to interact with wireless nodes.

2.1.7 Mobility:

Is the quality of being capable of movement or moving readily from place to place. Some time use mobility and roaming to describe the act of moving between access points (APs). some sorts of mobility management mechanisms are needed, Mobile IP has been widely considered as a solution for realizing mobility in IP networks. However, in WLAN, the speed of roaming is a big challenge, because it is critical for real-time applications such as voice over IP.

2.1.8 The Nature of Roaming in 802.11:

Wireless local area network (WLAN), provides users the mobility freedom to move and roam around within the local coverage area. and it simplifies the network by linking two or more computers or devices to enable communication between devices. In addition, WLAN simultaneously share resources within a broad coverage area, Using radio frequency (RF) technology, and it transmit and receive data over the air, without additional or intrusive wiring. The mobility and roaming capabilities gives user a freedom to be connected everywhere and anywhere. This also allowed users to move around rapidly.

Voice devices, like Wi-Fi phones and PDAs, are extremely sensitive to delay and jitter. When these VoIP clients roam between buildings and floors they can experience disruptions and dropped calls. Meanwhile standard PC clients may experience slower data transfers while Web browsing may be disrupted when roaming. [11]

Even if configuration settings for power output are set identically on every AP, 802.11 roaming still cause problems for VoIP handsets.

All Wi-Fi devices are designed to go through a series of steps in order to establish a connection with a new AP whenever the current AP reaches an unacceptably low service level. This process is called 802.11 roaming. Roaming occurs no matter what type of security is used over the Wi-Fi network. Whether Wired Equivalent Privacy (WEP), Wi-Fi Protected Access (WPA), WPA2, or (gasp!) no security is used, there will be a delay in network communication as handsets establish a connection to their new access point.

When handling the effect that handset roaming has on wireless VoIP, it's best to be familiar with both the network and the handset. Make sure that APs are at a good distance to handle roaming. It's well known that great distances between APs can lead to dead spots, but it's also important to know that closely placed APs may cause some handsets to roam too often.

From the handset perspective, find out what roaming mechanisms are built into your handsets. Since 802.11 roaming is initiated by the station and not the AP, handset manufacturers have a significant degree of control over how their phones behave. Low roaming thresholds are generally preferable in wireless VoIP environments because that causes 802.11 roaming to occur less frequently. Some vendors have even created dynamic roaming thresholds. These dynamic thresholds increase when the handset is idle so that the handset associates to the best possible AP. Once a call begins, the roaming threshold lowers so that the call is less likely to be compromised by802.11roaming[12].

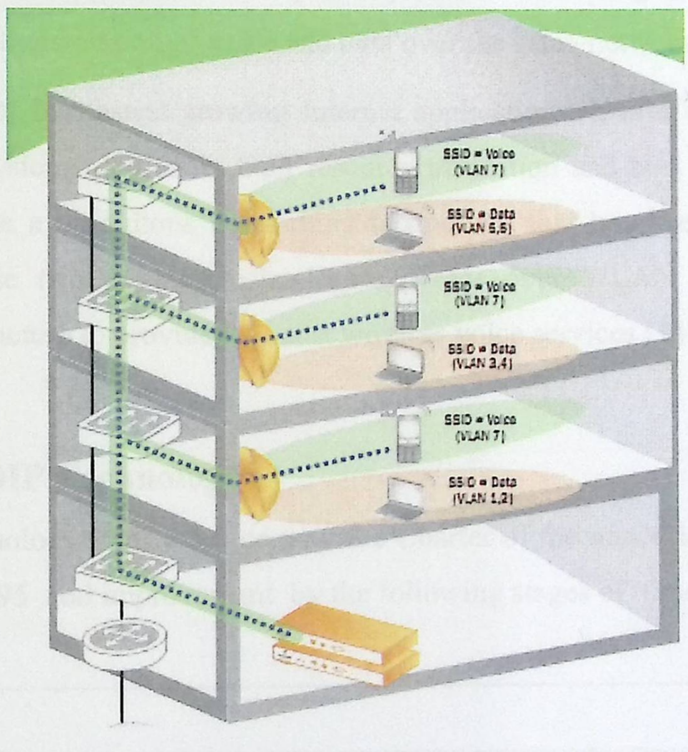

Figure 2.2: Roaming between floors

2.2.1 Introduction to VOIP:

VoIP stands for Voice over Internet Protocol. It is also referred to as IP Telephony or Internet Telephony. It is another way of making phone calls, with the difference of making the calls cheaper or completely free.

(VOIP) uses the Internet Protocol (IP) to transmit voice as packets over an IP network. This technology use the same protocols that the Internet uses .Where the protocol is a set of rules governing how data is transferred over networks, how they are compressed, how and how they are presented on the screen and so on . That means voice travels by way of the same protocols used on the Internet. [13]

One of the most important things to point out is that VoIP is not limited to voice communication. In fact VoIP means voice, video, data conferencing , voice conferencing ,by allowing the integrated transmission of voice and data over the same network.

(VoIP) is one of the fastest growing Internet applications. It is a viable alternative to the traditional telephony systems due to its high resource utilization and cost efficiency. Meanwhile, (WLANs) have become a ubiquitous networking technology that has been deployed around the world. Driven by these two popular technologies, Voice over WLAN (VOWLAN) has been emerging as an infrastructure to provide low-cost wireless voice services.[14]

2.2.2 History of VOIP Technology :

History of technology transfer voice over the Charter of the network (VoIP) shows that this technology began in 1995 and improvement by the following stages of time:

Year	Event
1995	Began technology of transfer voice over the Charter of the network(VoIP)
1996	The year of the IP Telephony Client.
1997	The year of the Gateway.
1998	The year of the Gatekeeper, the rate of use of VoIP traffic represent approximately 1% of all voice messages in the United States.
1999	The year of the Application in VOIP .

2003	Skype launches a peer-to-peer (P2P) VoIP telephony service
2005	The issue of ensuring the quality of voice takes priority over data transfer to become more reliable for the transfer of voice and clear phone calls without interruption.

Table 2.2: History of VOIP technology

2.2.3 Circuit Switching vs. Packet Switching:

In circuit-switching, this path is decided upon before the data transmission starts. The system decides on which route to follow, based on a resource-optimizing algorithm, and transmission goes according to the path. For the whole length of the communication session between the two communicating bodies, the route is dedicated and exclusive, and released only when the session terminates.

To be able to describe what packet switching is first a packet is a basic unit of communication over a digital network , it is also called a datagram, a segment, a block, a cell or a frame, depending on the protocol. When data has to be transmitted, it is broken down into similar structures of data, which are reassembled to the original data chunk once they reach their destination. Packets vary in structure depending on the protocols implementing them. VoIP uses the IP protocol, and hence IP packet.

In packet-switching, the packets are sent towards the destination irrespective of each other. Each packet has to find its own route to the destination. There is no predetermined path; the decision as to which node to hop to in the next step is taken only when a node is reached. Each packet finds its way using the information it carries, such as the source and destination IP addresses.

2.2.4 The principle of technical work in VOIP :

In a VOIP network, the voice signal is digitized, compressed and converted to IP packets and then transmitted over the IP network. VOIP signaling protocols are used to set up and tear down calls, carry information required to locate users and negotiate capabilities.

So VoIP converts analog voice signals into digital data packet and sent over the Internet, and then converted back into analog signals before reaching the phone receiver at the other end . VoIP calls can be made on the Internet using a VoIP service provider and standard computer audio systems. Alternatively, some service providers support VoIP through ordinary telephones that use special adapters to connect to a home computer network. Many VoIP implementations are based on the H.323 technology standard.[13]

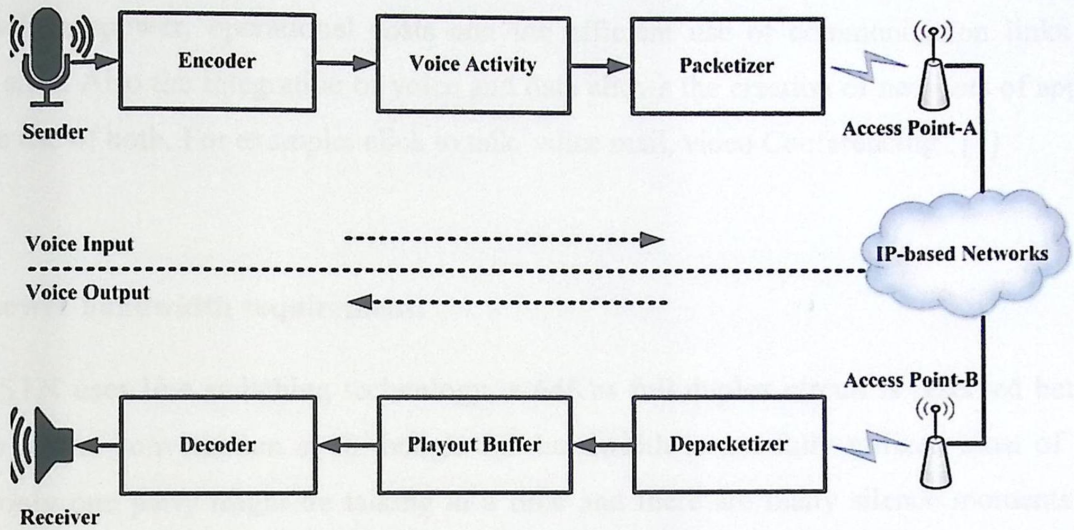

Figure 2.3 : VoIP over WLAN component

2.2.5 Voice over IP Advantages :

2.2.5.1 Cheap user hardware and software:

In VOIP you can do a call in cheap cost or free cost by using a simple software that download on your mobile phones or laptops . no need for external hardware to provided the VOIP service .

2.2.5.2 Save a lot of money:

Since VoIP uses the Internet as backbone, the only cost you have when using it is the monthly Internet bill to your ISP or by using the soft phone the call will be free .[8]

2.2.5.3 Integration of voice and data:

The use of one network to carry both voice and data allows savings of management and operational manpower, operational costs and the efficient use of communication links between different sites. Also the integration of voice and data allows the creation of new sets of applications that make use of both. For example: click to talk, voice mail, video Conferencing . [7]

2.2.5.4 Lower bandwidth requirement:

PSTN uses line switching technology, a 64Kbs full duplex circuit is reserved between the two ends of the conversation even though this bandwidth is not fully utilized most of the time, because only one party might be talking at a time and there are many silence moments during a conversation. Also, line switching does not allow the shared use of a valuable resource, namely communication lines between the different exchanges. In addition, the developments in compression technologies have reduced the bandwidth needed to carry voice to less than 7 kbps without a noticeable loss of voice quality.[7] IN addition of that it is known that about 50 % of a voice conversation is silence. VoIP fills the 'empty' silence spaces with data so that bandwidth in data communication channels is not wasted. In other words, a user is not given bandwidth when he is not talking, and this bandwidth is used efficiently for other bandwidth consumers. Moreover, compression and the ability to remove redundancy in some speech patterns add up to the efficiency.[8]

2.2.6 What Affects Voice Quality in VoIP Calls:

2.2.6.1 Bandwidth:

The Internet connection always tops the list of factors affecting voice quality in VoIP conversations. The available bandwidth have for VoIP is the key for voice quality. A broadband connection will work right, as long as it is not spotty, and not shared with too many other communication applications.

2.2.6.2 Hardware Noise and Equipments :

Certain levels of noise can be coupled into the conversational audio signals due to the hardware design. The source can be ambient noise or 60Hz noise from the power adaptor. The SPA hardware design minimizes noise coupling.

2.2.6.2 Phone frequencies:

The frequency of your IP phone may cause interference with other VoIP equipment. There are many cases where people using 5.8 GHz phones have been getting voice quality problems. When all troubleshooting tricks failed, changing the phone to one with a lower frequency (e.g. 2.4 GHz) solved the problem.

2.2.6.3 Weather Conditions:

At times, the voice is terribly distorted by something called static, which is a small 'dirty-weed' static electricity generated on broadband lines due to thunderstorms, heavy rain, strong gusts, electrical impulses etc. This static is not very much noticeable when you surf the net or download files, which is why we don't complain about it when we use the Internet for data despite it be here; but when you are listening to voice, it becomes disturbing. The effect of weather conditions on our connection is not something we can change. We can have some short-term relief in some cases, but most of the time, it is up to our service provider to do something. At times, changing the cables solves the problem completely, but this can be costly.

2.2.6.4 Location of your hardware:

Interference is a poison for voice quality during voice communication. Often, VoIP equipment interfere with each other thus producing noise and other problems.

2.2.6.5 Compression: the codec(code decompression code) used:

VoIP transmits voice data packets in a compressed form, so that the load to be transmitted is lighter. The compression software used for this are called codec's. Some codecs are good while others are less good. Put simply, each codec is designed for a specific use.

2.2.7 VOIP Quality Of services :

2.2.7 QOS :

The voice performance in the data network (WLAN) is measured using several parameters that include Delay, Jitter, Packet loss, Mean Opinion Score (MOS) and throughput. These parameters would help network designers to determine how the voice packets are handled in the network with varying levels of background traffic. It is finally observed that the increased background traffic or overloaded data links will result in decreased MOS (Mean Opinion Score) value.

Stands for Quality of Service. The most common definition we have of QOS is the differentiation between types of traffic and types of services so that the different types of service and traffic can be treated differently. This way, one type can be favored over another .QOS is an important tool for VoIP success. Through the years QOS mechanisms have become more and more sophisticated.

In VoIP, quality simply means being able to listen and speak in a clear and continuous voice, without unwanted noise. Quality depends on the following factors:

2.2.7.1 Packet Loss:

Packets transmitted over IP network may be lost in the network or arrived corrupted or late. Packets would be discarded, when they arrive late at the jitter buffer of the receiver or when there is overflow in jitter buffer or router buffer. Therefore packet loss is the total loss occurs due to network congestion and late arrival . In case of packet loss, the sender is informed to retransmit the lost packets and this is cause more delay and thus affecting transmission QOS. Furthermore, VoIP system can tolerate packet loss to some extend as 1% or less is acceptable for roll quality while for business quality 3% or less is acceptable. Hence, more than 3% of packet loss degrades the speech quality. Techniques and algorithms have been designed to resolve packet loss problem. MAC sub-layer at the sender side uses the acknowledgement scheme to retransmit lost packets but this technique is bandwidth consuming. In addition, forward error correction (FEC) is a mathematical technique that helps receiver to reconstruct lost packets from previously sent packets.

2.2.7.2 End-to-End Delay :

Delay can be defined as the total time it takes since a person, communicating another person, speaks words and hearing them at the other end. Unlike data applications, VoIP applications are very sensitive to delay although they can tolerate packet loss to some extent. End-to-end or mouth to ear delay is one of the main factors affecting QOS and should be less than 150ms for good network connection as defined by ITU G.114 while delay of less than 100ms is defined by the European Telecommunications Standard Institute (ETSI). In general Maximum acceptable delay limits for VoIP are considered to be 150-200 milliseconds (ms), depending on call quality requirements.[15] Following is table 3.1 that specifies the voice delay requirements as specified by the G.113[17]

Delay	Quality
0 to 150 *msec*	Acceptable to most applications
150 to 400 *msec*	Acceptable for international connections
> 400 *msec*	Acceptable for public network operation

Table2.3: Delay specification

Delay is mainly caused by network congestion which leads to a slow delivery of packets. Furthermore, delay is affected by several parameters or algorithms which can be categorized into: delay at the source, delay at the receiver, and network delay.

1) Delay at the source

The delay of the whole process performed at the sender side before transmitting the voice packet over the network is caused by several components: codec, packetization and process . Codec functions introduce some delays when processing the analogue-to-digital conversion. The more bits compressed, the less the bandwidth required, and the longer the delay added. For packetization delay, it's the time taken to place the chunks of frames in packets which would be transmitted across the network. The third component of source delay is when the computer passes the packets into the network for transmission to other side.[15]

2) Delay at the receiver

The reverse process that carried out at the sender is performed at the receiver adding more delay. process delay and decoding delay including decompressing delay. Additionally, Playback delay is incurred when playing out the voice stream which includes the jitter buffer delay as well.[15]

3) Network delay

Network delay in WLAN environment is the total delay of both WLAN and backbone networks. Queuing, transmission and propagation are other components of network delays. The propagation delay is the delay in the physical media of the network, while transmission delay includes router's delay and MAC retransmission delay .[15]

2.2.7.3 Jitter:

IP network does not guarantee of packets delivery time which introduces variation in transmission delay. This variation is known as jitter and it has more negative effects on voice quality. Since voice packets of the same flow are not received at the same time. Therefore, jitter buffer are introduced to diminish the jitter effect and make the conversation smoothly as it holds a number of packets in a queue before payload. The buffer queue size can be fixed or adaptive which varies based on network condition, voice character , for better performance. Buffer jitter adaptive techniques perform better as it reduces the possibility of buffer overflow and underflow.

2.2.7.4 MOS Value :

The Mean Opinion Score (MOS) provides a numerical indication of the perceived quality from a voice codec during and after the transmission and compression of voice data [11]. Factors that can affect MOS include packet loss, jitter, and end-to-end delay.

It has a numerical ranging from 1-5 , where a higher score is better . MOS 1 indicate very low quality but however present sound .

MOS value can be measured by the following mathematical equation :

$MOS=1+0.035R+(7/10^-6)R(R-60)(100-R)$ Equation (2.1)

In which R is the numerical value called the transmission rating factor , which is used to express the audio transmission quality , the following equation describe who R can be calculated :

$R=R-Is-Id- Ie + A$ Equation (2.2)

In which :

R : Signal to Noise Ratio .

Is : Impairment that occurs simultaneously with the voice signal , that might include quantization noise .

Id : Impairment that are due to effect caused by delay and echo .

Ie : Effective equipment impairment factor .

A : Advantage factor , used to provide compensation for the advantage a system might have a giants a conventional system.

The result of the equation (2.2) falls within the rang 0 and 100 , where higher number means better quality . The table 2.4 show the rang of R .

Ranging of R	*Speech transmission Quality Category*	*User Satisfaction*
90 = R = 100	**Best**	**Very Satisfied**
80 = R = 90	**High**	**Satisfied**
70 = R = 80	**Medium**	**Some Users Dissatisfied**
60 = R= 70	**Low**	**Many User Dissatisfied**
50 = R =60	**Poor**	**Nearly All User Satisfied**

Table 2.4 : Voice Quality vs. R- value

The following figure shows a plot for equation (2.1) :

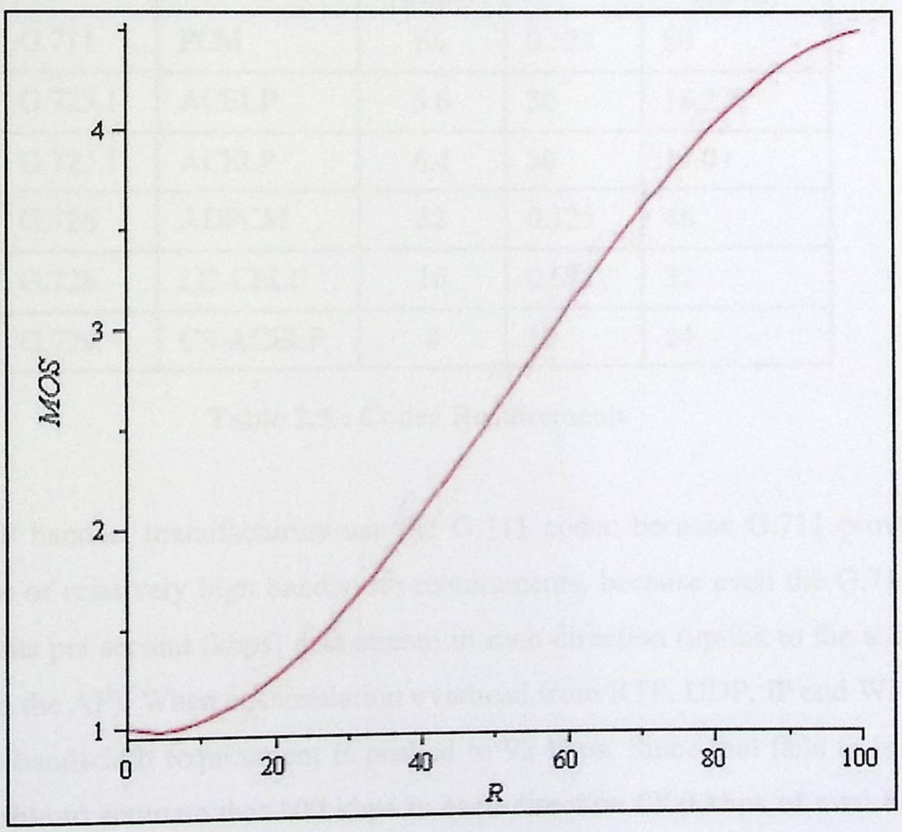

Figure 2.4 : MOS related to R-value

2.2.7.2 QOS requirement and how to improve:

QOS has become a critical issue ,because real-time applications, unlike data (non real-time) applications, are very sensitive to delay. Therefore, QOS of VoIP is an import concern to ensure that voice packets are not delayed, lost or dropped during the transmission over the network.[16]

In wireless network ,voice is digitized with the G.711 coding standard and transported at 64 Kbps while G.711 is the main digital code for toll quality voice service ,a number of more efficient codec's are used for both cellular and voice application ,in VOIP network voice codecs are placed into packet with duration of 5,10 or 20 msec of sample voice ,and these sample are encapsulated in VOIP packets ,the table bellow illustrate the various codecs and their corresponding bandwidth requirements for IPv4.[16]

Coding	Algorithm	Band-width (Kbps)	Sample (ms)	Typical IP bandwidth (Kbps)
G.711	PCM	64	0.125	80
G.723.1	ACELP	5.6	30	16.27
G.723.1	ACELP	6.4	30	17.07
G.726	ADPCM	32	0.125	48
G.728	LD-CELP	16	0.625	32
G.729	CS-ACELP	8	10	24

Table 2.5 : Codec Requirements

Many VoIP handset manufacturers use the G.711 codec because G.711 provides superior quality at the price of relatively high bandwidth requirements. because even the G.711 codec only requires a 64 kilobits per second (kbps) data stream in each direction (uplink to the access point AP and downlink from the AP). When encapsulation overhead from RTP, UDP, IP and Wi-Fi headers is accounted for .the bandwidth requirement is pushed to 92 kbps. Since that fails to include control traffic, it's reasonable to estimate that 100 kbps in each direction (200 kbps of total bandwidth) is necessary for each G.711 call. [17]

VoIP quality of service is measured based on different parameters like delay, jitter, packet loss. VoIP QOS is improved by controlling the values of these parameters to be within the acceptable range.

2.2.8 VOIP Protocols :

There are a number of protocols which is definition is mentioned above that may be employed in order to provide for VoIP communication services. There are many several VOIP protocols such as :

- **H.323** : An ITU Recommendation that defines "Packet-based multimedia communications systems". H.323 defines a distributed architecture for creating multimedia applications, including VoIP.
- **SIP** : Defined as IETF RFC 2543. SIP defines a distributed architecture for creating multimedia applications, including VoIP.
- **MGCP**: Defined as IETF RFC 2705. MGCP defines a centralized architecture for creating multimedia applications, including VoIP.
- **H.248** :An ITU Recommendation that defines "Gateway Control Protocol". H.248 is the result of a joint-collaborate with the IETF. H.248 defines a centralized architecture, and is also known as "Megaco". Megaco defines a centralized architecture.

The main VOIP protocols :

2.2.8.1 RTP: Real-Time Transport Protocol

Most data travelling over the Internet uses the Transmission Control Protocol (TCP) for the transport layer because it guarantees data delivery and integrity.

VoIP does not need the kind of delivery guarantee which TCP provides, so most VoIP transmissions use the faster User Datagram Protocol (UDP) as the transport layer.

RTP is the Internet protocol which transmits real-time data such as audio and video. RTP does not exclusively guarantee real-time delivery of data, but it does provide mechanisms for the sending and receiving applications to support streaming data.

As VoIP doesn't use TCP(Transmission Control Protocol), RTP runs on top of the User Datagram protocol (UDP) instead. VoIP uses UDP as the transport layer. The UDP protocol provides only a direct method of sending and receiving data over an IP network and offers very few error recovery services. UDP has no mechanisms in place to notify the application of any loss in transmission whilst delivering packets of data; it also sends data unordered with no guarantees of the data being presented in the receiving application. All re-ordering of data into the correct format it was sent, is handled by the RTP.[18]

2.2.8.2 H.323:

It is a protocol used for voice transmission over Internet. In addition to voice applications, H.323 provides mechanisms for video communication and data collaboration.

H.323 was originally developed for multimedia streams over a Local Area Network, and was widely accepted in this arena. The standards of H.323 have been widely received and the specification continues to evolve. It is related to a suite of protocols which individually handle things like security, call signaling, and determining the capabilities of each party.

H.323 and SIP differ significantly in design, with H.323 being a binary protocol, and with SIP being an ASCII-based protocol. H.323 was developed before SIP, and seems to be losing ground to SIP as a standard VoIP Protocol. One reason for this is that SIP is much simpler than H.323. However, saying that, H.323 is still one of the major VoIP Protocols in use today.[18]

2.2.8.3 SIP (Session Initiation Protocol):

It is a request-response protocol, dealing with requests from clients and responses from servers. It is becoming the standard for VoIP, and most VoIP service providers and soft phones use or at least offer this protocol. SIP was designed as a multimedia protocol that could take advantage of the architecture and messages found in popular Internet applications, such as voice, music and video. In addition to VoIP, SIP is used for videoconferencing and instant messaging.

When used for VoIP, SIP assigns each user a unique address. This address is independent of actual physical location, so the same SIP address can be used by one user anywhere in the world. SIP also defines standards for a number of different services including caller identification, conference calls, call forwarding, and user mobility.

SIP is a application layer control simple signaling protocol for VoIP implementations using the Redirect Mode. It is a textual client-server base protocol and provides the necessary protocol mechanisms so that the end user systems and proxy servers can provide different services: Call forwarding in several scenarios such as no answer, busy , unconditional, address manipulations, Called and calling number identification , Personal mobility , Caller and called authentication , Invitations to multicast conference Basic Automatic Call Distribution (ACD).

SIP transparently supports name mapping and redirection services, allowing the implementation of ISDN and Intelligent Network telephony subscriber services. These facilities also enable personal mobility which is based on the use of a unique personal identity.

SIP supports five facets of establishing and terminating multimedia communications: User location , User capabilities , User availability ,Call setup, and Call handling.[18]

2.2.9 IP Protocols :

2.2.9.1 The Internet Protocol (IP) Definition :

It is a network-layer (Layer 3) protocol that contains addressing information and some control information that enables packets to be routed. IP is documented in RFC 791 and is the primary network-layer protocol in the Internet protocol suite. Along with the Transmission Control Protocol (TCP), IP represents the heart of the Internet protocols. IP has two primary responsibilities: providing connectionless, best-effort delivery of datagram through an internetwork; and providing fragmentation and reassembly of datagram to support data links with different maximum-transmission unit (MTU) sizes.[19]

2.2.9.2 The IP Header

The term 'IP header' is used to refer to the combined IP, UDP and RTP information placed in the packet. The payload generated by the codec is wrapped in successive layers of information in order to deliver it to its destination. These layers are:

1) IP – Internet Protocol

2)UDP – User Datagram Protocol

3)RTP – Real-time Transport Protocol. see the figure 3.1

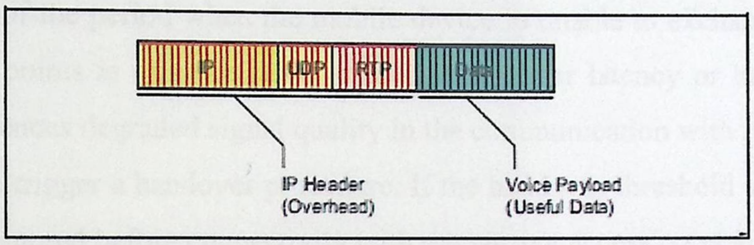

Figure 2.5: IP header of VOIP packet

RTP is the first, or innermost, layer added. This is 12 byte . RTP allows the samples to be reconstructed in the correct order and provides a mechanism for measuring delay and jitter.

UDP adds 8 byte , and routes the data to the correct destination port. It is a connectionless protocol and does not provide any sequence information or guarantee of delivery.

IP adds 20 byte , and is responsible for delivering the data to the destination host. It is connectionless and does not guarantee delivery or that packets will arrive in the same order they were sent.

In total, the IP/UDP/RTP headers add a fixed 40 byte to the payload. With a sample period of 20 ms, the IP headers will generate an additional fixed 16 kbps to whatever codec is being used the payload for the G.711 codec and 20 ms sample period calculated above is 160 byte , the IP header adds 40 byte . This means 200 byte , or 1,600 bits sent 50 times a second result 80,000 bits per second. This is the bandwidth needed to transport the Voice over IP only, it does not take into account the physical transmission medium.

There are other factors, which can reduce the overhead incurred by the IP headers, such as compressed RTP (CRTP). This can be implemented on point-to-point links and reduces the IP header from 40 to just 2 or 4 byte .

2.2.10 VoIP Over WLAN and QOS:

The IEEE 802.11 standard specifies that a mobile device can only be associated with one AP at a time [20], so there is a risk that the communication is interrupted while performing the handover.

The duration of the period when the mobile device in unable to exchange data traffic via its old and new access points is often referred to as the handover latency or handover delay. If the mobile device experiences degraded signal quality in the communication with its access point, it will at some point in time trigger a handover procedure. If the handover threshold value is configured so that a handover is triggered before connectivity with the current access point is lost, then the time to detect movement will not affect the total handover latency. To find candidate access points to re-associate with the mobile device will start to scan the different radio channels.

Since 802.11 networks were designed to carry data, not voice, 802.11 b/g standards have no QOS mechanisms built-in to tell the network to prioritize voice packets over data, so a surge in network traffic can disrupt voice calls. With voice being a real time application, QoS control is essential and without it may lead to end-to-end delays, jitter, out of sequence errors, packet losses and contention (resulting in people talking over each other or the sound breaking up).[21]

2.2.11 VoIP Bandwidth:

The amount of bandwidth required to carry voice over an IP network is dependent upon a number of factors. Among the most important are:

- Codec (coder/decoder) and sample period
- IP header
- Transmission medium

- Silence suppression

The codec determines the actual amount of bandwidth that the voice data will occupy. It also determines the rate at which the voice is sampled. The IP/UDP/RTP header can generally be thought of as a fixed overhead of 40 byte per packet, though on point-to-point links RTP header compression can reduce this to 2 to 4 byte (RFC 2508). The transmission medium, such as Ethernet, will add its own headers, checksums and spacers to the packet. Finally, some codecs employ silence suppression, which can reduce the required bandwidth by as much as 50 percent.[23]

2.2.11.1 The Codec bandwidth :

The conversion of the analogues waveform to a digital form is carried out by a codec. The codec samples the waveform at regular intervals and generates a value for each sample. These samples are typically taken 8,000 times a second. These individual values are accumulated for a fixed period to create a frame of data. A sample period of 20 ms is common. Some codecs use longer sample periods, such as 30 ms employed by G.723.1. Others use shorter periods, such as 10 ms employed by G.729a.

The important characteristics of the codec are:

- The number of bits produced per second
- The sample period this defines how often the samples are transmitted together, these give us the size of the frame. For example, take a G.711 codec sampling at 20 ms. This generates 50 frames of data per second. G.711 transmits 64,000 bits per second so each frame will contain 64,000‚50=1,280bits or 160 byte.

2.2.11.2 Frames and Packets :

Many IP phones simply place one frame of data in each packet. However, some place more than one frame in each packet. For example, the G.729a codec works with a 10 ms sample period and produces a very small frame (10 bytes). It is more efficient to place two frames in each packet. This decreases the packet transmission overhead without increasing the latency excessively.[23]

2.3 OPNET MODELER :

2.3.1 What is the OPNET Modeler :

OPNET MODELER is used to design and study communication networks, devices, protocols and applications. It provides a graphical editor interface to build models for various network entities from physical layer modulator to application processes[22]. It is also provides the flexibility to build very detailed customized models as well to perform general system analysis. Systems are built up in an object oriented way, compiling the models automatically generates discrete event simulations in C language [23].

OPNET is an extensive and powerful simulation software with wide variety of possibilities, enables the possibility to simulate entire heterogeneous networks with various protocols. OPNET is a high level event based network level simulation tool which operates at "packet- level". Originally built for the simulation of fixed networks . It can be used as a research tool or as a network design/analysis tool (end user).

2.3.2 The structure of OPNET

OPNET consists of high level user interface, which is constructed from C and C++ source code blocks with a huge library of OPNET specific functions.

Hierarchical structure, modeling is divided to three main domains:

1) Network domain : Networks + sub-networks, network topologies, geographical coordinates, mobility.
2) Node domain : Single network nodes (e.g., routers, workstations, mobile)
3) Process domain : Single modules and source code inside network nodes

2.3.3 OPNET Modeler Tools

OPNET supports model specification with a number of tools, called editors. These editors handle the required modeling information in a manner that is similar to the structure of real network systems. Therefore, the model-specification editors are organized hierarchically. Model specifications performed in the Project Editor rely on elements specified in the Node Editor, there are some of these tools:

- Source code editing environment
- Network model editor
- Node model editor
- Process model editor
- Antenna pattern editor
- Modulation curve editor (SNR – BER behavior)
- Packet format editor
- Analysis configuration tool
- Simulation tool
- ICI editor (Interface Control Information)
- Probe model tool (organization of result collection)
- Link model editor (properties of fixed link models)
- Path model editor (for routing and modeling virtual circuits)
- Demand model editor (wide scale application modeling)
- OPNET Animation viewer

2.3.3.1 Project Editor :

Project Editor is used to develop network models. Network models are made up of subnets and node models. This editor also includes basic simulation and analysis capabilities. The Project Editor is the main staging area for creating a network simulation. From this editor, we can build a network model using models from the standard library, choose statistics about the network, run a simulation and view the results. It is also possible to create node and process models, build packet formats, and create filters and parameters, using specialized editors that we can access from the Project Editor.

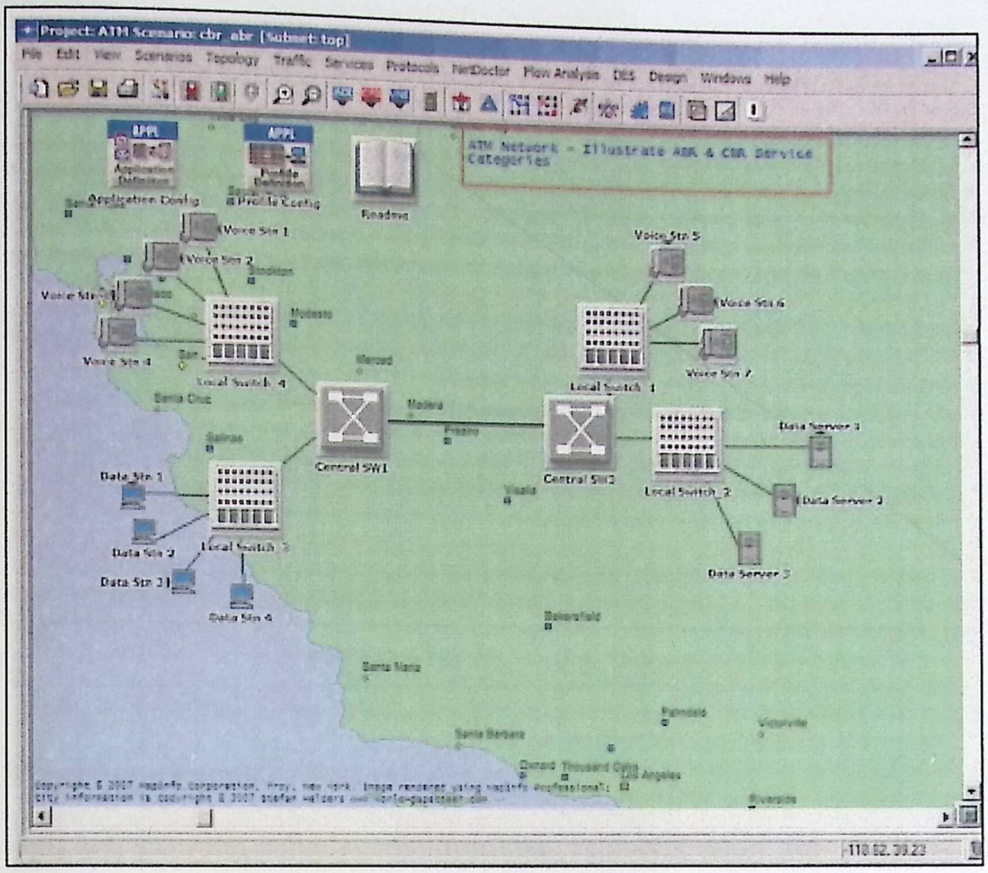

Figure 2.6 : Network Model in the Project Editor

2.3.3.3 Node Editor :

Node Editor is used to develop node models. Node models are objects in a network model. They are made up of modules with process models. The Node Editor lets us define the behavior of each network object. Behavior is defined using different modules, each of which models some internal aspect of node behavior such as data creation, data storage, etc. A network object is typically made up of multiple modules that define its behavior.

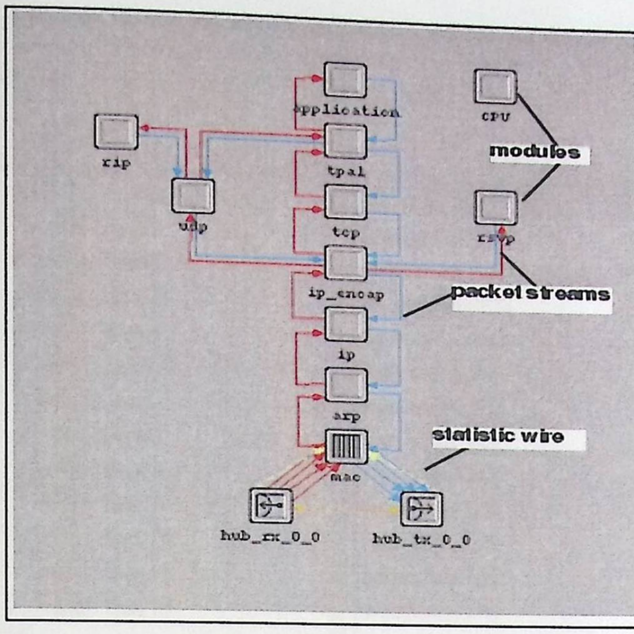

Figure 2.7 : The Node Editor

2.3.3.4 The Simulation Sequence Editor

Although we can run simulations from the Project Editor, we might want to specify additional simulation constraints in the Simulation Sequence Editor. Simulation sequences are represented by simulation icons, which contain a set of attributes that control the simulation's run-time characteristics.

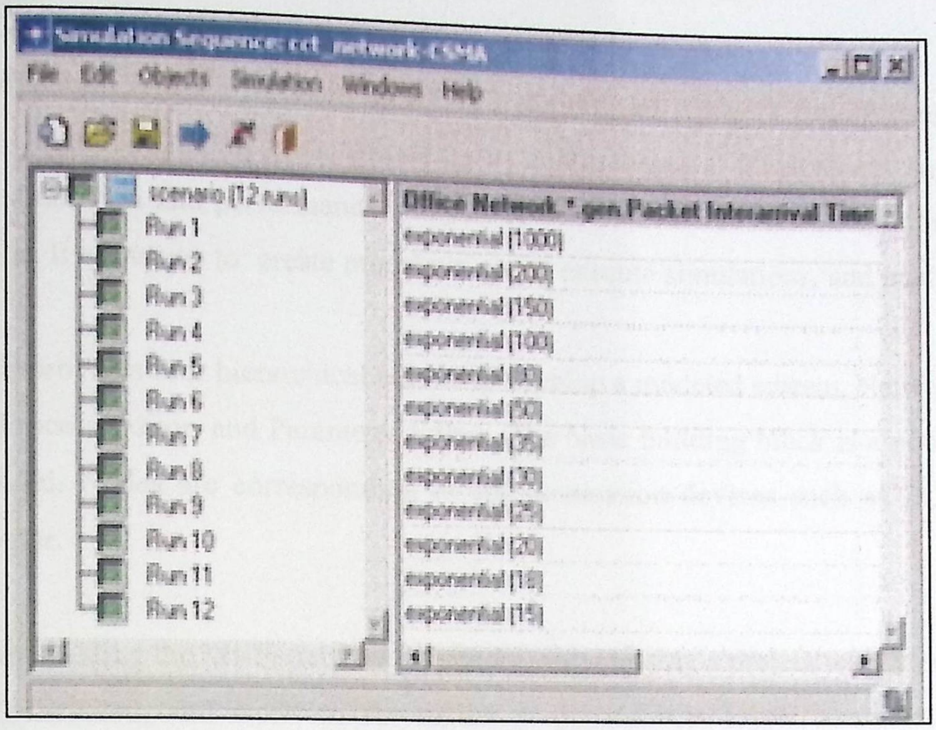

Figure 2.8 : Simulation Sequence Editor

2.3.4 OPNET Software :

OPNET solutions incorporate a high fidelity software model that accurately simulates the behavior of a real-world network. By changing the configuration, link capacity, traffic volumes, and characteristics of this virtual network model, professors and students can accurately predict the impact of these changes on the real network. This capability enables a broad range of studies including:

- Studying various wired and wireless routing protocols
- Visualizing TCP/IP mechanisms and variations
- Understanding LAN/WAN/MAN network architectures
- Designing reliable wireless networks
- Implementing efficient network security

2.3.5 OPNET Implementation

The simulation tool we are using is OPNET (Optimized Network Engineering Tool) . This tool is a set of decision support tools, providing a comprehensive development environment for specification, simulation and performance analysis of communication networks, computer systems and applications. It allows us to create models in great, execute simulations, and analyze the output data.

OPNET provides four hierarchical editors to develop a modeled system, Network Editor, Node Editor, Process Editor, and Parameter Editor. The basic building block is a node, which is an underlying model. Nodes are corresponding to communication devices such as PC, file server, printer, and router.

Example :

We start building the Wi-Fi networking model with creating a project with Model Family "wireless_lan" included, and work on the model at the network layer. A subnet is created to represent the office wireless network. Within the subnet, we put one or two Access Point (AP) as a wireless router to transmit wireless signals, and various numbers of workstations according to different scenarios. The AP is connected to a switch and then connected to a server which provides applications used for the workstations. We also need to define applications and profiles by adding a node for each, and we can associate the work station with the profiles in order to use the applications[10].

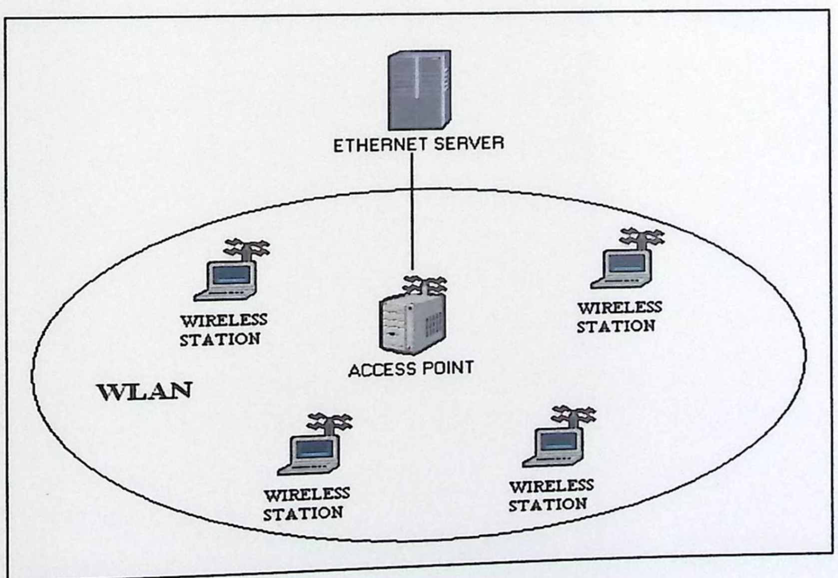

Figure 2.9 : A sample network

3

Conceptual Design

3.1 Work plan

This project will analysis the current state of PPU university wireless network , to study the possibility of providing VOIP technology over it , in order to make free and easy calls with a good quality .

This will be achieve by many steps which will be discussed later in this section , in general the project will study the current state of PPU wireless LAN to know the weakness point in order to be more suitable to provide VOIP application .

First of all the project will introduce the principle of studying the current state at building B , and since the communication between PPU buildings is fiber optic what standing for B will be apply on A and C so we will focus in our studying at network in building B.

The following steps describe the principle of work :

1) Determine the number and the position of the access point at each floor of the building : since the number of access point affect on coverage of wireless network .

2) Using inSSIDer software , to scans networks , tracks signal strength over time, it will show the converge at each floor , and determine the power of each accesses point to determine the dead point in the floors .

3) Some practical experiment at PPU building B during VOIP calls using Skype software in Wi-Fi mobiles in order to be familiar with the problems will faced VOIP calls . There are some of the experiment that will be done :

- Experiment one : A VOIP call will be made between two workstations belong to the same AP . The workstations was close to the access point then one of them moved gradually away from the AP .

  The result : When they close together the quality of voice was acceptable , but when one of them moving away from the AP the call quality gradually decrease , and voice getting noisy during moving away from AP.

- Experiment two : A VOIP call will be made between two workstation one of them used internet for browsing and load data during the call .

  Result : Will be noted that as we increase the use of wireless LAN the quality of call will decrease .

- Experiment three : A VOIP call between two workstation , at the first the belong to the same AP , then one of them moved towered the next AP means out of coverage with respect to the first AP .

- Experiment four : A VOIP call between two users during the time that is a large number of user in the same network and compare it with a call when there is fewer number of user , to resulted the effect of background traffic on the VOIP call quality .

4) then we will start the software part by building the current network in building B on OPNET simulator then we will do several experiment in different cases of traffic and different number of users and analysis it to find the QOS parameter (packet loss , delay and jitter) to know the quality of voice in VOIP calls in the present situation and to find the solutions to be better for free VOIP calls .

3.2 Work algorithms:

Our work divided in two parts the practical part and the OPNET software part.

3.2.1 Current Status Measurements and Performance Analyze algorithm:

1- determine the number of access point and its location in each floor in building B.
2- Scan the wireless network in building B and measure the power coverage by using insider software .
3- doing practical experiment in building B by using Skype software in Wi-Fi mobile to notice the quality of voice calls .
4- Take the result and give logical explain to the real situation in university to be ready to put suggestion to improve it .

3.2.2 Simulation Algorithm:

1- build the current situation of the network at building B in OPNET modeler .
2- define the application that we want in the OPNET network and do some experiment with different scenario.
3- represent the result in excel sheet and draw the result in graphs.
4-analysis the graphs and give logical explain

3.3 Experiment and result :

3.3.1 Coverage scans using inSSIDer:

To determine the real situation of the networks in the PPU building, we scan this it by using inSSIDer software which is **Wi-Fi network scanner** and know the coverage in each floor, and where is the dead point in the floors, This shown in the following figures :

Figure 3.1: Coverage at floor one

This figure show us the coverage at first floor, as shown the power level is between -50dB and -40dB, which mean the coverage is good in it , that because there is an AP in it.

Figure 3.2: Coverage at the second floor

This figure present the coverage situation at the second floor, as we can note the power level average is between -90 and -70 which is very bad , but we can note also that there is some region have a high power level , at this region the coverage was good and call can be with good quality.

3.3.2 VOIP Calls using Skype:

Some experiment using Skype were made , to determine where the packet loss and voice quality, there are some of experiments:

Experiment One : A call was made at the first floor and noted that the voice quality is good , but the call lost during the mobility , we can analyses the result of that , that we move away from AP so the coverage will be decrease , so we loss the call.

Experiment Two: We did a call between first floor and second floor, we note that the voice quality so bad and we can't hear anything clearly, and while movement toward the second floor we loss the connection to the AP so we loss the call, that was because of move away of AP which lays at the first floor, and since second floor has not any AP the coverage is so bad as we introduce in the figure we get from inSSIDer.

We have been done several experiments in PPU building B floors in different cases included the following experiments:

- Experiment one : a VOIP call has been made between two person in the first floor in building B , they were close to access Point, then during the call one of them went away from the AP , noted that VOIP call was unclear during the move far away from the AP and there is a delay on the voice .

- Experiment two : a VOIP call has been done between two persons one of them loaded files from the internet during the call , noted that as increase the traffic the voice become more bad and the call disconnected after few second, that mean this WLAN can't provide a good VOIP service with the usually traffic .

- Experiment three : VOIP call between two persons has been done ,one of them moved away from the first floor to the second floor ,the voice quality became bad and there is a delay in the voice .

- Experiment four :We tried to make a call during the presence of a large number of network users ,but we could not make a call ,because we couldn't log in to Skype software in this situation .

- Experiment five :We had a call between three persons at the same time , the first person was in the first floor ,the second was in the third floor ,and the third person was in the fifth floor ,here we faced a problem third person couldn't hear the sound well ,while the second person hear the voice with more acceptable quality ,that due to the distance between first and the third person.

Chapter Four

4

Modeling and Simulation

4.1 OPNETWLAN model

OPNET Modeler provides a graphical user interface, which enables modeling and simulating networks. OPNET provides four hierarchical editors to develop a modeled system, Network Editor, Node Editor, Process Editor, and Parameter Editor. The basic building block is a node, which is an underlying model. Nodes are corresponding to communication devices such as PC, file server, printer, and router. OPNET provides the flexibility to build very detailed customized models as well to perform general system analysis. Systems are built up in an object oriented way, compiling the models automatically generates discrete event simulations in C language . After simulation it is possible to gather and analyze results with some of the built-in performance statistics features provided by this package.

4.1.1 Object Palette Tree

We start building the Wi-Fi networking model with creating a project with Model Family "wireless_lan" included. A subnet is created to represent the PPU wireless network. Within the subnet there is a number of Access Point (AP) as a wireless router to transmit wireless signals, and various numbers of workstations according to different scenarios.

In order to build any WLAN network , we need number of nodes act as AP , workstations , switch , and server which can obtain from object palette see figure 4.1 .

The AP is connected to a switch and then connected to a server which provides applications used for the workstations. We also need to define applications and profiles by adding a node for each, and we can associate the workstation with the profiles in order to use the applications.

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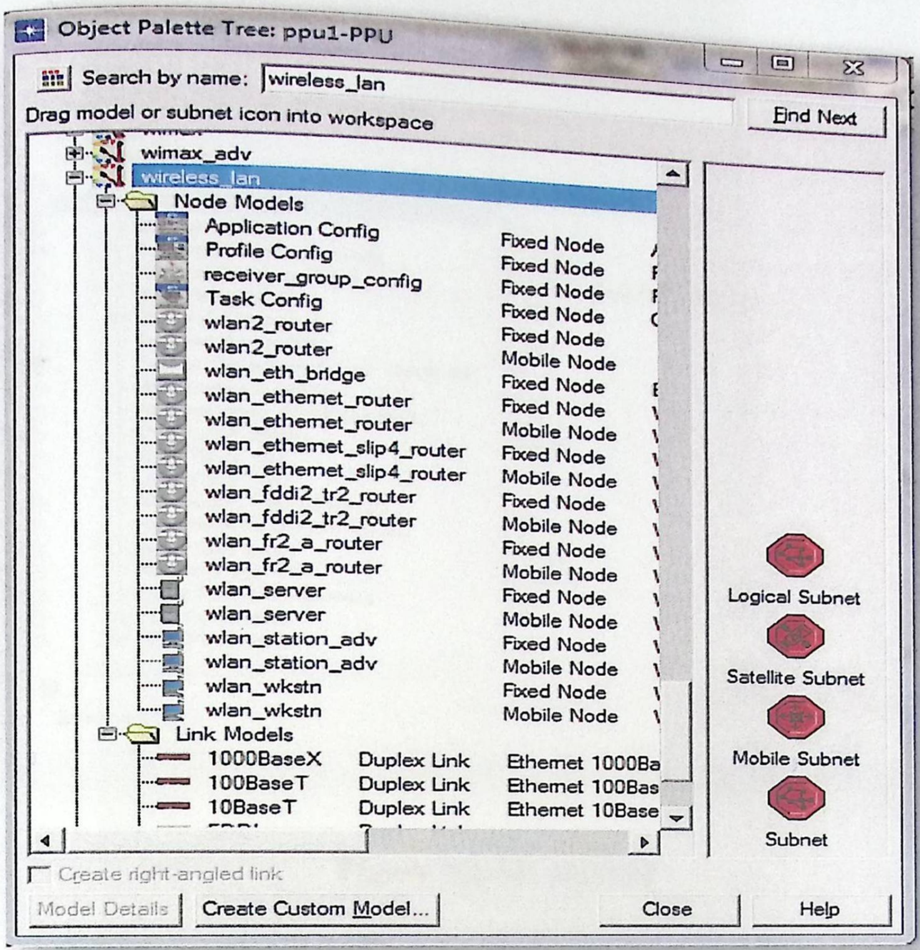

Figure 4.1 :object palette tree window

4.1.2 AP Attribute :

The AP model we use here is "wlan_ethernet_slip4_adv" .This window contain all properties that can be modified for AP , from it can modify on the name of node ,x- position , y - position , threshold , data rate … etc (see Figure 4.2) .

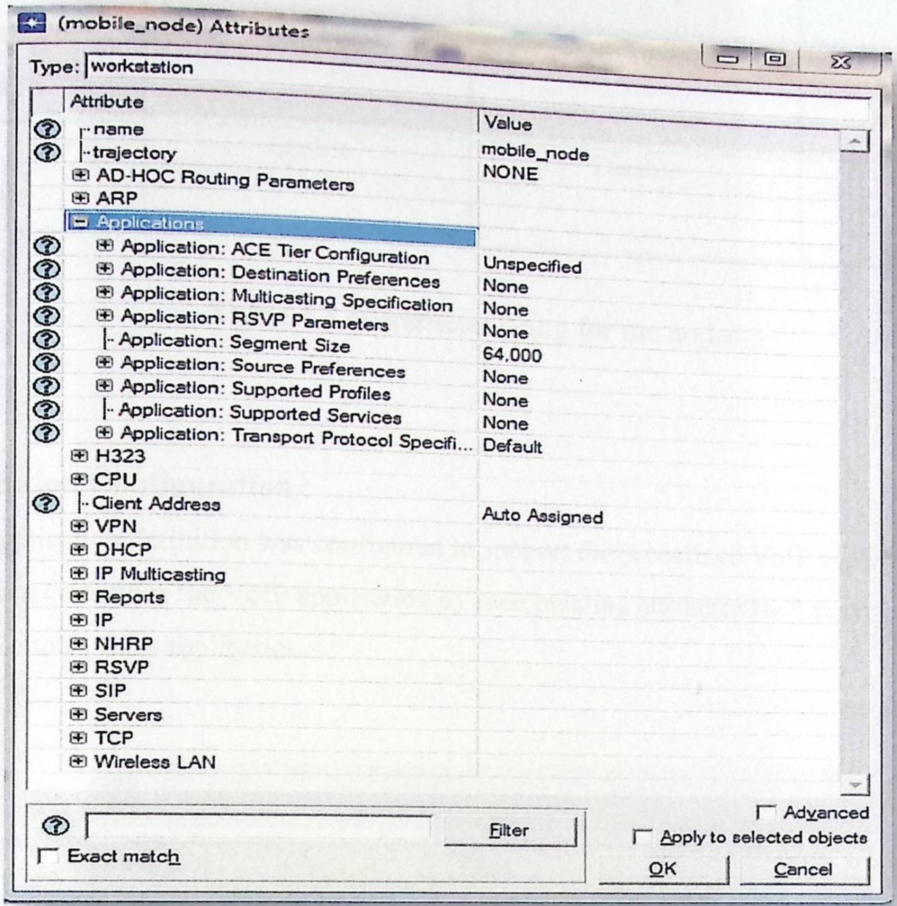

Figure 4.3 : Mobile Node Attribute

4.1.4 Typical simulation parameters used :

Parameter	Value For the AP	Value For the Workstation	Description
Physical characteristic	Extended Rate PHY	Extended Rate PHY	802.11 g
Data rate	54 Mpbs	54 Mpbs	802.11 g
Roaming capability	Enable	Enable	————

Transmit power	.005 W	.005 W	————
Packet reception power threshold	-85	-85	————
Access Point Functionality	Enable	Disable	————

Table 4.1 : Parameter using for the nodes

4.1.5 Application Configuration :

The application definition was configured to support the predefined VoIP application. In figure 4.4 , the user can customize the VoIP application by manipulating attributes such as types of services and encoder scheme to fit their application.

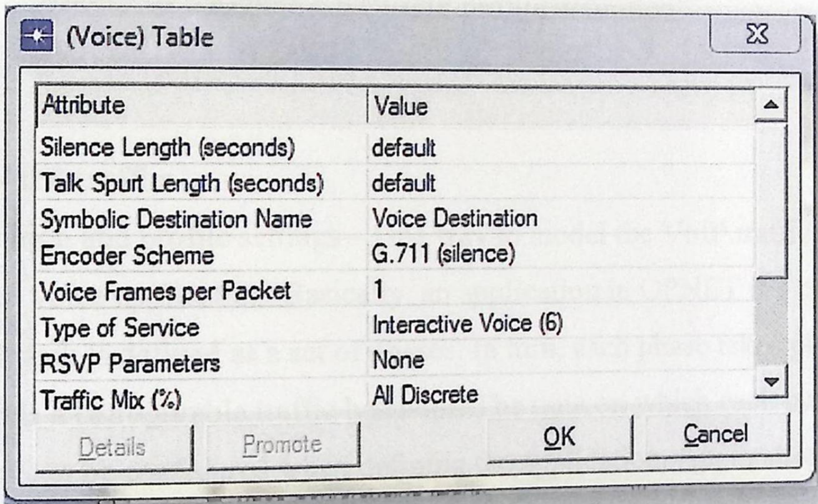

Figure 4.4 : VOIP application definition

4.1.6 Profile Definition :

The profile definition is built on top of the VoIP application where it specifies which node workstation will support VoIP services. Since OPNET only supports P2P or Client-Server relationship for VoIP application, three profile definitions were created as shown in figure 4.5 .

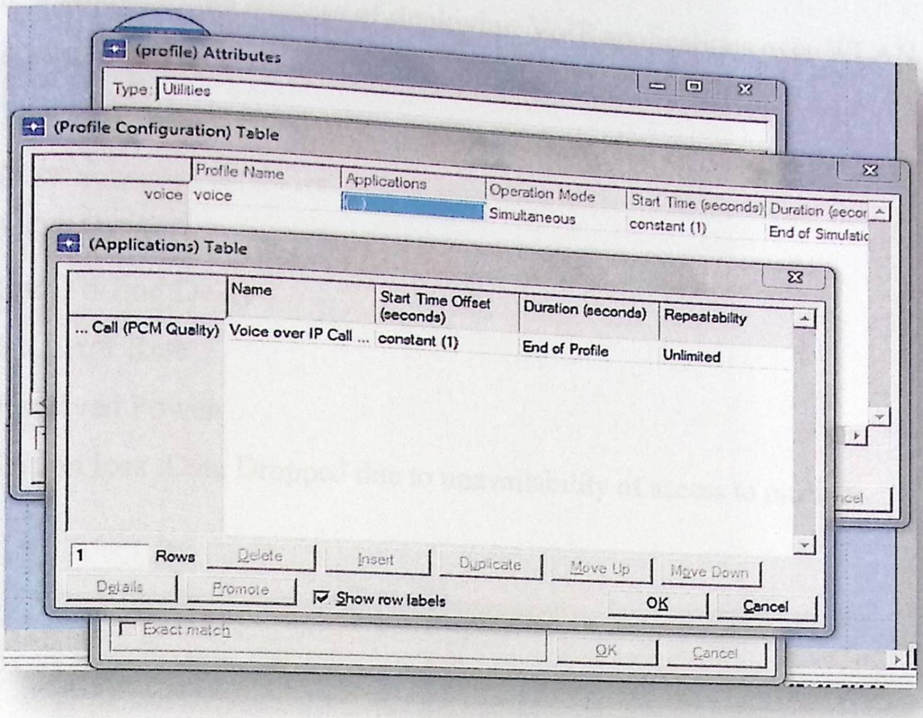

Figure 4.5 : VOIP profile definition

4.1.7 Generating VoIP traffic:

VoIP application and profile settings—One way to model the VoIP traffic in OPNET is to use the predefined voice application. Basically, an application in OPNET is a collection of tasks, of which each task is defined as a set of phases. In turn, each phase takes place between two endpoints and has a configurable traffic behavior. The time on which each task starts and the duration that it takes can be configured when defining the application. Applications can be defined and configured using the Application Definition node. The most important parameter is the '*Encoder Scheme*', which is set to G.711.

After defining and configuring the VoIP application, it is required to configure the way in which workstations will be implementing this application. In general, the behavior of a network workstation is defined through its Profile, which is basically a collection of applications that can be configured to control their start and end times, in addition to their repeatability. Profiles can be defined and configured through the Profile Definition node

4.1.8 QOS Component :

It is crucial to the success of deploying VoIP applications over WLAN to have the ability to support and provision QoS capabilities.

This project concerned on a specific QOS , which are clear at the figure 4.4 below

- ✓ Jitter
- ✓ MOS Value
- ✓ End-To-End Delay
- ✓ Bit Error Rate
- ✓ Received Power
- ✓ Packet loss :Data Dropped due to unavailability of access to medium.

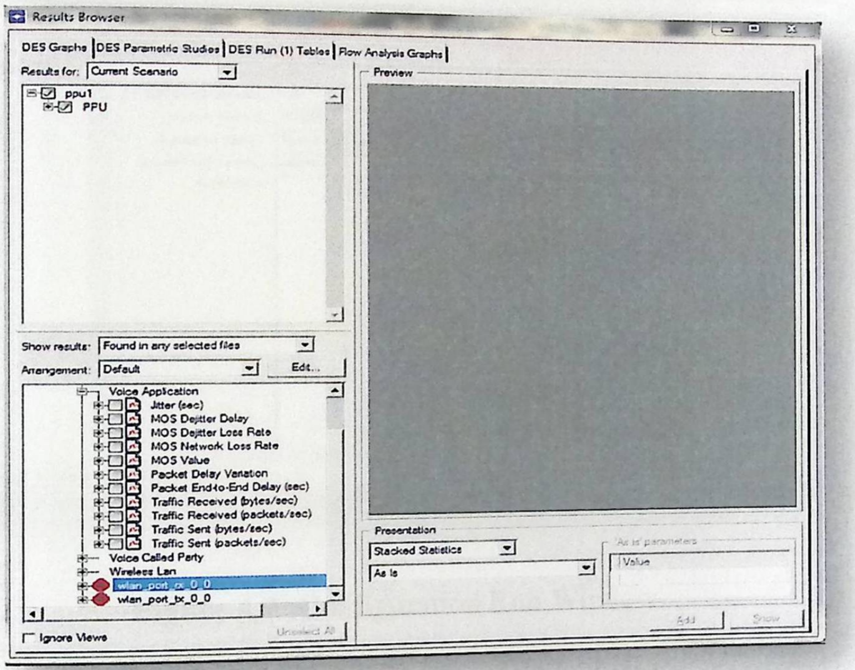

Figure 4.6 : Collected Result

4.1.9 Choose Statistics :

Afterwards and before running a simulation, it is necessary to choose the statistics we want to collect. OPNET does not automatically collect all statistics in the system because there are so many available that you may not have enough disk space to store them.

4.1.10 Run Simulation :

After the completion of building a network of WLAN and press the button of simulation will see the following window (Figure 4.7) , this window contain :

- **Duration** :the time take to simulation .
- **Seed** :random number generators .
- **simulation set name** : the name of senario or simulation .
- **Run** :Launch the simulation .

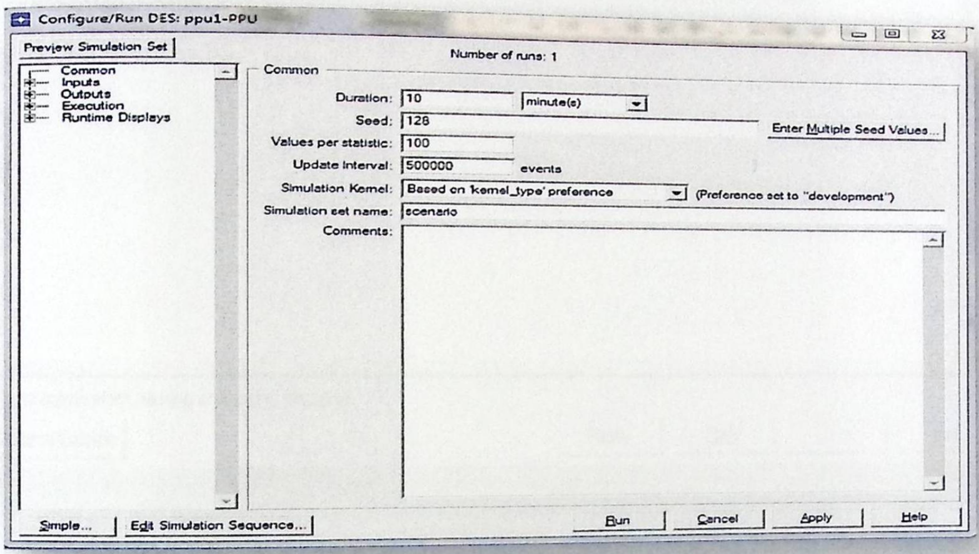

Figure 4.7 :Configuration Run Window

4.1.11 Simulation Progress :

After pressing the button run, it will appear list called Simulation Progress(see Figure 4.6) , this window contain:

- **Elapsed time** :pass away time of the simulation .
- **Estimate remaining time** :The remaining time of the simulation .

- **Simulation speed :** number of operation for one second .
- **Messages :**Description of the simulation process at each stage .

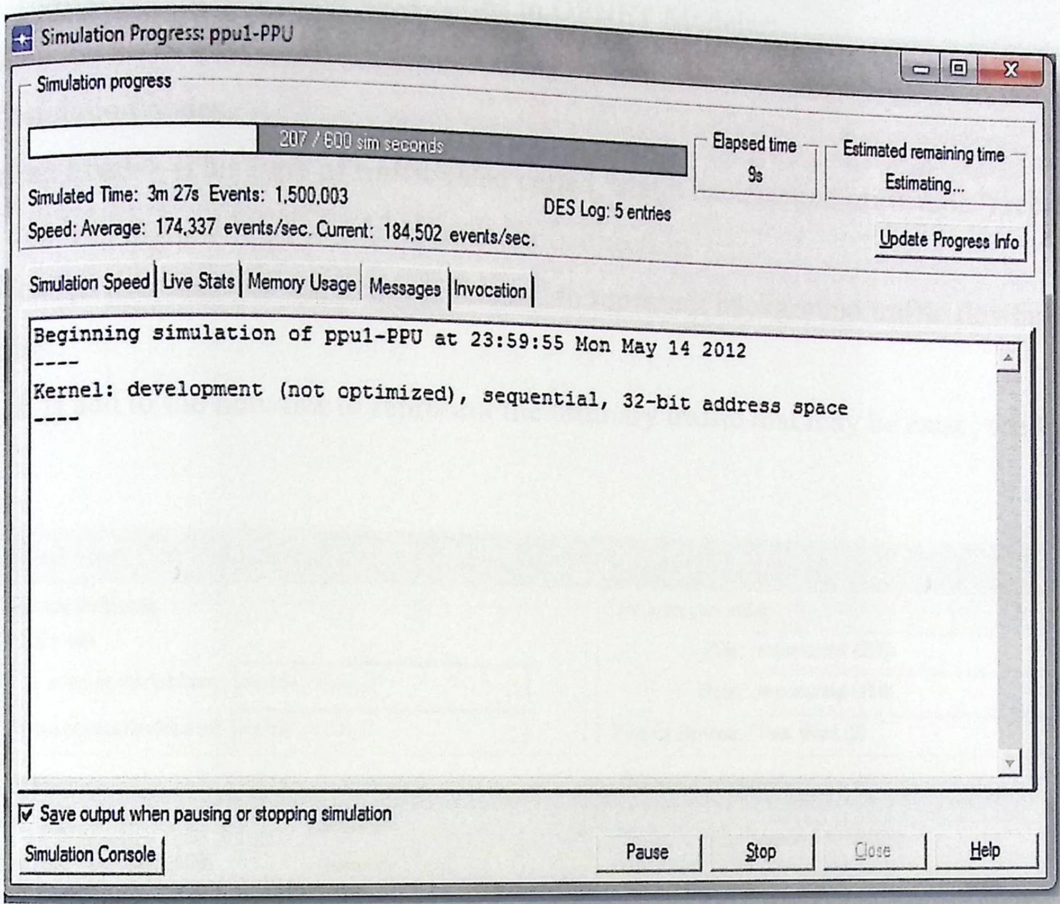

Figure 4.8 : Simulation Progress

4.1.12 View and Analyze Results :

It is the last step of simulation. The results can be watched from the Project Editor or from the Analysis Tool. The Analysis Tool provides the capability to extract data from simulation output files, and to manipulate and display it according to various plotting methods. Data can be manipulated through built-in operations in a different way to get wanted information.

4.1.13 Background Traffic :

Background traffic" is analytically modeled traffic that affects the performance of explicit traffic by introducing additional delays.

Background traffic takes three forms in OPNET Modeler:

1)Traffic Flows—A traffic flow describes an end-to-end flow of traffic from a source to one or more destination nodes.

2)Baseline Loads—This type of traffic (also called "static background utilization") represents traffic as a background load on a link, node, or connection

3) application demand :use application demand to represent background traffic flowing between two nodes .

It is add to the network to represent the ordinary traffic that may be exist , see figure 4.9 .

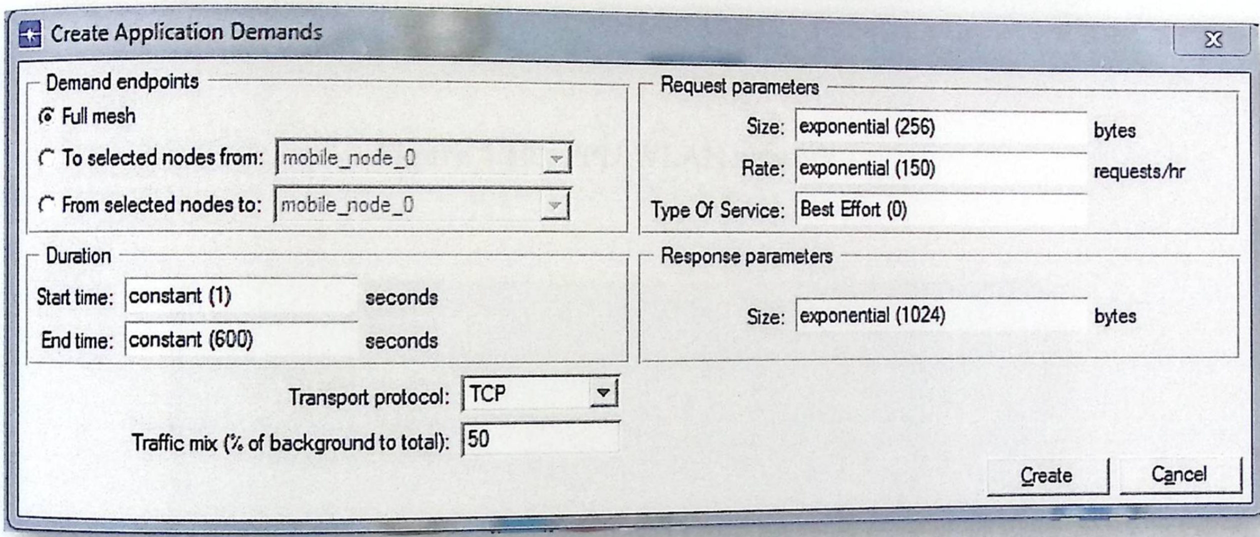

Figure 4.9 : Background traffic definition

4.2 Networks Modeling :

In our project we decide to make simulation for a numbers of scenarios to measure the QOS parameter mentioned at section 4.1.6 .

4.2.1 Network Structure:

The figure 4.10 , show the current PPU buildings WLAN network , and the Figure 4.11 illustrate building B in which the scenarios will analysis , in order to determine the ability of WLAN to provide VOIP application.

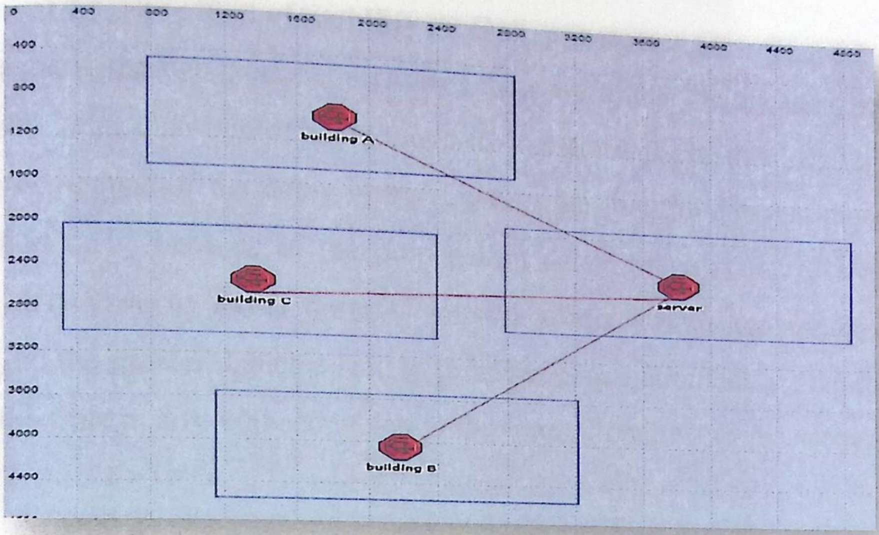

Figure 4.10 : PPU WLAN network

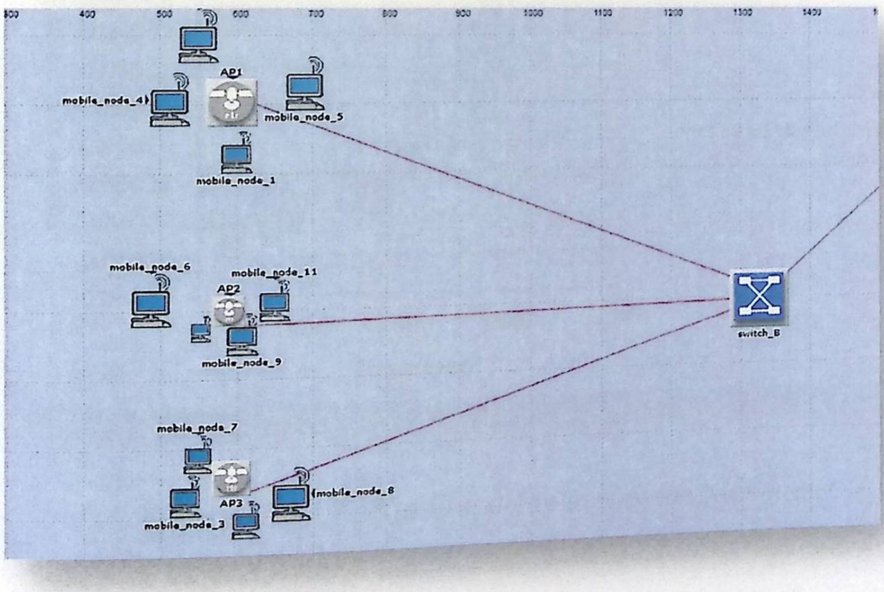

Figure 4.11 : PPU building B network

4.2.2Three AP and Multi-Workstations :

This section will discuss the weakness of the PPU wlan network . In which the network contain 3 AP , transmit power .001W , and -90 packet reception power threshold .

4.2.2.1 Experiment 1 : The effect of mobility on QoS parameter :

At this scenario the two workstation at first they are connect to the same access point and have a VOIP call , and then one of them move to connect with another AP , the figure 4.12 below show the difference of the delay in two cases , get that the delay at the second case is more than the first one , because of the propagation delay which is increase as the distance increase so the data have to travel through network router switch and each adding its own transport delay , so the greater distance lead to greatens propagation delay . the following graph indicate that at the time nearly 400 sec which is the time of connect to the second AP the delay increase .

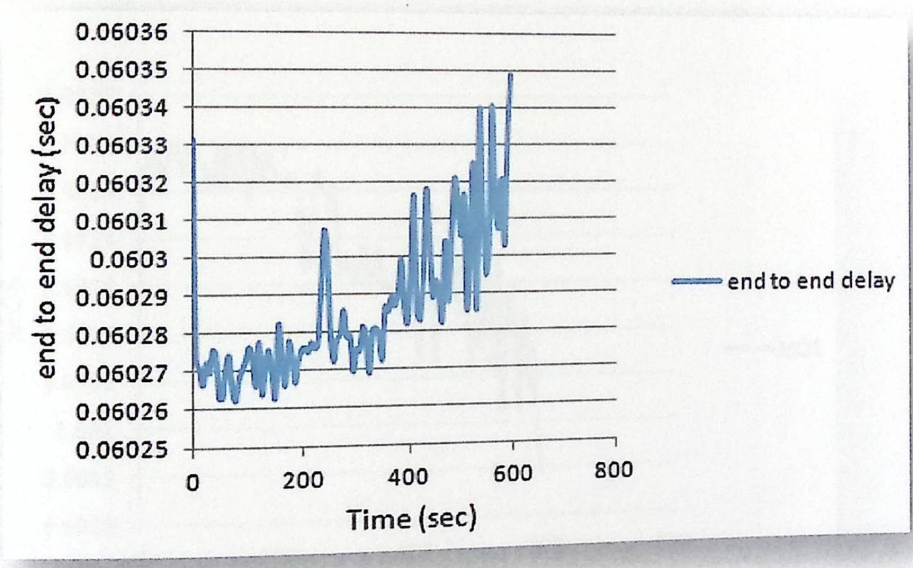

Figure 4.12 : End to End delay during the Movement

According to the jitter , from the next figure 4.13 , it is notice that the variation in packet arrival time (jitter) is usually the result of the network congestion or routing change .

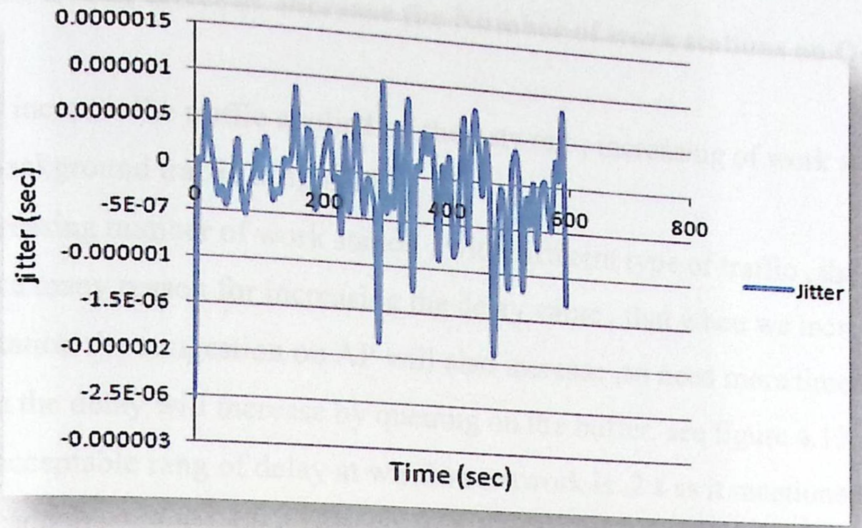

Figure 4.13 :Jitter during the Movement

For the MOS value can be notice that the value of it will be at the safe rang in both cases since the parameter that depends on change with slightly difference .

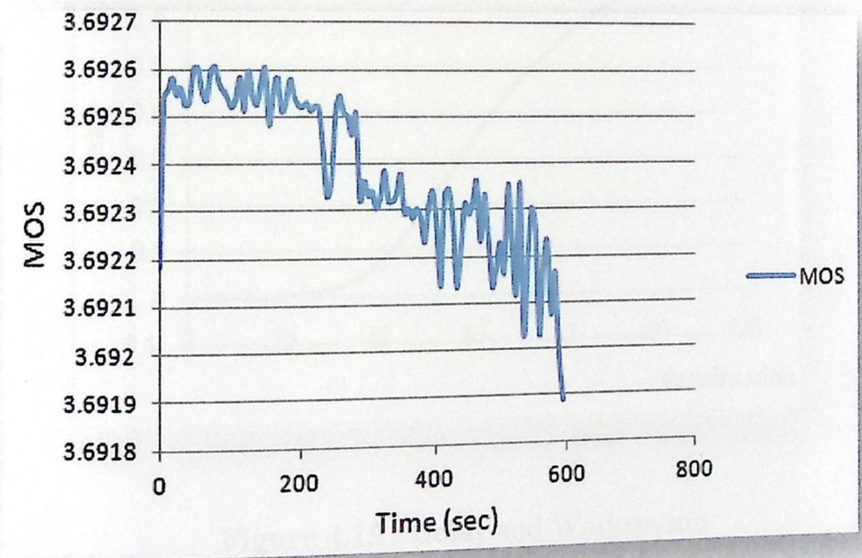

Figure 4.14 : MOS during the Movement

4.2.2.2 Experiment 2 :The effect of increase the Number of work stations on QoS parameter :

In order to increase the traffic applied on the network , increasing of work station with different type of background traffic is applied .

As the increasing number of work station , with different type of traffic , the delay will increase . There are many reason for increasing the delay value , that when we increase the number of work station the congestion on AP will also increase ,so need more time for data processing , mean the delay will increase by queuing on the buffer. see figure 4.15.

Since the acceptable rang of delay at wireless network is .2 s as it mentioned in chapter two , the figure reach this value with nearly 50 workstation , and then increase with high value , which indicate the worse state for the network .

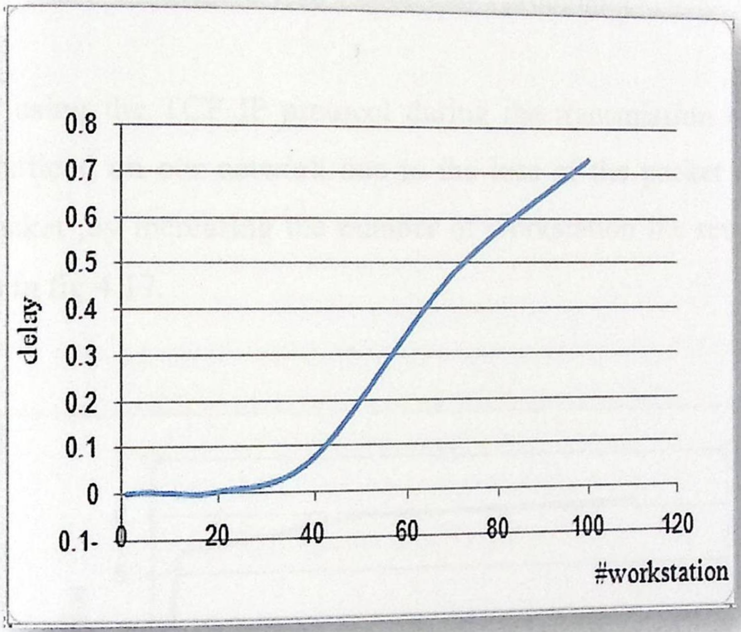

Figure 4.15 : Delay and Workstation

Throughput and Number of work stations :

By increasing the number of work station on the AP we notice that the load increased gradually and the throughput also gradually increase but due to the congestion on the AP the throughput was increased slowly and the drop was appear after that as we see in fig 4.16 .

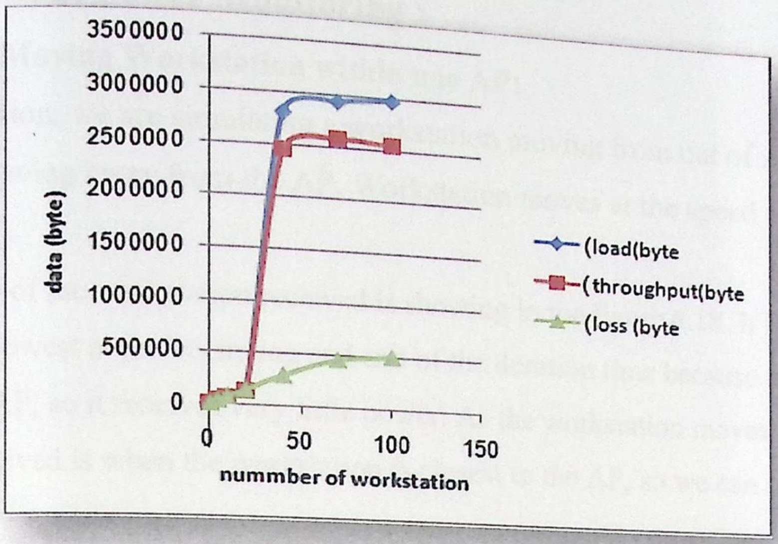

Figure 4.16 : Load and Throughput

Because of using the TCP IP protocol during the transmission of VOIP traffic there transmission was noticed on our network due to the loss of the packet and the TCP protocol retransmit those packet ,by increasing the number of workstation the retransmission value was increased as shown in fig 4.17.

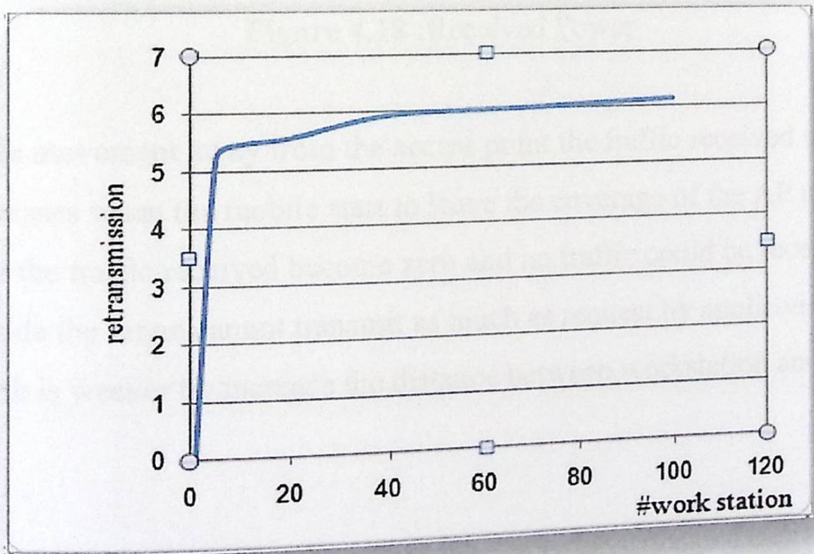

Figure 4.17 : Retransmission Attempt

4.2.3Experiment 3: Power Monitoring :

4.2.3.1 Power of Moving Workstation within one AP:

In this section, we are simulating a workstation moving from out of AP range towards AP and continuing moving away from the AP. Workstation moves at the speed of 0.5 meter per second.

The power of the workstation received is showing in the figure 4.18. It is as expected that the power is very the lowest at the beginning and end of the duration time because the workstation is at very far from the AP, so it receives very little power. As the workstation moves close to the AP, the highest power received is when the workstation is closest to the AP, so we can sees a peak on the graph.

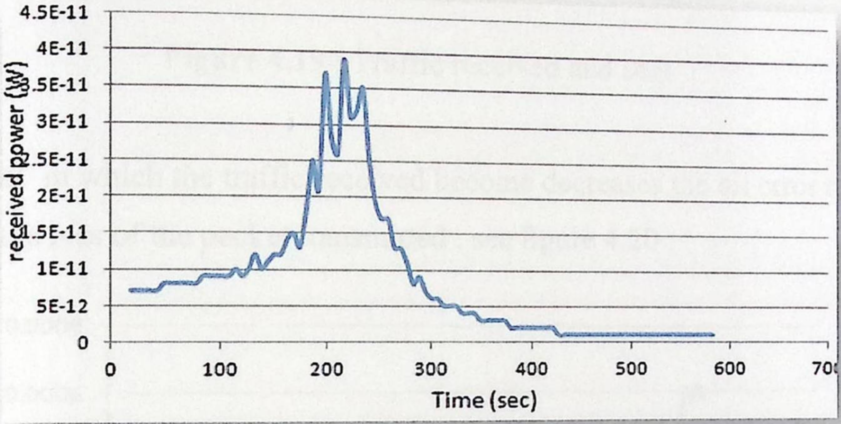

Figure 4.18 : Received Power

Due to the movement away from the access point the traffic received on the mobile workstation decreases when the mobile start to leave the coverage of the AP until reach the dead point in the floor the traffic received become zero and no traffic could be received .The workstation outside the range cannot transmit as much as request by application. This is due to the signal strength is weaker by increase the distance between workstation and AP.

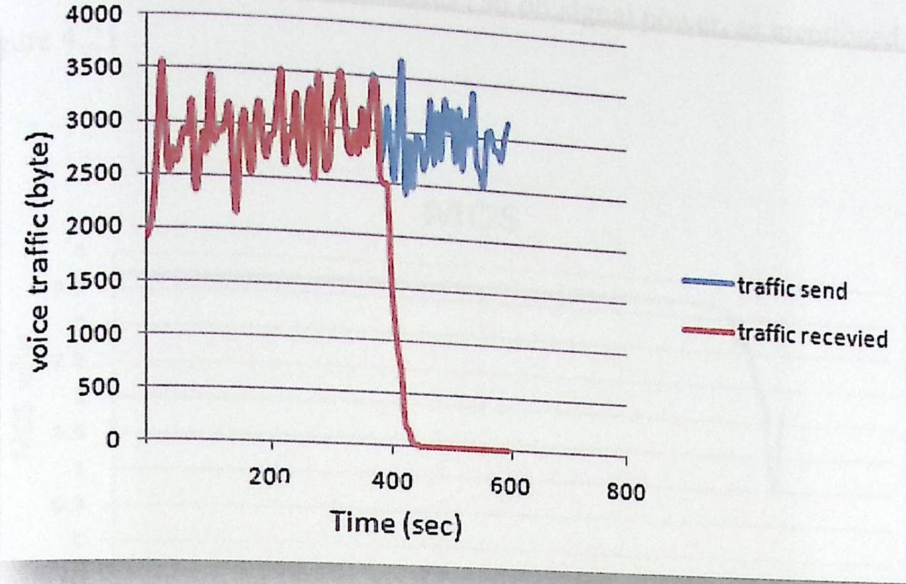

Figure 4.19 : Traffic received and sent

At the point in which the traffic received become decreases the bit error rate was increased due to the loss of the packet transmitted , see figure 4.20 .

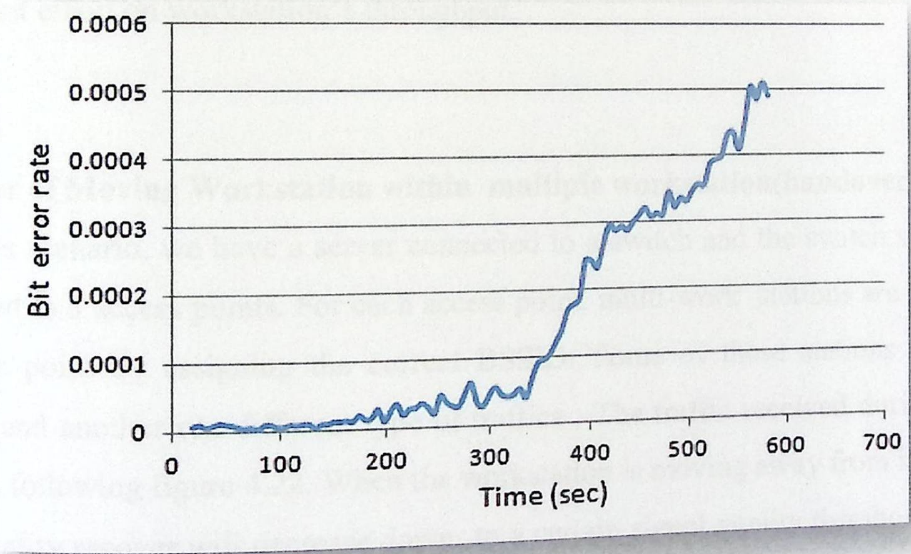

Figure 4.20 : Bit error Rate

When the mobile node existed in the AP region the quality of the call is good(MOS=3.7) then gradually decrease as the distance between the AP and the mobile node decrease since the

MOS value depends on Signal to Noise Ratio , so on signal power, as mentioned in equation (2.1) , see figure 4.21 .

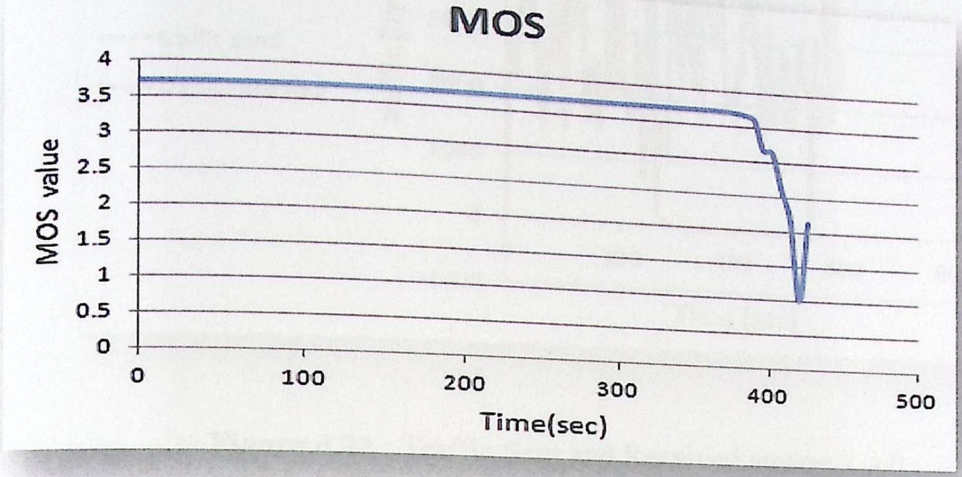

Figure 4.21 : MOS Value at One AP

By the previous scenario we found that distance between workstation and AP is a factor that significant effect on workstation's throughput.

4.2.3.2 Power of Moving Workstation within multiple workstation(handover) :

In this scenario, we have a server connected to a switch and the switch split into 3 links and connected to 3 access points. For each access point, multi-work stations are connected to its closet access point by assigning the correct BSSID. Some of these stations runs the VOIP application and another run different type of traffics . The traffic received during the mobility shown in the following figure 4.22. When the workstation is moving away from it is current AP , the signal quality recover will decrease down to a certain signal quality threshold , at this point the workstation triggers the hand over procedure .

The figure 4 also describe the value of the received power , on the workstation while moving through two AP.

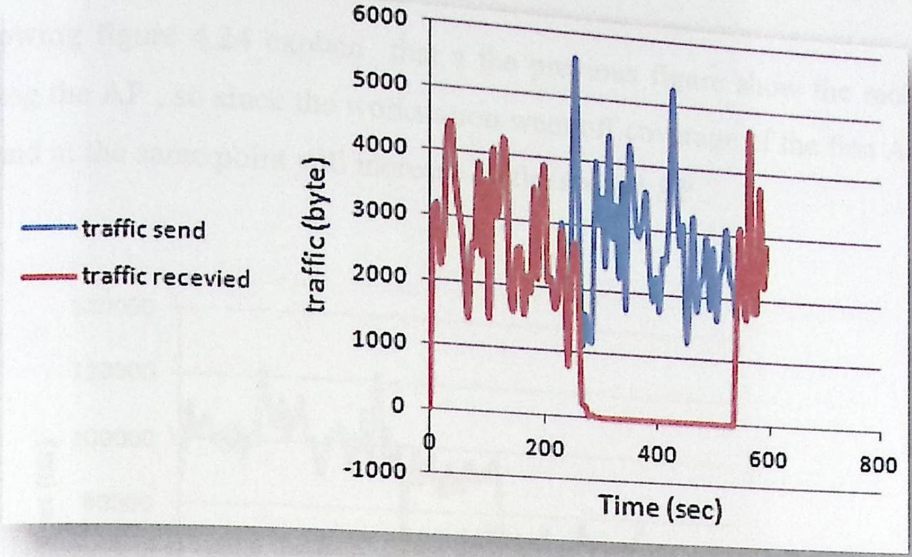

Figure 4.22 : Traffic Sent and Received among 2 AP

The power of the workstation received is showing in the following graph. As the figure 4.23illustrate , at the beginning of simulation the received power was low , and while moving toward AP the power will increase until it is reach the peak value , as the containing mobility far of the AP , toward the next one , the power will decrease gradually because the workstation reach the dead point (which have no coverage from surrounding AP) . With continuity of mobility the received power will increase gradually such the workstation enter the coverage region of the next AP .

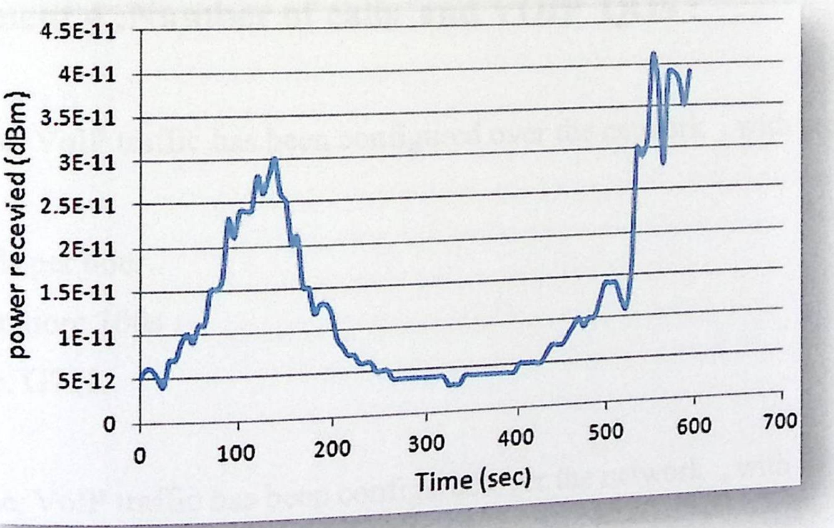

Figure 4.23 : Received Power

The following figure 4.24 explain that a the previous figure show the mobility of the workstation during the AP , so since the workstation went off coverage of the first AP the traffic on it decrease , and at the same point will increase on the second AP .

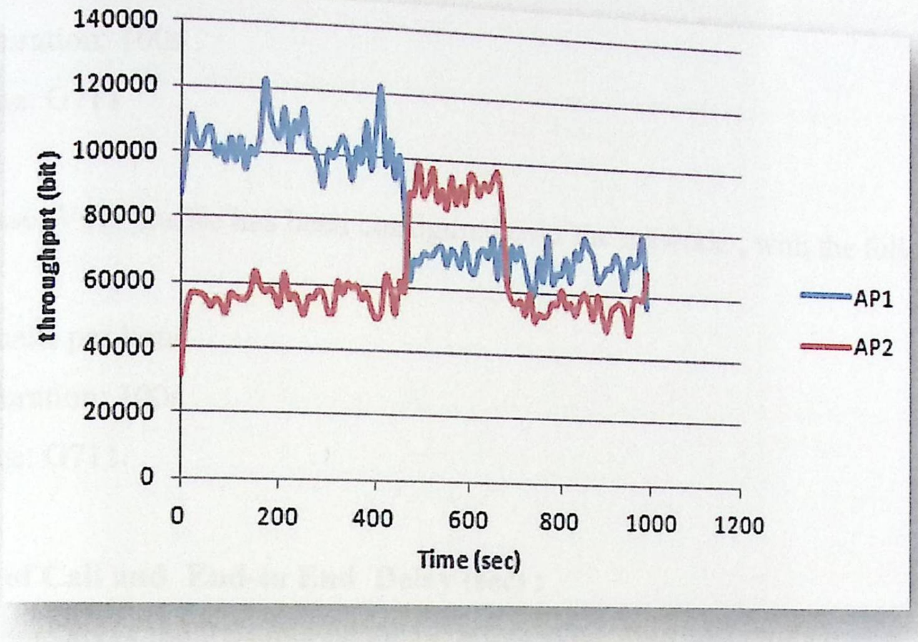

Figure 4.24 : Traffic on APs during Workstation Mobility

4.2.4 Experiment 4 :Number of calls and VOIP QOS :

Case 1 :

In this case, VoIP traffic has been configured over the network , with the following input parameters:

- Call rate: 10 calls per hour.
- Average call duration: 100s .
- Encoder scheme: G711.

Case 2 :

In this case, VoIP traffic has been configured over the network , with the following input parameters:

- Call rate:50 calls per hour.

• Average call duration: 100s .

• Encoder scheme: G711.

Case 3 :

In this case, VoIP traffic has been configured over the network , with the following input parameters:

• Call rate: 150 calls per hour.

• Average call duration: 100s .

• Encoder scheme: G711

Case 4 :

In this case, VoIP traffic has been configured over the network , with the following input parameters:

• Call rate: 300 calls per hour.

• Average call duration: 100s .

• Encoder scheme: G711.

4.2.4.1Number of Call and End-to End Delay (sec) :

This statistic gives the End-to-End delay refers to the time taken for a packet to be transmitted across a network from source to destination .In figure 4.25 a and b , below show that as the number of calls increase the end to end delay and the jitter increase , that refer to the amount of traffic that be transmitted using AP , which lead to more time to processing ,and as the traffic increase on the AP queuing delay will be increase .

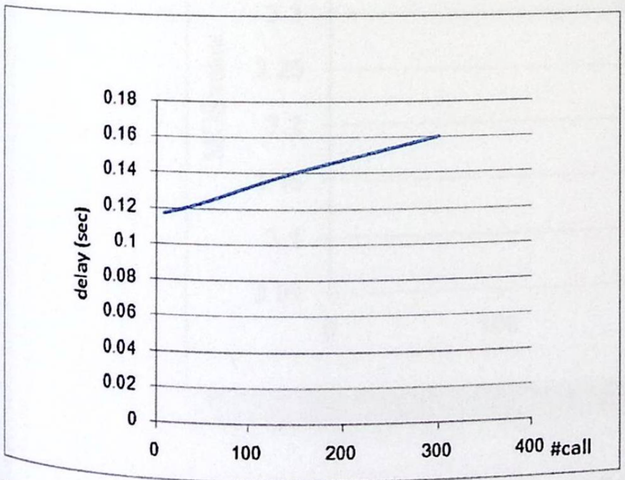

Figure 4.25 (a) :Call Number and Delay

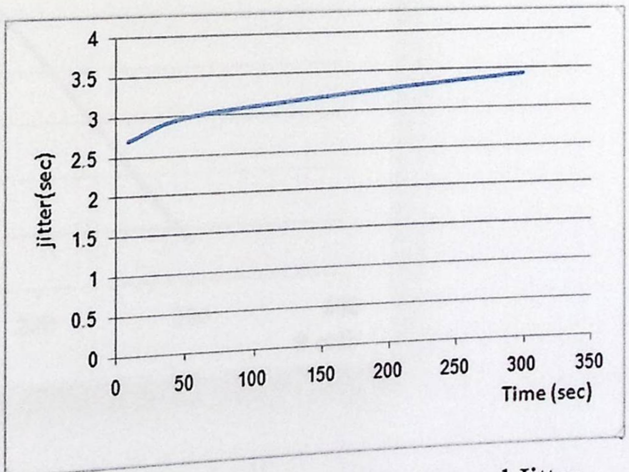

Figure 4.25(b) : Call Number and Jitter

4.2.4.2 Number of call and MOS :

In voice and video communication, quality usually dictates whether the quality is a good or bad one. There is a numerical method of quality voice and video. It is called Mean Opinion Score (MOS). It is gives a numerical indication of the perceived quality of the media received after being transmitted and eventually compressed using codec.MOS is expressed in one number, from 1 to 5.1 being the worst and 5 the best.

MOS Value	Description
5	Perfect. Like face-to-face conversation or radio reception
4	Fair. Imperfections can be perceived, but sound still clear. This is (supposedly) the range for cell phones.
3	Annoying
2	Very annoying. Nearly impossible to communicate
1	Impossible to communicate

Table 4.2 : MOS Value

According to the previous discuss , and related to the following figure 4.26 , which illustrate that as the increasing of the calls number , that will lead to decrease the quality of voice call.

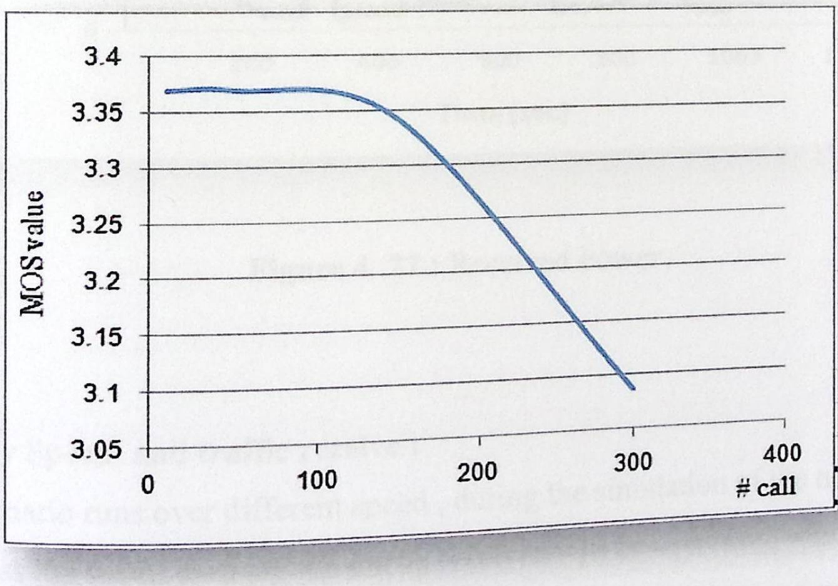

Figure 4.26 : MOS and Number of call

4.2.5 Experiment 5: Five AP and Multi-Workstations :

This section will discuss another improvement WLAN network . In which the network contain of 5 AP , transmit power .005W , and -85 packet reception power threshold , some scenario simulated over this state to get VIOP QOS , and to provide hand over through the network .

4.2.5.1 Received Power :

In this scenario , the workstation move during the five access point , with VOIP called , and with speed of 1 m/s , the figure 4.27 can explain how the received power change during the mobility in which it will increase at five point , when the workstation is coming near the AP .

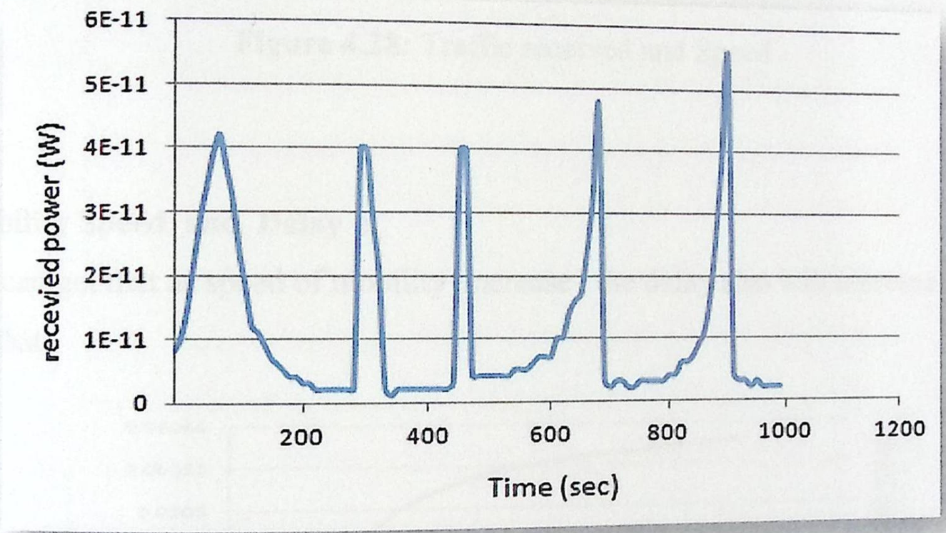

Figure 4 .27 : Received Power

4.2.5.2 Mobility Speed and traffic receive :

This scenario runs over different speed , during the simulation of the network . The figure 4.28 show that as the speed increase the traffic received will be decrease , which can be discuss that as the speed increase then the probability of hand over will increase , so more traffic data are loss during the reception .

By increasing the mobility speed of the workstation with the movement far away from the AP , the mobile arrive to the dead point quickly then the traffic received will gradually decrease with the increase of the speed .

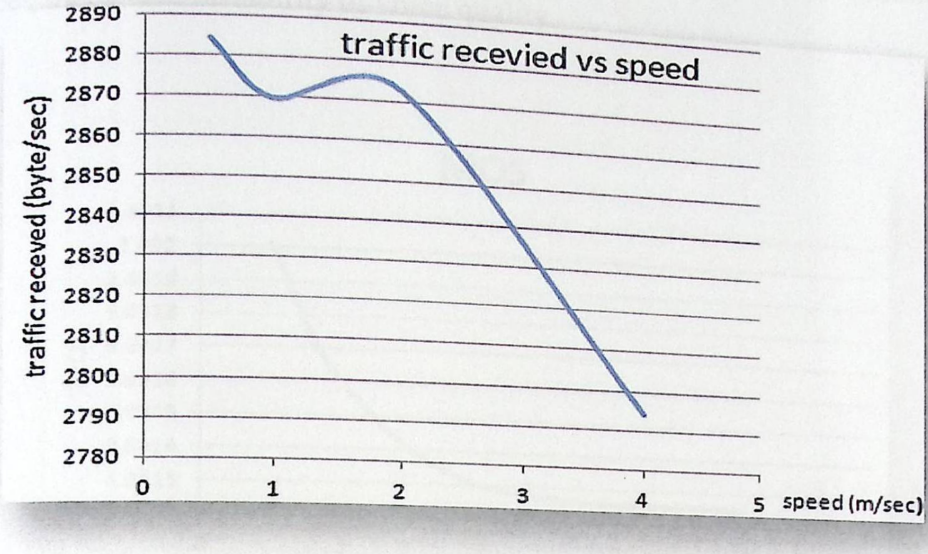

Figure 4.28: Traffic received and Speed

4.2.5.3 Mobility Speed and Delay :

We can get that as speed of mobility increase , the delay also will increase . The figure 4.29show that .

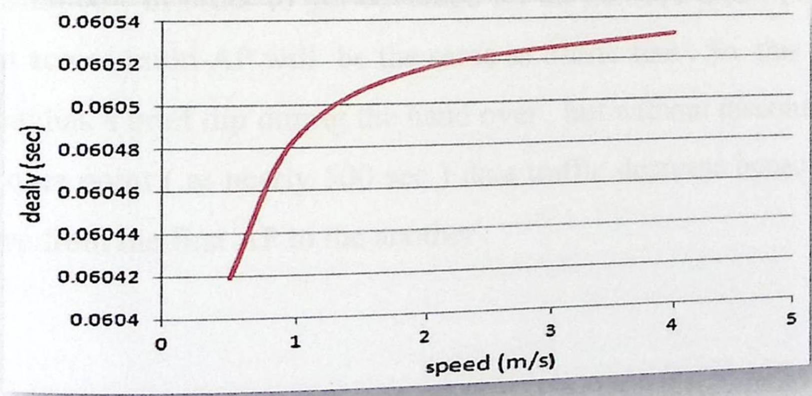

Figure 4.29: Mobility Speed and Delay

4.2.5.4 Mobility Speed and MOS :

As the figure 4.30 illustrate , as the speed of mobility increase the quality of voice call decrease , that because as the speed increase the power received to the mobile will variance multiple time , that cause instability of voice quality.

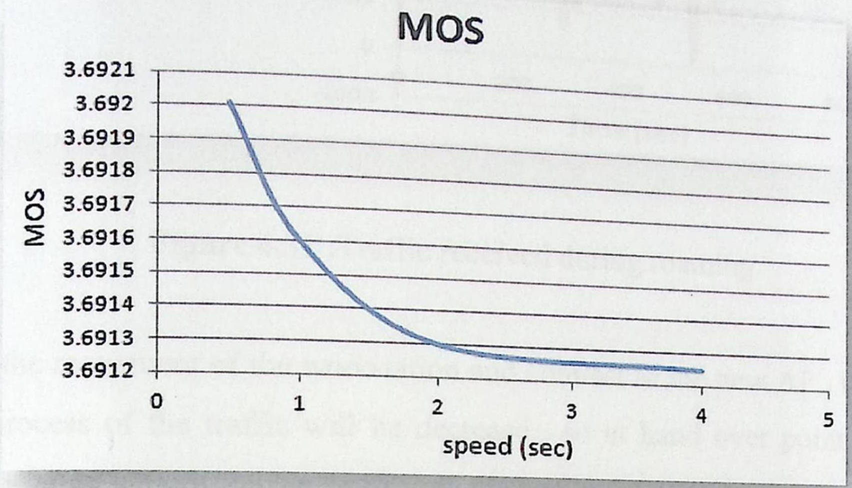

Figure 4.30 : Mobility Speed and MOS

4.2.5.5 Hand Over :

To improve the hand over during the mobility in the network , more AP are add with more power transmitted , in order to get coverage for all network area . So the traffic received during movement across multi AP will be the same as traffic sent . So this figure 4.31 indicate that the throughput has a brief dip during the hand over , but without disconnect of the data , just during the hand over point (at nearly 500 sec) data traffic decrease because the drop of some packet while move from the first AP to the another .

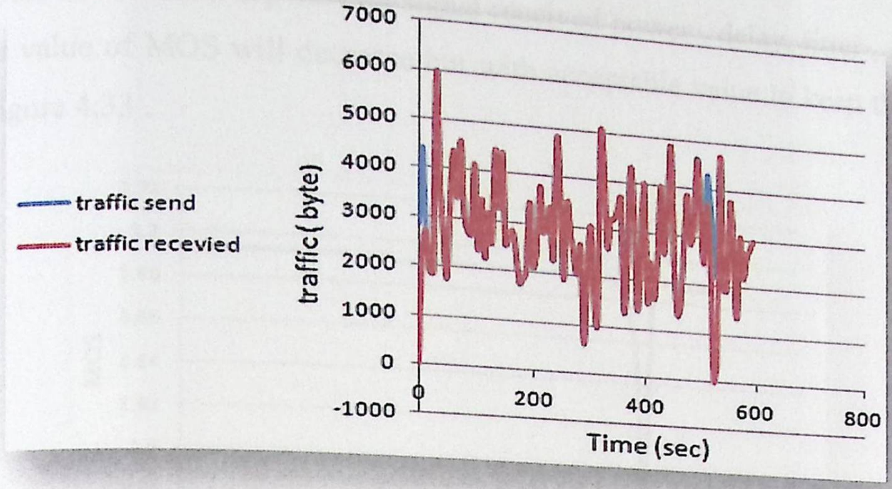

Figure 4.31 :Traffic received during roaming

During the movement of the workstation and connect to the new AP , this process need a time , so the process of the traffic will be decrease , so at hand over point the delay will be increase , because of waiting of the traffic at the buffer lead to loss some packet , cause the increasing of the bit error rate .figure 4.32 .

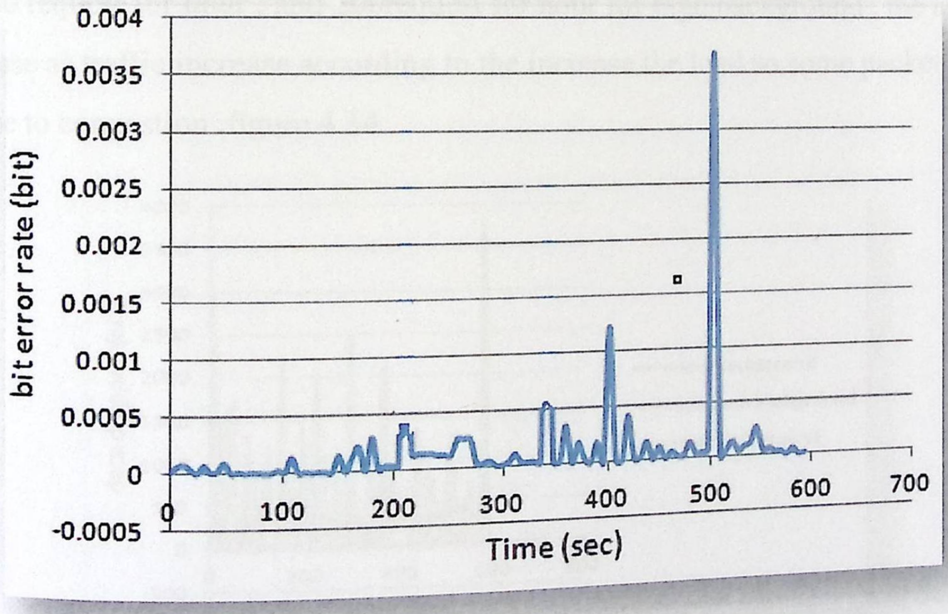

Figure 4.32 :Bit error rate

Since the MOS value depends on signal received power , delay, jitter , and packet loss . that mean the value of MOS will decrease but with acceptable value to keep the call clear .that show in the figure 4.33 .

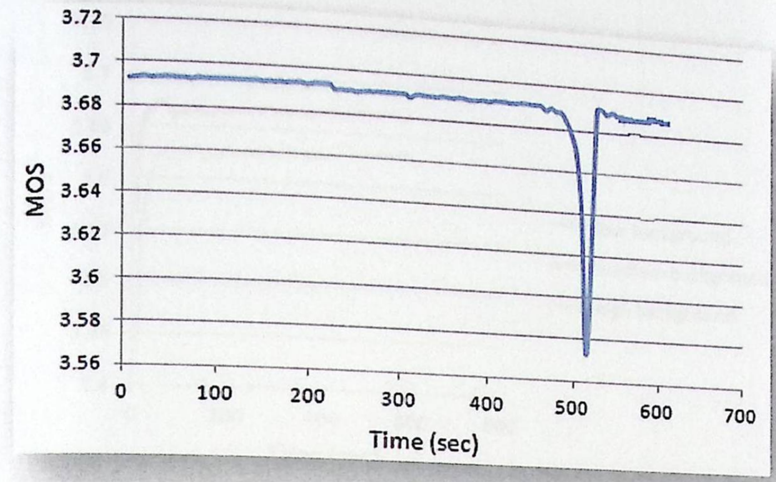

Figure 4.33 :MOS value

4.2.6 Experiment 6: The effect of Background Traffic on QOS :

By applying different amount of traffic with low background 100 request for hour , medium 500 request for hour , and 200request for hour for high background , the amount of data drop increase as traffic increase according to the increase the load so some packet will be dropped due to congestion .figure 4.34

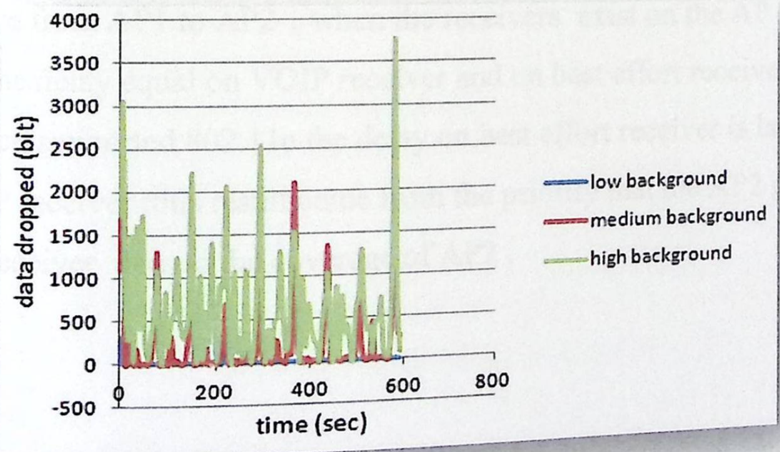

Figure 4.34 :Data Dropped for different background

Figure below , show the light variation of the MOS value , which lead to that background traffic on the network have minimal effect on the packet end to end delay and the voice speech quality .

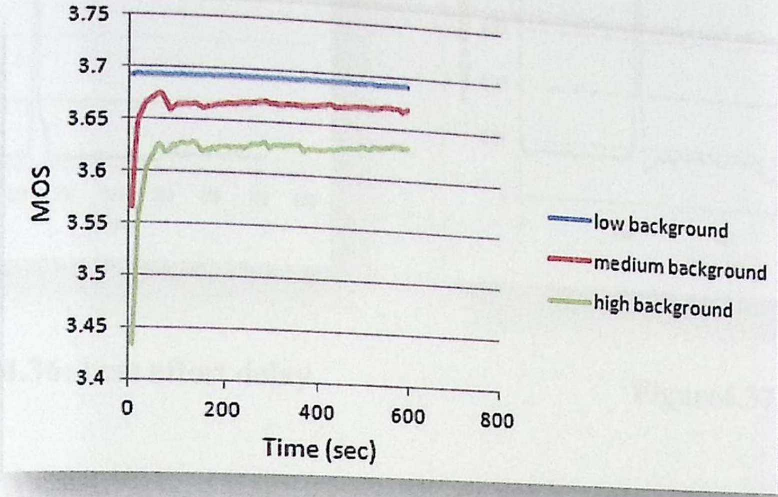

Figure 4.35 :MOS value with respect to different background

4.2.7 Experiment 7 : the effect of adding 802.11e to the access point :

This scenario include two access point the first AP1 does not applied 802.11 e on it and the second AP2 support 802.11e , and it contain 4 mobile node the first 2 node making a VOIP call and the receiver moving from AP1 to AP2 and the next two mobile node have a best effort traffic and also they move from AP1 to AP2 , when the receivers exist on the AP 1 which dose not support 802.11e the delay equal on VOIP receiver and on best effort receiver , but when they move to AP2 which supported 802.11e the delay on best effort receiver is larger than the VOIP delay on the VOIP receiver ;this result come from the priority that the AP2 give priority to voice When t=200 the receiver pass to the coverage of AP2 .

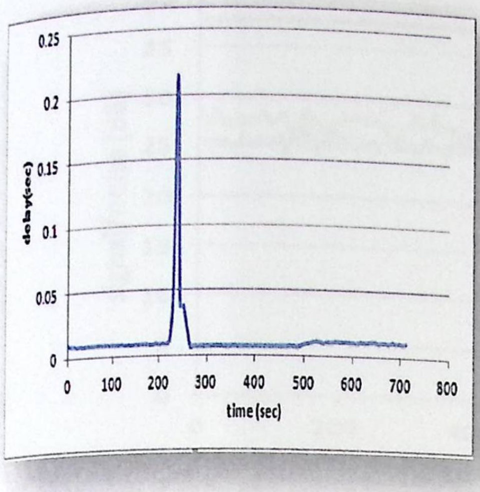

Figure4.36: best effort delay

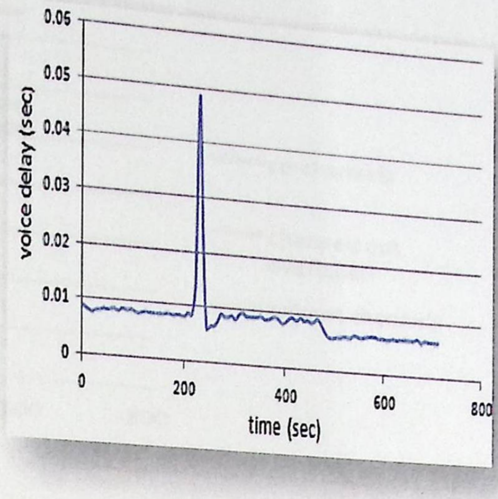

Figure4.37:voice

4.2.8 Experiment 8 : channels interference :

This scenario include 3 AP , we change the entire channel configuration as shown in table:

	Channels not overlap	Co-channels	Adjacent channels
AP1	1	1	1
AP2	6	1	2
AP3	11	1	3

Table 4.3 :AP at different channels

When the channel are not overlapped the signal to noise ratio will be larger than uses adjacent or cochannel due to the high interference on the adjacent and cochannel as shown in fig 4.38 .

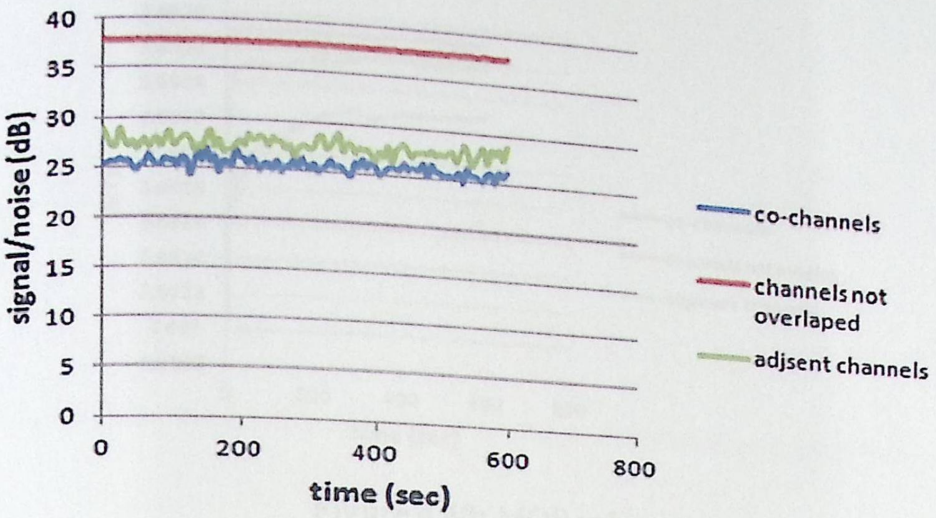

Figure 4.38: signal to noise ratio

at the case of co_channel the end to end delay is higher than the adjacent and the non overlap channels as shown in fig 4.39 .

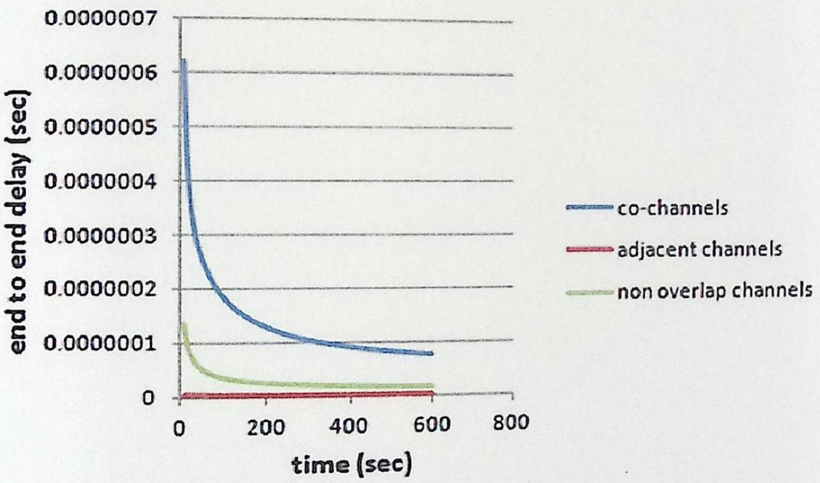

Figure 4.39: end to end delay .

The MOS at the non-overlap case is most better than other, that because of decreasing delay and jitter .

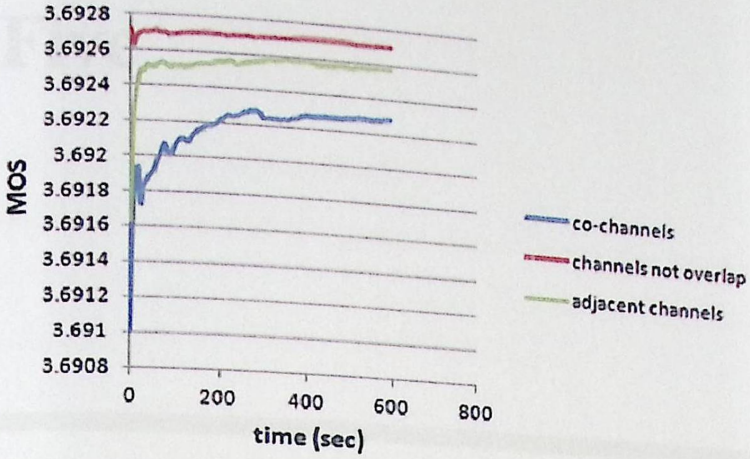

Figure 4.40: MOS value

Chapter Five

5

Conclusions and Future Work

5.1 Conclusions On The Experiments' Results:

5.1.1 Effect of mobility on QOS parameter :

During the movement between the access points when the receiver and sender on the same access point the quality of VOIP call is better than when one of them move away and connect to another access point .

This shown on the delay and the MOS and jitter, where the delay increase when one of them go far to another access point and the MOS value become small than when they on the same AP so the quality of the voice become bad.

5.1.2 Effect of the number of workstation on the QOS parameter on the AP

By increase the workstation around the AP that affect the QOS parameter of VOIP calls ,this increase mean increasing the traffic received on the AP so the load will increase but the throughput will be constant after period of time because the load is very high on it after that the drop will appeared and by the increasing of workstation number the delay and jitter will increase gradually and the MOS will decrease and the quality of the VOIP call become bad .

5.1.3 Effect of increase the number of access point:

By increasing the number of Aps on PPU building we solve the dead point problem in the power coverage ,so when the power decrease on the mobile node it will connected to the strongest AP power and the roaming will applied and the call will finished without any cut .

5.1.4 Effect of call number on VOIP QOS :

The increase of call number mean increasing of load on the AP so in period of time the AP cannot serve all the calls so some of them will blocked and the loss will appeared and the delay will increase also the MOS value will decrease .

5.1.5 Effect of increasing speed on VOIP QOS:

By increasing the speed of the mobile node the VOIP QOS affected in different side ,the traffic received reduced; because the mobile nodes will pass to poor coverage early .and the delay will increase and the MOS will decreasing so the quality of VOIP call will be bad.

5.2 Recommendations :

We hope from the officials to take into consideration the following recommendations:

1. Increase the number of access point to cover the dead point area , and to provide the roaming between the floors, and increase the power of those Aps .
2. Put a tower between the building to cover the area and provide a VOIP call during the movement between the building in the university .
3. apply 802.11e on the AP to give priority to VOIP calls .

5.3 Future work :

This project only considered peer to peer voice calls. VOIP conferencing are suggested as a future research .

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Appendix

The Contact between two workstation :

Time (sec)	End to end delay (sec)	Jitter (sec)	MOS
0	0.060332	-2.53369E-06	3.69199
6	0.060273	3.73873E-08	3.692538
12	0.060271	8.9937E-09	3.692537
18	0.060266	6.55823E-07	3.692183
24	0.060272	2.06338E-07	3.692538
30	0.06027	-7.00021E-08	3.692555
36	0.060276	-3.49855E-07	3.692581
42	0.060275	-2.57079E-07	3.692545
48	0.060263	-4.50006E-07	3.692561
54	0.060263	-2.41685E-07	3.692524
60	0.060271	-4.33762E-08	3.69253
66	0.060274	2.58109E-07	3.692601
72	0.060264	-3.09994E-07	3.692602
78	0.060262	-5.3591E-07	3.692553
84	0.060268	-2.28783E-07	3.692534
90	0.06027	-1.03291E-07	3.692592
96	0.060273	1.53828E-07	3.692607
102	0.060276	-1.40204E-07	3.692571
108	0.060273	3.00926E-07	3.692556
114	0.060266	6.15465E-08	3.692541
120	0.060278	-3.6549E-07	3.69252
126	0.060264	-2.58505E-07	3.692537
132	0.060272	-2.90815E-07	3.692583
138	0.060275	7.71715E-08	3.692512
144	0.060268	1.3332E-09	3.692597
150	0.060263	1.52851E-07	3.692544
156	0.060282	8.64938E-07	3.692526
162	0.060271	6.52109E-08	3.692569
168	0.060266	-1.4617E-07	3.692601
174	0.060278	4.22031E-07	3.692483
180	0.060275	3.59004E-07	3.692551
186	0.060267	-4.24976E-07	3.692582
192	0.060272	-2.20328E-07	3.692512
198	0.060276	6.77567E-08	3.692529
204	0.060276	2.91938E-07	3.692579
210	0.060275	4.09598E-07	3.692543
216	0.060277	-1.13616E-07	3.692525
222	0.060276	-6.87429E-08	3.692521
228	0.060277	-6.79377E-07	3.692529
234	0.060291	3.6781E-07	3.692513
240	0.060307	-4.45528E-07	3.692523
246	0.060303	2.50717E-08	3.692518
252	0.060279	6.14668E-07	3.692428
258	0.060272	1.50927E-08	3.692331

264	0.060278	-4.67577E-07	3.692357
270	0.06028	-6.6315E-08	3.692504
276	0.060286	-2.00896E-07	3.692544
282	0.060279	-4.18065E-07	3.692508
288	0.060279	-1.81239E-06	3.6925
294	0.060269	9.77597E-07	3.692462
300	0.060275	-4.11267E-07	3.692505
306	0.060275	-2.40982E-07	3.692324
312	0.060282	-3.54722E-07	3.692363
318	0.060278	5.6537E-07	3.692331
324	0.060269	-6.8704E-07	3.692341
330	0.06028	1.98902E-07	3.692303
336	0.060281	7.79253E-07	3.692328
342	0.060279	-1.2426E-06	3.692385
348	0.060273	1.0938E-07	3.692319
354	0.060287	5.75137E-08	3.692318
360	0.060285	4.88565E-07	3.692335
366	0.060289	2.54103E-07	3.692377
372	0.060287	4.63247E-07	3.692293
378	0.06029	-4.8168E-09	3.692303
384	0.060299	-9.02158E-07	3.692282
390	0.060287	1.25079E-07	3.692302
396	0.060282	-2.36996E-07	3.692284
402	0.060296	2.33972E-07	3.692231
408	0.060317	-8.32995E-07	3.692311
414	0.060285	-7.08767E-07	3.692341
420	0.060283	1.1129E-07	3.692262
426	0.060294	6.57716E-07	3.692137
432	0.060318	-6.94569E-07	3.692333
438	0.060306	3.52612E-07	3.692346
444	0.06029	7.74859E-07	3.692282
450	0.060294	3.4143E-07	3.692135
456	0.06029	-1.58808E-07	3.692208
462	0.060283	4.50125E-07	3.692313
468	0.060304	-1.71052E-07	3.69229
474	0.060288	-1.90881E-06	3.692315
480	0.060304	-6.4997E-07	3.69236
486	0.060321	-5.78215E-08	3.692231
492	0.060314	-3.14339E-07	3.692332
498	0.060306	-1.16133E-06	3.692235
504	0.060316	1.60439E-07	3.692133
510	0.060286	3.66809E-07	3.692175
516	0.060307	7.56512E-08	3.692229
522	0.060325	-1.17042E-06	3.692164
528	0.060286	-1.57993E-07	3.692353
534	0.06034	-4.45463E-07	3.692223
540	0.060314	2.0024E-08	3.692117
546	0.060295	-7.09049E-07	3.692357
552	0.060302	-6.73257E-08	3.69203
558	0.060341	5.46174E-08	3.692186
564	0.060314	-2.82045E-07	3.692303
570	0.060307	-4.7661E-07	3.692263
576	0.060321	2.7404E-07	3.692028

Moving Workstation within one AP:

Time (sec)	Power received (w)	Traffic send (byte)	Traffic received (byte)	End to end delay (sec)	Jitter (sec)	MOS	Bit error rate (bit)
0	6E-12	1913.333	1913.333	0.060238	2.18E-06	3.69275	4.49E-06
6	6E-12	2006.667	2006.667	0.060241	-6.1E-07	3.692735	8.29E-06
12	6E-12	2286.667	2286.667	0.060249	-4E-07	3.692683	1.09941E-05
18	6E-12	2864	2864	0.060252	2.44E-06	3.692667	6.96429E-06
24	7E-12	3558.667	3558.667	0.06024	-2.5E-06	3.692741	5.40326E-06
30	7E-12	3000	3000	0.060253	-3.3E-08	3.69266	5.9735E-06
36	8E-12	2544	2544	0.060237	2.53E-07	3.69276	5.85634E-06
42	8E-12	2738.667	2738.667	0.060241	-3.6E-07	3.692737	7.21501E-06
48	8E-12	2624	2624	0.060238	-1E-06	3.692755	6.76574E-06
54	8E-12	2777.333	2777.333	0.060243	9.57E-08	3.69272	5.43703E-06
60	8E-12	2928	2928	0.060245	-2.7E-07	3.692711	8.91899E-06
66	8E-12	2889.333	2888	0.060246	-8.2E-07	3.692705	1.07871E-05
72	9E-12	3181.333	3182.667	0.060236	-2.2E-07	3.692765	3.59114E-06
78	8E-12	2380	2380	0.060255	1.85E-06	3.692647	3.59114E-06
84	8E-12	2561.333	2561.333	0.060247	6.92E-07	3.692701	6.07118E-06
90	8E-12	2896	2896	0.060241	-1.4E-06	3.692733	4.76463E-06
96	9E-12	2733.333	2733.333	0.060242	-4.7E-07	3.692726	5.75064E-06
102	9E-12	3434.667	3434.667	0.060244	-1.9E-07	3.692716	3.10228E-06
108	1E-11	2828	2828	0.060242	3.04E-07	3.692726	2.23498E-06
114	1E-11	2928	2925.333	0.060241	5.02E-07	3.692735	7.96022E-06
120	9E-12	2909.333	2912	0.06024	-1.7E-06	3.692741	3.23916E-06
126	1E-11	3008	3008	0.060237	-2.8E-07	3.692756	7.12773E-06
132	1E-11	3164	3164	0.060239	-2E-06	3.692748	6.33899E-06
138	1.1E-11	2596	2596	0.060237	-7.2E-07	3.692758	9.11644E-06
144	1.3E-11	2164	2164	0.060241	8E-12	3.692738	3.99644E-06
150	1.2E-11	2893.333	2893.333	0.060229	-4E-07	3.692807	8.95047E-06
156	1.3E-11	3113.333	3113.333	0.060252	-1.6E-06	3.692666	1.07451E-05
162	1.5E-11	2772	2772	0.06024	8.12E-07	3.692739	4.98986E-06
168	1.7E-11	2541.333	2541.333	0.060241	6.24E-07	3.692734	1.77191E-05
174	1.6E-11	2997.333	2997.333	0.060259	-2.5E-06	3.692625	6.89365E-06
180	2.2E-11	3206.667	3206.667	0.060245	-2.5E-07	3.692713	5.51296E-06
186	2.6E-11	2865.333	2865.333	0.060245	-6.6E-07	3.692713	5.78704E-06
192	2.2E-11	2689.333	2689.333	0.060244	-5.2E-07	3.692719	1.65281E-05
198	2.9E-11	2894.667	2894.667	0.060252	-4.1E-07	3.692669	2.37113E-05
210	3.6E-11	3212	3212	0.060238	1.33E-06	3.692755	4.33165E-05
216	3.3E-11	3498.667	3498.667	0.060241	3.99E-07	3.692735	2.57275E-05
222	3.1E-11	2662.667	2662.667	0.060254	3.28E-07	3.692654	2.80584E-05
228	3.2E-11	2882.667	2882.667	0.060239	-1.8E-06	3.692747	2.84934E-05
234	2.7E-11	2896	2894.667	0.060242	2.59E-07	3.692729	3.64474E-05
240	3.1E-11	3297.333	3298.667	0.060239	-7.7E-07	3.692746	3.15477E-05
246	1.8E-11	2738.667	2737.333	0.060237	1.73E-07	3.692758	1.92326E-05
252	1.6E-11	2648	2649.333	0.060243	-1.7E-06	3.692725	3.0597E-05

258	1.5E-11	3149.333	3148	0.060243	-1.3E-06	3.692723	5.4439E-05
264	1.1E-11	3324	3324	0.060232	5E-07	3.692787	3.23437E-05
270	1.4E-11	2504	2505.333	0.060242	5.03E-07	3.692726	4.991E-05
276	1E-11	3484	3484	0.060247	-3.8E-08	3.692698	2.26391E-05
282	9E-12	3069.333	3069.333	0.060246	-4.6E-07	3.692707	4.33165E-05
288	7E-12	2597.333	2597.333	0.060243	1.4E-11	3.692725	4.63957E-05
294	7E-12	2668	2668	0.060241	-1.4E-07	3.692738	5.10039E-05
300	7E-12	3205.333	3205.333	0.06025	-1.8E-07	3.692681	3.24051E-05
306	6E-12	3305.333	3304	0.060282	1.17E-06	3.692485	3.48725E-05
312	6E-12	3534.667	3536	0.060255	-1.6E-06	3.692652	2.99993E-05
318	5E-12	3208	3208	0.060283	-4.2E-07	3.69248	5.40035E-05
324	5E-12	2825.333	2825.333	0.060319	-3.9E-07	3.692259	6.20153E-05
330	4E-12	2761.333	2761.333	0.060287	5.92E-07	3.692453	3.70887E-05
336	4E-12	2962.667	2962.667	0.060344	5.84E-06	3.692106	6.18687E-05
342	3E-12	2769.333	2769.333	0.060414	1.06E-06	3.69168	6.70414E-05
348	4E-12	3189.333	3189.333	0.060451	-3E-07	3.691453	8.4611E-05
354	3E-12	2849.333	2845.333	0.060539	1.99E-05	3.690918	7.32294E-05
360	3E-12	3150.667	3146.667	0.06078	3.05E-06	3.68945	7.58658E-05
366	3E-12	3510.667	3489.333	0.061175	6.44E-05	3.685776	0.000105641
372	3E-12	3413.333	3370.667	0.061283	2.03E-06	3.681595	0.000119195
378	3E-12	2726.667	2560	0.063311	2.27E-05	3.651788	0.000110385
384	2E-12	2873.333	2494.667	0.063536	-2.3E-05	3.637039	0.000162599
390	2E-12	3213.333	2490.667	0.065492	-5.5E-05	3.579696	0.000168661
396	2E-12	2796	1678.667	0.071497	0.00015	3.488196	0.000200505
402	2E-12	2529.333	1213.333	0.092391	1.74E-06	3.222688	0.00023266
408	2E-12	3094.667	909.3333	0.111525	0.001357	2.969178	0.000249968
414	2E-12	3658.667	685.3333	0.160385	0.005703	2.395826	0.0002706
420	2E-12	2413.333	213.3333	0.111737	0.019204	2.57825	0.000266043
426	2E-12	2905.333	161.3333	0.222194	0.020703	1.817417	0.000290014
432	1E-12	2465.333	37.33333	0.375524	0.04418	1.398747	0.00030279
438	1E-12	2957.333	30.66667	0.331817	0.067534	1.29756	0.00031105
444	1E-12	2928	8	#N/A	#N/A	#N/A	0.00031105
450	1E-12	2648	1.333333	#N/A	#N/A	#N/A	0.00031105
456	1E-12	2776	1.333333	#N/A	#N/A	#N/A	0.00031105
462	1E-12	3312	0	#N/A	#N/A	#N/A	0.00031105
468	1E-12	2706.667	0	#N/A	#N/A	#N/A	0.00031105
474	1E-12	3196	0	#N/A	#N/A	#N/A	0.00031105
480	1E-12	2756	0	#N/A	#N/A	#N/A	0.00031105
486	1E-12	3328	0	#N/A	#N/A	#N/A	0.00031105
492	1E-12	3045.333	0	#N/A	#N/A	#N/A	0.00031105
498	1E-12	3225.333	0	#N/A	#N/A	#N/A	0.00031105
504	1E-12	2742.667	0	#N/A	#N/A	#N/A	0.000311697
510	1E-12	3249.333	0	#N/A	#N/A	#N/A	0.000305111
516	1E-12	2669.333	0	#N/A	#N/A	#N/A	0.00029453
522	1E-12	3152	0	#N/A	#N/A	#N/A	0.000307794
528	1E-12	2920	0	#N/A	#N/A	#N/A	0.000312631
534	1E-12	3414.667	0	#N/A	#N/A	#N/A	0.000332556
540	1E-12	2740	0	#N/A	#N/A	#N/A	0.00325282
546	1E-12	2660	0	#N/A	#N/A	#N/A	0.000325282
552	1E-12	2525.333	0	#N/A	#N/A	#N/A	0.000344618
558	1E-12	3037.333	0	#N/A	#N/A	#N/A	0.000344618
564	1E-12	3065.333	0	#N/A	#N/A	#N/A	0.000345474

Power of Moving Workstation within multiple workstation(handover) :

Time (sec)	Power received (W)	Traffic send (byte)	Traffic received (byte)	Throughput AP1	Throughput AP2
0	3E-12	786.6667	0	101376	32736
6	4E-12	3186.667	3186.667	124704	84192
12	4E-12	2733.333	2733.333	116736	35136
18	3E-12	4266.667	4266.667	101664	62880
24	3E-12	2986.667	2986.667	87072	67584
30	2E-12	1666.667	1666.667	80448	59424
36	2E-12	2400	2400	115296	63744
42	2E-12	2733.333	2733.333	68352	73152
48	2E-12	2853.333	2853.333	99648	60384
54	3E-12	1986.667	1986.667	97056	72768
60	2E-12	4160	4160	106848	47616
66	2E-12	3080	3080	109824	64320
72	2E-12	2773.333	2773.333	85440	79104
78	2E-12	1200	1200	104544	53472
84	3E-12	4613.333	4613.333	118368	94848
90	2E-12	4693.333	4693.333	77568	70560
96	2E-12	4040	4040	117408	62112
102	1E-12	3386.667	3386.667	133824	49536
108	1E-12	1120	1120	79680	72192
114	1E-12	1400	1400	89184	55104
120	2E-12	3186.667	3186.667	97056	77856
126	1E-12	3546.667	3546.667	112320	66720
132	1E-12	1413.333	1413.333	101472	68736
138	1E-12	4680	4680	91584	77088
144	1E-12	2146.667	2146.667	107712	69408
150	1E-12	3066.667	3066.667	124128	70272
156	1E-12	2440	2440	72192	41568
162	1E-12	3853.333	3853.333	140448	72960
168	1E-12	1933.333	1933.333	91968	54432
174	1E-12	3560	3560	80352	91488
180	1E-12	1306.667	1306.667	108384	60192
186	1E-12	2426.667	2426.667	116160	61536
192	1E-12	2546.667	2546.667	118944	73536
198	1E-12	1506.667	1506.667	71808	83136
204	1E-12	5826.667	5826.667	92640	86688
210	1E-12	3960	3960	86112	92640
216	1E-12	2293.333	2293.333	88224	74784
222	1E-12	3533.333	3506.667	127776	59424
228	1E-12	3920	3746.667	114144	70848
234	1E-12	2920	2813.333	66144	78336
240	1E-12	1920	1493.333	112896	68544
246	1E-12	3666.667	2813.333	92256	51360
252	1E-12	1866.667	720	62688	71040
258	1E-12	1973.333	693.3333	85152	80832
264	1E-12	3120	693.3333	84384	69120
270	1E-12	3493.333	573.3333	79008	94176

276	1E-12	1786.667	93.33333	66432	88800
282	1E-12	2093.333	13.33333	71040	123168
288	1E-12	2440	26.66667	93312	103488
294	1E-12	1826.667	26.66667	64128	131424
300	1E-12	3040	13.33333	57888	99264
306	1E-12	2920	0	84768	131232
312	0	2280	0	74976	119328
318	1E-12	3306.667	0	79968	80160
324	1E-12	4440	0	81792	99264
330	1E-12	1480	0	66624	97632
336	1E-12	1600	0	58752	91584
342	1E-12	466.6667	0	101664	105984
348	1E-12	3106.667	0	71328	104544
354	1E-12	2093.333	0	47808	99648
360	1E-12	3333.333	0	54912	97824
366	1E-12	5426.667	0	51840	115968
372	1E-12	4240	0	78336	89472
378	1E-12	3293.333	0	59808	115488
384	1E-12	2973.333	0	66720	85248
390	1E-12	2373.333	0	71136	105408
396	1E-12	3880	0	89184	139872
402	1E-12	2280	0	81216	132960
408	1E-12	3920	0	62208	79104
414	1E-12	3120	0	82656	61056
420	1E-12	4880	0	69120	48192
426	0	3266.667	0	81216	90816
432	0	1973.333	0	73536	57504
438	0	3506.667	0	55296	65664
444	0	1613.333	0	49152	52608
450	0	2386.667	0	58368	68544
456	0	2560	0	58080	49728
462	0	1693.333	0	94176	73056
468	1E-12	2133.333	0	66624	78912
474	0	2773.333	0	63264	74496
480	0	2533.333	0	70848	61728
486	0	2360	0	54720	68064
492	0	2600	0	95904	67968
498	0	3746.667	0	71424	58080
504	0	2386.667	0	76608	64512
510	0	5360	0	66720	86688
516	0	3413.333	0	65472	73920
522	0	2000	0	73152	83904
528	0	2680	0	58176	88128
534	0	2733.333	0	58176	42336
540	1E-11	1760	1773.333	62496	71808
546	1E-11	2746.667	2746.667	67488	88224
552	1E-11	3053.333	3053.333	69024	47232
558	1.1E-11	3666.667	3666.667	86688	56640
564	1.1E-11	3400	3400	46176	59040
570	1.2E-11	4186.667	4186.667	92832	56928
576	1.2E-11	3053.333	3053.333	62016	88992
582	1.2E-11	3066.667	3066.667	72288	68256
600	#N/A	#N/A	#N/A	#N/A	#N/A

Mobility Speed and traffic receive :

Time (sec)	Traffic received (byte) at speed= 1m/s	Traffic received (byte) at speed = 1.5 m/s	Traffic received (byte) at speed =2m/s	Traffic received (byte) at speed =4m/s
0	0	0	0	0
6	520	1733.333	520	1133.333
12	1386.667	2240	4826.667	4880
18	3800	5093.333	2600	3426.667
24	3773.333	3440	2386.667	2560
30	1800	3306.667	3626.667	1333.333
36	1600	1493.333	3400	2360
42	1680	1973.333	2640	2480
48	2680	2560	1493.333	2560
54	4053.333	4093.333	2800	2360
60	4773.333	2146.667	3760	2440
66	706.6667	2773.333	2026.667	760
72	3440	2413.333	3426.667	3133.333
78	2200	1186.667	2026.667	1960
84	2813.333	3973.333	1546.667	1373.333
90	4466.667	1893.333	2880	946.6667
96	3160	3600	2880	1853.333
102	3346.667	3733.333	1160	3333.333
108	1760	2906.667	4146.667	4066.667
114	4200	2493.333	2893.333	3013.333
120	4306.667	3826.667	2826.667	3680
126	3440	3000	3746.667	493.3333
132	1813.333	4506.667	4693.333	2826.667
138	5106.667	2120	3826.667	3773.333
144	3293.333	1920	3173.333	2720
150	2640	1733.333	3280	2106.667
156	4986.667	2720	3026.667	2266.667
162	2426.667	2773.333	2826.667	2653.333
168	2640	2613.333	1586.667	3480
174	1386.667	1613.333	1613.333	2866.667
180	3826.667	3386.667	2053.333	2866.667
186	4026.667	1880	2146.667	1906.667
192	1693.333	2240	1426.667	5013.333
198	1813.333	2720	5133.333	4746.667
204	1493.333	3440	4053.333	2680
210	1960	3760	2333.333	2240
216	2320	2773.333	3773.333	2160
222	2213.333	4560	3626.667	1613.333
228	1933.333	3093.333	2586.667	2320
234	2853.333	2440	4093.333	4253.333
240	3760	4040	3013.333	933.3333

Mobility Speed and Delay :

Time (sec)	Delay (sec) at speed= 1m/s	delay (sec) at speed = 1.5 m/s	Delay(sec) at speed =2m/s	delay(sec) at speed =4m/s
6	0.060173	0.06018	0.060176	0.060181
12	0.060181	0.060181	0.06019	0.060181
18	0.060174	0.060176	0.060213	0.060192
24	0.060188	0.060175	0.060186	0.060424
30	0.060185	0.060187	0.060214	0.060532
36	0.060179	0.060177	0.060172	0.060507
42	0.060172	0.060174	0.060222	0.060529
48	0.060187	0.060193	0.06023	0.060569
54	0.060173	0.06018	0.06054	0.060539
60	0.060187	0.060216	0.06053	0.060536
66	0.060186	0.060172	0.060537	0.060608
72	0.060185	0.060193	0.060567	0.060548
78	0.060178	0.060187	0.060563	0.060694
84	0.060172	0.060179	0.060505	0.060516
90	0.060178	0.060177	0.060519	0.060559
96	0.060197	0.060187	0.060541	0.060551
102	0.060172	0.060383	0.060547	0.060539
108	0.060201	0.060566	0.060558	0.06059
114	0.060172	0.060523	0.06052	0.060512
120	0.060187	0.060553	0.06056	0.06053
126	0.060176	0.06051	0.060528	0.06053
132	0.060195	0.060568	0.060548	0.060554
138	0.060193	0.060578	0.060555	0.060614
144	0.060175	0.060524	0.060532	0.060519
150	0.06018	0.060534	0.06058	0.060518
156	0.060183	0.060506	0.060513	0.060525
162	0.060212	0.060549	0.06057	0.060546
168	0.06022	0.060531	0.060529	0.060522
174	0.060177	0.060554	0.06054	0.060521
180	0.060219	0.060556	0.060604	0.060547
186	0.060172	0.060519	0.060519	0.060533
192	0.060201	0.060535	0.060558	0.060561
198	0.060183	0.060538	0.060523	0.060539
204	0.060172	0.060527	0.06054	0.060522
210	0.060459	0.060549	0.060575	0.060539
216	0.060523	0.060517	0.060515	0.060536
222	0.060534	0.060551	0.060564	0.060526
228	0.060537	0.060554	0.060577	0.060538
234	0.060531	0.060548	0.060527	0.060569
240	0.060581	0.060598	0.060526	0.060519
246	0.060527	0.060505	0.060538	0.06052

Mobility Speed and MOS :

Time (sec)	MOS at speed= 1m/s	MOS at speed = 1.5 m/s	MOS (sec) at speed =2m/s	MOS at speed =4m/s
0	3.693194	3.693173	3.693149	3.693118
6	3.693137	3.693194	3.693132	3.693181
12	3.693154	3.693137	3.693104	3.693128
18	3.693174	3.693202	3.693161	3.692176
24	3.693122	3.692699	3.693028	3.69167
30	3.693202	3.692493	3.693202	3.69174
36	3.693183	3.692537	3.693036	3.691665
42	3.693188	3.692543	3.69291	3.691358
48	3.693234	3.692665	3.691569	3.691626
54	3.693129	3.692547	3.691295	3.691366
60	3.693187	3.692785	3.691602	3.691436
66	3.693169	3.692729	3.690931	3.691527
72	3.693162	3.692656	3.690967	3.691053
78	3.693234	3.692879	3.691294	3.691491
84	3.693102	3.692792	3.691086	3.691385
90	3.693049	3.692832	3.691254	3.691389
96	3.693211	3.692036	3.691313	3.691479
102	3.693125	3.691197	3.691159	3.69139
108	3.693214	3.691312	3.691335	3.691613
114	3.693165	3.691238	3.691069	3.691557
120	3.693058	3.691408	3.691163	3.691466
126	3.692876	3.691049	3.690976	3.691541
132	3.692888	3.691065	3.690926	3.691209
138	3.692942	3.691493	3.691204	3.691634
144	3.692851	3.691534	3.690851	3.691582
150	3.692986	3.691513	3.691228	3.691618
156	3.692702	3.691206	3.690973	3.691415
162	3.692883	3.691331	3.691246	3.69157
168	3.692998	3.691431	3.691189	3.691686
174	3.692838	3.691376	3.690891	3.691425
180	3.693056	3.691507	3.691201	3.691456
186	3.692979	3.691499	3.691141	3.691478
192	3.692996	3.691374	3.691335	3.691345
198	3.69306	3.691482	3.691285	3.691519
204	3.691743	3.691188	3.691213	3.691498
210	3.691528	3.691292	3.691286	3.691451
216	3.691481	3.691399	3.690885	3.691494
222	3.691613	3.691492	3.691087	3.691509
228	3.691468	3.691473	3.691225	3.691516
234	3.691427	3.691164	3.691113	3.69162
240	3.691549	3.691532	3.691241	3.691507
246	3.6915	3.691074	3.691044	3.691327
252	3.691037	3.691336	3.690945	3.691634
258	3.691623	3.691314	3.691181	3.691722

264	3.691503	3.691365		
270	3.691764	3.691386	3.691075	3.691383
276	3.691327	3.691266	3.691132	3.691526
282	3.691404	3.691511	3.690889	3.691612
288	3.691564	3.691481	3.691131	3.691457
294	3.691522	3.691336	3.6912	3.691421
300	3.691463	3.691382	3.69102	3.691424
306	3.691463	3.691335	3.691265	3.691637
312	3.691214	3.691267	3.690945	3.691712
318	3.691513	3.691371	3.690965	3.691528
324	3.691583	3.691175	3.691183	3.691459
330	3.691642	3.691161	3.691136	3.691506
336	3.69167	3.690987	3.691275	3.691549
342	3.691249	3.691284	3.691115	3.691511
348	3.691536	3.691545	3.691104	3.691416
354	3.691285	3.691185	3.690966	3.691586
360	3.691458	3.691515	3.691021	3.69131
366	3.691506	3.691151	3.691199	3.691497
372	3.691268	3.691203	3.691103	3.691368
378	3.691616	3.691509	3.690999	3.691096
384	3.691565	3.691522	3.691158	3.691511
390	3.691633	3.691261	3.691068	3.691367
396	3.691366	3.69123	3.691169	3.691606
402	3.691385	3.691659	3.691095	3.691435
408	3.691358	3.691643	3.691011	3.691655
414	3.691272	3.691471	3.691185	3.69148
420	3.691397	3.691544	3.69125	3.691428
426	3.691329	3.691563	3.691335	3.691557
432	3.691487	3.691385	3.690969	3.691475
438	3.691511	3.691449	3.69098	3.691311
444	3.691458	3.691278	3.691417	3.691461
450	3.691582	3.691575	3.691106	3.691559
456	3.691335	3.691232	3.691163	3.691703
462	3.691403	3.691312	3.691102	3.691556
468	3.691438	3.691554	3.691067	3.69158
474	3.691503	3.691264	3.69111	3.691494
480	3.69147	3.691525	3.691031	3.691102
486	3.691347	3.691325	3.691139	3.691549
492	3.691384	3.691059	3.691179	3.691364
498	3.691613	3.691607	3.691234	3.691393
504	3.691363	3.691245	3.691189	3.691458
510	3.691463	3.691537	3.691136	3.69137
516	3.691156	3.691112	3.691175	3.691434
522	3.691532	3.691398	3.691033	3.691391
528	3.691476	3.691576	3.691006	3.691384
534	3.691397	3.691238	3.69114	3.691546
540	3.691545	3.69154	3.691255	3.69155
546	3.691386	3.691228	3.691377	3.691751
552	3.691432	3.691294	3.691344	3.691643
558	3.691408	3.691304	3.691127	3.691416
564	3.691049	3.691165	3.691314	3.69161
570	3.691598	3.691398	3.691199	3.691668
576	3.691403	3.69142	3.69131	3.691766
			3.690823	3.69139

Handover :

Time (sec)	MOS	Traffic send (byte)	Traffic received (byte)	Bit error rate (bit)
0		2840	0	
6	3.692487	4400	2360	9.92E-06
12	3.692813	2706.667	2706.667	6.06E-05
18	3.693155	1893.333	1893.333	7.79E-05
24	3.693155	1853.333	1853.333	4.02E-05
30	3.692894	5946.667	5946.667	0
36	3.693112	3653.333	3653.333	6.92E-05
42	3.693155	2720	2720	8.91E-06
48	3.693155	1760	1760	1.97E-05
54	3.693155	3400	3400	9.21E-05
60	3.693054	4373.333	4373.333	0
66	3.693049	3413.333	3413.333	1.25E-05
72	3.692569	4573.333	4573.333	0
78	3.693156	3053.333	3053.333	4.92E-06
84	3.693156	2746.667	2746.667	3.02E-06
90	3.693018	2680	2680	0
96	3.693148	4133.333	4133.333	0
102	3.693039	2386.667	2386.667	2.52E-05
108	3.69311	3426.667	3426.667	7.85E-06
114	3.693156	2226.667	2226.667	0.000134
120	3.693102	3160	3160	0
126	3.693156	2560	2560	6.56E-06
132	3.693079	4453.333	4453.333	0
138	3.693156	3266.667	3266.667	3.36E-06
144	3.693156	4413.333	4413.333	3.98E-06
150	3.692738	2533.333	2533.333	0
156	3.693156	2706.667	2706.667	0.000143
162	3.692875	2826.667	2826.667	0
168	3.693156	2426.667	2426.667	0.000125
174	3.692883	1760	1760	0.000218
180	3.693156	1853.333	1853.333	1.98E-06
186	3.693156	2026.667	2026.667	0.000295
192	3.693156	3440	3440	1.08E-05
198	3.692514	2120	2120	4.15E-05
204	3.693155	2413.333	2413.333	4.15E-05
210	3.693155	3773.333	3773.333	4.15E-05
216	3.693155	2840	2840	0.003692
222	3.693155	2946.667	2933.333	0.000139
228	3.690868	3346.667	3360	0.000139
234	3.691126	2306.667	2306.667	0.000139
240	3.690776	4813.333	4813.333	0.000139
246	3.691136	3626.667	3626.667	0.000124
252	3.690362	1906.667	1906.667	0.000124
				0.00012

258	3.690835	3493.333	3493.333	0.000261
264	3.691138	2960	2960	0.000261
270	3.691029	2360	2360	0.000261
276	3.691155	2040	2040	3.38E-05
282	3.691105	2613.333	2613.333	3.38E-05
288	3.691183	626.6667	626.6667	5.48E-05
294	3.691159	2000	2000	1.23E-05
300	3.691162	3053.333	3053.333	1.56E-05
306	3.691126	1546.667	1546.667	3.92E-05
312	3.689498	1026.667	1026.667	0.000109
318	3.690973	4986.667	4986.667	3.17E-05
324	3.691136	4200	4200	3E-05
330	3.69079	2640	2640	3.33E-05
336	3.690855	3226.667	3226.667	2.02E-05
342	3.690616	3226.667	3226.667	0.002703
348	3.69114	3413.333	3413.333	2.01E-05
354	3.691022	3680	3680	1.29E-05
360	3.690654	1333.333	1333.333	0.000345
366	3.691121	2533.333	2533.333	2.38E-06
372	3.691124	4320	4320	0.000187
378	3.6905	3426.667	3426.667	3.8E-05
384	3.690369	1400	1400	6.48E-07
390	3.690584	2506.667	2506.667	0.00018
396	3.690925	4026.667	4026.667	2.38E-06
402	3.690331	1586.667	1586.667	0.001216
408	3.691074	2400	2400	9.04E-06
414	3.691025	1640	1640	1.27E-06
420	3.69103	2413.333	2413.333	0.000411
426	3.69056	4000	4000	1.21E-05
432	3.690491	2613.333	2613.333	3.22E-05
438	3.690975	3440	3440	0.000167
444	3.69092	4853.333	4853.333	2.88E-06
450	3.690674	2586.667	2586.667	0.000119
456	3.690654	1346.667	1346.667	7.87E-06
462	3.690479	1573.333	1573.333	7.02E-05
468	3.688366	2746.667	2746.667	2.86E-05
474	3.689698	3866.667	3866.667	2.61E-05
480	3.688081	2853.333	2853.333	0.000146
486	3.687973	3200	3200	4.13E-05
492	3.68373	4640	4600	5.12E-05
498	3.677795	3640	3560	6.8E-05
504	3.667425	2586.667	2453.333	0.003692
510	3.638515	4266.667	3693.333	0
516	3.573683	3800	2520	5.90E-05
522	#N/A	3026.667	0	0.000139
528	3.690654	2240	320	4.86E-05
534	3.690479	4706.667	4706.667	9.30E-05
540	3.688366	1933.333	1933.333	0.00012
546	3.689698	1933.333	1933.333	0.000261
552	3.688081	3773.333	3773.333	5.10E-05
558	3.687973	2720	2720	6.55E-05
564	3.68373	1360	1360	3.38E-05
570	3.677795	3053.333	3053.333	3.30E-05

The effect of Background Traffic on QOS :

Time (sec)	Data dropped low background	Data dropped medium background	Data dropped high background
6	0	1101.467	1735.733
12	307.2	2757.467	3042.133
18	0	0	753.6
24	0	33.73333	307.2
30	0	349.8667	1493.6
36	0	0	47.86667
42	0	183.6	1600.133
48	0	0	413.3333
54	0	21.33333	1721.333
60	0	16	211.8667
66	0	0	64
72	0	58.66667	310.2667
78	0	395.0667	569.6
84	0	1306.8	1092
90	0	0	127.2
96	0	0	14.4
102	0	48	833.3333
108	0	0	216.8
114	0	328.5333	483.7333
120	0	16	599.7333
126	0	53.33333	1234.4
132	0	16	509.2
138	0	0	307.2
144	0	32	383.8667
150	0	338.4	547.6
156	0	1389.867	2213.067
162	0	0	368
168	0	0	26.66667
174	0	159.4667	1023.2
180	0	5.333333	66
186	0	292.1333	606.8
192	0	0	253.6
198	0	26.66667	1399.467
204	0	32	480.5333
210	0	0	48
216	0	50.66667	925.2
222	0	152.6667	889.3333
228	286.6667	702.4	2057.2
234	0	0	409.6
240	0	0	0
246	0	53.33333	1067.867
252	0	0	146.9333
258	0	53.33333	301.4667
264	0	16	97.06667
270	0	163.0667	1081.733
276	0	37.33333	154.6667

282	0	0	16
288	0	48	263.6
294	0	150	1102
300	0	1412.133	2519.067
306	0	16	341.7333
312	0	0	16
318	0	61.46667	386.9333
324	0	0	69.73333
330	0	80	476.6667
336	0	265.2	520
342	0	42.66667	1293.867
348	0	16	617.4667
354	0	0	143.3333
360	0	344.5333	180.9333
366	0	110.4	1257.733
372	0	2105.467	1522.4
378	0	262.5333	988.4
384	0	0	0
390	0	50	973.7333
396	0	0	181.6
402	0	48	671.0667
408	0	0	423.7333
414	0	26.66667	248
420	0	16	205.3333
426	0	0	16
432	0	109.7333	503.0667
438	0	767.4667	733.0667
444	0	1336	1031.867
450	0	0	308.9333
456	0	0	0
462	0	121.0667	870
468	0	5.333333	48
474	0	5.333333	759.8667
480	0	37.33333	334.9333
486	0	48	416.1333
492	0	16	146.2667
498	0	0	0
504	0	90.8	602.1333
510	0	757.0667	725.0667
516	0	198.1333	1300.8
522	0	5.333333	618.1333
528	0	0	354.2667
534	0	26.66667	397.7333
540	0	456.4	85.33333
546	0	10.66667	452.1333
552	0	16	457.4667
558	0	37.33333	726.5333
564	0	32	101.3333
570	0	0	32
576	0	16	72.13333
582	0	534	1144.533
588	0	1251.2	3756.133

Time (sec)	MOS low background	MOS medium background	MOS high background
18	3.692436	3.564866	3.433201
24	3.692645	3.620354	3.485825
30	3.692809	3.648941	3.553567
36	3.692678	3.65545	3.571093
42	3.692608	3.662515	3.589043
48	3.692686	3.666235	3.604277
54	3.692681	3.668067	3.609912
60	3.692639	3.670089	3.615035
66	3.692642	3.671801	3.620323
72	3.692702	3.673988	3.626289
78	3.692687	3.674161	3.624567
84	3.692601	3.669972	3.6191
90	3.692542	3.665619	3.618819
96	3.692668	3.659411	3.620797
102	3.692571	3.661228	3.626104
108	3.692773	3.663439	3.62402
114	3.692601	3.664239	3.626376
120	3.692783	3.664191	3.627971
126	3.692692	3.664592	3.627517
132	3.692619	3.664567	3.626721
138	3.692772	3.664463	3.627003
144	3.692637	3.665123	3.628964
150	3.692605	3.665431	3.628001
156	3.692724	3.664078	3.624913
162	3.692586	3.662885	3.620482
168	3.692714	3.661823	3.62064
174	3.69276	3.662906	3.621821
180	3.692785	3.66333	3.62155
186	3.69262	3.664032	3.624248
192	3.692628	3.664512	3.622327
198	3.692658	3.66527	3.624464
204	3.692718	3.665723	3.625141
210	3.692625	3.665881	3.624877
216	3.692747	3.666701	3.625977
222	3.69284	3.667022	3.624364
228	3.69278	3.666161	3.625162
234	3.692693	3.666542	3.625263
240	3.692866	3.667025	3.624776
246	3.692706	3.667895	3.626361
252	3.692703	3.667921	3.624083
258	3.692743	3.668304	3.62611
264	3.692714	3.667474	3.626877
270	3.692614	3.668561	3.62887
276	3.692741	3.668574	3.627758
282	3.692764	3.669133	3.62837
288	3.692575	3.6697	3.628899

294	3.692657	3.669591	3.630839
300	3.692741	3.670143	3.629111
306	3.692593	3.668732	3.625147
312	3.692703	3.668334	3.627506
318	3.692648	3.668568	3.628832
324	3.69269	3.667738	3.627975
330	3.692684	3.667832	3.629702
336	3.692685	3.66741	3.627498
342	3.692706	3.668449	3.629298
348	3.692607	3.668426	3.628626
354	3.692707	3.66807	3.629526
360	3.692612	3.668613	3.629517
366	3.692611	3.667813	3.630094
372	3.692637	3.667923	3.62851
378	3.692766	3.667173	3.623865
384	3.692772	3.667774	3.625811
390	3.692765	3.667621	3.626711
396	3.692629	3.668161	3.62568
402	3.692728	3.668649	3.626814
408	3.692726	3.668473	3.625537
414	3.692714	3.669034	3.625951
420	3.69261	3.669415	3.626209
426	3.69271	3.669382	3.626023
432	3.692641	3.670149	3.627121
438	3.692753	3.670136	3.626842
444	3.692726	3.669595	3.62656
450	3.692546	3.667746	3.626963
456	3.692605	3.668344	3.627773
462	3.692778	3.668412	3.62871
468	3.692714	3.668888	3.628471
474	3.692712	3.668895	3.62928
480	3.692721	3.668856	3.629425
486	3.69247	3.669343	3.628899
492	3.692623	3.669745	3.629156
498	3.692486	3.669597	3.629564
504	3.692584	3.669932	3.63139
510	3.692533	3.669602	3.631184
516	3.692627	3.669426	3.629895
522	3.692779	3.668876	3.628456
528	3.692659	3.669097	3.631222
534	3.692573	3.669292	3.631943
540	3.692438	3.669398	3.631002
546	3.692556	3.66974	3.63143
552	3.692489	3.670274	3.631481
558	3.692581	3.670281	3.631903
564	3.692588	3.670276	3.632294
570	3.692501	3.670274	3.633174
576	3.692634	3.670136	3.633656
582	3.69261	3.669916	3.632826
588	3.692618	3.669792	3.633398
594	3.692518	3.667393	3.631675
598	3.692792	3.668842	3.632601
600	3.692378	3.669446	3.632838